AkAbak - Exercises

The exercises demonstrate how to use AkAbak with reference to simple, practical examples, The '\AkAbak\Scripts\Examples' contains example scripts for the exercises. The first exercise enters into particular detail and discusses the operation of the program at length[1].

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First steps

...in the script editor

In AkAbak, the data for the simulation are entered via the script. The script is a text describing the data of the device to be simulated. AkAbak displays the script in a window in which the text can be entered and modified. This window works like a text editor. The text can be saved and loaded from the hard disk at a later time, or printed. If you are used to working with a text editor, you will feel entirely at home. Naturally, the AkAbak editor does not provide all the functions of a full word processor, but only those required for editing the data. Once the data have been entered, you can start the simulation. AkAbak interprets the script of the active window, since it is possible to work with several editors simultaneously. If an error occurs the program issues a message, otherwise it generates the diagram of the desired simulation.

Now to a few preliminary exercises to get to know some of the editor’s functions. First create a new script:

Menu: 'File/New...'

Now you see an empty window. The caption bar shows a default file name 'Script1.aks', which you should replace with your own when you save the script.

You can write in the empty area. Enter any text. Backspace (←) erases the text again. Delete (Del) erases characters in front of the cursor. Enter (↵, Return) forces a new line. You can use the cursor keys ←↑↓→ to move backwards and forward and up and down in the entered text. If you press the control key (Ctrl) while operating the cursor keys, the distance you move increases. Try it...

It is important to be able to use the clipboard. The clipboard can temporarily save text and images and even insert them anywhere in the program. It is used in the same way in all Windows programs. AkAbak makes intensive use of it.

To get to know how to use the clipboard, type a few words. To copy the text into the clipboard, you first have to mark it. Press the shift key (⇑) together with one of the cursor keys. The text is shown inversely. Try it out. The cursor keys, Home, End, PgUp, PgDn etc. all work in this way. Control (Ctrl) is also still active. You can also use them for marking. Press the left-hand mouse key and keep it pressed down. If you move the mouse while doing this, you will mark everything under the mouse cursor.

You have now marked the text. In the Windows programs, commands associated with the clipboard are all in the 'Edit' menu. If you open this menu, you can see the commands and, at the right-hand side, the shortcut keys for the commands. If you move through the commands with the cursor, a short explanation is given in the status line.

Copy	Copies the marked text into the clipboard (Ctrl + C)
Paste	Inserts from the clipboard at the cursor position (Insert)
Cut	The same as Copy, except that the marked text is erased (Ctrl + X)

Activate the "Copy" command. To demonstrate this, delete the marked text in the script by pressing any cursor key. Activate the 'Paste' command or press Ins. The text from the clipboard is inserted at the cursor position. Let us assume you don’t want to do that, but instead you want to reverse the process. Press the Alt-Backspace combination (Alt+←) or actuate the menu command: 'Edit/Undo' and the former state is restored.

On the subject of marking, please note that the marked text is in danger - very great danger!. Mark a few words and then enter some text. See how the previous marked text disappears and is replaced by the new text. That also applies for inserting (pasting). Marked text is replaced by the new text from the clipboard.

Scripts accompanying the exercise

In the following, you can either follow the exercises exactly, entering the data, or load the script files accompanying this exercise. In the description, it is assumed that you type in the data.

The exercise scripts are numbered consecutively. The first number represents the exercise, the second the script version within an exercise.

Loading the script for the exercise (optional):

Open the dialog file for loading script files (menu: File/Open script diagram).

– In the list of directories, set the directory: „\AkAbak**Scripts\Examples**' in the center of the dialog (double click on 'Examples' with the mouse).

– In the left-hand list of file names, double click on, for example, the entry **Ex_11.aks**'. The first exercise script for the first exercise is loaded and displayed in the window. The following exercise scripts have corresponding names 'Ex_12.aks' etc.

Just snooping around...

To gain an impression of how the simulation works, just load one of the script files from the directory: '\AkAbak**Scripts\Examples**'. For example:

script file: **Ex_17.aks**

This script describes a small two-way speaker with a simple passive crossover network, as is constructed in the following exercise.

Use the simulation of the sound pressure level curve. When the script has been loaded and the script window is active, issue the menu command: '**Sum/ Acoustic Pressure...**'. The control dialog for this simulation appears. Leave all the settings as they are and press Enter ↵. In the diagram you now see the sound pressure curve from three listening angles in the vertical.

If you want to close one of the two windows, press the Ctrl+F4 combination.

1. Designing a Two-Way Loudspeaker

Design data

We will now design a small two-way loudspeaker system with a crossover network made from passive components. There are two loudspeaker chassis present. Their Thiele/Small parameters are determined with free radiation, i.e. uninstalled:

Bass loudspeaker

Resonance frequency fs=65Hz, equivalent volume to the compliance of the diaphragm suspension Vas=12.5L, mechanical quality Qms=1.25, electrical quality Qes=0.6, d.c. resistance of the voice coil Re=5.5ohm, inductance of the voice coil Le=2.35mH, diaphragm diameter dD=12.75cm, diameter of the dust cap dD1=1.5cm, depth of the diaphragm cone tD1=3.5cm, control-frequency of diaphragm-mass-reduction fp=1700Hz (measured without baffle and housing).

Maximum possible linear diaphragm excursion from peak to peak Xmss=8mm. Nominal electrical loading capacity Pelmax=80W.

Tweeter (dome)

Resonance frequency fs=1.71kHz, mass of the oscillating system, incl. air load Mms=0.33g, mechanical quality Qms=0.7, electrical quality Qes=1.2, d.c. resistance of the voice coil Re=5ohm, inductance of the voice coil Le=140uH, diaphragm diameter dD=28mm, height of the dome tD1=7mm, depth of a small horn surrounding the dome t1=4mm, control-frequency of diaphragm-mass-reduction fp=8kHz (measured on the baffle).

Nominal electrical loading capacity Pelmax=100W in the frequency range 3kHz to 20kHz

We will first simulate the system using the simplest possible elements. The element `BassUnit` is used for the bass and the element `Speaker` for the tweeter. These elements, together with their definitions, `Def_BassUnit` and `Def_Speaker`, are also know as 'instant' elements. They are used for quick simulation or as an introduction to the design.

Step 1: Generating a new script

Menu: 'File/New...'

Now are presented with an empty window. In the caption bar is a default file name '...\Script1.aks', which you can replace with your own file name when you save the file.

Step 2: Entering the parameters of the bass driver

Script file: Ex_11.aks

The `BassUnit` element, which you will be installing in the network later, always has associated with it a definition `Def_BassUnit`. This receives the loudspeaker parameters and is at the start of the script. The definition has to be given a name - any name - and as many `BassUnit` elements as required can therefore refer to this definition. The `BassUnit` network element contains information about the position in the network and about the position on the baffle. This structure has proved useful and is used for all driver types that the program supports.

'Def_BassUnit / Calculator' dialog

Now enter the parameters of the definition `Def_BassUnit`. The parameters can be entered directly into the script. Choose the method most convenient for you and open the dialog 'Def_BassUnit Calculator' (Def/Def_BassUnit... menu).

One of the biggest dialogs in AkAbak opens up. The Def_BassUnit dialog is used not only for comfortable data input, but is also a tool for designing bass speakers.

Non-modal dialogs

First a few points about the dialogs. The Def_BassUnit dialog is a so-called non-modal dialog. That means you can leave it and re-activate it. There are also other commands in the system menu of the dialog (top left in the corner of the window). Press the Alt+space combination and the system menu drops down. If you just want to terminate non-modal dialogs, press the Alt+F4 combination (**Esc does not work here!**).

Input

First please enter the aforementioned parameters of the bass driver. The cursor is flashing in the first input box. The input boxes are miniature text editors and work like the script editor. The clipboard can also be used (not via the menu, however, but via the keyboard). Now enter the parameters. Jump from box to box using the Tab key (Tab or shift+Tab), or using the mouse. Boxes whose names are followed by three stops (e.g. Qms...) have subdialogs in which alternative parameters can be entered. The subdialogs open when the cursor keys Alt+↑↓ are pressed or the right mouse key pressed. You don’t need this function just yet.

How do you enter the data of Vas=12.5L? Quite simply: Type ‘12.5L’ in the Vas box. In the box for the diaphragm diameter dD enter ‘12.75cm’. AkAbak understands that. You can enter subunits in all numerical input boxes unless the unit is compound (e.g. cm3, mm3, L for volumes). Take care to distinguish upper and lower case for units (to be able to make a distinction between, for example, Mohm and mohm). For volumes, [L] can be used for liter or [in3] for cubic inch and, for distances, [in] can be used for the inch dimension. Liters and inches cannot be further subdivided. If you only enter the value without a unit, AkAbak uses the appropriate SI unit. Compound units, for example Ns/m^5 or Tm can only be entered in so-called scientific (SCI) notation. A small 'e' is used here to indicate powers of ten.

All numerical input fields have the so-called spin function. Press the cursor keys ↑↓ to spin the value by steps of the E96 row or, when you press additionally the Ctrl-key, the step size is of the E12 row.

All the data of the driver have now been entered in the appropriate boxes. However, we still need an appropriate bass enclosure. The design aids of this dialog are not described until one of the following exercises. In the 'Enclosure' group, enter 8L for the enclosure volume in the input box for 'Vb'.

We still need another two values. Jump to the box for mb and enter the value 0.95. This takes into account the modified radiation conditions when the loudspeaker is installed in an enclosure.

Enter a value of 0.1 in the Qb/fo box. Qb/fo is a factor for the acoustic losses in the enclosure.

Before you return to the script, another entry has to be made. `Def_BassUnit` is not yet the network element, but the associated definition. Before it can link these two, the definition requires a name. The name may contain any characters (max. 20 characters). Enter it in the 'Identification' box.

Inserting parameters into the script

At the bottom right-hand side in the dialog is the button 'Copy to clipboard and close'. Activate it and AkAbak closes the 'Def_BassUnit Calculator' dialog and any others involved. When this is done, the entered parameters are correctly formatted and copied into the clipboard.

The cursor is flashing in the script again. Press Ins (or menu: Edit/Paste) and the parameters appear in the script.

The first word in the first line of the definition is the `Def_BassUnit` keyword. It follows the name in quotation marks ('...' or "...") and the parameters in the following lines. All values have units. Unlike in the input boxes, the value must be followed by the unit in the script.

```
Def_BassUnit   'B1'
  fs=65Hz  Vas=12.5L
  Qms=1.25  Qes=0.6  Re=5.5ohm  Le=2.35mH
  dD=12.75cm  |Piston
  Xmss=8mm  mb=0.95
  Vb=8L  Qb/fo=0.1
  |Performance closed cabinet:
  |  fc           Qtc       fD        f3
  |  98.8Hz       0.639     857.1Hz   110.5Hz
  |  Lwmax        Pelmax    Uorms     t60       Ripple
  |  99.4dB       23.5W     11.37V    13.45ms   0
```

The last lines each start with the comment character **|**. The reproduction characteristics are documented here, as calculated in the 'Def_BassUnit Calculator' dialog. If you do not require them, you can simply erase them. Mark the lines starting with the **|** character and press the Del key. The **|** character introduces a comment. The interpreter ignores everything following it. The comment extends as far as the line end or as far as the next **|** in the same line.

Step 3: Saving the script

Next save the script to the hard disk. In Windows, commands for saving, printing, etc. are usually in the 'File' menu. Activate the command 'File/Save as'. The file dialog for saving scripts appears.

In the list to the right of this dialog, you can see the directory structure of the drive currently active. Select the directory in which you want to save the file.

The input box for the file name contains '*.aks'. Replace the text with your own, for example Test1. If an input box is marked, you only need to type into it. The first character erases the old text.

Don’t forget that the operating system only allows 8 characters for the file name. The file name extension 'aks' is appended automatically. The file is saved when the actuate the 'OK' button.

Step 4: Building up the network for the bass speaker

So far, the script only contains a definition. The currents do not know yet how they are supposed to flow. What you need is a framework in which the structure of the network and the positions of the radiators can be specified. In the AkAbak, this framework is called `System`. All network and filter elements following this keyword in the text form part of a network. The next network starts with the keyword `System` again. Each System should be named, for example 'S1', 'S2' or so, because only named System's and, also only named components, are displayed in the network list of the simulations.

Input in the 'BassUnit' dialog

The parameters of the `BassUnit` network element can be either entered manually or a dialog used. Use the dialog (Net/ Transducer/ BassUnit menu). This dialog is much smaller than the dialog of Def_BassUnit. The cursor is flashing in the input box for the element name. The name is optional here. Leave the box empty and jump directly to the first box of the network node **s**. Enter a 1. The input voltage of the network is always at node 1 and ground node at zero. The 0 in the box of for node **t** remains.

The right-hand list contains the names of all `Def_BassUnit` definitions. In this case only the one that you entered when naming `Def_BassUnit`. Click on the entry with the mouse or use the cursor keys ↑↓, to copy the entry into the box above it. Please do not use the 'Position of radiation center...' button yet. A sub-dialog opens to establish the position of the bass speaker on the baffle. The position is changed later. At first, the driver is located at the origin of the baffle co-ordinate system. Since all entries have been made, close the dialog again using the 'Copy and close' button.

The data are now in the clipboard and are inserted at the cursor position in the script when you press Ins.

The script now looks like this:

```
Def_BassUnit   'B1'
  fs=65Hz  Vas=12.5L
  Qms=1.25  Qes=0.6  Re=5.5ohm  Le=2.35mH
  dD=12.75cm  |Piston
  Xmss=8mm  mb=0.95
  Vb=8L  Qb/fo=0.1


System  'Bass'
  BassUnit   Def='B1'
    Node=1=0
    x=0  y=0  z=0  HAngle=0  VAngle=0
```

Step 5: Simulating the sound pressure level

You have now made all the preparations to allow you to simulate a complete bass system. First look at the on-axis sound pressure curve of this bass unit at an input voltage of 1 volt.

Simulation - sound pressure level

Start the simulation using the ‘Sum/Acoustic pressure' command (keyboard shortcut: F5). The 'Acoustic pressure' control dialog appears. The dialog has some control elements. First leave everything just as it is and press the 'OK' button or simply Enter ↵ (the button outlined in black always responds immediately to Enter). The dialog disappears and a diagram window opens (Fig. 1). A curve is drawn in the diagram, which corresponds to the sound pressure level at an input voltage of 1Vrms at a distance of 1m.

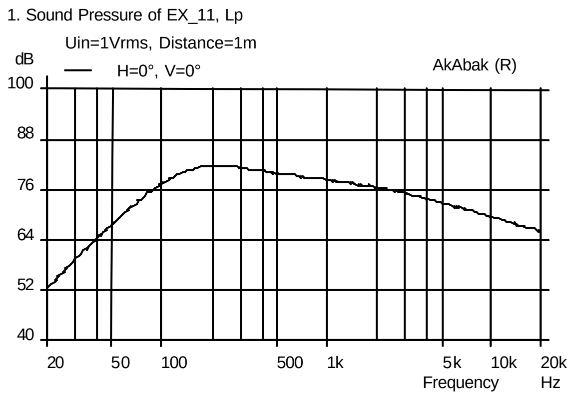


Fig. 1 Sound pressure level of bass-loudspeaker on-axis.

Diagram

Before you proceed, investigate the things you can do with the diagram. Double click on the area of the ordinate numbers. The ordinates are adapted so that you can see the whole curve.

Double click on that area again and the old state is restored. You can also zoom out a portion of the curve. First a zoom window has to be pulled out. If you press the left mouse key, hold it down and then move the mouse to the right the abscissa will be zoomed. When you move the mouse downwards the ordinate will be zoomed. A double click on the abscissa or ordinate area restores the old state. The network is not recalculated during zooming. The zoom depth is therefore limited. There is a trick for increasing the resolution in the zoom window, which you will learn about later on.

Phasing in the marker

Click on the graphs with the mouse or press one of the cursor keys. A small cross and a window containing values appears. The upper value is the abscissa value and the lower one the ordinate value of the marker position. The border of the box has the colors of the graph on which the marker is located. The cursor keys ←, →, End and Home move the marker on the graph. Its speed increases if you press Ctrl at the same time. The +

and - keys locate the maximum and minimum of the graph. Since a diagram can represent up to four graphs simultaneously, the ↑↓ are reserved for changing from one graph to the other. The marker disappears when you press Esc.

You can use the relative marker for dimensioning the curves. This functions exactly in the same way as the absolute marker, except that you have to additionally hold down shift. The marker panel then shows the difference between the two markers in brackets.

By the way: If your monitor does not display the marker very well, you can change its color (File/ Preferences/ Screen diagram style... menu).

'Diagram Edit Range' dialog

Double click on the area of the diagram region or simply press Enter↵ or else use the Edit/Range... menu. The 'Diagram Edit Range' dialog opens. Here, you can enter the ordinates and the abscissa manually.

Step 6: Simulation of the acoustic power

Leave the diagram of the sound level on the screen and activate the script window again (simply click anywhere on the area of this window or press the Ctrl+F6 or Ctrl+Tab key).

Now activate the 'Sum/ Acoustical Power...' menu command. The 'Acoustic Power' control dialog opens.

In the 'Integration' group the switch is set to 'Cross'. The acoustic power is calculated by integration of the sound pressure on an envelope surface around the speaker. The 'Cross/Area' switch is used to control the integration. If it is set to 'Cross', all sound pressures of a horizontal and a vertical meridian on the integration sphere are added together. If the switch is set to 'Area', AkAbak adds up the sound pressure on the entire sphere. Since, at present, only one radiator is exists, which is symmetrical with respect to the origin of the baffle co-ordinate system, leave the setting as it is. Press 'OK', therefore, to start the simulation. You will see that the curve grows more slowly than the sound level curve. By the way, as you can see from the status line, the computation can be terminated at any time with Esc.

When the power simulation is finished the directivity factor Q is displayed which is the ratio of the on-axis pressure level to the power level.

Step 7: Comparison of curves

It is conspicuous that the curve of sound power falls off earlier at higher frequencies than the sound pressure curve (Fig. 2). For a better comparison, place one window above the other. To do this, activate the Window/Tile Diagrams menu command.

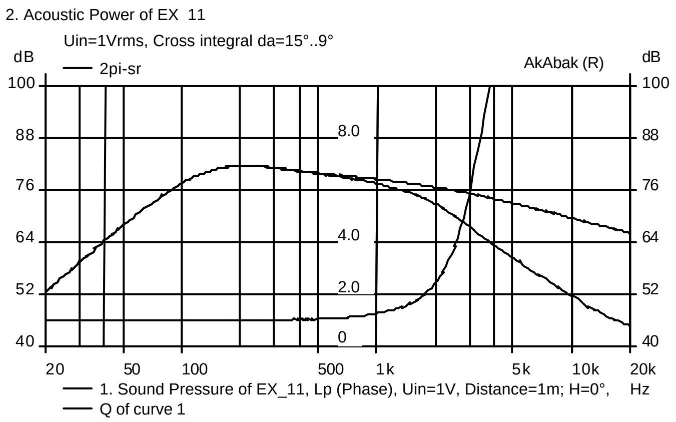


Fig. 2 Sound pressure level as in Fig. 1,
level of acoustical power of bass loudspeaker and
Directivity Factor Q

Copying graphs

You can also go a step further and copy the graphs from one diagram into the diagram of the other window (Fig. 2). To do this, click on the legend of the graph for acoustic power and keep the mouse key pressed. The legend is shown in inverse display and the mouse cursor changes. Move the cursor into the diagram area of sound level and let go. Two ordinates are now shown. The right-hand ordinate forms part of the so-called 'guest graph', in this case the acoustic power. The legends of the guest graphs are always below the diagram.

The curve for power has to decrease earlier since, from a certain frequency, (directivity frequency), the radiated sound pressure decreases towards the side which is also displayed by the curve of the Directivity Factor. It is only directly in front of the loudspeaker, i.e. on-axis, that the sound pressure remains constant. The pitch of the curve is at the expense of voice coil induction.

Step 8: LP at different listening angles

To be able to investigate the radiation behavior in greater detail, the example in the exercise simulates the sound pressure curve from three different listening angles. Activate the diagram of the sound pressure simulation and press **Alt+Y** (menu: Calc/ Simulate again...). In the 'Acoustic Pressure' control dialog, leave everything as before, except in the 'Listening angles' group. Here click on the 'Horizontal' switch. The three input boxes below it are activated. You can enter the listening angles in the horizontal plane here. Leave everything as it is and press 'OK'.

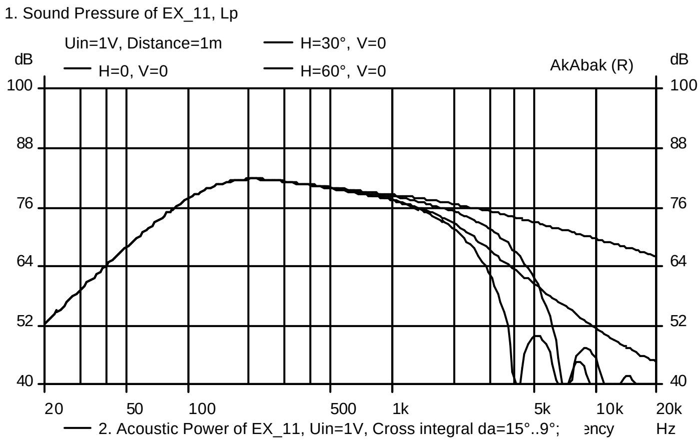


Fig. 3 Sound pressure level curves at the listening angles of 0°, 30° and 60° and the acoustic power of the bass loudspeaker (thin)

You now see that three graphs are drawn in the diagram (Fig. 3). The first is the sound pressure curve directly in front of the loudspeaker again. For the next, the listening point is displaced to the left by 30°. (seen from the speaker). The third indicates the curve at 60°. You can now easily see why the power is reduced.

Step 9: Recomputing changes in the script directly

Script file: Ex_12.aks

AkAbak has a very useful function that allows graphs to be recomputed. All settings of the diagram are preserved. You can therefore observe the effects of changes in the script immediately. Leave everything just as it is and activate the script. Now additionally enter the following in the `Def_BassUnit` line containing the diaphragm diameter dD=12.75cm (you can delete the commentary '|piston'):

dD1=1.5cm tD1=3.5cm fp=1.7kHz

The definition then appears as follows:

```
Def_BassUnit   'B1'
  dD=12.75cm  dD1=1.5cm  tD1=3.5cm  fp=1.7kHz
  fs=65Hz  Vas=12.5L
  Qms=1.25  Qes=0.6  Re=5.5ohm  Le=2.35mH
  Xmss=8mm  mb=0.95
  Vb=8L  Qb/fo=0.1
```

These entries describe the diaphragm shape in detail. So far it was a flat disk, or piston, since only the diaphragm diameter `dD=` had been entered. Most bass speakers have a conical shape. In this case `dD1=` is the diameter of the dust cap or the inner diaphragm and `tD1=` is the depth of the cone from the edge of the suspension as far as the dust cap. This leaves the position of the driver on the baffle unaffected.

The frequency `fp=` is used to control the effect of the so-called mass reduction of the diaphragm. At high frequencies, not the entire cone vibrates, but only a part that reduces in size with increasing frequency. With the exception of the partial vibrations that normally occur, AkAbak takes into account the modified effective mass and the reduced diaphragm shape. In the case of the conical diaphragm, the outer diameter is reduced. In the case of the dome, a hole is formed in the center of the diaphragm, so that an annular radiator is produced. The diaphragm area reduction has an effect on the radiation characteristics and on the radiation impedance.

If you have entered the two parameters, activate the 'Calc/ Simulate again' menu command or press **Ctrl**+**Y**.

In the status line, you see the message 'long computation', and the mouse cursor is an hourglass. Wait until the message has disappeared. What is happening? AkAbak is now recomputing the simulation for each diagram window having the same name as the script window. Since in this case one diagram window is the acoustic power, the computation takes a particularly long time.

You can now recalculate a single diagram. For this purpose the diagram window has to be enabled before you can issue the command for recomputation. That means that, when the script window is activated, all the associated diagrams are recomputed. If only one diagram window is active, only this diagram is recalculated. Reduced diagrams and guest graphs are not included in the recomputation.

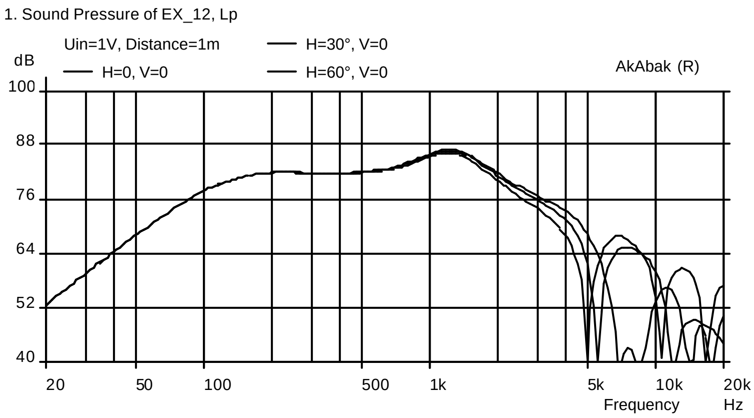


Fig. 4 Sound pressure curves at listening angles of 0°, 30° and 60°

What can you see now in Fig. 4? The acoustic pressure level now falls off more steeply at higher frequencies. Interferences resulting from the diaphragm shape cause this falling off. If the wavelength is small compared with cone depth, the sound causes extinctions. The acoustic center of the radiator migrates inwards. At extremely high frequencies, virtually only the inner diaphragm radiates sound. In practice, the natural vibrations of the diaphragm in this frequency range are also to be added. However, AkAbak cannot simulate these.

Step 10: Entering the Def_Speaker definition

Script file: Ex_13.aks

How do we proceed now? Of course you can investigate the bass unit further. (see, for example, the simulations in the Inspect/ menu). The next step in designing the two-way speaker is to add the tweeter, since the sound level curve shows that the speaker at present only radiates uniformly up to maximum 2kHz.

First enter the parameters of the tweeter, as described above, as a definition `Def_Speaker`. Like the BassUnit element, the speaker element is an 'instant' element. It is a very compact and practical means of installing complete loudspeakers. Your tweeter is in this category since it is rigidly installed in a housing.

You should note that the speaker element implements a very simple model. For example, some dome mid-range units are not correctly simulated by the speaker element, since the sound from the diaphragm reverse is directed via an acoustic vent into the enclosure behind the magnet. A Helmholtz resonator is produced here, whose resonance is reflected in the sound level curve. If you want to simulate such a mid-range unit, use the elements `Duct` and `Enclosure`.

Def_Speaker dialog

The `Def_Speaker` definition is again entered with the aid of a dialog, 'Def/Def_Speaker' menu. Fill in the boxes with the data as given for the dome tweeter described above. Don’t forget the obligatory name here. When you reach the box 'diaphragm dimensions', press the cursor keys Alt+↑↓ or click on the input box with the right-hand mouse key.

'Diaphragm' sub-dialog

The 'Diaphragm' sub-dialog pops up. You can use this dialog for entries relating to the radiation diaphragm. First choose the diaphragm shape. In the 'Outer dimensions' group, choose 'Circular' and in the 'Inner dimensions' group, choose the diaphragm shape 'Convex dome'.

Then enter the diaphragm diameter dD. Since the dome is slightly recessed, the input box contains `t1=4mm`. Enter the fp=8kHz in the 'Frequency of mass reduction' input box. With a convex dome-shaped diaphragm, only the dome height tD1 is required. The area of the inner diaphragm is zero. Enter the dome height tD1 in the box at the bottom right. Push the OK button. The dialog is closed and its entries are saved in the `Def_Speaker` dialog.

When you have entered everything, close the dialog again using the 'Copy and close' button. Then place the cursor in an empty line in the script above the `System` keyword and after the `Def_BassUnit` definition. Press Ins and the parameters of `Def_Speaker` are inserted.

It does no harm to save the script from time to time (File/Save menu or Ctrl+**S** key).

Step 11: Building up the network for the tweeter

Since the example in the exercise constructs a crossover network from purely passive components, you basically only needed one system, i.e. only one network. Since, however, low-pass and high-pass of this crossover are not networked together, the example utilizes the advantages of two systems. Each of these networks has at first only one element, namely the loudspeaker. First of all, therefore, please design the crossover network abstractly using the filter elements. AkAbak can then synthesize the passive filter network.

After the parameters of `BassUnit`, enter the `System` keyword again and open the 'Network Element Speaker' dialog via the 'Net/ Transducer/ Speaker...' menu. As with the BassUnit dialog, enter the data in the dialog and close it using 'Copy and close'. Enter the parameters from the clipboard into the script below the `System` keyword.

Before you start the simulation, you have to change something else in the BassUnit element. At present, both loudspeakers radiate from the same position. The bass therefore has to be moved downwards by 15cm. Change the `BassUnit` element by moving its vertical position to `y=-15cm`. The script now looks like this:

```
Def_BassUnit   'B1'
  fs=65Hz  Vas=12.5L
  Qms=1.25  Qes=0.6  Re=5.5ohm  Le=2.35mH
  dD=12.75cm  dD1=1.5cm  tD1=3.5cm  fp=1.7kHz
  Xmss=8mm  mb=0.95
  Vb=8L  Qb/fo=0.1


Def_Speaker  'S1'
  fs=1.71kHz  Mms=0.33g
  Qms=0.7  Qes=1.2  Re=5ohm  Le=140uH
  dD=2.8cm  tD1=7mm  |Convex dome
  t1=4mm  fp=8kHz
  mb=1


System  'Woofer'
  BassUnit   Def='B1'  Node=1=0
    x=0  y=-15cm  z=0


System  'Tweeter'
  Speaker   Def='S1'  Node=1=0
    x=0  y=0  z=0
```

Step 12: Simulation of woofers and tweeters

Simulate the sound pressure level again from different listening angles and the acoustic power as above. Look at the sound pressure curve from different listening angles.

As you see, the frequency range 1kHz...3kHz has a regular ripple of the sound pressure and power levels.

Using Labels

To see both System's independently together with the total response activate the simulated diagram of the sound pressure and issue 'Calc/ Simulate again...' (Alt+Y). Click on 'Multi-Labels' and switch on the boxes below 'Graph'. Select then in the first list '<all>', in the second list 'Bass' and in the last list select 'High'. Press 'Ok' and then the three curves display in the first curve the total response and the other curves show the sound pressure of the Bass and High System only.

The Label-feature can be extended also to individual radiators to investigate the system-response in great detail.

Step 13: Diaphragm excursion of the tweeter

Activate the script and start the simulation of the diaphragm excursion (Inspect/ Excursion). In the control dialog select in the network list 'High, 1->0, Speaker S1'. The network list displays all System's and elements which has given a name.

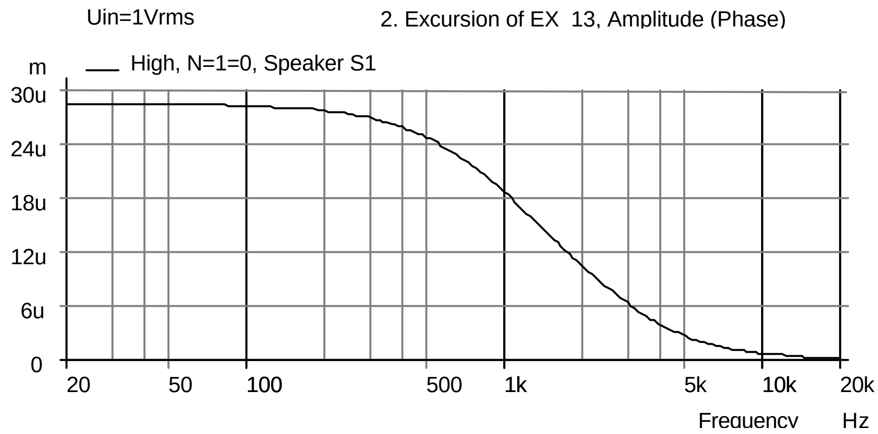


Fig. 5 Diaphragm excursion curve of the tweeter

Click on the 'OK' button and the diagram is drawn. However, you do not see a graph. This is because the excursion of the tweeter is extremely small. Double click on the ordinate area. The diaphragm excursion of the tweeter is displayed as a function of frequency (peak value, but not peak-to-peak). As you can see, the curve is that of a low-pass function. At low frequencies, its curve is constant and starts to fall off at approx. 300Hz. At approx. 1.5kHz, the excursion is only half. If you know the maximum excursion of the loudspeaker, it is easy to calculate the loading capacity with the aid of this diagram.

Step 14: Designing the crossover network

Script file: Ex_14.aks

A crossover network that keeps low frequencies away from the tweeter and high frequencies away from the tweeter will improve the reproduction and performance.

'Filter' dialog

Set the crossover frequency to 3kHz and look for an appropriate filter circuit. To do this, open the filter dialog in the 'Filter/Filter Dialog...' menu. This dialog contains some tools to allow you to generate and investigate certain kinds of filters and characteristic curves.

The cursor is flashing in the 'Identification' input box. Leave this empty and jump into the next box, with the name 'Filter pole frequency fo'. Enter the crossover frequency 3kHz here.

Sub-dialog 'Standard lowpass functions'

In the next step, you can have AkAbak calculate a transfer function. Click on the 'Standard lowpass functions...' button. A sub-dialog opens. In the first input box, you can enter the order of the filter. Leave the 2 where it is, since we want to generate a 2nd order transfer function. The list contains a selection of typical characteristic curves. Below these are some of particular interest for constructing crossover networks. Choose the Butterworth characteristic curve. Now click on the 'OK' button. The dialog closes and the 2nd order Butterworth transfer function is now entered in the filter dialog in the 'Transfer 2' group.

At the top is the numerator polynomial, and below it the denominator polynomial. Before this transfer function can be exported as a script element, it has to be in the 'Transfer 1'input box. Therefore click on the 'Copy to 1' button. The multi-line input box then contains:

```
b0=1;
a2=1;  a1=1.414214;  a0=1;
```

`b0=` is the numerator coefficient, `a2=`, `a1=` and `a0=` are the denominator coefficients. The index is the same as the power of the frequency variables. The entries here have to be separated by a semicolon. There is a reason for this: The entries of the filter coefficients of the `Filter` element may also be expressions. For example, instead of `a1=1.414214` you can also enter: `a1=sqrt(2)`, where 'sqrt' is the square root.

In the next step, generate the symmetrical high pass for this function. Click on the 'Lowpass to highpass' button in the 'Transfer 2' group. The transfer function in the 'Transfer 2' group is reflected. For Butterworth functions, not very much happens, only the numerator is changed.

Diagram in 'Filter' dialog

The dialog now contains two transfer functions. The low pass is in the group 'Transfer 1' and the high pass in the group 'Transfer 2'. Click on the 'diagram' button and the modulus of the transfer function is drawn. You can see how the low pass and high pass intersect at 3kHz. Their level in this case is -3dB. The third curve results from the sum of the other two. Although that doesn’t look good. the example in the exercise is intended to show that, at the crossover frequency, complete extinction takes place, i.e. the phases of the two signals are 180° apart at this point. To correct this problem, please proceed as in the filter dialog. (Simply click on its area. You do not need to close the diagram window). Jump into the 'Transfer 1' input box and place a minus sign before the one of b0 (`b0=-1`). Click on the 'diagram' button again. Now the diagram looks better. Instead of an extinction, there is a slight emphasis of the crossover frequency.

Copying the coefficients into the script

Please close the diagram window now, or activate the filter dialog. Copy the transfer function of the 'Transfer 1' group into the script. To do this, click on the 'Copy function 1 to clipboard and close'. Then place the cursor in an empty line in the script below the `BassUnit` parameters, i.e. before the second System and press **Ins** to insert the filter parameters.

What does AkAbak do with this Filter element? The filter element has no node entries. There is thus no network in the System. The program multiplies together all filter elements listed within a system. The result is used to weight the input voltage of the network U1.

You can generate the high-pass filter for the tweeter yourself (Filter dialog → Filter pole frequency fo (3kHz) → Standard lowpass functions... → 2nd order Butterworth → Lowpass to highpass → Copy to 1 → Copy and close). Enter this filter into a free line below the `Speaker` parameters, i.e. at the script end. You can also generate the high-pass filter more quickly, by the way, by marking the low-pass filter, copying it into the clipboard (Edit/Copy menu) and then inserting it again. The high-pass transformation in this case is very simple. Replace the numerator coefficients `b0=-1` with `b2=1`.

The script now has the following text:

```
Def_BassUnit   'B1'
  fs=65Hz  Vas=12.5L
  Qms=1.25  Qes=0.6  Re=5.5ohm  Le=2.35mH
  dD=12.75cm  dD1=1.5cm  tD1=3.5cm  fp=1.7kHz
  Xmss=8mm  mb=0.95
  Vb=8L  Qb/fo=0.1

Def_Speaker  'S1'
  fs=1.71kHz  Mms=0.33g
  Qms=0.7  Qes=1.2  Re=5ohm  Le=140uH
  dD=2.8cm  tD1=7mm  |Convex dome
  t1=4mm  fp=8kHz
  mb=1

System  'Woofer'
  BassUnit   Def='B1'
    Node=1=0
    x=0  y=-15cm  z=0
  Filter
    fo=3kHz
    {b0=-1;
     a2=1;  a1=1.414214;  a0=1; }
```

```
System  'Tweeter'
  Speaker   Def='S1'
    Node=1=0
    x=0  y=0  z=0
  Filter
    fo=3kHz
    {b2=1;
     a2=1;  a1=1.414214;  a0=1; }
```

Simulation with Filter-elements

The script now describes a complete speaker with crossover network, which could even have been built up with active filters. Now simulate the sound level from different angles and the acoustic power of this circuit. As you can see, the result is still not satisfactory. At the transfer frequency, there are severe notches. How do we correct this? As you have seen above in the analysis of the transition from low-pass to high-pass, the phase response in the summation of low-pass and high-pass plays a significant role. Try removing the minus sign from the numerator coefficient `b0=` of the low pass (`b0=1`) and simulate it again. (Ctrl+Y). Now the response curve looks more promising. At the vertical listening angle there is severe rippling in the sound level curve. This ripple, however, is unavoidable with this type of design. It is caused by interferences resulting from the difference in travel time caused by the displacement of woofer and tweeter.

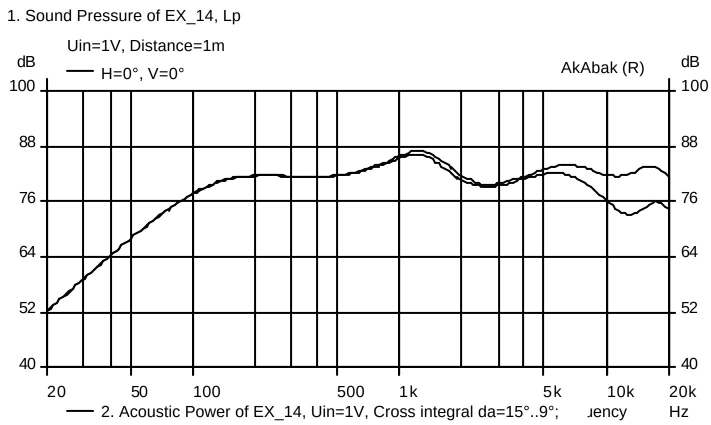


Fig. 6 On-axis sound pressure curve and
acoustic power curve for woofer and tweeter with filters

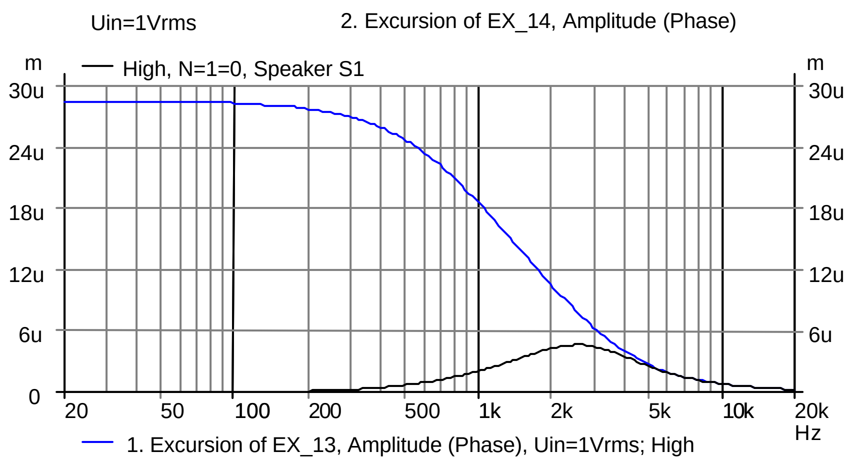


Fig. 7 Diaphragm excursion curve of the filtered and unfiltered tweeter (guest graph)

Diaphragm excursion

The result of the diaphragm excursion simulation for the tweeter is also interesting (Fig. 7). The diaphragm excursion reaches a maximum at approx. 2.6kHz and the excursion is damped by approximately a sixth of the maximum of the unfiltered tweeter (guest graph in Fig. 7).

5. Step: Impedance compensation

Script-file: Ex_15.aks

It is useful to create a new script for the passive version of the design. First save the currently active script without closing it. Leave the simulations of the sound level and the power on the screen. You will need them later as a comparison.

Create a new script (File/New menu). The activate the old script window and mark the entire text. To do this, jump to the text start (Ctrl+Home), keep shift depressed and additionally press the Ctrl+End combination. After these maneuvers, the entire text is shown inverse. Copy the text into the clipboard (Edit/Copy menu or Ctrl+C). Activate the new window and insert the text (Ins key). You now have two scripts with the same text. Save the new script under a suitable name (File/Save menu).

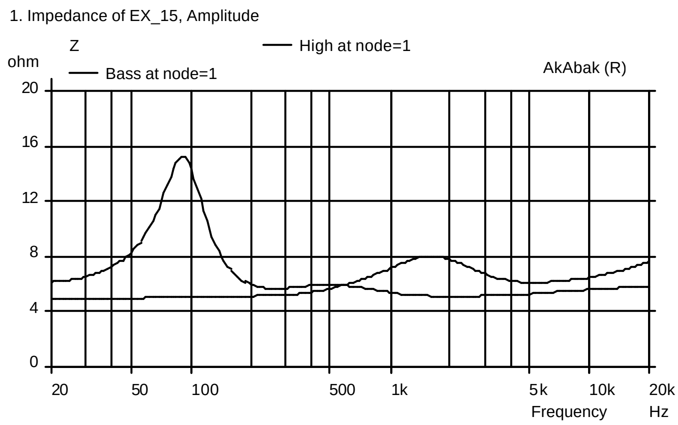


Fig. 8 Driving point impedance of woofer and tweeter

Driving-point impedance

For the synthesis of the passive network from the abstract filter element, you require the value of the terminating resistor of the network. The voltage transfer function of the synthesized network is exactly the same as the transfer function of the filter, if the latter resistor is real and frequency-independent. The driving point impedance of dynamic drivers, however, is anything but that. The are two ways of circumventing this problem.

The first way is to connect a dual network before the driver and the following elements, so that the driving point impedance is constant and real, or at least a good approximation. This method is often very complicated. For very steep filter characteristics and critical settings, such a network is unavoidable.

The second way can be used if the steepness of the filter curves is moderate and the impedance gradient is to some extent constant. In this case the value of the input impedance of the respective driver is estimated from the diagram of the input impedance.

First display the curve of the driving point impedance of woofer and tweeter. Activate the 'Inspect/Network impedance...' menu command. In the control dialog, assign System 'Bass' to Graph 1 and System 'High' to Graph 2 and start the simulation. Double click on the ordinate area to bring the graphs into the display (Fig. 8).

The frequency of the crossover network in the example of the exercise is 3kHz. The impedance curve of the woofer and tweeter in this frequency range is anything but constant. The woofer has a high voice-coil inductance and the resonance of the tweeter is very close to the cross over frequency. Before you can design the passive crossover network, you must first try to smooth the impedance curves in the frequency range around 3kHz.

Impedance compensation network

AkAbak has a tool for smoothing the impedance curve of electrodynamic drivers: 'Tools/Impedance Compensation' menu.

Before you can apply this tool, simulate the driving point impedance at the specific node where the compensating network shall be connected in parallel (Inspect/ Network impedance'). When the node is somewhere within your network then cut away all connections which form not part of the transducer (specify any temporary node number). Only the first impedance curve of a diagram is launched by the tool. The Bode format should be of 'Real and Imaginary' or 'Amplitude and Phase' where the real and imaginary format is best suited to observe the compensation process.

In the next step, open the 'Tools/ Impedance Compensation' dialog menu. The diagram of this dialog shows the amplitude response of the driving point impedance of the active diagram previously simulated, with real and imaginary or amplitude and phase components. In the 'Target Resistance' input box contains the value (5.5ohm) that the loudspeaker ought ultimately to have when connected in parallel with the compensation network.

Bass loudspeaker

In the bass loudspeaker, the impedance curve only has to be compensated in the upper frequency range. Therefore, please switch off the 'Equalize at resonance' switch and click on the 'Estimate' button. Two further curves now appear in the diagram. These curves show the real and imaginary components of the driving point impedance of the loudspeaker when connected in parallel with the compensation network. These curves shows the quality of compensation. In the frequency range around the cross over frequency of 3kHz, the real component should have the constant value of the target resistance RL=5.5ohm. The imaginary component should converge to zero.

With 'Estimate', you have first generated only the starting values for the optimization now following. Press the 'Optimize' button. You now see how the curves are smoothed. Stop the optimization using **Esc** and close the dialog with 'Copy and close'.

Place the script cursor before the `BassUnit` element and press Ins. The `Capacitor` element is inserted. The capacitance C corresponds to the value of Ce and the serial loss resistance Rs of this element corresponds to Rs=RL+Re (see compensation circuit in the chapter Tools/ Impedance Compensation. Re is not equal to the voice coil resistance here). The compensation network has the same node numbers as the `BassUnit` element, and is thus connected in parallel.

Tweeter

Repeat the procedure for the impedance of the tweeter, with the difference than in this case the impedance curve is also to be smoothed in the vicinity of the resonance.

Open the control dialog of the simulation of the network driving point impedance (Inspect/Network impedance menu), set the 'Mode' switch to 'Real and Imag.' and assign System 2 (tweeter) to Graph 1.

After the simulation, open the 'Tools/ Impedance Compensation' dialog again. Since in this case the impedance is also to be smoothed at resonance, its value has to be entered in the 'Resonance fs' input box. In the diagram, click on the maximum of the impedance hump of the curve. In the display panel, at the top left-hand side of the diagram, are displayed the abscissa and ordinate values of the marker position. Click on the upper value and keep the mouse key pressed. Now move the mouse cursor across the 'fs' input box and release the key. The value is entered in the box.

First press the 'estimate' button to obtain the starting value for the optimization and then press 'Optimize'. You will see how the curve is smoothed. When the process slows down, terminate the optimization with **Esc**. Repeat the optimization procedure with one of the 'Fix...' switches switched off. It is usually better to switch off only one of the 'Fix...' switches. When one of the 'Fix...' switches is on, its elements do not participate in the optimization.

When the values are stop improving, close the dialog with 'Copy and close' and insert the elements from clipboard before the `Speaker` element.

The script now looks like this:

```
...

System  'Woofer'
  |Impedance compensation RL=5.5ohm
```

```
    Capacitor  Node=1=0  C=28.274uF  Rs=6.056ohm
    BassUnit   Def='B1'    Node=1=0
      x=0  y=-15cm  z=0
    Filter
      fo=3kHz
      {b0=1;
       a2=1;  a1=1.414214;  a0=1; }

  System  'Tweeter'
      |Impedance compensation RL=5ohm
    Resistor   Node=1=2  R=5ohm
    Capacitor  Node=2=0  C=2.668uF  Rs=3.49ohm
    Capacitor  Node=2=0  C=16.94uF  Rs=8.168ohm  Ls=0.563mH
    Speaker   Def='S1'    Node=1=0
      x=0  y=0  z=0
    Filter
      fo=3kHz
      {b2=1;
       a2=1;  a1=1.414214;  a0=1; }
```

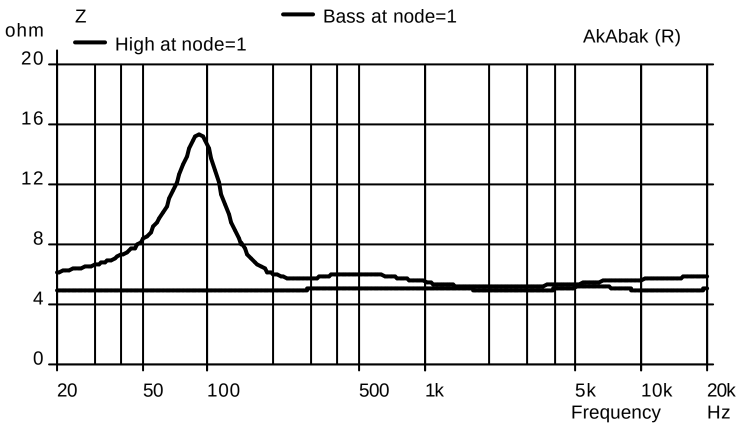


Fig. 9 Driving point impedance of woofer and tweeter, compensated in the upper frequency range

Fig. 9 shows the compensated impedance curves of woofer and tweeter. The components connected in parallel with the drivers only smooth the curves in the range of the cross over frequency 3kHz.

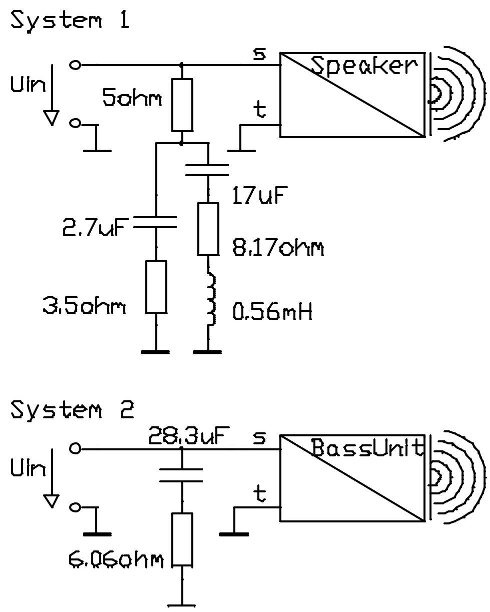


Fig. 10 Circuit diagram of the woofer and tweeter with impedance compensation

Fig. 10 shows the circuit diagram with the drivers and their compensation network. In the script, the components RL, Re, Ce, Lm, Rm and Cm are combined in `Capacitor` elements. The resistances and the inductance are assigned to the loss parameters of the capacitor. The individual elements could also be entered discretely by means of the elements `Resistor` and `Coil`.

16. Step: Synthesis of passive filter network

Script-file: Ex_16.aks

In the script, place the cursor somewhere within the filter element of the bass system. Start the synthesis using the 'Filter/LCR synthesis...' menu command. The control dialog of the synthesis appears. At the top you can see the transfer function again. The cursor is flashing in the 'RL-Loading resistor' input box. Here, enter the value of 5.5 ohm you have just determined. Since the box is already correctly filled in for the filter frequency, you only need to press Enter↵ (Network type 1 button). The left-hand list is filled in with the network; a coil, a capacitor and a resistor. The resistor is the loading resistance as entered.

In practice there are no ideal coils. All coils have at least one wire resistor in series with the inductance. In the case of coils having a core, the core losses are added. For crossover networks for loudspeakers this series resistance has an undesirable effect in that the impedance level of the entire network is very low. Each ohm costs power and falsifies the filter characteristics. Although capacitors also do not represent pure capacitances, their losses can usually be neglected in conventional crossover circuits.

AkAbak’s synthesis method can generate networks with dissipative inductances and non-dissipative capacitances. To do this enter in the input box 'Quality of coils', some values for the quality of the coils at 3kHz: For example Q=5, 10, etc. Start the synthesis again. As you see, the values have changed and the entry for the coil is followed by the value of the loss resistance that generates the entered quality. With the introduction of dissipative coils, the network damps within the transmission band.

In the 'Quality of coils' input box, enter a value of Q=15 and evaluate the network.

Press the 'Copy and close' button and enter the elements before the `BassUnit` element and after the `System` keyword.

Delete now the Filter element since otherwise two filters would be calculated. The transfer function and the loading resistor is documented in `SynthesisInfo`. To re-synthesize the nework move the script cursor in the lines of SynthesisInfo and issue 'Search/ Current element' (Ctrl+E) and the LCR-Synthesis dialog opens.

The parameters of this system now have the following appearance:

```
...

System  'Woofer'
  Coil       Node=1=2  L=0.434mH  Rs=0.55ohm
  Capacitor  Node=2=0  C=7.158uF
  SynthesisInfo
    Passive  FirstNode=1  RL=5.5ohm  QL=15.0  |Damping=0.82dB
    fo=3kHz  vo=1
    {b0=1;
     a2=1;  a1=1.414214;  a0=1;  }
    |Impedance compensation  RL=5.5ohm
  Capacitor  Node=2=0  C=28.274uF  Rs=6.056ohm
  BassUnit   Def='B1'    Node=2=0
    x=0  y=-15cm  z=0  HAngle=0  VAngle=0

System  'Tweeter'
...
```

The driving point voltage is at nodes 1 and 0 (ground). The coil leads to the next node (2). From there, one capacitor leads to ground (0) and the `BassUnit` is also connected there. Simulate the curve of the sound level and the acoustic power again. Compare the curves with the simulations in the previous script.

Synthesis of the tweeter part

Place the cursor of the script in the `Filter` element of the tweeter so that the synthesis dialog can read in the data. Open this dialog (Filter/LCR synthesis menu). Enter 5ohm for the loading resistor and Q=15 for the quality. Start the synthesis and close the dialog. Subsequently insert the elements into the script before the `Capacitor` and after the word `System`.

Adapting the node numbers

So far, the speaker element and the compensation network have been connected at the network input. To move them behind the passive high-pass and damping element, the nodes have to be renumbered. To do this, place the script cursor before the parameter `Node=1=2` of the `Resistor` element of the high-pass compensation network. Activate the 'Edit/Move Nodes' menu and enter into it an offset of 2. This shifts the node numbers by 2.

The script now looks like this:

```
...
System  'Woofer'
  Coil       Node=1=2  L=0.434mH  Rs=0.55ohm
  Capacitor  Node=2=0  C=7.158uF
  SynthesisInfo
    Passive  FirstNode=1  RL=5.5ohm  QL=15.0  |Damping=0.82dB
    fo=3kHz  vo=1
    {b0=1;
     a2=1;  a1=1.414214;  a0=1;  }
  BassUnit   Def='B1'    Node=0=2
    x=0  y=-15cm  z=0  HAngle=0  VAngle=0

System  'Tweeter'
  Capacitor  Node=1=2  C=7.874uF
  Coil       Node=2=0  L=0.395mH  Rs=0.5ohm
  SynthesisInfo
    Passive  FirstNode=1  RL=5ohm  QL=15.0  |Damping=0.11udB
    fo=3kHz  vo=1
    {b2=1;
     a2=1;  a1=1.414214;  a0=1;  }
     |Impedance compensation  RL=5.0ohm
  Resistor   Node=2=3  R=5ohm
  Capacitor  Node=3=0  C=2.668uF  Rs=3.49ohm
  Capacitor  Node=3=0  C=16.94uF  Rs=8.168ohm  Ls=0.563mH
  Speaker   Def='S1'    Node=2=0
    x=0  y=0  z=0  HAngle=0  VAngle=0
```

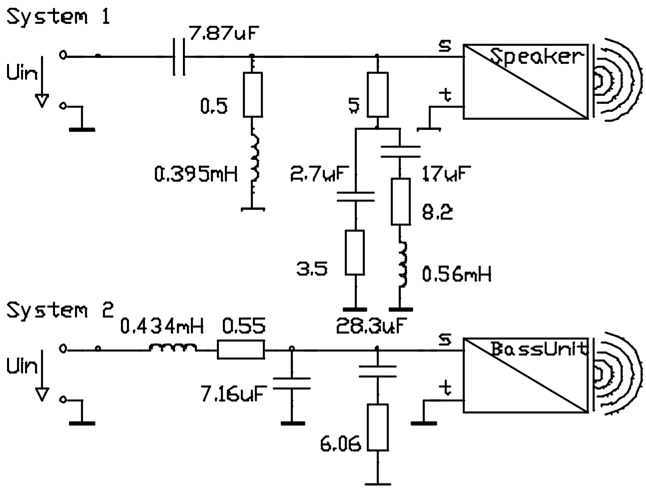


Fig. 11 Circuit diagram of the passive crossover and the impedance compensation

17. Step: Radiation environment

Script-file: Ex_17.aks

Taking into account the radiation environment

So far we have assumed that the tweeter and woofer are embedded in an infinite baffle. If this were true, the speaker would be embedded in a wall.

In practice, however, the radiation conditions are predominantly mixed. At high frequencies the enclosure acts as an infinite baffle for the radiators, at low frequencies, on the other hand, the sound is diffracted around the enclosure and free radiation conditions prevail. In the latter range the acoustic pressure (-6dB) and the power (-3dB) is reduced to half.

AkAbak can take into account the diffraction effects at the edges of the finite baffle and up to three reflectors: wall, room edge and room corner.

First enter the width and height of the baffle: `WEdge`=20cm `HEdge`=30cm. Enter these parameters after the position parameters of the tweeter and woofer.

Carry out the simulation again (Fig. 12). Now you see the reproduction of the enclosure when the loudspeaker is located somewhere freely in the room, remote from reflecting walls. In the bass range, the sound level has dropped by 6dB and the acoustic power by 3dB. At high frequencies, the waves reflected from the baffle edges cause a slight ripple in the sound level curve.

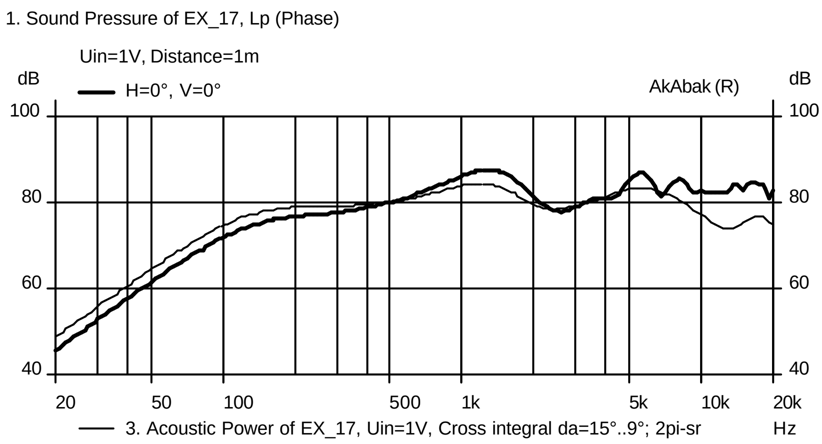


Fig. 12 Sound pressure and power level (thin line) inclusive edge diffraction

Reflections

In practice, the speaker will be located close to walls that reflect the sound. Reflective walls near to the loudspeaker are, so to speak, part of the sound source. The other walls are part of the listening room.

The positional data of the speaker with respect to the walls are saved at the start of the script, in the `Def_Reflector` definition. Each radiator that is expected to take part in the reflection is given the keyword `Reflection` as parameter. Please enter the keyword `Reflection` after the parameters `WEdge=20cm HEdge=30cm`.

Then open the Def_Reflector dialog (Def/ Def_Reflector... menu). Using this dialog you can shift your speaker in front of a wall or into an edge or corner and rotate it. Take the time to try out the various possibilities. The slider controls adjust the positional angle. The switches decide the basic arrangement. The input boxes determine the distance (perpendicular) of the origin of the baffle coordinates with respect to the particular wall. When you have entered something here, press the 'Repaint' button to draw the changes.

Leave the speaker positioned in front of a horizontal room edge ('Horizontal Edge'). The distance to the bottom ('to bottom') is 25cm and to the rear wall ('to top') is 100cm. Set both angles to zero degrees, so that the speaker is flat parallel with the bottom and with the rear wall. Press the 'Copy and close' button and insert the definition at the start of the script. The script now looks like this:

```
Def_Reflector  HorizEdge
  Bottom=25.0cm  Top=100.0cm HAngle=0  VAngle=0

Def_BassUnit   'B1'
  fs=65Hz  Vas=12.5L  Qms=1.25  Qes=0.6  Re=5.5ohm  Le=2.35mH
  dD=12.75cm  dD1=1.5cm  tD1=3.5cm  fp=1.7kHz
  Xmss=8mm  mb=0.95  Vb=8L  Qb/fo=0.1

Def_Speaker  'S1'
  fs=1.71kHz  Mms=0.33g  Qms=0.7  Qes=1.2  Re=5ohm  Le=140uH
  dD=2.8cm  tD1=7mm  |Convex dome
  t1=4mm  fp=8kHz  mb=1


System  'Woofer'
  Coil       Node=1=2  L=0.434mH  Rs=0.55ohm
  Capacitor  Node=2=0  C=7.158uF
  SynthesisInfo
    Passive  FirstNode=1  RL=5.5ohm  QL=15.0  |Damping=0.82dB
    fo=3kHz  vo=1
    {b0=1;
     a2=1;  a1=1.414214;  a0=1;  }
  BassUnit   Def='B1'   Node=0=2
    x=0  y=-15cm  z=0  HAngle=0  VAngle=0
    WEdge=20cm  HEdge=30cm  Reflection

System  'Tweeter'
  Capacitor  Node=1=2  C=7.874uF
  Coil       Node=2=0  L=0.395mH  Rs=0.5ohm
  SynthesisInfo
    Passive  FirstNode=1  RL=5ohm  QL=15.0  |Damping=0.11udB
    fo=3kHz  vo=1
    {b2=1;
     a2=1;  a1=1.414214;  a0=1;  }
    |Impedance compensation  RL=5.0ohm
  Resistor   Node=2=3  R=5ohm
  Capacitor  Node=3=0  C=2.668uF  Rs=3.49ohm
  Capacitor  Node=3=0  C=16.94uF  Rs=8.168ohm  Ls=0.563mH
  Speaker   Def='S1'   Node=2=0
    x=0  y=0  z=0  HAngle=0  VAngle=0
    WEdge=20cm  HEdge=30cm  Reflection
```

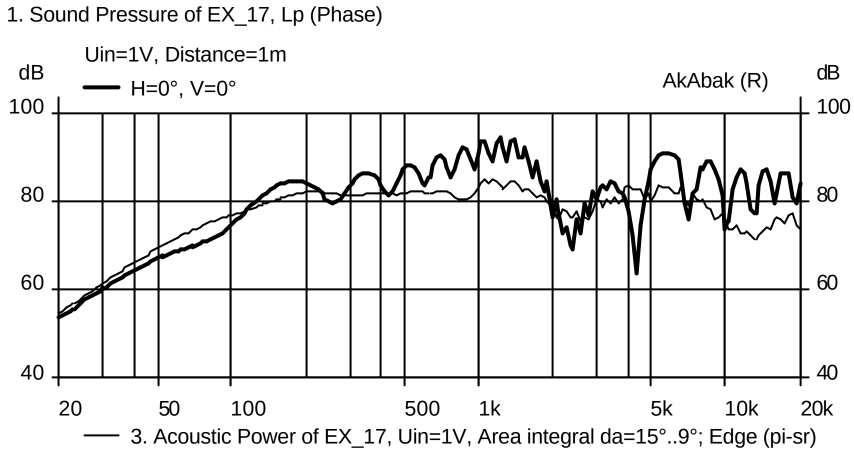
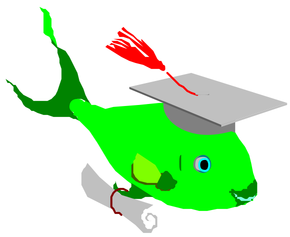


Fig. 13 Sound pressure level and power level (thin line) inclusive reflection of room edge

The reproduction of the sound pressure (Fig. 13) has a great deal of ripple. In reality, the up and down would be more damped, especially in the upper frequency range, since the walls never completely reflect the sound. But the reflector simulation gives a good impression of the effect of the mounting position on the reproduction. Since speakers like these are often used for public address systems, you can use the reflector simulation to find a suitable mounting position and, naturally, to avoid an unsuitable one.

In most cases it is not advisable to smooth the ripple in the sound-pressure curve due to reflections by means of the crossover. In this case, a better clue is given by the acoustic power, which is an important parameter in room acoustics (Fig. 13 thin line).



2. Design of a Bass-Speaker Enclosure

Script file: **Ex_21.aks**

This exercise shows you how to use the `Def_BassUnit` dialog, with its extensive design aids. We will design a high-pass-filtered reflex speaker with the following driver. The `Def_BassUnit` dialog takes into account a two-pole active high-pass filter as described in [Pan1].

We start with the following bass speaker

Resonance frequency	fs	30Hz
Equivalent volume to the compliance of the diaphragm suspension	Vas	130L
Mechanical quality factor	Qms	1
Electrical quality factor	Qes	0.5
D.C. resistance of the voice coil	Re	5ohm
Inductance of the voice coil (ExpoLe=0.6)	Le	2mH
Diaphragm diameter	dD	30cm
Diameter of the dust cap	dD1	10cm
Depth of the diaphragm cone	tD1	7cm
Frequency of diaphragm mass reduction	fp	750Hz
Maximum linear peak diaphragm excursion	Xms	5mm
Electrical load rating	Pelmax	300W

(measured without baffle and enclosure)

Step 1: Entering the loudspeaker parameters

Open the script 'BassUnit.aks' in the directory \AkAbak\Scripts\Examples' and place the script cursor in the `Def_BassUnit` definition. Activate the 'Def/ Def_BassUnit' menu command or just press the Ctrl+**B** combination.

As with exercise 1, the input dialog of `Def_BassUnit` opens. In this exercise, the design part of this dialog is treated in greater detail than the input functions.

The given parameters are typical for a high-performance bass loudspeaker. The parameters have been determined without a baffle or enclosure. To take into account the increased mass due to the increased air load in the installed state, enter a value of mb=0.95 in the 'Massload factor mb' input box.

Cable resistances or the like create a generator resistance, which is taken into account by an entry of 1ohm in the 'Generator resistance Rg' input box. The entries for the enclosure and filter are neglected for the time being. However, switch on the 'Vented' switch to tell the dialog that a vented cabinet is to be designed.

Step 2: Alignments catalogue

When you have entered all details of the driver and switched on the 'Vented' switch, press on the 'Alignments' button. Another dialog opens. The dialog has the header'Vented Cabinet Alignments' to indicate that the alignments are those of a reflex enclosure.

The intermediate results are repeated in the first line:

The resonance frequency, incl. air load, is fsb=28.5Hz.

The overall quality factor, incl. generator resistance and air load is Qtsbg=Qtr=0.395.

The second line contains the parameters that are listed by the alignments on the basis of the driver parameters.

Since a high-pass filtered system is to be computed, please switch on the 'high-pass filtered' switch.

Now press the 'Alignment list' button. In the top right-hand side you see a display, then values are entered in the list.

To limit the number of list entries, enter the following in the 'query' input boxes:

Vb:	<120L	(all alignments with an enclosure volume less than 120Liter)
f3:	<50Hz	(all alignments with a cut-off frequency less than 50Hz)
Lwmax	>100dB	(all alignments with a max. sound level greater than 100dB)
Qb/fo	<0.5	(all alignments with a loss factor less than 0.5)

Press the 'Alignment list' button again. You now see a subset of alignments that meet the above criteria.

Step 3: Diagram of the alignments

The individual alignments can be displayed in the diagram. To accomplish this, press the 'Diagram' button. The diagram shows the reproduction curve of the entire system, the transmission of the HP filter and the relative diaphragm excursion of the driver. The curves belong to the alignment currently selected in the list.

If your computer-screen is too small to tile all dialogs, activate the Alignments dialog again without closing the dialog. Move the Alignments dialog so far down the screen that the diagram is visible and also the Alignments list accessible. With this configuration, you can readily see the behavior of the various settings. Each time you select different alignments, the diagram is redrawn.

In the diagram dialog, enter values for the distances to the reflecting walls ('left wall', 'right wall', 'floor'). Switch on 'reflection' and press 'repaint'. You can now see how the power reproduction behaves in the presence of reflectors.

If you want, you can close the diagram dialog with the Alt+F4 keys (or double click with the mouse on the system menu of the diagram dialog).

Activate the alignments dialog again and look for the alignment with the greatest output power (Lwmax=110.5dB). At the top right-hand side of the dialog, you can see the alignments identifier: 'QB6 II m=1.353'. This is thus a 6th order Quasi-Butterworth alignment of 2nd class with the alignment factor m=1.353.

To obtain the data of this alignment, the given driver requires an enclosure with a volume of Vb=118.4L. The loss factor of the enclosure should be Qb/fo=0.164. Since the Helmholtz frequency is fb=35.6Hz, this results in a enclosure quality at fb of Qb=(Qb/fo)·fb=0.164/35.6=5.84. After the speaker has been completed, this value can be adjusted with insulating material in the enclosure interior. The enclosure quality is measured using the method from [Pan1].

The lower cut-off frequency of the reproducing of the acoustic output power is f3=38Hz. The maximum output power of this system is Lwmax=110.5dB. This values is for free radiation and for the maximum possible diaphragm excursion of the driver, as entered in the Def_BassUnit dialog (Xms=0.5cm). Lwmax is an absolute peak value. If it is exceeded, the diaphragm is deflected at any frequency within the transmission band further out than the entered value of Xms. The maximum operating mean sound power level is approx. 10dB less, i.e. Lw=100dB in this example.

Since a speaker rarely radiates freely, the sound pressure in the lower frequency range is increased by reflections, as could be seen in the diagram for the alignments. In the presence of reflecting walls, the maximum acoustic power, is thus some decibels higher and the cut-off frequency f3 moves towards lower frequencies. The exact frequency curve can be simulated with the aid of the definition `Def_Reflector`.

The electrical high-pass filter has a quality of Qe=0.632 at a filter pole frequency of fe=37.1Hz. This filter can only be implemented by means of an active electrical filter, as described in, for example, [Pan1].

Step 4.: Copying the alignments parameters into the Def_BassUnit-Dialog

Once the alignment described above has been selected, press the 'Copy alignment to Def_BassUnit Calculator' button. This activates the Def_BassUnit dialog and the table parameters are entered into the corresponding input boxes.

You can also calculate the reproduction and power parameters here. Press the 'Evaluate' or 'Diagram' button. You can now change some values to test, for example, the sensitivity of the bass system to one parameter or another. Change, for example, the Helmholtz resonance first to fb= 30Hz and then to fb=40Hz. The diagram clearly shows how sensitively the reproduction reacts to an increment of the Helmholtz frequency. It is also useful to carry out a test with a changed generator resistance Rg.

Step 5: Copying the parameters into the script

`Def_BassUnit` is a definition. Therefore also give a name in the 'Identification' box, e.g. 'B1'. Close the dialog with 'Copy to clipboard and close'. The parameters are now in the clipboard and can be inserted into the script.

```
Def_BassUnit   'B1'
  fs=30Hz  Vas=130L
  Qms=1  Qes=0.5  Re=5ohm  Le=2mH
  dD=30cm  dD1=10cm  tD1=7cm   |Cone
  fp=750Hz
  Xms=0.5cm  Rg=1ohm  mb=0.95
  Vb=118.37L  fb=35.6Hz  Qb/fo=0.16
  Qe=0.632  fe=37.1Hz
  |Performance of vented cabinet:
  |  fsb         Qtr       fD        f3
  |  28.5Hz      0.395     364.3Hz   38.1Hz
  |  Lwmax       Pelmax    UoRms     t60       Ripple
  |  110.5dB     246.0W    38.42V    93.88ms   0
```

Analyzing temperature-dependent voice coil resistance

Script file: **Ex_22.aks**

Finally, we will analyze the effect of the voice coil temperature on the curve for the acoustic pressure. Please open the script **Ex_22.aks**.

```
Def_Const
  { T   = 200;    |temperature of the voice coil wire
                  |in degrees Celsius
    Reo = 5;      |Voice coil resistance at T=20°C
                  |temperature coefficients of copper
    a   = 3.9e-3; |1st order
    b   = 0.6e-6; |2nd order  }

  |the parameter Qes cannot be used here, since Qes is dependent on Re:
  |  Qes=Re/Bl²*sqrt(Mms/Cms)

Def_BassUnit   'B1'
  fs=30Hz  Vas=130L
  Qms=1
  Bl=16.9Tm  | Driver constants instead of Qes
    |Taylor temperature curve of resistance Re
  Re={ Reo*(1 + a*(T - 20) + b*(T - 20)^2) }
  Le=2mH
```

```
    dD=30cm  dD1=10cm  tD1=7cm   |Cone
    fp=750Hz
    Xms=0.5cm  Rg=1ohm  mb=0.95
    Vb=118.37L  fb=35.6Hz  Qb/fo=0.16
    Qe=0.632  fe=37.1Hz

  System  'S1'
    BassUnit   Def='B1'
      Node=1=0
      x=0  y=0  z=0  HAngle=0  VAngle=0
```

At maximum loading of the loudspeaker, it is possible for the voice coil wire to reach temperatures of up to 200° Celsius. The temperature increase increases the voice coil resistance Re. The resistance changes. This script shows the use of the formula parser for analyzing the effects of the temperature on the reproduction. The temperature dependence of a conductor is given by:

$$R_{(T)} = Ro \cdot \left(1 + \alpha \cdot (T - 20) + \beta \cdot (T - 20)^2\right)$$

where
Ro: resistance at 20°C
T: temperature in degrees Celsius
α: temperature coefficient (copper: α=0.0039K^{-1})
β: temperature coefficient (copper: β=0.6·$10^{-6}K^{-2}$)

The temperature is entered centrally in the definition `Def_Const`. The formula itself in this case stands for the parameter Re. In this example, it is important that the parameter of the electrical quality Qes is not entered, since Qes also depends on Re, and thus on the temperature. Qes could, of course, also be weighted with the temperature formula. Instead, the driver constant B1 is entered here. Fig. 14 shows the sound level curve. First at a voice coil temperature of 20°C and then at a temperature of T=200°C.

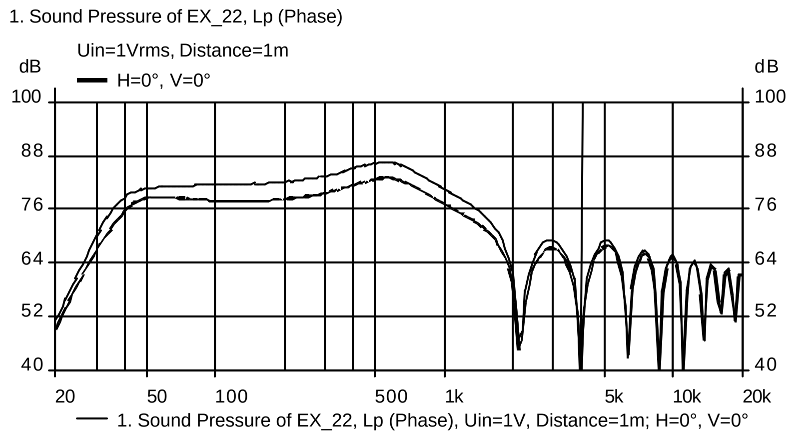


Fig. 14 Sound pressure level curve for the bass unit
The upper curve shows the temperature of the voice coil T=20°C
The lower curve shows the temperature of the voice coil T=200°C

3. Determining Dyn. Driver Parameter

This exercise demonstrates the use of the tool for determining the motor parameter for electrodynamic drivers. We start from the curve measured for amplitude+phase of the input impedance of a bass loudspeaker with a diaphragm diameter of dD=17cm. The measurement curve is stored as an ASCII file 'Bassimp.spe' in the '\AkAbak\Import' directory.

Step 1: Loading the measurement data

Open the dialog for determining the TS parameter of electrodynamic drivers. Menu: 'Tools/ Dyn. Driver Parameter'. Click on 'Import Script' and the Tools Import Script Editor opens. There is an example script prepared for the tools called 'ImpExTo.aki'. Please load this file (File/Open). Scroll to the 2nd example 'Example for dyn. driver parameter determination'. There are two Def_Import definitions. The first relates to a scalar measurement and the second to a vector measurement ('BassVec'). Move the script cursor into the lines of the latter and click Import... in the menu. The impedance curve is displayed in the Real and Imaginary format.

Step 2: Starting values

First the value of the voice coil resistance Re should be tested. The program uses the first curve point for this value. Often the suggested value is somewhat too great. Press the 'Estimate' button, AkAbak now looks for the resonance point and measures the curve to obtain the starting values for the subsequent optimization. There are cases in which the program cannot find the resonance, or it is necessary to use a different value. In this case enter the resonance frequency fs manually or use drag and drop of the marker panel.

Step 3: Optimizing the voice coil reactance

It is advisable first to adapt the voice-coil induction Le by the optimization process. To do this, switch the switch 'Fix parameter Le, ExpoLe' off and the other switches in the group on, and press 'Optimize'.

As you see, the red curve of the diagram slowly approaches the imported one. When the two curves lie one on top of the other and the diagram is only infrequently redrawn, the optimization is complete. Then press **Esc** again to terminate the process.

Step 4: Optimizing the voice coil resistance

The voice coil resistance becomes frequency dependent at high frequencies due to eddy currents and other effects. Fix now Le, ExpoLe again and free fre and ExpoRe. Press 'Optimize' and see how the fit process.

Step 5: Optimizing the qualities of Qms and Qes

To obtain an even better agreement between the model cure and the measured curve, the quality factors Qms and Qes will now be adapted. Free now only Qms and Qes. Start the optimization. You can now see how the model curve approaches the measured curve. When the process slows down, terminate the process with **Esc**.

Step 6: Optimizing the resonance frequency fs

This step is not always absolutely necessary, particularly if you have entered the resonance frequency manually. But in this case you can obtain an additional improvement with the resonance optimization. In the 'Fix parameter' group, switch 'Qms and Qes' on again and 'fs' off. Start the optimization. When you can see that there is no further improvement, stop the process with **Esc**.

Inserting the parameter in the script

Click on the 'Copy and close' button with the mouse. The dialog is closed and the parameter are in the clipboard. Generate a new script (File/New menu). Press Ins. The parameter are inserted into the script: All that is now lacking is the name of a driver definition and some more parameter.

```
Def_Driver
  fs=65Hz  Qms=5.131  Qes=0.405
  Re=2.59ohm  fre=691.5Hz  ExpoRe=0.833  Le=2.88mH  ExpoLe=0.641


Def_Driver   'Drv1'
  dD=17cm
  fs=29.98Hz  Qms=1.97  Qes=0.499
  Re=5ohm  fre=4.8kHz  ExpoRe=0.974  Le=1.5mH  ExpoLe=1.201
```

Only one parameter has to be determined yet: either the mass Mms, the compliance Cms or the volume Vas.

Determining the Mass Mms

Since a drive is a transmitter, not all the relevant parameter can be read from the driving point impedance. At least one parameter has to be determined from the transfer function. For the electrodynamic driver, it is easiest to determine the mass of the oscillating system, Mms. AkAbak has a tool for determining these parameter. Activate the menu 'Tools/ Mms, Cms, Vas parameter'. This dialog supports three different processes for calculating Mms. In the example, the 'Added mass' method will be used. For convenience first place the script cursor in the definition 'Drv1'. In this way necessary data are read in automatically.

Select 'Added mass' and enter the weight of 10g and the corresponding new resonance of 51Hz. Then press 'Evaluate'. The result is: Mms=16g. The weight is to be measured. Use the dyn. driver parameter dialog to measure the new resonance fs'. Now we have gathered all necessary parameter to start with simulation:

```
Def_Driver  'Drv1'
  dD=17cm Mms=16.016g
  fs=65Hz  Qms=5.131  Qes=0.405
  Re=2.59ohm  fre=691.5Hz  ExpoRe=0.833  Le=2.88mH  ExpoLe=0.641
```

4. Driver - Enclosure - Radiator

This two-part exercise illustrates how the compact bass unit element from Exercise 1 is fitted into the `Driver - Enclosure` and `Radiator` combination. In the second part of the exercise, a further bass loudspeaker is inserted.

Part 1: Driver, Enclosure and Radiator elements

Script file: **Ex_41.aks**

For reference, open the **Ex_41.aks** script. The part relevant to the bass channel is (Fig. 15):

```
Def_Driver   'B1'
  Meas_Dipole
  dD=12.75cm  dD1=1.5cm  tD1=3.5cm   |Cone
  fp=1.7kHz
  fs=65Hz  Vas=12.5L
  Qms=1.25  Qes=0.6  Re=5.5ohm  Le=2.35mH  ExpoLe=0.618
...
System  'Bass'
    |Lowpass: Damping: 0.821dB, Q=15, RL=5.5ohm
  Coil       Node=1=2  L=0.434mH  Rs=0.55ohm
  Capacitor  Node=2=0  C=7.158uF
    |Impedance compensation  RL=5.5ohm
  Capacitor  Node=2=0  C=28.274uF  Rs=6.056ohm
  Driver    Def='B1'    Node=2=0=3=4
  Radiator  Def='B1'    Node=3            |Diaphragm front
    x=0  y=-15cm  z=0  HAngle=0  VAngle=0
    WEdge=20cm  HEdge=30cm
  Enclosure    Node=4                     |Diaphragm rear
    Vb=8L  Qb/fo=0.1  Lb=5cm
...
```

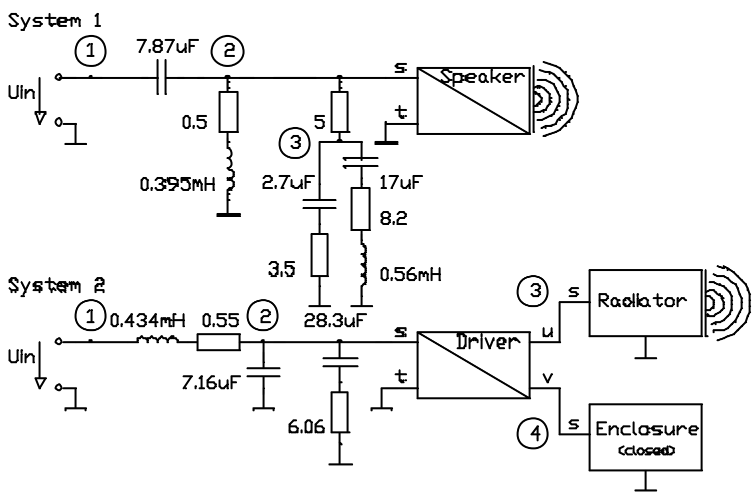


Fig. 15 Circuit diagram for part 1 of the exercise

Def_Driver

Instead of the `Def_BassUnit` definition, the definition part now contains `Def_Driver`. In `Def_Driver`, all parameter relating to the driver are listed. Instead of the mb factor which, in `Def_BassUnit`, takes into account the additional air load due to installation in the enclosure, the program is informed of the altered air load by the `Meas_Dipole` keyword. Since `Meas_Dipole` is the default setting, it could also be left out here (for further information see chapter Introduction/ Transducer). The other parameter are the same as in the `Def_BassUnit` definition.

Driver

The `Def_Driver` definition includes the network element `Driver`. `Driver` does not radiate. This element transforms electrical into acoustic energy. The two first poles of `Driver` are at the same point as the BassUnit element of Exercise 1. They are at the output of the frequency divider (node 2.0). The pole at node 3 represents the front of the diaphragm and the pole at node 4 the rear.

Radiator

The sound is radiated via this element. `Radiator` therefore also has a position parameter. The position is the locate of the diaphragm of the driver on the baffle. In addition, with the `Radiator` element, an impedance is also inserted into the network This impedance corresponds to the radiation impedance of the driver diaphragm. The `Radiator` element 'imports' the diaphragm dimensions from the `Def_Driver` definition via the reference `Def='B1'`. `Radiator` is connected to the diaphragm-front of the driver and is therefore at node 3. The `Radiator` element is always grounded. One of the poles is therefore connected to ground.

Enclosure

The diaphragm-rear of the driver radiates into a sealed enclosure, which is now assigned to node 4 as the `Enclosure` network element. The other pole is connected to ground. The parameter of `Enclosure` correspond to the details of the enclosure as given in the `Def_BassUnit` definition of exercise 1. In addition, the parameter Lb=5cm specifies a rectangular enclosure shape with a depth of 5cm.

Simulation

Carry out the simulation of the structure of the script **Ex_41.aks**.

For example:

Sum/Acoustic Pressure

Sum/Acoustical Power Level

Compare the curves with the simulation from Exercise 1, Ex_17.aks. There is virtually no difference between the two. The simple enclosure design of the example can be described more easily with the compact model of `Def_BassUnit`. However, with `Def_BassUnit` one quickly reaches the limits of the model if the acoustic structure is to be extended or certain parameter are to be investigated.

Part 2: Adding a second woofer

Script file: **Ex_42.aks**

We will now expand the design of part 1 with a loudspeaker in the bass range, so that the speaker can be more heavily loaded. The model of `Def_BassUnit` describes an isolated driver in the enclosure and cannot therefore be used. If several drivers radiate into the same enclosure, the bass unit has to be simulated with the `Driver`-, `Enclosure` and `Radiator` elements. (Circuit diagram Fig. 16). The bass part of the script (Ex_42.aks) is now as follows:

```
...
System  'Bass'
    |Damping: 0.821dB, Q=15,  RL=11ohm
  Coil       Node=1=2  L=0.868mH  Rs=1.09ohm
```

```
   Capacitor  Node=2=0  C=3.579uF
     |Impedance compensation  RL=11ohm
   Capacitor  Node=2=0  C=14.945uF  Rs=12.156ohm
   Driver    Def='B1'   Node=2=3=4=6
   Radiator  Def='B1'    Node=4            |Diaphragm front
     x=0  y=-15cm  z=0  HAngle=0  VAngle=0
     WEdge=20cm  HEdge=50cm
   Driver    Def='B1'    Node=3=0=5=6
   Radiator  Def='B1'    Node=5            |Diaphragm front
     x=0  y=-30cm  z=0  HAngle=0  VAngle=0
     WEdge=20cm  HEdge=50cm
   Enclosure    Node=6                     |Diaphragm rear
     Vb=16L  Qb/fo=0.1  Lb=21cm
...
```

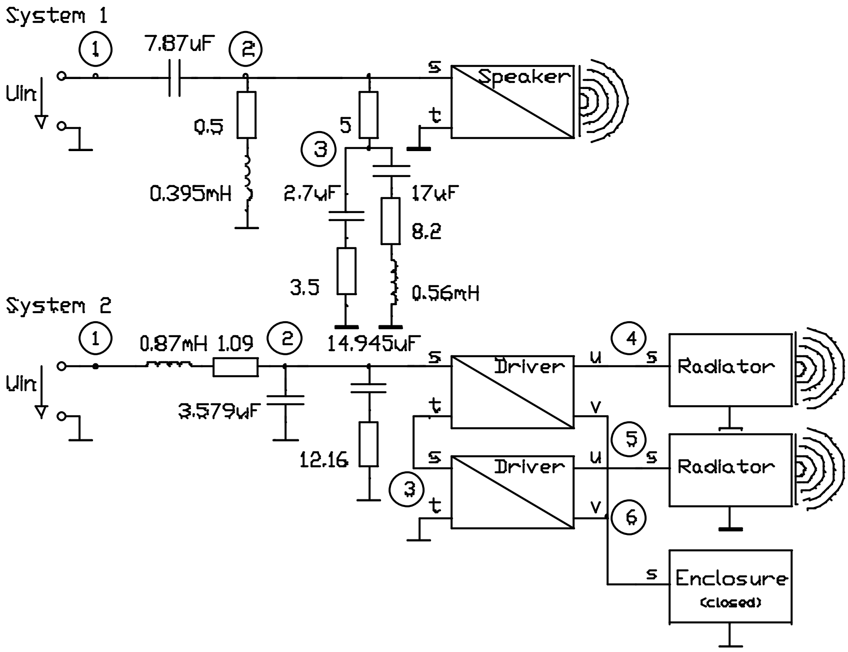


Fig. 16 Circuit diagram for Part 2 of the exercise (2 bass speakers)

Driver

There are now two `Driver` elements in the 'Bass' system. Both are the same, since they both refer to the same `Def_Driver` definition. The two drivers are in series at the electrical side. At the acoustic side the pole of the diaphragm reverse is connected to node 6, to which the enclosure is also connected.

Radiator

Since both drivers radiate with the diaphragm front, two `Radiator` elements are also present. At the baffle, the two woofers are arranged one above the other (y=-15cm and y=-30cm). The `HEdge=50cm` parameter is has been increased somewhat to take into account the new radiation envi ronment.

Crossover

The crossover has the same transfer function as that from Exercise 1, except that the values are adapted to a resistance of RL=11ohm.

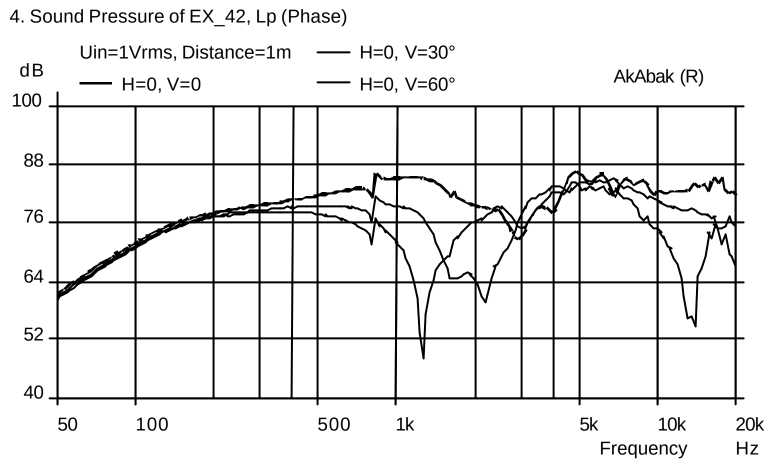


Fig. 17 Sound pressure level curve from three vertical listening angles in the enclosure with 2 bass loudspeakers, including sound diffraction and standing waves in the enclosure.

Sound pressure level curve

Fig. 17 shows the sound-level curve of the enclosure with two bass loudspeakers mounted one above the other. The strong ripple can be attributed to interferences caused by the vertical arrangement of the two bass speakers. The step in the curve at 800Hz is caused by the eigenfrequency of the enclosure.

Acoustic power

Fig. 18 shows the curve of acoustic power. For comparison, the curve here is illustrated with a bass loudspeaker. (Script: Ex_41.aks). The peak due to the conical diaphragm shape of the loudspeaker in the frequency range of 1kHz is damped slightly due to interference effects of two displaced radiators.

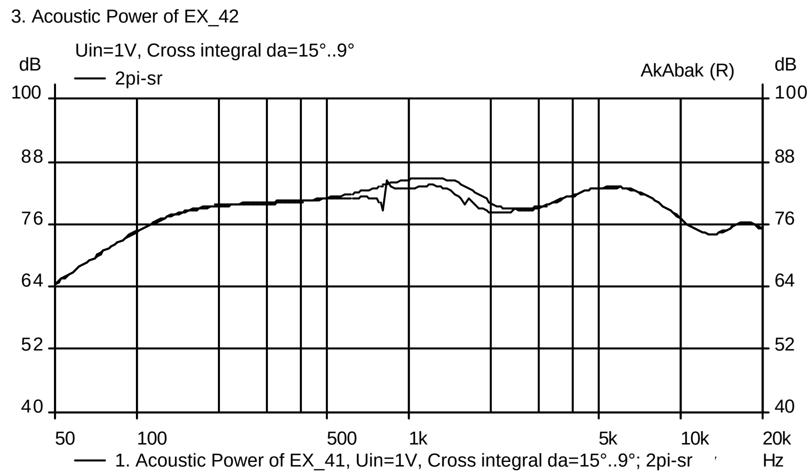


Fig. 18 Curve oft he acoustic output power
Enclosure with one bass loudspeaker and Enclosure with two bass loudspeakers (curve with trough at 800Hz)

5. Three-Way Speaker with Passive Crossover Network

This exercise shows the construction of a simple three-way loudspeaker with a passive crossover network. It is based on the design from exercise 4, part 2 and the bass speaker design from exercise 2. This exercise does not show how to design a loudspeaker that is a valid prototype from the point of view of its reproduction and power characteristic. It rather shows how to handle the important basic steps in a relatively complex structure. Finally, simulation of the extremes and some components of the sound pressure simulation are discussed.

Crossover for bass range and mid-range

Script-file: **Ex_51.aks**

The script is the same as in Exercise 4, part 2, except that the bass loudspeaker from Exercise 2 and two additional filter elements have been added. The filter elements form the crossover between the bass and mid-range. The cross over frequency is ft=300Hz. The two filter elements have been generated using the filter dialog (Filter/Filter Dialog menu). Both filters are of 4th order and have a Linkwitz-Riley characteristic. The high-pass filter of the mid-range unit damps in the pass range by 3db (`b4=0.707`) in order to adapt the level of the mid-range unit to the bass.

The low-pass filter of the two mid-range units and the tweeter are at first unchanged.

```
Def_BassUnit   'B1'
  fs=30Hz  Vas=130L
  Qms=1  Qes=0.5  Re=5ohm  Le=2mH
  dD=30cm  dD1=10cm  tD1=7cm   |Cone
  fp=750Hz
  Xmss=1cm  Rg=1ohm  mb=0.95
  Vb=118.37L  fb=35.6Hz  Qb/fo=0.16
  Qe=0.632  fe=37.1Hz
  |Performance vented cabinet:
  |  fsb      Qtr         fD          f3
  |  28.5Hz   0.395       364.3Hz     38.1Hz
  |  Lwmax    Pelmax      Uorms       t60         Ripple
  |  110.5dB  245.6W      38.53V      99.15ms     0


Def_Driver   'M1'
  dD=12.75cm  dD1=1.5cm  tD1=3.5cm   |Cone
  fp=1.7kHz
  fs=65Hz  Vas=12.5L
  Qms=1.25  Qes=0.6  Re=5.5ohm  Le=2.35mH


Def_Speaker  'S1'
  fs=1.71kHz  Mms=0.33g
  Qms=0.7  Qes=1.2  Re=5ohm  Le=140uH
  dD=2.8cm  tD1=7mm  |Dome
  t1=4mm  fp=8kHz
  mb=1


System   'Bass'
  BassUnit   Def='B1'
    Node=1=0
```

```
    x=0  y=-55cm  z=0  HAngle=0  VAngle=0
    WEdge=40cm  HEdge=90cm
    |low pass Linkwitz-Riley 4.Order
  Filter
    fo=300Hz
    {b0=1;
     a4=1;  a3=2.828427;  a2=4;  a1=2.828427;  a0=1;  }


System  'Mid'
    |Low pass: ft: 3kHz, Damping: 0.821dB, Q=15,  RL=11ohm
  Coil       Node=1=2  L=0.868mH  Rs=1.09ohm
  Capacitor  Node=2=0  C=3.579uF
    |Impedance compensation  RL=11ohm
  Capacitor  Node=2=0  C=14.945uF  Rs=12.156ohm
  Driver   'M11'  Def='M1'
    Node=2=3=4=6
  Radiator  'R1'  Def='M1'
    Node=4            |diaphragm front side
    x=0  y=-15cm  z=0  HAngle=0  VAngle=0
    WEdge=40cm  HEdge=90cm
  Driver   'M12'  Def='M1'
    Node=3=0=5=6
  Radiator  'R2'  Def='M1'
    Node=5            |diaphragm front side
    x=0  y=-30cm  z=0  HAngle=0  VAngle=0
    WEdge=40cm  HEdge=90cm
  Enclosure
    Node=6            |diaphragm rear side
    Vb=16L  Qb/fo=0.1  Lb=21cm
  off Filter          |low pass is switched off
    fo=3kHz
    {b0=1;
     a2=1;  a1=1.414214;  a0=1;  }
    |high pass Linkwitz-Riley 4.Order
  Filter
    fo=300Hz
    {b4=0.707;
     a4=1;  a3=2.828427;  a2=4;  a1=2.828427;  a0=1;  }


System  'High'
    |Q=15, RL=5ohm
  Capacitor  Node=1=2  C=7.874uF
  Coil       Node=2=0  L=0.395mH  Rs=0.5ohm
    |Impedance compensation  RL=5.0ohm
  Resistor   Node=2=3  R=5ohm
  Capacitor  Node=3=0  C=2.668uF  Rs=3.49ohm
  Capacitor  Node=3=0  C=16.94uF  Rs=8.168ohm  Ls=0.563mH
  Speaker   Def='S1'
    Node=0=2
    x=0  y=0  z=0  HAngle=0  VAngle=0
    WEdge=40cm  HEdge=90cm
  off  Filter        |Filter is switched off
    fo=3kHz
    {b2=1;
     a2=1;  a1=1.414214;  a0=1;  }
```

The filter elements listed in the script and disabled with `Off` are used for documentation, since they have already been implemented and installed as a passive version. The loudspeaker diaphragms are each located one above the other. The origin of the baffle coordinate system is congruent with the position of the tweeter. The values of diffraction-edge-parameter `WEdge=40cm HEdge=90cm` has been increased corresponding to the enlargement of the cabinet. Simulate the circuit:

Sum/Acoustic Pressure

Sum/Acoustical Power Level

Inspect/Driving Point Impedance (Z1 of System 'Bass')

The level curves are not particularly flat, but in this exercise that should not worry us. It can be seen from the impedance curve of the bass system that, in the frequency range around 300Hz, the impedance is approximately constant (at approx. 5ohm), so that impedance compensation of the bass speaker can be dispensed with below.

Passive synthesis of the bass-channel filter

Script file: **Ex_52.aks**

```
Def_BassUnit   'B1'
  fs=30Hz  Vas=130L
  Qms=1  Qes=0.5  Re=5ohm  Le=2mH
  dD=30cm  dD1=10cm  tD1=7cm   |Cone
  fp=750Hz
  Xmss=1cm  mb=0.95
  Vb=118.37L  fb=35.6Hz  Qb/fo=0.16
  Qe=0.632  fe=37.1Hz
...
System   'Bass'
      |Low pass: Damping: 2.559dB, RL=6.55ohm, Q=10
    Coil       Node=1=2  L=8.264mH  Rs=1.56ohm
    Capacitor  Node=2=0  C=0.119mF
    Coil       Node=2=3  L=3.639mH  Rs=0.69ohm
    Capacitor  Node=3=0  C=30.815uF
  BassUnit   Def='B1'
    Node=3=0
    x=0  y=-55cm  z=0  HAngle=0  VAngle=0
    WEdge=40cm  HEdge=90cm
    |low pass Linkwitz-Riley 4th order (switched off)
  off Filter
    fo=300Hz
    {b0=1;
     a4=1;  a3=2.828427;  a2=4;  a1=2.828427;  a0=1;  }
...
```

As discussed at the end of section 1, for the bass speaker, you do not need impedance compensation to convert the abstract filter element exactly into a passive network. The passive network was generated using the synthesis dialog (Filter/LCR Synthesis menu). The load resistance is RL=6.55ohm. With a coil quality of Q=10, the passive low-pass filter damps in the pass range by approx. 2.56dB. This is a very high level, since about the half of the input power is available at the terminals. For high-performance loudspeakers, it is better to implement the lower part of the crossover network with active filter circuits. In the `Def_BassUnit` definition the generator resistance Rg is set to zero ohm, since in this circuit the generator resistance is formed by the resistances in the crossover network. Simulate the sound level curve of this circuit again.

Passive synthesis of the mid-range-filter

Script file: **Ex_53.aks**

In this step, the last remaining filter element is transformed into a network with passive elements.

The problem here is that the mid-range unit contains two filter functions. The low pass separates at the frequency ft=3kHz and the high-pass at ft=300Hz. The overall band-pass characteristic is not symmetrical. In the passive filter version, the filter circuits cannot, unfortunately, be so simply cascaded as in the active case. The transfer function would be completely different from that determined by the filter elements.

The passive band-pass filter has to be synthesized in one piece. To do this you have to multiply out the two filter elements of the mid-range system. Mark the two filter elements completely (the keyword `off` is ignored) and activate the 'Filter/Product->Polynomial" menu command. The multiplied-out filter element is now in the clipboard and can be used in the script. In this case it is only used for intermediate calculation for the synthesis and can then be erased.:

```
Filter              | Low-pass of mid-range/tweeter crossover
  fo=3kHz
  {b0=1;
   a2=1;  a1=1.414214;  a0=1;  }
```

multiplied by:

```
Filter              | High-pass of bass/mid-range crossover
  fo=300Hz
  {b4=0.707;        | damping by 3dB
   a4=1;  a3=2.828427;  a2=4;  a1=2.828427;  a0=1;  }

Filter               | asymmetrical band-pass function
  fo=948.683Hz
  { b4=-70.700007;
    a6=10.000001;  a5=53.665649;  a4=144.000023;
    a3=108.225699;  a2=44.100003;  a1=9.391485;  a0=1;  }
```

Before the network can be generated, the impedance curve of the mid-range unit should be analyzed to see if it has a constant curve in the frequency range of the lower cross over frequency of ft=300Hz. If not, the synthesized network would deviate too far from the specification. In the present case, the deviation is not excessive and an impedance compensation at the resonance frequency could be dispensed with. We will nevertheless carry it out in this example.

Script text of the mid-range part

```
...
System  'Mid'
    |Band pass: ft=948.68Hz, RL=11ohm, Q=20
  Capacitor  Node=1=2  C=37.776uF
  Coil       Node=2=0  L=3.338mH  Rs=0.99ohm
  Capacitor  Node=2=3  C=55.593uF
  Coil       Node=3=0  L=25.128mH  Rs=7.49ohm
  Coil       Node=3=4  L=0.816mH  Rs=0.24ohm
  Capacitor  Node=4=0  C=2.924uF
  Resistor   Node=4=5  R=4.41ohm
  Resistor   Node=5=0  R=16.46ohm
    |Impedance compensation  RL=11ohm
  Resistor   Node=5=6  R=11.0ohm
  Capacitor  Node=6=0  C=16.365uF  Rs=1.299ohm
```

```
  Capacitor  Node=6=0  C=0.165mF  Rs=6.505ohm  Ls=14.842mH
  Driver   'M11'  Def='M1'
    Node=5=7=8=10
  Radiator  'R1'  Def='M1'
    Node=8            |diaphragm front side
    x=0  y=-15cm  z=0  HAngle=0  VAngle=0
    WEdge=40cm  HEdge=90cm
  Driver    'M12'  Def='M1'
    Node=7=0=9=10
  Radiator  'R2'  Def='M1'
    Node=9            |diaphragm front side
    x=0  y=-30cm  z=0  HAngle=0  VAngle=0
    WEdge=40cm  HEdge=90cm
  Enclosure
    Node=10            |diaphragm rear side
    Vb=16L  Qb/fo=0.1  Lb=21cm
      |Band-pass filter.
      |formed using the 'Product->Polynomial' function from
      | the two following filters.
      |(switched off, since already implemented by passive elements)
  off Filter
    fo=948.683Hz
    {b4=70.700007;
     a6=10.000001;  a5=53.665649;  a4=144.000023;
     a3=108.225699;  a2=44.100003;  a1=9.391485;  a0=1;  }
    | Original low-pass filter (switched-off)
  off Filter
    fo=3kHz
    {b0=1;
     a2=1;  a1=1.414214;  a0=1;  }
    | Original high-pass filter (switched off)
  off Filter
    fo=300Hz
    {b4=0.707;    | damping by 3dB
     a4=1;  a3=2.828427;  a2=4;  a1=2.828427;  a0=1;  }

System  'High'
    |Q=15, RL=5ohm
  Capacitor  Node=1=2  C=7.874uF
  Coil       Node=2=0  L=0.395mH  Rs=0.5ohm
    |Impedance compensation  RL=5.0ohm
  Resistor   Node=2=3  R=5ohm
  Capacitor  Node=3=0  C=2.668uF  Rs=3.49ohm
  Capacitor  Node=3=0  C=16.94uF  Rs=8.168ohm  Ls=0.563mH
  Speaker   Def='S1'
    Node=0=2
    x=0  y=0  z=0  HAngle=0  VAngle=0
    WEdge=40cm  HEdge=90cm
  off  Filter        |Filter is switched off
    fo=3kHz
    {b2=1;
     a2=1;  a1=1.414214;  a0=1;  }
```

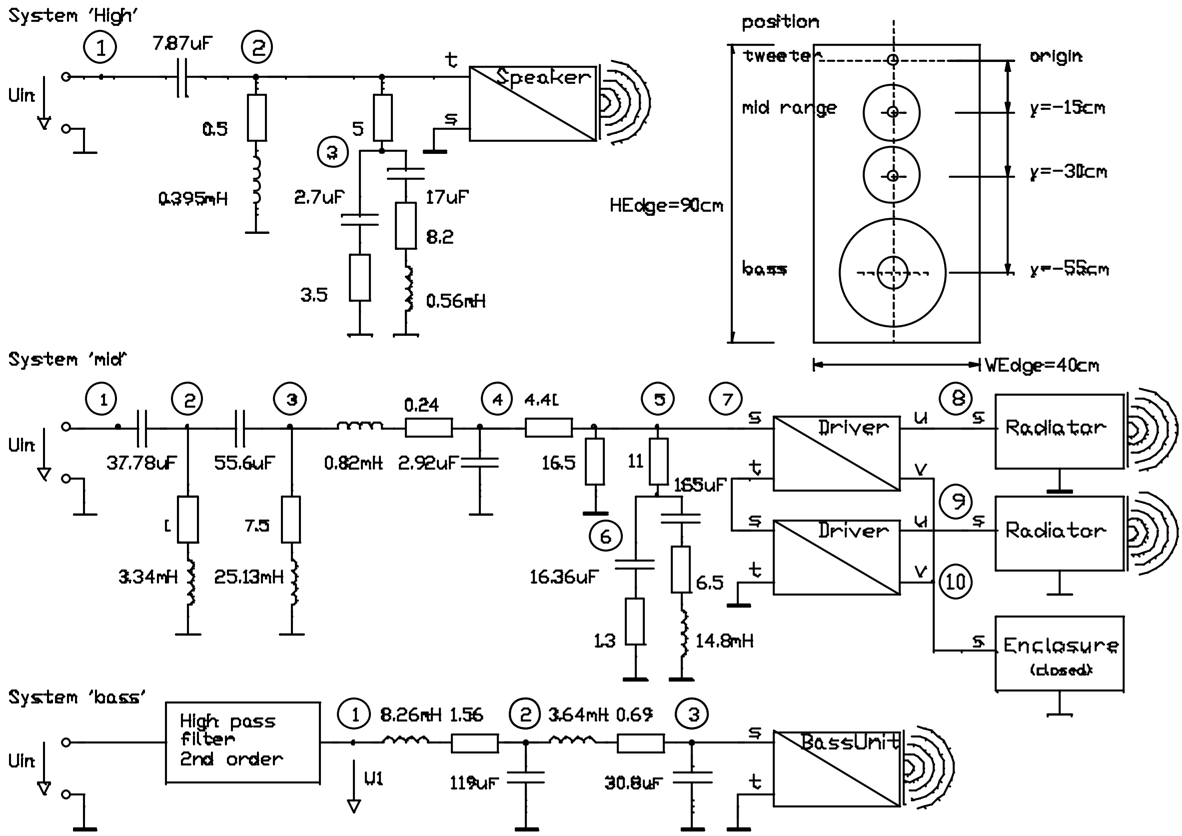


Fig. 19 Circuit of loudspeaker system

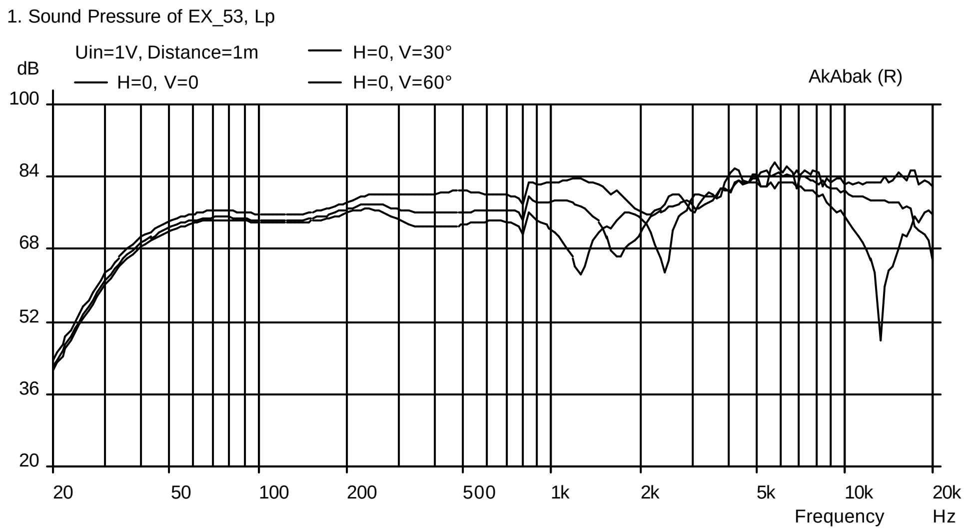


Fig. 20 Sound pressure level under three different vertical listening angles

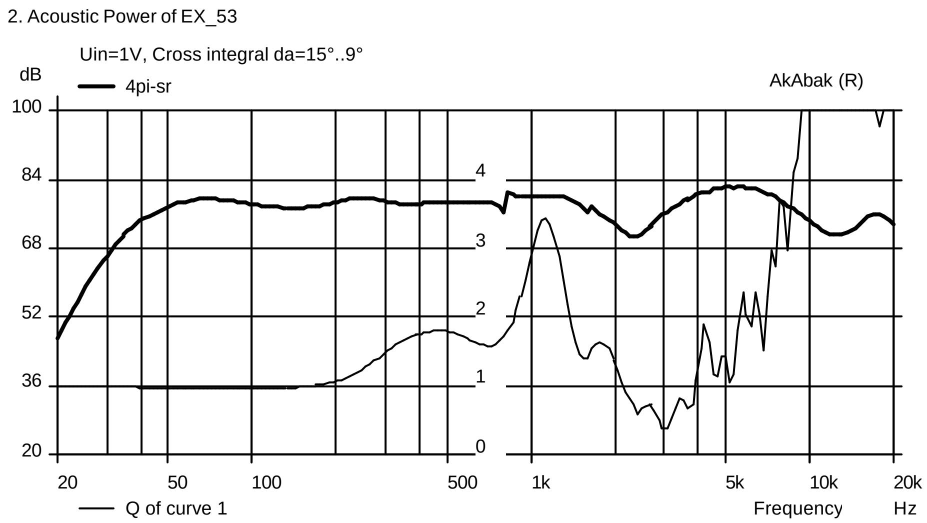


Fig. 21 Acoustical power level and directivity factor Q

Circuit diagram

Fig. 19 shows the circuit diagram of the loudspeaker unit. For the passive filter networks to have the desired transmission properties, their inputs should be connected to an ideal voltage source.

Do not forget the two-pole high-pass filter of the bass part. It can only be implemented by means of an active filter. This filter is located before the power amplifier to whose output the inputs of the crossover network are connected. You could also include this filter in the circuit, implementing it with, for example, operational amplifiers or transistors, including the power amplifier.

Sound pressure level

Fig. 20 shows the sound-level curve from three vertical listening angles. Fig. 21 shows the curve of the acoustic power. The deviations of the axial sound pressure are approx. +4dB. At other listening angles in the vertical plane there are intense troughs. The reproduction could be improved here by using mid-range speakers with a smaller diaphragm and moving the drivers closer together on the baffle. There are some tricks here regarding the arrangement and choice of the crossover network, which, under certain conditions generate a broader radiation characteristic. This will not be discussed here, however.

At very low frequencies, a slight peak is apparent. This rise can be explained by the increased generator resistance due to the loss resistances in the low pass of the bass crossover. In the design of the bass unit (Exercise 2), a generator resistance of Rg=1ohm was taken into account. As is clear from Fig. 19, the bass loudspeaker is exposed to more than twice that resistance. With passive crossover networks and especially if high losses are to be expected, the bass box should be designed with a large generator resistance.

Another interesting point is the reproduction behavior of the speaker in the presence of reflectors (not shown here). It may be possible to adjust the strong bass reproduction by reducing the quality Qe of the electrical high-pass filter in the bass part.

In the last stage, the poling of the tweeter was reversed, with the effect that the sound-pressure curve is somewhat smoother at the upper frequency band.

Extreme values and power density

As we saw in Exercise 2, the maximum loading capacity of the bass loudspeaker due to the maximum diaphragm excursion (Xms=5mm) is approx. Uorms=38.53V. This voltage is not an operating value, since the value must not be exceeded. The maximum operating value is regarded as a value 10dB low, i.e. approx. $Uorms_{mean}$=12.2V. For both voltages, the loading factors should now be calculated.

Extreme values (Uin=38.53V$_{rms}$)

For the script **Ex_53.aks**, start the simulation menu: 'Sum/Extreme values'. In the driving-point voltage box, enter 38.53V$_{rms}$. Switch on all switches.

Extreme calculation for 54V:

```
Rms  input voltage Uin:  38.53V
Frequency range:         20Hz ... 20kHz
Frequency points/octave: 20.1

Max. diaphragm excursion (peak) -------------------------
1. Bass      3->0   BassUnit B1           Xmax: 4.671mm
2. Mid range 18->20 Driver (diaph) M1     Xmax: 0.265mm
2. Mid range 19->20 Driver (diaph) M1     Xmax: 0.265mm
3. High      0->2   Speaker S1            Xmax: 0.176mm

Max. current in coils and transformers (peak) -----------
1. Bass      1->2   L=8.26mH              Imax: 7.55A
1. Bass      2->3   L=3.64mH              Imax: 6.597A
2. Mid range 12->0  L=3.34mH              Imax: 6.066A
2. Mid range 13->0  L=25.13mH             Imax: 0.822A
2. Mid range 13->14 L=0.82mH              Imax: 6.011A
3. High      2->0   L=0.4mH               Imax: 5.24A

Max. voltage across capacitors (peak) -------------------
1. Bass      2->0   C=0.12mF              Umax: 48.311V
1. Bass      3->0   C=30.82uF             Umax: 48.178V
2. Mid range 11->12 C=37.78uF             Umax: 75.042V
2. Mid range 12->13 C=55.59uF             Umax: 33.501V
2. Mid range 14->0  C=2.92uF              Umax: 63.86V
2. Mid range 16->0  C=16.37uF             Umax: 33.719V
2. Mid range 16->0  C=0.17mF              Umax: 33.719V
3. High      1->2   C=7.87uF              Umax: 69.212V
3. High      3->0   C=2.67uF              Umax: 47.276V
3. High      3->0   C=16.94uF             Umax: 47.276V

Max. power in resistors and voice coils -----------------
1. Bass      3->0   BassUnit B1           Pmax: 130.789W
2. Mid range 14->15 R=4.41ohm             Pmax: 76.071W
2. Mid range 15->0  R=16.46ohm            Pmax: 46.538W
2. Mid range 15->16 R=11ohm               Pmax: 35.836W
2. Mid range 15->17 Driver (in) M1        Pmax: 22.88W
2. Mid range 17->0  Driver (in) M1        Pmax: 22.88W
3. High      2->3   R=5ohm                Pmax: 99.105W
3. High      0->2   Speaker S1            Pmax: 261.997W
```

The calculation shows the extreme values with an input voltage that, with the bass loudspeaker, leads to the maximum possible diaphragm excursion.

It should also be checked whether the mid-range and tweeter used can handle the values of Xmax.

Capacitors are very sensitive to peak load. If the field intensity in the dielectric is too high, the capacitor is destroyed. The maximum voltage loading capacity should be at least the values from the table. It also doesn't do any harm to have something in reserve.

As can be seen from the table, the voltages in the network may be much greater than the driving-point voltage. That is not an arithmetical error, but is related to the quality of the resonant circuits. There are filter characteristics with very high qualities that generate peaks of 10...20 times the voltage in some branches of the network.

In the coils, there are very high peak currents. From a certain current density, coils with cores are saturated and the inductance decreases. Although the coil is not destroyed here, other components may be indirectly forced into sympathy. In any case, intense distortions occur. Coils without cores, on the other hand, have a current-independent inductance.

Power density (Uin=12.2V$_{rms}$)

For loudspeaker design, a maximum operating driving point voltage should be determined. This voltage should be chose so that the components still operate with linear characteristics, even at a excess range of 10dB, and are not overloaded. A driving-point voltage of Uin=12.2V$_{rms}$ is approx. 10dB below the absolute voltage of Uin=38.53V$_{rms}$ (see above). The important features of the maximum operating point are the power values in resistors and voice coils, since, in contrast to a peak voltage, an operating voltage is permanently present. Start the simulation menu: 'Sum/Power density...'. The measure of noise is the rms-voltage within a frequency range. The result of this simulation is the integral power within the frequency range in resistive electrical elements. Optional the noise shape can be choosen.

Take the IEC 268 curve, which simulates common music signal.

```
Rms  input voltage Uin:  12.2V
Frequency range:         20Hz ... 20kHz
Frequency points:        200

Power in resistors and voice coils ----------------------

(Power distribution norm IEC 268)
1. Bass      3->0   BassUnit B1           P: 2.115W
2. Mid range 14->15 R=4.41ohm             P: 1.962W
2. Mid range 15->0  R=16.46ohm            P: 1.179W
2. Mid range 15->16 R=11ohm               P: 0.649W
2. Mid range 15->17 Driver (in) M1        P: 0.469W
2. Mid range 17->0  Driver (in) M1        P: 0.469W
3. High      2->3   R=5ohm                P: 0.174W
3. High      0->2   Speaker S1            P: 1.404W
```

Interesting is the power dissipation in resistor R=4.41ohm and R=16.46ohm of P=1.9W and P=1.18W. This power has to be converted to heat.

Time response of the sound-pressure curve

To calculate the time response we need an already simulated spectrum. Activate the diagram with the sound pressure levels, click on the legend of graph 1 (on-axis) and issue 'Calc/ Spectrum to time'. In the control dialog leave the settings and press 'Ok'. In the new generated diagram the discrete inverse Fourier transform of the spectrum is displayed. To zoom it out, move the mouse in front of the main peak, press the button and pull the inverted drawn rectangle across the peak. Then you should get a picture like Fig. 22.

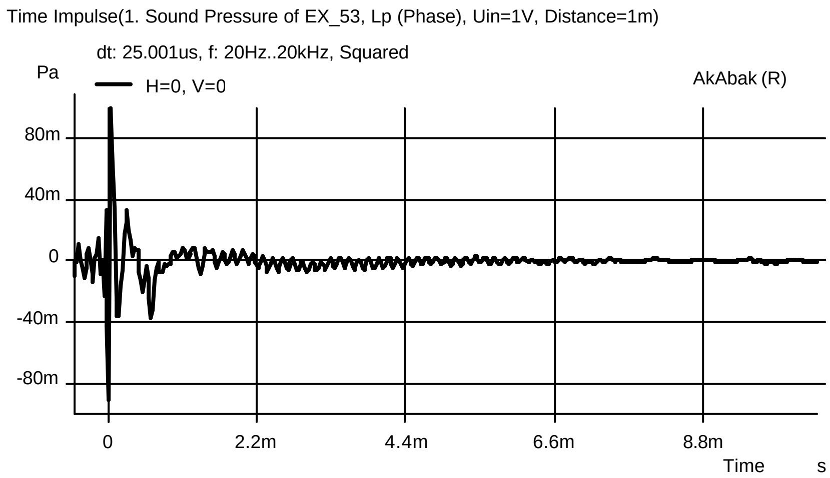


Fig. 22 Time response of sound pressure level

The fine rippling is produced by the FFT due to sharp windowing. Since the program re-calculates the phase to the origin of the baffle there is no time lag due to the distance. An alternative way to look to the time response is to calculate the ETC. Fig. 23 displays the energy density on-axis which demonstrates that the acoustical energy arrives approximately at the same time.

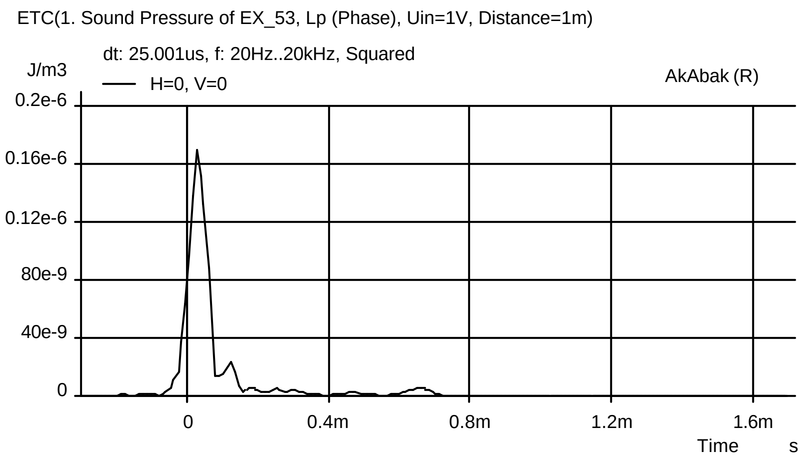


Fig. 23 ETC of the sound pressure on-axis

Fig. 24 shows the same calculation but derived from a sound pressure spectrum under a vertical listening angle of -60° (down wards). Since all drivers are mounted in the vertical line time traveling of the waves lead to a time smearing effect which can be seen in Fig. 24.

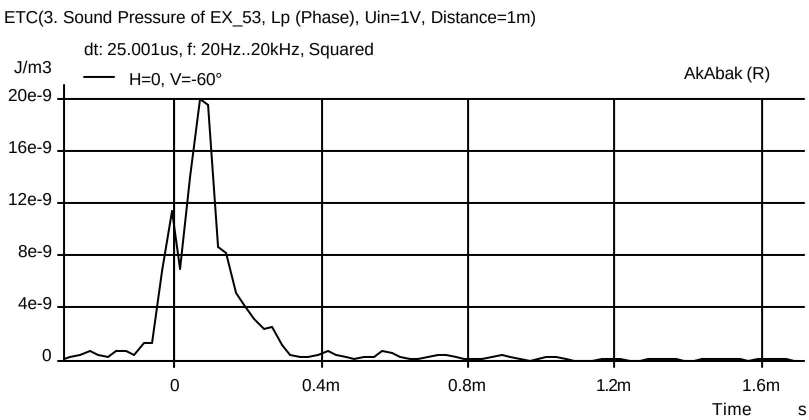


Fig. 24 ETC of sound pressure with listening angle 60° down

Inserting a generator resistance

Script file: Ex_54.aks

In the design of passive crossover networks, it is assumed that the network components are decoupled from one another at the driving point by an ideal voltage source. This idealization corresponds to the subdivision of the script into individual system. In practice there is a line resistance of lead cables and transition resistance of plug connections between the voltage source (power amplifier) and the input of the crossover network. To test the effect of this generator resistance, all the components have to be located in one system of the script.

Load the **Ex_54.aks** script and simulate the curve of the sound-pressure level again. Fig. 25 shows the effect of a generator resistance of 5ohm on the reproduction. Apart from the general damping, there is hardly any distortion of the level curve here. This can be attributed to the dissipative coils in the crossover network. The reproduction curve reacts the more sensitively to the generator resistance the smaller the losses in the components and the higher the pole qualities of the filters.

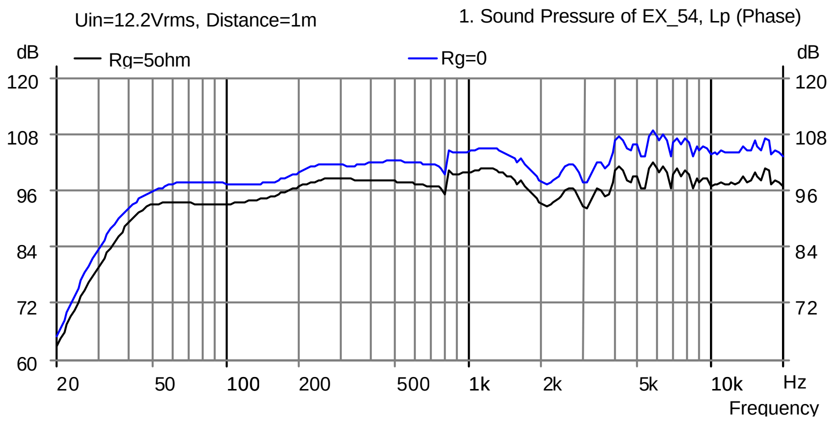


Fig. 25 Effect of a generator resistance on the sound-level curve
Top curve: Without generator resistance. Bottom curve: With generator resistance Rg=5ohm

6. Synthesis with Active Components

This exercise illustrates the conversion of a transfer function into an active filter circuit with operational amplifiers and transistors.

Generating and decomposing the transfer function

Script file: **Ex_61.aks**

Generate a new script and open the filter dialog (Filter menu). For the filter frequency, enter fo=1kHz. Push the 'Standard lowpass functions' button to generate a low-pass function. In the sub-dialog that opens, enter 3 for the order and switch to 'Butterworth'. Push 'OK' and then copy the coefficients into the large input box ('Copy to 1'). The activate 'Copy to clipboard and close' and insert the filter element into the script (Ins).

If you want to simulate this filter function, enter, before the filter element, the keyword `System` and start 'Inspect/Voltage'. Select the 1st system.

```
|3rd order Butterworth low-pass system
|generated with filter dialog

Filter
  fo=1kHz
  {b0=1;
   a3=1;  a2=2;  a1=2;  a0=1;  }
```

This transfer function will be implemented in the exercise by means of a Sallen-Key circuit. For this purpose it will be decomposed into 1st and 2nd order blocks. Locate the script cursor in the text of the filter element and activate the 'Filter/Polynomial-> Product' command. After the calculation has finished the result is in the clipboard. Insert it into the script. To simulate the filter correctly, enter the `System` keyword before the new filters as well. If you want to check the analysis, start 'Inspect/Voltage'. Assign the first graph to System 1 and the second graph to System 2.

```
    |Decomposed transfer function
System  'S1'
  Filter
    fo=1kHz   |  fpD=1kHz  QD=1
    {b0=1;
     a2=1;  a1=1;  a0=1;  }
  Filter
    fo=1kHz   |  fpD=1kHz
    {b0=1;
     a1=1;  a0=1;  }
```

The analysis routine 'Polynomial -> Product' outputs as commentary the pole frequencies **fpD** and pole qualities **QD**. These parameter are important for many methods for synthesizing active filter circuits.

Using the Active Filter Synthesis tool

For transfer functions of first and second order a set of common filter circuits can be synthesized with the help of the tool 'Filter/ Active Synthesis'. Each of the decomposed transfer function has to be realized separately. Move therefore the script cursor in the lines of the first Filter and issue 'Filter/ Active Synthesis'. Click on the tab 'Sallen+Key **MQ**'. Enter in the fields 'C2' the value 15nF and in 'R5' the value 10kohm. The results are displayed immediately. For convenience try the spin function of the number-fields. Press Ctrl+up/down than the values spin within the E12 row.

In the upper part of the dialog the amplification vo=1 is displayed. Since this kind of circuit amplificates in the pass band a voltage divider is used in front of the circuit (R11, R12). The list of possible Def_OpAmp definitions is empty because our script has no definition yet. Leave this entry blank. In this case the simulator calculates with OpAmp's with default values of vo=10^6 and Rg=1Ω. Click Copy and Close. Then the circuit is formatted and copied into the clipboard. Insert the line `System 'Syn'` behind the system's with the Filter elements and insert the clipboard. The script should look like this:

```
...
System  'Syn'
  Resistor  'R11'  Node=1=2  R=21.22kohm
  Resistor  'R12'  Node=2=0  R=21.22kohm
  Capacitor  'C2'  Node=2=5  C=15nF
  Resistor  'R3'  Node=2=3  R=10.61kohm
  Capacitor  'C4'  Node=3=0  C=15nF
  OpAmp  'Op1'  Def=''  Node=3=4=5
  Resistor  'R5'  Node=4=0  R=10kohm
  Resistor  'R6'  Node=4=5  R=10kohm
  SynthesisInfo
    Active  FirstNode=1  Def=''
    Sys={LPSKMQ 15nF 15nF 10kohm}
    fo=1.0kHz  vo=1
    {b0=1;
     a2=1;  a1=1;  a0=1;  }
```

Simulate the response at the output stage of OpAmp Op1 (node 5) with Inspect/ Voltage - Level. To compare the curve with that of the Filter elements choose the for the first graph 'S1-U1' and for the second 'Syn-5->0 OpAmp (out) Op1'. The level has a slightly overshot at the filter frequency. To obtain a BU-response of 3rd order we have to synthesis also the transfer function of first order.

Therefore move the script cursor into the lines of the second Filter element (1st order) and repeat the process. In this case we have to take into account that the two circuits shall be chained. Therefore enter in 'First node' the number 5 (In case you have forgotten this and need to move the numbers use 'Search/ Move nodes').

```
...
  Resistor  'R6'  Node=5=6  R=10.61kohm
  Capacitor  'C6'  Node=6=0  C=15nF
  OpAmp  'Op6'  Def=''  Node=6=8=8
  SynthesisInfo
    Active  FirstNode=5  Def=''
    Sys={15nF 0}
    fo=1.0kHz  vo=1
    {b0=1;
     a1=1;  a0=1;  }
```

When we now simulate with Inspect/ Voltage at 'Syn, 8->0, OpAmp (out) Op6' the curves are identical. The SynthesisInfo term has not only the purpose to document the filter function but it is also a means to re-synthesize the circuit. To try this, move the cursor into the lines of SynthesisInfo and issue 'Search/ Current Element' (Ctrl+E). Then the associate dialog for synthesis of active circuits opens. The same mechanics works also with the filter dialog. In this case issue 'Filter/ Filter Dialog' (Ctrl+F).

The formulae for this kind of circuit are:

$$C1 \cdot R1 = \frac{1}{2\pi \cdot fpD} \quad \text{and} \quad C2 \cdot R2 = \frac{1}{2\pi \cdot fpD} \quad \text{(pole frequencies)}$$

$$Ra / Rb = 2 - 1/QD \quad \text{(pole quality factors)}$$

This type of Sallen-Key circuit amplifies in the pass band 3 - 1/QD. Calculation formula for Sallen-Key filters are to be found, for example, in [Hor].

Using the formula parser

Script file: **Ex_63.aks**

The **Ex_63.aks** script has a similar structure as the **Ex_62.aks** script, except that the numerical values have been replaced by formulae. The filter frequency and the quality of the two-pole filter block are entered centrally in the `Def_Const` definition. To simplify the example the voltage divider at the input of the first block has been omitted here. Therefore the pass band amplification depends on the quality factor QD.

```
  |Centralized entry of the filter parameter
Def_Const
  { fo = 1000;    |filter frequency in Hz
    QD = 1;       |quality of the 2nd order filter block
    C1 = 15e-9;   |entry of a capacitor value in farad
  }
Def_OpAmp  'Op1' |TL071
    vo=200e3  Rg=250ohm
    ft=3.0MHz  Rdiff=1Tohm  Rin=1Tohm

  |for comparison, the filter with filter elements
System  'Abstract'
  Filter
    fo={ fo; }
    {b0=1;
     a2=1;  a1=1/QD;  a0=1;  }
  Filter
    fo={ fo; }
    {b0=1;
     a1=1;  a0=1;  }

  |Sallen-Key filter circuit
  |The values are calculated with the formula parser.
System  'OpAmp'
    |2nd order filter block
  Resistor    Node=1=2   R={ 1/(2*pi*fo*C1); }
  Capacitor   Node=2=5   C={ C1; }
  Resistor    Node=2=3   R={ 1/(2*pi*fo*C1); }
  Capacitor   Node=3=0   C={ C1; }
  OpAmp  'Op11' Def='Op1'   Node=3=4=5
  Resistor    Node=5=4  R={ 10e3*(2 - 1/QD); }
  Resistor    Node=4=0  R=10kohm

    |1st order filter block
  Resistor    Node=5=6   R={ 1/(2*pi*fo*C1); }
  Capacitor   Node=6=0   C={ C1; }
  OpAmp  'Op12' Def='Op1'   Node=6=7=8
  Resistor    Node=7=8  R=10kohm
```

The inline formula can replace the numerical entries. The formulae are, as shown, in braces. The filter frequency fo, the pole quality QD and a setting for the capacitors C1 and C2 are entered centrally via the `Def_Const` definition. You can also display the results of the individual formulae. Place the script cursor in the text of the element you want to evaluate, e.g. the `Resistor` element and press Ctrl+E (Search/ Current element menu). This opens the associated input dialog, which contains the calculated values. Before the input dialog of an element is opened, AkAbak evaluates any existing formulae, including the `Def_Const` definition. If an error occurs, no error message appears.

Implementation with transistors

Script file: **Ex_64.aks**

The 3rd order Butterworth low pass can also be implemented with the aid of transistors. A simple circuit is shown in Fig. 26. The filter frequency fo and the pole quality QD are set as with the operational amplifier.

```
Def_Transistor  'Q1' |npn and pnp
  h11e=3.5kohm  h12e=2e-4  h21e=270  h22e=32uS
  ft=200MHz     Ccb=3.5pF

  |Sallen-Key-Filter
System  'Transistor'
    |Filter block 2nd order
  Resistor    Node=1=2   R=10.61kohm
  Capacitor   Node=2=6   C=15nF
  Resistor    Node=2=3   R=10.61kohm
  Capacitor   Node=3=0   C=15nF
  Transistor  'Q11'  Def='Q1' Node=3=4=5  |Base-collector-emitter
  Resistor    Node=4=0  R=5.6kohm
  Resistor    Node=5=0  R=2.2kohm
  Transistor  'Q12' Def='Q1'  Node=4=6=0
  Resistor  Node=6=5  R=2.2kohm

    |Filter block 1st order
  Resistor    Node=6=7   R=10.61kohm
  Capacitor   Node=7=0   C=15nF
  Transistor  'Q13' 'Q1'  Node=7=8=9
  Transistor  'Q14' 'Q1'  Node=8=9=0
  Resistor    Node=8=0  R=5.6kohm
  Resistor    Node=9=0  R=2.2kohm

    |Mark the output stage
  Potential 'Out' Node=9
```

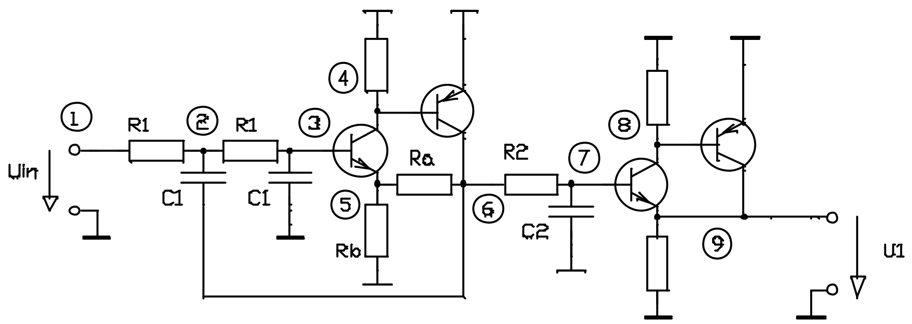


Fig. 26 Sallen-Key circuit with Darlington transistor circuit

7. Compression Driver

This exercise introduces some ideas to compression driver measurement and simulation. It should be pointed out that the simulation of a compression driver is a very complicate matter (see also Introduction/ Transducer/ Modeling a transducer). There will be only reliable results obtained when the design of the individual driver is clearly understood. This exercise demonstrates one method of determining the driver parameter of a horn driver. The formulae are evaluated with a formula script of the '**Abakus**' calculator. If the equivalent circuit parameter of a horn driver are not known, they have to be measured. However, this is not as simple as with a conical loudspeaker. For modeling with the acoustic elements, the 'inner' equivalent circuit parameter are required.

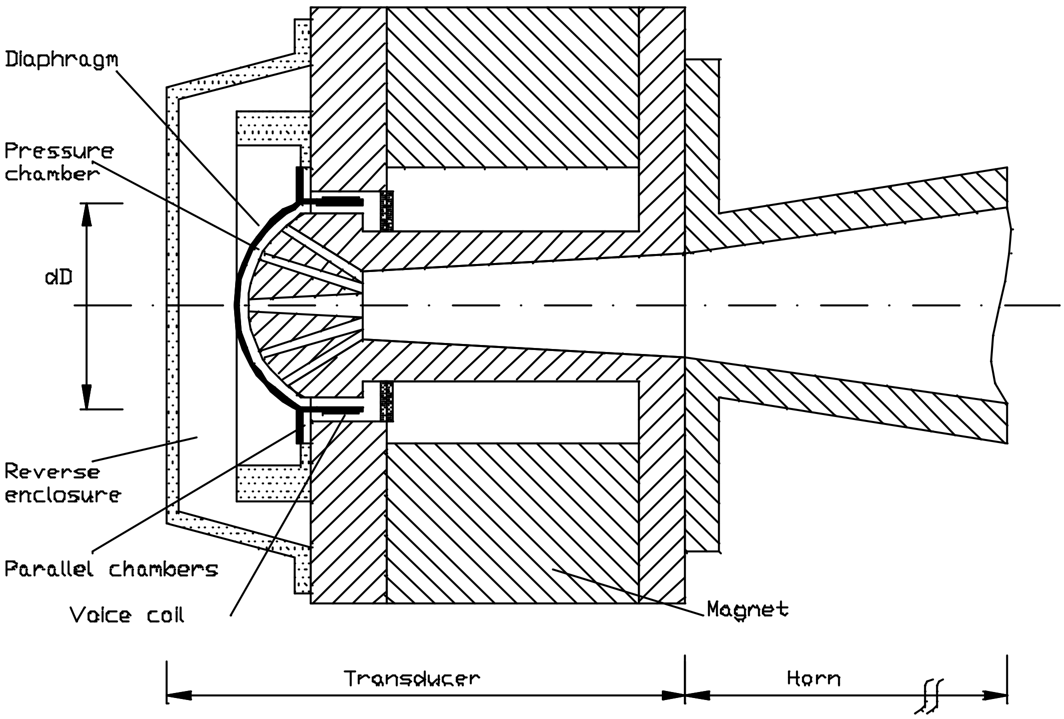


Fig. 27 Diagram of the cross-section of a conventional medium-frequency horn driver

Fig. 27 shows a diagram of the cross-section of a conventional horn driver. The diaphragm radiates with the concave side into the horn stub, whose walls at the same time form one of the magnet poles. The input area of the horn stub is subdivided, so that at high frequencies there are no differences in travel time. Between the diaphragm and the input of the horn stub is the pressure chamber. This is produced automatically, since the area of the horn stub is smaller than the diaphragm area[2]. The mechano-acoustics of this chamber - if viewed in detail - is extremely complex. On the one hand the daphragm is vibrating in its eigen-modi, coupled to the spherical room which itself posses eigen-modi. On the other hand huge impedance variations, which are reflected by the horn-bell, takes place at the phase plug.

At the dome base is the voice coil. The magnet gap and the unit for fastening the diaphragm have hollow spaces whose volume and shape participate in the overall transmission. It is often difficult to reproduce these hollow spaces, since mechanical components are also involved. The voice coil itself works like a pump. Air is pressed through the slit and energy gets lost in friction and turbulence. What can clearly be observed is the high sensitivity of the overall transfer functions to the dimensions of the voice coil path, as described in the related example in chapter 'Introduction/ Transducers'.

The convex diaphragm side radiates into a sealed enclosure whose stiffness is added to the stiffness of the diaphragm suspension.

[2] Since the diaphragm does not behave like a pulsating surface, the diaphragm area is in a simple ratio to the diaphragm diameter

Determining the equivalent circuit parameter

It is difficult to determine the parameter of a horn driver, since the impedance curve is not that of a driver that radiates freely or into a sealed enclosure. If you use the `Def_Driver` definition in all cases include the `Meas_DoNotModify` keyword (see Introduction/ Transducer). Because the parameter are already freed from any radiation impedance.

Vacuum - method

This is best. All acoustical impedances are zero. Only mechanical values are measured. The measurement setup is identically to that of a normal direct-radiating driver. The results can directly entered into the script.

Damped duct - method

One way of getting round this problem is to attach an acoustic duct of the horn at the orifice of the horn stub. This duct is long enough (1m...2m) and is filled with insulating material. This duct should not have any standing waves and should offer the following acoustic resistance to the diaphragm:

$$Ra = \rho \cdot c / S$$

where

ρ: air density, c: velocity of sound, S: cross-sectional area

In this measurement the measured mechanical resistance Rms contains the mechanical counterpart to Ra ($Rm = SD^2 \cdot Ra$, where SD is the diaphragm area). This value should be subtracted from the measured Rms value.

Sealed method

This way of determining the parameter may be easier to carry out but is not so accurate. In this case the orifice of the horn stub is sealed. To achieve this, push a piece of Pertinax or cardboard between the horn flange and driver. The impedance curve now corresponds to the curve of a loudspeaker in the sealed enclosure. The volume is formed from the volume of the horn stub, the pressure chamber and the hollow spaces. Since the rear volume is part of the driver, it should be attached during the measurement so that this volume is already contained in the parameter. Measure the value of the vibrating mass Mms by the method 'adding an additional mass' (see Chapter 'Tools/ Mms, Cms, Vas Parameter'). If the inner dimensions of the horn driver are not known, there is no alternative but to dismantle the driver and measure all the parts. These dimensions are also needed for the simulation. Calculate the equivalent volume Vas:

$$Vas = \frac{1}{(2\pi \cdot fc)^2 \cdot M - \frac{1}{V}} \quad \text{where} \quad M = \frac{Mms}{SD^2 \cdot \rho \cdot c^2}$$

fc	resonant frequency with closed horn stub [Hz]
V	volume of the horn stub, pressure chamber and hollow spaces [m3]
Mms	measured mass of the vibrating system [kg]
SD	diaphragm area [m2]
ρ, c	air density [kg/m^3], velocity of sound [m/s]

The equivalent circuit parameter of the driver without horn stub, pressure chambers and hollow spaces are then:

$$\alpha = \sqrt{1 + Vas / V}$$

$$fs = fc / \alpha, \qquad Qes = Qec / \alpha, \qquad Qms = Qmc / \alpha$$

where

Qec	Measured electrical quality with closed horn stub
Qmc	Measured mechanical quality with closed horn stub

These formulae can be evaluated with a script of the 'Abakus' calculator:

Start the 'Abakus' calculator (Calc/ Abakus menu) and load the Abakus with the formula script **HDrv.aba** (\AkAbak\Formula directory).

Abakus script file: **HDrv.aba**

```
|Input volumes:
|-----------------------------------------------
Vf= 1e-6;    |pressure chamber [m3]
Vq= 10e-6;   |hollow spaces [m3]
Vh= 12.4e-6; |horn stub [m3]

|Input driver data and measured data.
|For measurement of the resonance, fc and fcc,
|the horn stub is closed.
|-----------------------------------------------
dD=  4.5e-2; |diaphragm diameter [m]
fc=  991;    |resonance [Hz]
Qmc= 5.44;   |mech. quality
Qec= 0.702;  |electr. quality
m=   3.1e-3; |additional mass [kg]
fcc= 436;    |resonance with m  [Hz]

V= Vf + Vq + Vh;
Mms= m/(sqr(fc/fcc) - 1);     |vibrating mass
A= sqr(fc)*Mms/(2220*dD^4);

|Result:
|-----------------------------------------------

Vas= 1/(A - 1/V); |equivalent volume  [m3]

alpha= sqrt(1 + 1/(A*V - 1));

fs= fc/alpha;      |internal resonance [Hz]
Qms= Qmc/alpha;    |internal mechanical quality
Qes= Qec/alpha;    |internal electrical quality
```

Vf, Vq and Vh are the volumes of the pressure chamber, the hollow spaces and the horn stub in [m3].

Also enter the diameter of the diaphragm dD, the measured resonance frequency fc, the mechanical quality Qmc, the electrical quality Qec, the weight of the additional mass m used and the resonance fcc that is generated when the additional mass is attached.

The values fc, fcc, Qec and Qmc are each determined with the horn stub closed. Use the tool for this: 'Tools/ Dyn.Driver Parameter' menu.

When all data have been entered, press Ctrl+Enter ↵ to start the evaluation. The results appear in the output list (SI -units):

```
Qes = 0.480047
Qms = 3.720019
fs = 677.672648
alpha = 1.462358
Vas = 26.640686e-6
A = 80.271614e3
Mms = 0.744077e-3
...
```

8. Multiple Driver Modeling

Abstract[3]

The dodecahedron loudspeaker, a design widely used in acoustic measurements, is modeled using components available in a computer-based electroacoustic network simulator. The acoustic radiation and driver behavior simulation includes specifying the physical position and radiation angle of each of the twelve drivers. Comparison is made with anechoic chamber measurements on a real loudspeaker.

0. Introduction

This paper demonstrates the simulation of the response involving multiple drivers mounted on the surface of a spherical body radiating in all directions. This special loudspeaker design is widely used in acoustic measurements where it serves as an omnidirectional sound source. The simulation of the transfer and loading parameters not only helps in designing such a loudspeaker but also the simulation leads to a sound understanding of the fundamental effects. The paper also explains how the classical lumped element method can be applied successfully using an analyzer for electroacoustical networks. The analyzer used is an available computer program called AkAbak® running under Windows™ 3.x.

1. Loudspeaker enclosure

The radiators in the system under study are mounted each in the center of the 12 faces of a regular dodecahedron. The dodecahedron is a so-called Platonic body (Fig. 28) consisting of 12 regular pentagons joined at their edges. Due to the regular form and high number of faces the dodecahedron is acoustically a close approximation to the radiation from a sphere.

Table 1 Driver parameter

Eff. outer diaphragm diameter dD	93.5 mm
Eff inner diameter dD1	30 mm
Depth of cone tD1	11 mm
Frequency of mass reduction fp	4.5 kHz
Mass of vibrating assembly Mms	4.95 g (incl. air load)
Compliance of suspension Cms	0.6 mm/N
Motor conversion factor Bl	2.1 Tm
Voice coil resistance Re	0.93 Ω
Voice coil inductance factor Le (approx. "squared reactance")	120 µH

Fig. 28 Photo of dodecahedron

2. Loudspeaker drive units

There are 12 equal drive units. Each driver has an electro- dynamical motor and a conical shaped diaphragm. The parameters of the diaphragm and the motor are given in Table 1. The mass of the vibrating assembly together with the suspension compliance yield to a fundamental free field resonance of fs=92Hz. The parameter

[3]Paper of 102nd AES Convention Munich 1997: 'Multiple Driver Modeling with a Modern Lumped Element Simulation Program'; Jörg W. Panzer, Panzer and Partner, Steinstr. 15, D-81667 Munich, Germany and Richard H. Campbell Worcester Polytechnic Institute 100 Institute Rd., Worcester, MA 01609, USA

of the frequency of mass reduction fp is used by the simulator to take into account that at high frequencies only part of the diaphragm radiates sound.

The voice coil reactance differs from the curve of a pure inductance. A good approximation is to take the square root: Xe(jω)=sqrt(ωLe). This is not exactly what the simulator does, but without going into details here, it is a good way to understand this specification in principal.

3. The simulator

The simulator used is a Windows™ program called AkAbak which is a synonym for "Acoustic Abakus". This software provides an integrated development system for analyzing and designing electroacoustic devices such as loudspeaker drivers and systems.

To be able to describe the structure under investigation by means of AkAbak, the structure must be first separated into a set of lumped elements and one-dimensional waveguides. The components are wired with a node-based system, called a network. Based on linear system theory, the analysis is carried out in the frequency domain by solving the node-potential-matrix.

AkAbak offers special components which are dedicated to especially loudspeaker design. There are several complete models for transducers, waveguides, enclosures and radiation which are used to simulate the response when multiple radiation sources are involved.

Before arranging all of the components into a full system it is necessary to understand how AkAbak handles the radiation of cone shaped diaphragms, and other computational aspects of the simulator.

4. Loading and radiation into free space

4.1 Self-radiation impedance

Each diaphragm is acoustically loaded by the so-called radiation impedance. The radiation impedance is the ratio of the driving velocity of the air in the very near of the diaphragm surface and the pressure reacting back onto it. If we imagine us the diaphragm surface separated into many small sub-areas then the force on one of this sub-areas is created by the generated sound pressure of all the other sub-areas. The total self-radiation impedance is then obtained by surface-integration over the diaphragm.

The acoustical power output is proportional to the real part of the radiation impedance, Re(Z_A). The impedance function, Re(Z_A), resembles that of a high-pass filter having a typical cut-off frequency where one wavelength of the radiated sound is equal to the perimeter of the circular diaphragm. This frequency is sometimes called the *directivity frequency*, fD, or the *rim resonance*. In Figure 2 the directivity frequency is at approx. fD=1.2kHz.

The magic of flat direct-loudspeaker response lies in the cooperation of the velocity of the diaphragm and the radiation resistance (Fig. 30). The band-pass curve of the velocity is centered at the fundamental resonance of the driver. At frequencies above this point, the so-called mass controlled region, the roll-off of the velocity curve is compensated with respect to the radiated power by the rising radiation resistance. Above the directivity frequency, fD, where the radiation resistance is approximately constant, the well known drop of the power output can be recognized.

Cones, which belong to the class of concave-formed diaphragms, cause standing wave patterns to exist which increases the radiation impedance. The strongest effect on the power output is caused by an amplification of the radiation resistance in the region of the directivity frequency, fD (Fig. 29). Nearly all cone shaped diaphragms have an enhanced power response in the vicinity of the directivity frequency (Fig. 31).

Near and above the directivity frequency mechanical eigen-vibration of the diaphragm also effects the radiation impedance. Without going into details one major effect can be extracted. With rising frequency the acceleration of the whole vibrating assembly is so strong that only a part of the diaphragm comes into controlled motion. To be able to simulate this reduction of radiating area and vibrating mass in principal, we use the estimation parameter fp. The frequency fp can be regarded as a cut off frequency where only half of the cone-diaphragm area is vibrating. At higher frequencies the radiating part of the cone becomes smaller until only a small ring around the voice coil is radiating.

4.2 Mutual-radiation impedance

The force, reacting back to any point of the diaphragm surface, naturally can be caused by any other radiating source, in the same way as the diaphragm does it to itself. This source can be any other driver or reflecting wall as well. Within the latter is included the baffle in which the diaphragm is mounted.

The mutual-radiation impedance between diaphragms can be understood in the same way as the self-radiation impedance. The sound pressure radiated to each sub-area of the other diaphragm creates a force there forming part of the total radiation impedance.

The closer the other radiator is the stronger is the coupling. The strongest coupling is exhibited by the baffle in which the diaphragm is mounted. An infinite baffle doubles the radiation resistance. A finite baffle causes diffraction at the baffle edges. At high frequencies the radiation resistance is double the amount as at low frequencies.

The simulator implements a radiation impedance element in the network which takes into account mutual coupling caused by reflecting walls and by the baffle including diffraction effects. The effects of the feedback of the mutual coupling into the network is usually very small because of the extremely low acoustical energy which is received when radiation takes place in free space.

4.3 Radiation

The sound pressure at any listing point outside the source is the sum-total of generated sound of all diaphragms including diffraction and reflections.

4.3.1 Radiation of diaphragm

Because the dimension of a single diaphragm is within the range of the radiated wavelength, an interference phenomenon causes the well-known directivity or beaming effect (Fig. 32). Consider that if the diaphragm surface separated into many small sub-areas, the pathlength from each of this sub-areas to the listening point is different. When this difference is comparable to the wavelength or greater interference occurs. When the listening point is sufficiently far away from the source the rays from the radiating surfaces to the listing point can be regarded parallel, a condition known as far-field. For example, a piston-like diaphragm will cause no interference on-axis in the far-field. But when the test point is moved off-axis, interference of the closer and more far away sub-areas of the diaphragm causes directivity or a drop of output power, respectively.

Diaphragm shapes other then the piston will exhibit interference in the far-field even on-axis. The reason for this is that the radiating sub-areas of the vibrating diaphragm are not all located in the plane normal to the direction of the listening point.

The simulator calculates the directivity of a single diaphragm in the far field and takes into account the form of the diaphragm. The cone is divided into many rings. From each ring the directivity is calculated and then numerically integrated over the entire surface of the diaphragm. The integration limits are derived from the effective diaphragm area which is frequency dependent as mentioned earlier. Compared to a rigid diaphragm the directivity of a frequency controlled diaphragm decreases at high frequencies (Fig. 32).

4.3.2 Diffraction at the baffle edges

The finite baffle size causes diffraction. The sound pressure level is twice as high at high frequencies than at low frequencies (Fig. 33). The fundamental effect of diffraction is interference. Using a simplified model imagine the finite baffle formed of two parts: the actual baffle in which the diaphragm is mounted, and the extended area behind the edges in the same plane as the baffle itself. As soon as the traveling wave is behind the edges a pressure gradient is produced toward the reverse side of the baffle, thus creating a second radiator. The sound generated by this "outer" diaphragm adds to the direct radiated sound of the source, but it is time delayed.

The simulator calculates the diffraction effects using the mirror model. For the "outer" diaphragm one or two additional radiators are installed by the program. Only one mirror radiator is needed when the baffle edges are approximately equidistant from the source daphragm. A second mirror radiator is added when the baffle is rectangular or the source is mounted asymmetrically. The mirror-radiators are fed a time-delayed signal by the source and with a radiation angle of 90°. With this arrangement and because every diaphragm possesses directivity, the diffraction effect is diminished at high frequencies, which is exactly what we observe on real baffles. At high frequencies the diaphragm becomes its own baffle.

This simple implementation of the diffraction effect leads to surprisingly good results and is a good compromise between exactness, calculation speed and parameter specification.

4.3.3 Reflection

There are two cases of reflection simulated by AkAbak. Firstly, the main reflector is always the baffle in which the diaphragm is mounted or at high frequencies the diaphragm itself. Secondly, depending on the wavelength, a sufficiently large and rigid wall distant to the source will cause interference due to the traveling time of the sound wave from the source to the reflector and from the reflector to the listening point. In the first case the reflected wave interferes with the source modifying the radiation impedance. In the second case the reflected wave interferes with the direct radiated sound at the listing point. In closed listening rooms, a standing wave pattern further modifies the radiation impedance and the observed sound at the listening point.

AkAbak takes into account a maximum of three orthogonal reflecting walls affecting the radiation impedance and the radiated sound, again using the mirror radiator model. For each wall and for each original source additional mirror radiators of first, second and third order are installed automatically.

4.4 Radiation from multiple sources

When multiple radiators are involved the sum-total of complex sound pressure of all radiators including diffraction and reflection is summed at the listing point.

In the simulation-result the sum-total of sound pressure of all radiators is a near-field condition and a far-field condition with respect to the radiation of a single diaphragm.

The acoustical power output of the assembly is obtained by surface-integration over a sphere outside the loudspeaker.

To be able to calculate the directivity and the level with respect to the listening point the exact position and mounting angle of each radiator must be specified. Further it is required to provide entries for the effective baffle size and the distance and specification of the reflecting walls.

5 Loading and radiation into the enclosure

The reverse side of the diaphragm is usually loaded by an enclosure. The acoustics of small enclosures, where the dimension of the enclosure is of the order of a wavelength, is totally different from the radiation into free space. The standing wave-pattern in normal loudspeaker enclosures can be categorized in three frequency ranges. In the low frequency range, where the wavelength is much larger then the dimensions of the enclosure, the air in the cavity is reacting like a spring and a small mass to an applied force. In the medium frequency range, where the wavelength is in the range of the largest extent of the enclosure the cavity can be regarded as an one-dimensional waveguide. In the high frequency range, the wave-equation must be solved with time dependency in all dimensions.

Beside the simple closed rectangular cavity the enclosure can be of any form. It may be tube or horn like, with bends and obstacles. There might be multiple driving (or passive) diaphragms or radiating holes.

To simulate the acoustic response of such applications with lumped elements and one-dimensional waveguides it is necessary to investigate and separate the structure into modules. These modules are then interconnected in a network similar to the analysis of electric circuits. Because of the reflecting boundaries the sound pressure within an enclosure is usually much higher than in free space. In simulations careful investigation is necessary to be certain that linear system theory is still valid at high levels.

In the docecahedron loudspeaker 12 diaphragms are radiating in the same small closed enclosure of nearly spherical form. Naturally this form is impossible to separate exactly into three independent dimensions and because of this one can only approximate the exact acoustic wave-pattern inside the enclosure. At low frequencies the air reacts like a spring to all diaphragms. At approx. 700 Hz the first eigen-frequency of the cavity appears and modifies the loading of the drivers. At high frequencies multiple modes spread all over the inner of the enclosure to amplify and damp one or the other driver. Fortunately the higher modes have much less energy at a single point then the fundamental and because of this their effect can be seen as small peaks and dips in the curves.

In this simulation we model the enclosure with Duct-components as an approximation of the three dimensional wavefield. The Duct element is a one-dimensional waveguide with constant cross-section. We connect the Duct-fourpoles in a star-like arrangement with one port of each Duct connected with the reverse side of each diaphragm. The other end is connected with all other Ducts to an Enclosure element in the center of the star. The cross-section and length of each of the 12 Ducts are derived from the effective inner volume and dodecahedron dimensions. The Enclosure element itself is also a single waveguide taking into account the fundamental mode in the largest dimension. It is vented here to model some losses which are apparently present due to air leaks between the driver flange and the enclosure shell. In this way not only the correct loading but also the interaction of opposite-located drivers is maintained.

6. Data specification and simulation

AkAbak's input system is a text script edited by an built-in word-processor (see addendum). The elements are written in paragraphs. The parameters can be entered as a constant value or in the form of a formula system. Node-numbers connect the components. Drivers must be specified at the beginning of the script and are then used in the networks as often as needed.

In this example, most of the parameters in the acoustical network portion of the dodecahedron loudspeaker depend on the geometry of the Platonic body. Therefore, the Def_Const definition at the beginning of the script is used to calculate all the needed positions, mounting angles and parameters of the duct elements of the enclosure. In this way it is easy to control the input of many elements by only one or two stated variables. The specification starts with the edge length of the pentagonal face. In this way, the response of a larger or smaller dodecahedron loudspeaker can be computed simply by altering one value.

After entering all data, the script is compiled into binary format and then the spectrum analysis is carried out. There are numerous ways of inspecting the network and displaying the simulation. A loudspeaker should always be simulated at least at its input and output port. Figure 8 to 11 are examples for several simulations including measurement curves. The measurements are carried out in a sound absorbing chamber using the MLSSA system, DRA Laboratories, USA.

The drivers are mounted in such a way that one of the diaphragms has its mounting point on the z-axis. The z-axis of the baffle coordinate system is the "on-axis" reference. The y-axis points upwards and the x-axis to the left of the baffle (seen from the baffle). The distance to the listening point is measured from the center of the dodecahedron, i.e. six speakers are offset toward the on-axis listening point and six speakers are positioned behind the origin facing rearward.

One of the advantages of the lumped element simulation is the high calculation speed. For example, it takes about two minutes on a modern PC to compile and calculate the extensive script given in the addendum and the spectrum of Figure 9 with a high resolution.

Figure 11 displays the sound power output (SWL) calculated by surface integration. The other graph is the on-axis SPL. The third is the directivity factor Q relating to the on-axis response (abbreviated). The directivity factor is $Q = 1$ at low frequencies and rises at high frequencies due to power loss caused by interference. Both curves are under free field condition without any reflector. Note that the SPL drops by 6 dB but the SWL only by 3 dB radiating into 4π space.

6. Discussion

A lumped element simulator is a computer-aided design tool which helps to investigate, understand and enhance a loudspeaker system. The lumped element method is simple, but forces an understanding of the structure under investigation. Because it concentrates exclusively on the main effects, this analysis method is both practical and fast.

In the real world, resonance points and interference effects are more or less unsharp, damped or distorted. Because AkAbak uses only a limited set of components, resonances and interference effects are displayed with more pronounced effect. To the loudspeaker designer, the curves might be not so pleasant at first glance but highly informative.

The simulation of the dodecahedron loudspeaker reveals an interesting effect which can be worked out with this analysis method.

Seen from the on-axis listening point the centers of the 12 diaphragms are located in four planes. On the nearest plane one speaker is mounted. In the next plane five speakers are arranged in a circle around the z-axis radiating toward the side. On third plane five loudspeakers radiate rearward to the side, and on the last plane a single driver is opposite to the front speaker. Because of the regular offset a sharp interference occurs on-axis which can be seen clearly in the SPL curve of Figure 9 (1.3 kHz).

Oddly enough, another resonance at the same frequency seems to compensate the notch. This resonance is produced by the 2nd eigen-frequency of the enclosure modifying the acoustical load of the drivers. The next mode is at 2.6 kHz. In the simulation script the length of the Ducts which model the enclosure is directly derived from the dodecahedral geometry. The length of each Duct is equal to the radius of an inscribed sphere (Constant "b" in the Def_Const specification of the script).

It is essential to compare the simulation with both the measurement of a transfer function and the measurement of a driving point parameter. Here the transfer function is the SPL curve and the driving point is the electrical impedance curve. The curves in Figure 8 have multiple peaks. The pronounced one at approx. 250 Hz belongs to the mechanical resonance of the drivers. The peak below is caused by leakages of the enclosure which makes it actually a lossy, vented box. In reality the losses are more smeared with respect to frequency. At 700 Hz and 1.3 kHz the varying acoustic load on the drivers caused by the standing waves in the enclosure are fedback to the motional impedance.

Investigating the time domain response, a further interesting result can be obtained. From the spectrum the so-called energy time curve (ETC) can be calculated. The ETC is the magnitude of the analytical time transfer function. Derived from the SPL spectrum, it displays the time arrival of acoustic energy at the listening point. Figure 12 compares the ETC curves of a single driver and of the whole assembly. The peaks belong to the different driver mounting offsets. When only the front driver is radiating (which is only possible in the simulation) all energy is concentrated at approximately -330µs (calculated from the baffle origin). When all drivers are radiating, a cascade of sound energy arrives at the listening point carrying the same information. The result is a noticeably smeared time signal.

It is proper to say that the dodecahedron has omnidirectional radiation, but it is improper to say that it has spherical radiation characteristics. Even if an ever increasing multiplicity of faces and drivers were used, in the limit leading to an essentially spherical radiating surface, the presense of the hard shell causes unavoidable diffraction and interference from the spatial radiatiing elements distributed around its surface. An expanding spherical wave (with the same radius of curvature) in free space which originated at a point, has no shell and no diffraction, and no arrival time smear.

7. Conclusion

The coupling of a powerful calculation module to an accurate acoustic modeling program has significant benefits as seen in this paper. The translation and rotation of the 12 drivers as well as certain volume and diffraction values are all calculated from an embedded script which has a single given value at the begining -- the length of one edge of a pentagonal face. This same technique can be used to develop complex horn shapes using piece-wise linear segments.

8. References

1. Hill, F. S.: "Computer Graphics", Collier Macmillian Canada Inc.
2. Beranek, L. L.: “Acoustics”; American Institute of Physics, Inc., New York, USA, 1986
3. Morse, P., Ingard, K.: “Theoretical Acoustics”, Princeton, New Jersey, USA, Princeton University Press, 1986
4. Vanderkooy, J.: "A Simple Theory of Cabinet Edge Diffraction", JAES, Vol. 39, No. 12, Dec. 1991

9. Addendum

Script-file: **Dodec.aks**

```
|AkAbak Script - Dodecahedron model

Def_Const {                        |Dodecahedron geometric calculations
a=0.11;                           |in meters, length of an edge of a
pentagonal face
p=5;  q=3;                        |Schlafli symbols {p,q} for regular
polyhedra
gr=2*arcsin((cos(pi/q)/sin(pi/p))); |dihedral angle after Coxeter
grc=pi/2 - gr/2;                  |dihedral center angle
r=(a/2)*(1/tan(pi/5));            |radius of an inscribed circle in one face
doR1=r; doR2=doR1;                |distance to edge for diffraction
b=r*tan(gr/2);                    |length of a vector from the geo- to a face
center
zed=b*cos(2*grc);                 |displacement of center points
yed=b*sin(2*grc);                 |projection on the y-coordinate
Vb=9.5e-3;                        |in m³, eff. inner volume of body
Len=b;                            |length of Duct elements forming the
enclosure
Q_fo=0.5;                         |quality factor of enclosure

|Mounting positions:
x0=0;                  y0=0;                   z0=b;
x1=yed*sin(rad(0));    y1=-yed*cos(rad(0));    z1=zed;
x2=yed*sin(rad(72));   y2=-yed*cos(rad(72));   z2=zed;
x3=yed*sin(rad(144));  y3=-yed*cos(rad(144));  z3=zed;
x4=yed*sin(rad(216));  y4=-yed*cos(rad(216));  z4=zed;
x5=yed*sin(rad(288));  y5=-yed*cos(rad(288));  z5=zed;
x6=yed*sin(rad(-36));  y6=-yed*cos(rad(-36));  z6=-zed;
x7=yed*sin(rad(36));   y7=-yed*cos(rad(36));   z7=-zed;
x8=yed*sin(rad(108));  y8=-yed*cos(rad(108));  z8=-zed;
x9=yed*sin(rad(180));  y9=-yed*cos(rad(180));  z9=-zed;
x10=yed*sin(rad(252)); y10=-yed*cos(rad(252)); z10=-zed;
x11=0;                 y11=0;                  z11=-b;

|Mounting angles:
VA0=0;  HA0=0;
VA1=deg(arctan(y1/sqrt(sqr(x1) + sqr(z1))));
HA1=deg(arctan(x1/z1));
VA2=deg(arctan(y2/sqrt(sqr(x2) + sqr(z2))));
HA2=deg(arctan(x2/z2));
VA3=deg(arctan(y3/sqrt(sqr(x3) + sqr(z3))));
HA3=deg(arctan(x3/z3));
VA4=deg(arctan(y4/sqrt(sqr(x4) + sqr(z4))));
HA4=deg(arctan(x4/z4));
VA5=deg(arctan(y5/sqrt(sqr(x5) + sqr(z5))));
HA5=deg(arctan(x5/z5));
VA6=deg(arctan(y6/sqrt(sqr(x6) + sqr(z6))));
HA6=-(180-deg(abs(arctan(x6/z6))));
VA7=deg(arctan(y7/sqrt(sqr(x7) + sqr(z7))));
HA7=180-deg(abs(arctan(x7/z7)));
VA8=deg(arctan(y8/sqrt(sqr(x8) + sqr(z8))));
HA8=180-deg(abs(arctan(x8/z8)));
VA9=deg(arctan(y9/sqrt(sqr(x9) + sqr(z9))));
HA9=180-deg(abs(arctan(x9/z9)));
```

```
VA10=deg(arctan(y10/sqrt(sqr(x10) + sqr(z10))));
HA10=-(180-deg(abs(arctan(x10/z10))));
VA11=0;  HA11=180;
}
|Drive unit, type: Bose B901
Def_Driver  'Drv1'
  dD=9.35cm  dD1=30mm  tD1=11mm fp=4.5kHz
  Mms=4.95g  Cms=0.6e-3m/N  Rms=1Ns/m  Bl=2.1Tm
  Re=0.93ohm  Le=120uH  ExpoLe=0.618

System 'dodec'
  | Forward radiating loudspeakers
  Driver 'D1' Def='Drv1'  Node=10=11=100=31
  Radiator 'R1' Def='Drv1'  Node=100
    x={X0} y={Y0} z={Z0} HAngle={HA0} VAngle={VA0}  WEdge={doR1}
HEdge={doR2}
  Driver 'D2' Def='Drv1' Node=11=12=101=32
  Radiator 'R2' Def='Drv1'  Node=101
    x={X1} y={Y1} z={Z1} HAngle={HA1} VAngle={VA1}  WEdge={doR1}
HEdge={doR2}
  Driver 'D3' Def='Drv1' Node=12=13=102=33
  Radiator 'R3' Def='Drv1'  Node=102
    x={X2} y={Y2} z={Z2} HAngle={HA2} VAngle={VA2}  WEdge={doR1}
HEdge={doR2}
  Driver 'D4' Def='Drv1'  Node=13=14=103=34
  Radiator 'R4' Def='Drv1' Node=103
    x={X3} y={Y3} z={Z3} HAngle={HA3} VAngle={VA3}  WEdge={doR1}
HEdge={doR2}
  Driver 'D5' Def='Drv1'  node=14=15=104=35
  Radiator 'R5' Def='Drv1'  node=104
    x={X4} y={Y4} z={Z4} HAngle={HA4} VAngle={VA4}  WEdge={doR1}
HEdge={doR2}
  Driver 'D6'  Def='Drv1'  node=15=0=105=36
  Radiator 'R6' Def='Drv1'  node=105
    x={X5} y={Y5} z={Z5} HAngle={HA5} VAngle={VA5}  WEdge={doR1}
HEdge={doR2}
  | Rearward radiating loudspeakers
  Driver 'D7' Def='Drv1' node=10=21=106=37
  Radiator 'R7' Def='Drv1'  node=106
    x={X6} y={Y6} z={Z6} HAngle={HA6} VAngle={VA6}  WEdge={doR1}
HEdge={doR2}
  Driver 'D8' Def='Drv1'   node=21=22=107=38
  Radiator 'R8' Def='Drv1'  node=107
    x={X7} y={Y7} z={Z7} HAngle={HA7} VAngle={VA7}  WEdge={doR1}
HEdge={doR2}
  Driver 'D9' Def='Drv1' node=22=23=108=39
  Radiator 'R9' Def='Drv1'  node=108
    x={X8} y={Y8} z={Z8} HAngle={HA8} VAngle={VA8}  WEdge={doR1}
HEdge={doR2}
  Driver 'D10' Def='Drv1'  node=23=24=109=40
  Radiator 'R10' Def='Drv1'  node=109
    x={X9} y={Y9} z={Z9} HAngle={HA9} VAngle={VA9}  WEdge={doR1}
HEdge={doR2}
  Driver 'D11' Def='Drv1'  node=24=25=110=41
  Radiator 'R11' Def='Drv1'  node=110
    x={X10} y={Y10} z={Z10} HAngle={HA10} VAngle={VA10}  WEdge={doR1}
HEdge={doR2}
  Driver 'D12' Def='Drv1'  node=25=0=111=42
  Radiator 'R12' Def='Drv1'  node=111
```

```
    x={X11} y={Y11} z={Z11} HAngle={HA11} VAngle={VA11}  WEdge={doR1}
HEdge={doR2}

  |Enclosure build up by Duct elements
  Duct 'Du1'  node=30=31  SD={Vb/(Len*13)}  QD/fo={Q_fo}  Len={Len}
  Duct 'Du2'  node=30=32  SD={Vb/(Len*13)}  QD/fo={Q_fo}  Len={Len}
  Duct 'Du3'  node=30=33  SD={Vb/(Len*13)}  QD/fo={Q_fo}  Len={Len}
  Duct 'Du4'  node=30=34  SD={Vb/(Len*13)}  QD/fo={Q_fo}  Len={Len}
  Duct 'Du5'  node=30=35  SD={Vb/(Len*13)}  QD/fo={Q_fo}  Len={Len}
  Duct 'Du6'  node=30=36  SD={Vb/(Len*13)}  QD/fo={Q_fo}  Len={Len}
  Duct 'Du7'  node=30=37  SD={Vb/(Len*13)}  QD/fo={Q_fo}  Len={Len}
  Duct 'Du8'  node=30=38  SD={Vb/(Len*13)}  QD/fo={Q_fo}  Len={Len}
  Duct 'Du9'  node=30=39  SD={Vb/(Len*13)}  QD/fo={Q_fo}  Len={Len}
  Duct 'Du10' node=30=40  SD={Vb/(Len*13)}  QD/fo={Q_fo}  Len={Len}
  Duct 'Du11' node=30=41  SD={Vb/(Len*13)}  QD/fo={Q_fo}  Len={Len}
  Duct 'Du12' node=30=42  SD={Vb/(Len*13)}  QD/fo={Q_fo}  Len={Len}
  Enclosure  'E1'  Node=30  Vb={Vb/13}  Qb/fo={Q_fo}  Lb={Len}
    Len=1mm  SD=5cm2  QD/fo=0.007  WEdge={doR1} HEdge={doR2}
```

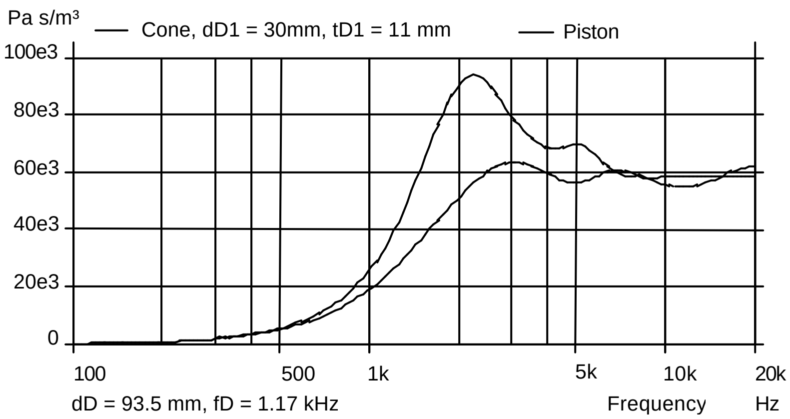


Fig. 29 Radiation impedance of piston and cone diaphragm

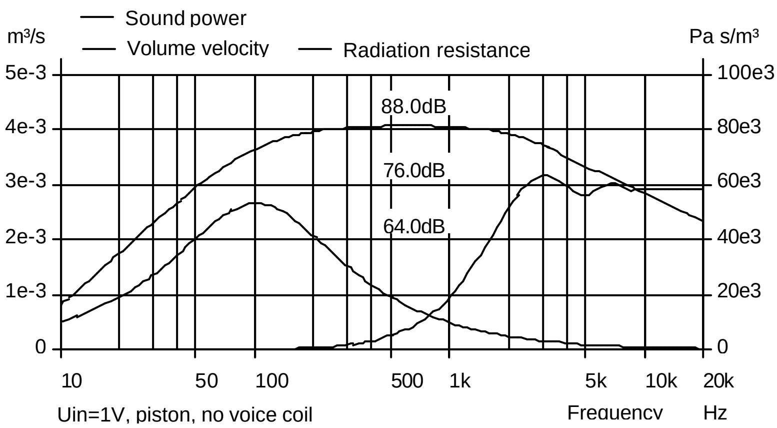


Fig. 30 Velocity, radiation resistance and sound power level of piston diaphragm

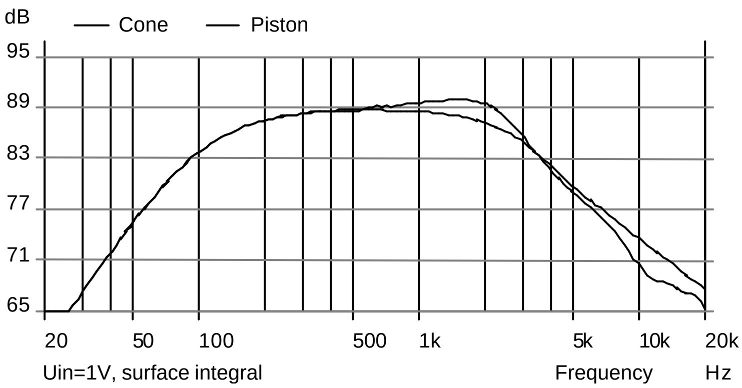


Fig. 31 Sound power level of piston and cone diaphragm

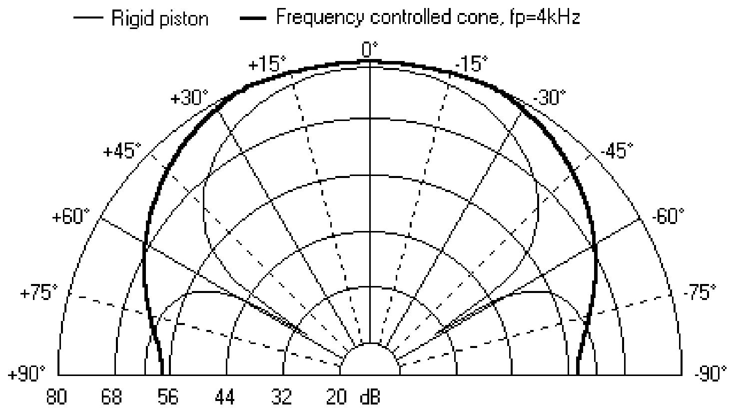


Fig. 32 Directivity pattern. Rigid cone and frequency controlled cone

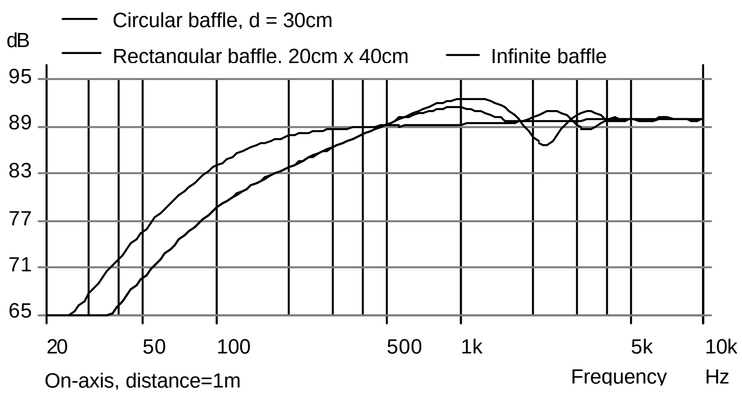


Fig. 33 SPL. Single driver mounted in the center of a circular and a rectangular baffle (1:2).
(piston and voice coil reactance switched off to demonstrate the effect)

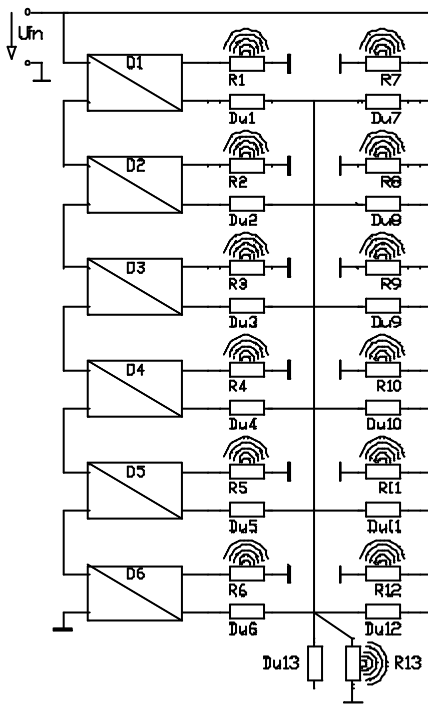


Fig. 34 Equivalent circuit. Drivers: D1...D12. Ducts: Du1...Du13. Radiation impedance and radiation: R1...R13

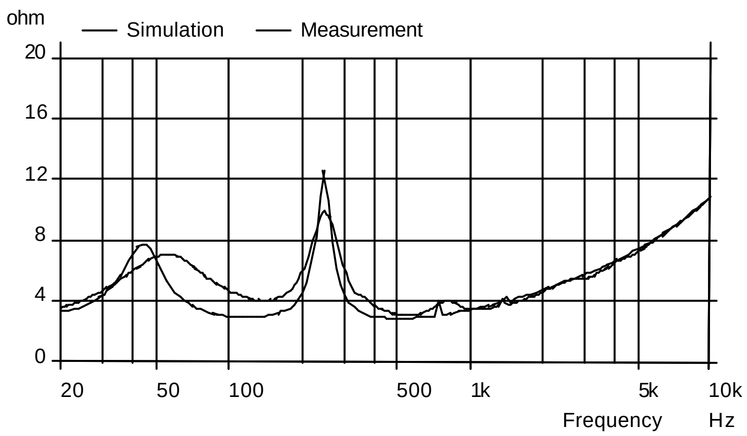


Fig. 35 Electrical input impedance, measurement and simulation

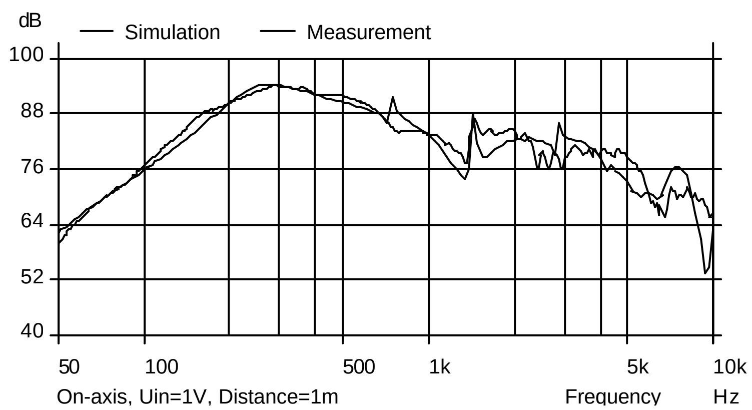


Fig. 36 On-axis SPL of dodecahedron loudspeaker, simulation and measurement

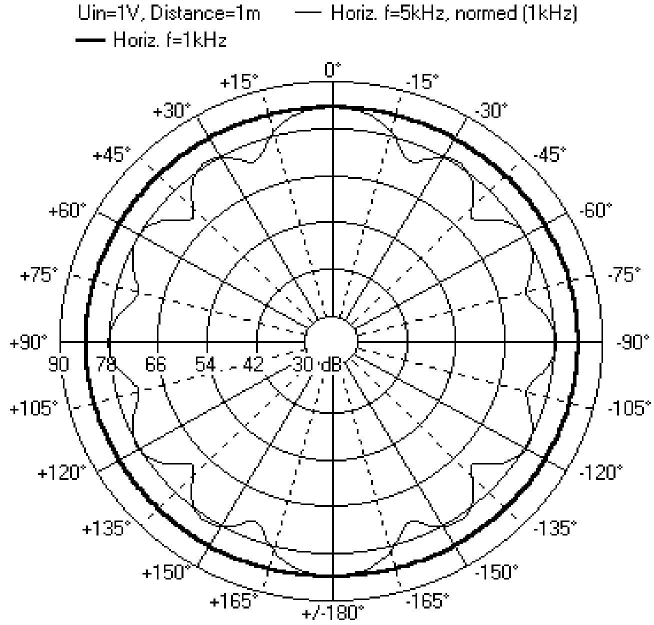


Fig. 37 Directivity of dodecahedron loudspeaker

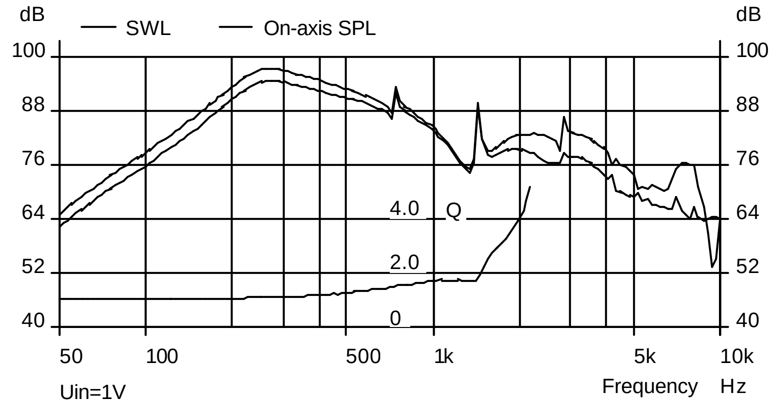


Fig. 38 Sound power level (SWL), Sound pressure level (SPL) and directivity factor (Q) of dodecahedron loudspeaker

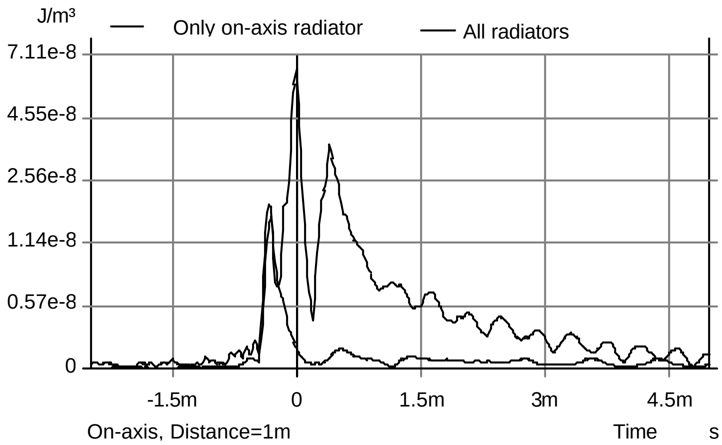


Fig. 39 ETC of SPL. Radiation of a single driver and of multiple drivers