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AKABAK (R) - AN ELECTROACOUSTIC NETWORK SIMULATOR

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1. INTRODUCTION

AkAbak is an abbreviation for "Akustik Abakus". This software simulator, running under Windows, calculates and displays transmission characteristics of electro-mechano-acoustic structures.
AkAbak is especially intended for the investigation and design of loudspeaker systems. AkAbak simulates a complete system: from the voltage source through to any listening point - including all filters, networks, radiation elements and the nearby radiation environment.
The flexible concept of AkAbak makes it possible to use the program also for a wide range of other mechano-acoustical applications such as microphone design, analysis of dampers, sirens etc.

The philosophy behind the program is to provide a simple, easy-to-use framework, even for complicated and extensive systems.
To be able to describe the device under investigation by means of AkAbak, the structure must be first separated into a set of lumped elements and one-dimensional waveguides. After these components have been carefully connected up, the program is able to build and solve the corresponding network matrix.
At the electrical side this separation-procedure is usually a simple affair, especially in the low frequency range. But in the mechanical and acoustical domain it can be difficult and in some cases impossible to describe complicated physical relations just with elements from the one-dimensional and linear world. The main problem here is that the wavelength is in the range of the dimensions of the components.

Due to its one-dimensional approach, there are limits to AkAbak's accuracy of simulation. It is precise enough to provide good and clear results for many common problems. While, for very complex structures, a good estimation can be achieved.
In any case, it is stimulating to work with the modules and usually this leads to a sound understanding of the physics of the investigated object.

2. THE PROGRAM

The AkAbak program provides an integrated development system for analysing and designing electroacoustic devices. It comprises a comfortable script-driven input system, powerful analysis and synthesis procedures and a wide range of output and display facilities. Additionally, a set of practical tools are implemented which are very handy for designing loudspeaker systems.

2.1 Network analysis

The components are wired with a node-based system, called a network. Being based on linear system theory, the analysis is carried out in the frequency domain by solving the node-potential-matrix.
Ten independent networks with a maximum of 54 nodes can each be solved simultaneously. Additionally, abstract filter elements can be connected with the network to form, for example, feedback loops.

The simulator input for the network components is organised as a script which can be edited with the in-built word-processor. For the simulation, the script is interpreted and compiled.
The syntax of the script is designed in such a way that on the one hand the specification is as comfortable and safe as possible and on the other hand it is self-documenting.

2.2 Filter synthesis
In AkAbak the filter element is the centre-point of an extensive and powerful range of functions related to filter analysis and synthesis. The feedback-free filter element is principally a rational transfer function of maximum 30th order. There are features for creating, scaling, transforming, decomposing and multiplying out transfer-functions.
The filter-element forms the basis of procedures for synthesising passive and active filter circuits.
The passive synthesising procedure creates ladder networks including coil losses of low pass, band pass and high pass transfer functions up to 30th order. All pass circuit can be created in two versions.
The active synthesising procedure uses the AkAbak function of decomposing transfer functions. A wide range of first and second order filter block-circuits with operational amplifiers are offered.

2.3 Parameter determination
AkAbak's components are designed to allow parameters to be entered from normal datasheets. The model can be extended by adding parameters to increase its accuracy.
AkAbak offers also several tools for determining parameters from measurement curves by the complex least square error method.
In addition, there are generators for filter-alignments and bass loudspeaker design, which create lists from which you can choose a specific alignment.

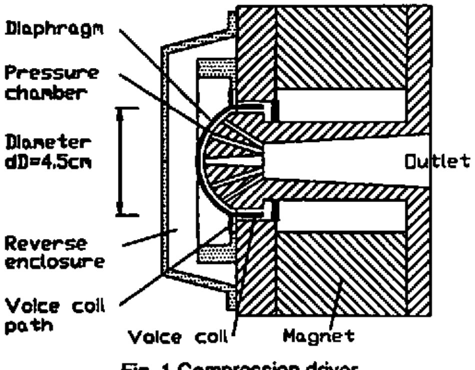

Fig. 1 Compression driver

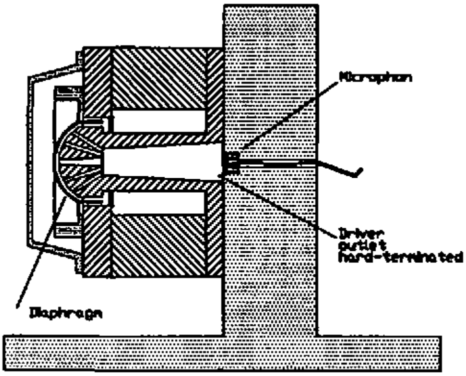

Fig. 2 Plate

3. EXAMPLE

An example will illustrate how the simulation is carried out.

3.1 Compression Driver

Let us, for an example, investigate a conventional mid-range compression driver as sketched out in Figure 1. Because the compression driver is a four-pole, we have to calculate and measure curves for both the electrical and acoustical sides. Across the electrical poles we simulate and measure the driving point impedance and at the acoustical side we determine the sound pressure. We will simulate and measure the driver under for different acoustical radiation environments to evaluate and verify the simulation.

3.1.1 Plate mounting. We close the output aperture of the driver and measure the sound pressure directly at the closing plate as sketched in Figure 2

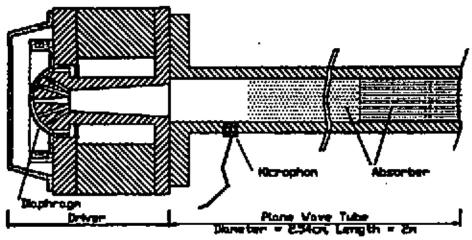

Fig. 3 Plane wave tube

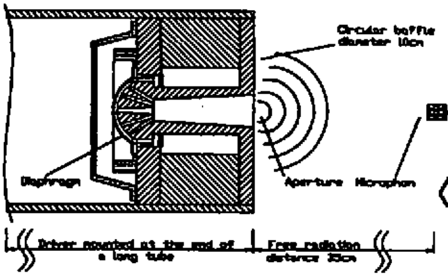

Fig. 4 Free radiation

3.1.2 Plane wave tube mounting, Figure 3. The acoustical power generated by the transducer is conducted to and absorbed by the so-called plane wave tube. This special tube is reflection free and offers a radiation impedance proportional to a plane wave. The sound pressure is measured at the entrance of the tube.

3.1.3 Free radiation, figure 4. The output radiates into free space. The microphone is placed 35cm in front of the aperture. The latter is placed in a circular baffle of 10cm diameter.

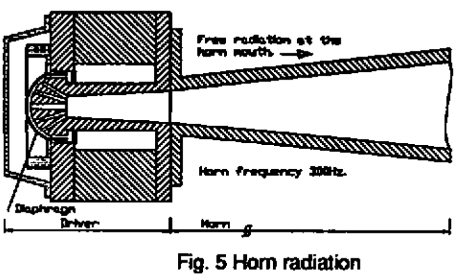

Fig. 5 Horn radiation

3.1.4 Horn-mounting, Figure 5. The sound wave is radiated into free space by a radial midrange horn. The microphone is placed in front of the unbaffled horn at a distance of 95cm measured from the flange.

3.2 Modelling

The concave diaphragm of our compression driver radiates via a thin layer of air into a duct system at whose other end the horn is mounted (Fig. 1). Due to the diminishing cross section, air particles are accelerated to enhance the efficiency of the transducer.

The force of the voice coil drives the diaphragm at the outer rim. At the junction of diaphragm rim and voice coil former is fixed the suspension. The suspension of most compression drivers with a concave dome functions itself as a diaphragm radiating into cavities which are acoustically connected to the compression chamber. These cavities along the voice coil former are called the voice coil path and form an impedance, to which the system response is highly sensitive. The reverse side of the diaphragm is loaded by a small closed enclosure.

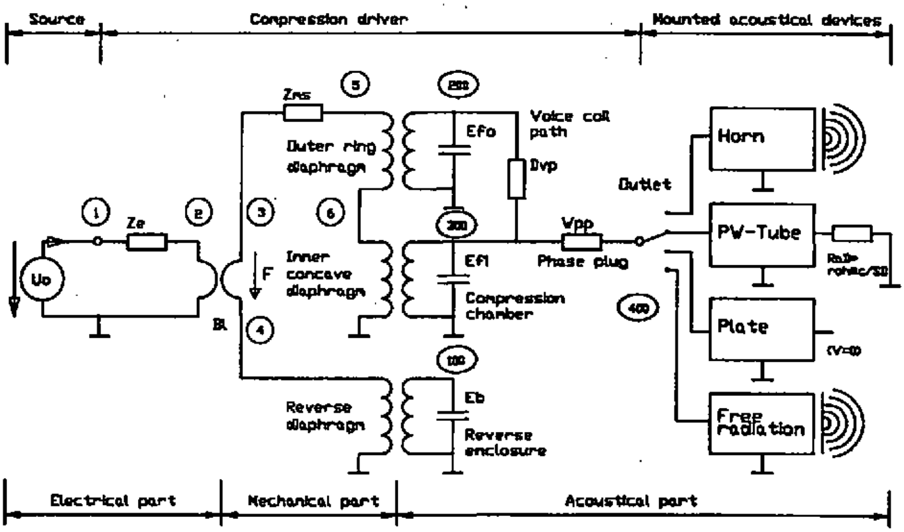

Fig.6 Equivalent circuit

To model the driver we need components from the electrical, mechanical and acoustical domain. Figure 6 is the equivalent circuit. The centre-part of the schematic comprises lumped elements of the transducer. To the left we have the driving voltage source and to the right the four radiation devices which can be mounted at the acoustical output of the driver.

The force generated by the magnetic fields of the voice coil and the permanent magnet is symbolised by the gyrator element. The voice coil impedance Ze is resistive at low frequencies. At high frequencies not only the imaginary part of Ze becomes inductive but also the real part rises due to eddy currents in the magnet pole piece.

```
| AkAbak example script - Compression driver

Def_Const
  {  Ree=13;       |Voice coil resistance [ohm]
     Le=0.5e-3;    |Voice coil inductance [H]
     Bl=8;         |Motor conversion factor F=Bl*i [Tm]
     Mms=1.1e-3;   |Mechanical mass [kg]
     Rms=0.4;      |Mechanical resistance [Ns/m]
     Cms=140e-6;   |Mechanical compliance [N/m]
     dDi=4.5e-2;   |Diameter centre diaphragm [m]
     dDa=5.3e-2;   |Diameter outer diaphragm [m]
     Hi=1e-3;      |Distance between diaphragm and phase plug
     Ha=13e-3;     |Eff. height of outer chamber under the ring
     ds=0.2e-3;    |Slit between voice coil and magnet
     ls=15e-3; }   |Length of voice coil path

System 'Windermere'
  |Voice coil  (frequency-non-linear resistance and reactance)
  Impedance  'Ze'  Node=1=2  Z={ Ree*(1 + f/50e3) + j*(w*Le)^0.6; }
  |Motor
  Gyrator  'Cyl'  Node=2=0=3=4  Bl={Bl}

  |Reverse side with enclosure
  Coupler  Node=4=0=100  dD={dDa}
  Enclosure  'Eb'  Node=100  Vb=120cm3  Qb/fo=0.1

  |Mechanical part (Mms frequency depending, cut-off at 5kHz)
  Impedance  'Zms'  Node=3=5
    Z={ wo=2*pi*5000;  Zms=Rms + j*(w*Mms/(1 + w/wo) - 1/(w*Cms)) }

  |Outer diaphragm (ring) with cavity
  Coupler  Node=5=6=200  SD={ pi*(sqr(dDa/2)-sqr(dDi/2)) }
  Enclosure  'Efo' Node=200  Vb={ SD=pi*(sqr(dDa/2)-sqr(dDi/2));  Vb=SD*Ha }

  |Centre diaphragm with compression chamber
  Coupler  Node=6=0=300  dD={dDi}
  Enclosure  'Efi'  Node=300  Vb={ SD=pi*sqr(dDi/2);  Vb=SD*Hi }

  |Voice coil tunnel between outer ring cavity and compression chamber
  Duct  'Dvp'  Node=200=300  SD={ U=pi*dDi; SD=U*ds }  Len={ ls }  QD/fo=0.01

  |Horn inside compression driver
  Waveguide  'Wpp' Node=300=400  dTh=1.5cm  dMo=2.24cm  Len=4cm  Conical

|Different acoustic devises (switch not used elements off)

  |1.Free radiation from the end of a long tube
off  Radiator  'R1'  Node=400  Def='W1'  dEdge=10cm

  |2.Plane wave tube
off Duct 'Du2'  Node=400=410  dD=1in  Len=3.8cm
off AcouMass  'AM1'  Node=410=420  Ma=10kg/m4
off AcouResistance  'AR1'  Node=420=0
    Ra={ roh=1.187; c=343.3; dD=2.54e-2; SD=pi*sqr(dD/2); Ra=0.6*roh*c/SD; }

  |3.Output closed
off AcouResistance  'AR1'  Node=400=0   Ra=1e15Ns/m5

  |4.Radiation via radial horn
  Horn  'H1'  Node=400
    T=1 dTh=1in  WMo=53.5cm  HMo=23cm  Len=47cm
    HArc=11cm  LenTh=10cm
    WEdge=59cm  HEdge=27cm
```

Fig.7 Equivalent AkAbak script

The mechanical part is dominated by the fundamental resonance frequency formed by the suspension compliance and the mass of the vibrating assembly. Further there are more or less strong eigen-frequencies of the mechanical structure. It is possible to model some of them with AkAbak but for the sake of clarity this is omitted here.
We would like our mechanical impedance Zms to include the so-called mass reduction of the diaphragm. At high frequencies the acceleration is so strong that only parts of the diaphragm are able to follow the imprinted force.

The transformation from the mechanical to the acoustical domain is performed by the diaphragm, symbolised by the coupler element. The front side of the diaphragm has to be split into two areas. The concave side of the dome radiates directly into the compression chamber. Part of the suspension ring radiates into the voice coil path. The reverse side of the diaphragm is coupled to the reverse enclosure.

In the acoustical domain the cavities are modelled by acoustical compliances. Passages where standing waves are expected are formed by waveguides. For the sake of clarity we are only including the most important modules in the script. We do not include resistive losses or dissipative Helmholtz resonators, for example.
Here the phase plug model, which is in reality a sophisticated duct system, is formed by a simple waveguide with variable cross section.
The acoustical outlet of the driver must be loaded by one of the radiation conditions already mentioned.

3.3. Entering the parameters
All elements of the schematic of Figure 6 have to be entered together with the parameter of each component. Further the components must be wired in such a way that the program can build and solve the node potential matrix of the network.
The AkAbak script shown in Figure 7 is equivalent to the schematic of Figure 6.

3.4 Simulation
As already mentioned we have to investigate both the electrical input and the acoustical output, because our object is a four pole. All the following diagrams display sets of two curves. The broader line is the simulated curve and the thin line shows the measurement. From the electrical impedance response the real- and imaginary components are shown. The unit is "Ohm". The acoustical pressure curve is normed to the threshold of hearing and is displayed as level. The unit is "Decibel".

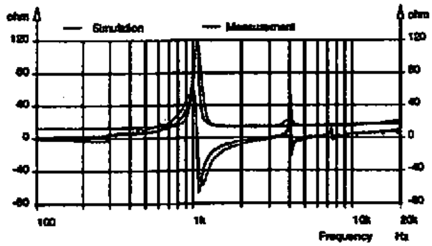

Fig. 8 Electrical impedance, real and imaginary: Plate

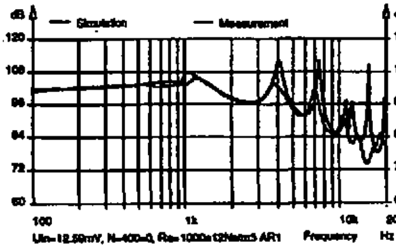

Fig. 9 Sound pressure level at the plate

3.4.1 Let us consider the plate mounting. The normal velocity is zero at the plate because of the hard termination.
The impedance curve of Figure 8 shows clearly the fundamental resonance at 1kHz. The peaks at 4kHz and 8kHz belong to the standing wave pattern of the phase plug duct. The measurement indicates stronger losses than the simulation. At very high frequencies it is obvious that not only the imaginary part is rising but also the real part as a result of the voice coil impedance.
The associated simulation and measurement of sound pressure at the plate demonstrates that this response is proportional to the input current. (Figure 9).

In comparison to the plate mounting all other radiation environments reveal a strong eigen-vibration of the diaphragm. We can see the effect in sharp up and downs of the impedance measurement curves.

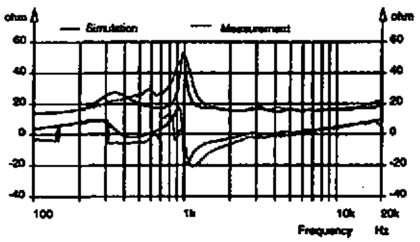

Fig. 10 Electrical impedance, real and imaginary: Plane wave tube

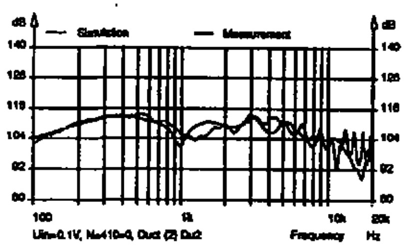

Fig.11 Sound pressure level in the plane wave tube

3.4.2 Without changing the driver parameter, we mount the driver on the plane wave tube.
At this stage it is not clear, why the imaginary part of the measurement impedance curve jumps so abruptly in the lower frequency range.
The sound pressure curve is approximately proportional to the velocity curve because of the more or less constant acoustical impedance with which the driver is loaded.

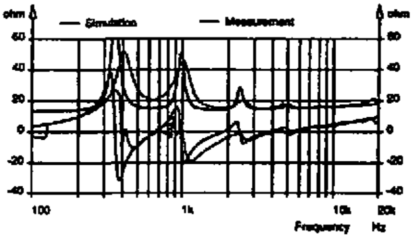

Fig. 12 Electrical impedance, real and imaginary: Free radiation

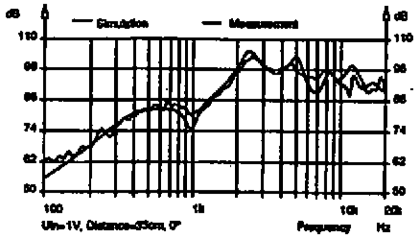

Fig.13 Sound pressure level: Free radiation

3.4.3 The free radiation environment condition is obtained when no device is mounted on the driver. The aperture is in the center of the circular magnet, which has a diameter of 10cm. To simulate this condition we apply a radiator element to the output port of the driver. The radiator implements the radiation impedance including diffraction and reflection effects and is also the means for radiation into free space. Some of the rippling in the upper frequency range of the SPL curve shown in Figure 13 is caused by diffraction of the free radiated sound at the circular baffle edge.

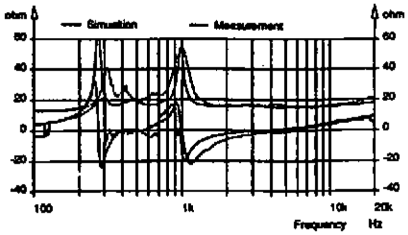

Fig. 14 Electrical impedance, real and imaginary: Horn radiation

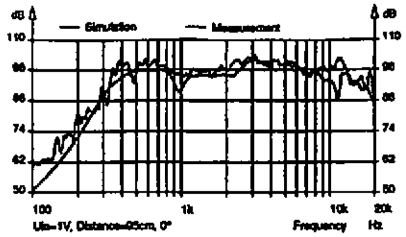

Fig.15 Sound pressure level: Horn radiation

3.4.4 The last loading example is radiation into free space via a horn for which the driver is originally constructed. The horn is a typical 300Hz-radial horn with an exponential flare.
AkAbak offers a special horn element, which comprises a four pole waveguide and a spherical radiation source. The transmission and radiation pattern is thus also dependent on the horn form. In this way not only the on-axis response can be calculated but also a good evaluation of the directivity for a wide range of conventional horn forms can be achieved.

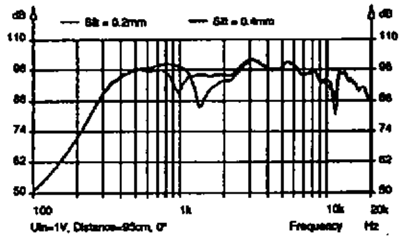

Fig.16 Sound pressure level: Horn radiation. Doubling the width of the voice coil path

3.4.5 For example one of the most surprising details we can learn from our compression driver model is the high sensitivity of the response to the dimensions of the voice coil path. Figure 16 demonstrates the effect of doubling the width of the slit between the voice coil former and the magnet.

4. Conclusion
AkAbak is an electroacoustic network simulator using lumped elements and one-dimensional waveguides. The tools for filter analysis and synthesis, parameter determination and bass loudspeaker design make AkAbak the ideal platform for the investigation and design of loudspeaker systems.
The program philosophy offers great freedom in modeling, so that even unusual and complicated designs can be simulated.

In the above example we have demonstrated how a complex structure such as a compression driver can be modeled and we have made comparisons with equivalent measurements.

In conclusion, I would like to express my thanks to Prof. R.H. Campbell, Steve Wood and Philipp Göppl for their help.