

SPEAK

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Getting Started

The Basics

Background

SPEAK is a sophisticated loudspeaker analysis program aimed at the knowledgeable professional or advanced hobbyist. This is not to say that the novice cannot use it, indeed a novice will learn very quickly what could take years of lab testing to learn, but it does assume a basic knowledge of loudspeaker design and acoustics. However, once this fundamental hurdle has been surpassed, *SPEAK* offers an extensive modeling capability well beyond what is commonly available in the marketplace today. Its capabilities in polar response simulation, new enclosure and driver designs, harmonic distortion simulations, and waveguides, just to name a few are unsurpassed. It is extremely fast, due to the use of an ultra efficient FORTRAN 95 compiler in the numerical engines. (FORTRAN 95 is the compiler of choice for high-speed execution of complex matrix arithmetic).

SPEAK is the evolution of more than thirty years of loudspeaker simulation experience by its author Dr. Earl R. Geddes - a well known authority in loudspeaker and enclosure design. Its predecessors *Speak* (DOS based) and *WinSpeak* were little known but widely used simulation programs. The current version is a complete rewrite utilizing a new matrix approach to calculations and the full use of 32 bit code throughout for flexibility and speed.

The philosophy of *SPEAK* is to be the best practical simulation possible. The author uses it on almost a daily basis to design loudspeaker systems for his clients. In this scenario it is more important to be accurate than easy to use. Ease of use is important, but accuracy is king. For this reason *SPEAK* can be very complicated and difficult to understand in some areas (arrays, motor structure design, etc.), but this complication is necessary for accuracy. If these complications are avoided, however, it can be very easy to use.

This manual will take you through some of the fundamental uses of *SPEAK* and give you a flavor for its capabilities. However, it would be far too complex a task to attempt to teach the user all aspects of loudspeaker design. Although, the capability to learn almost all there is to know about loudspeaker design is available within this package.

Basics

The core of *SPEAK* is the *Project* file. The *Project* file contains information on the *System Definition*, *Array definition*, *Constants*, and *Plot* specifications.

The numerical engine evaluates the unique combinations of the parameters defined by the *Project* file and associated *Parameter* databases. There are three *Parameter* databases: *Driver*, *Enclosure*, and *Crossover*.

The results of an analysis are then displayed on the *Plot* form, which has many options.

A First Project

Start by opening the *Project* file "*Default*". This project must be always be present in the directory that *SPEAK* resides in because new projects are created based on this project. Editing its features yields the "default" project settings. Do not make project files read only as *SPEAK* will refuse to open them.

A child window called the *Plot* form is then created after running the analysis.

Selecting any of the buttons on the *Plot Toolbar* (bottom of the new *Plot* child form) to show the sound pressure , cone displacement , input impedance , voltage  or current  into the system or the “port” displacement or velocity . In *SPEAK* a “port” follows the dictionary definition: “a hole in an enclosure”. This hole can contain a *duct*, a *passive radiator* or an *Acoustic Lever*TM placed in it.

Press the *Harmonic Distortion* button . There is no change in the plot because the levels of the harmonics are too low. Drag the mouse over the plot with the left button pressed to note the level of the curve. Press the “down” arrow on the keyboard to change to “curve 2”. Note the level of the second harmonic in the upper left corner.

Now open the *Project* form with  on the *Main Toolbar* (or with the menu item *Database/Project*, or *ctrl P*). Change the *Source Voltage* to 10 volts and press *OK* to close the *Project form*. Update the plot with the  button and re-select the harmonic distortion mode. Note the presence of the second and third harmonics.

Select *Iteration (Design sensitivity) Mode* with the  button on the *Plot Toolbar*. The *Design Sensitivity* form opens. Select “*Driver*” from the upper list box and “*Cone Mass*” from the lower list box. Select *OK*. The curves now show the system with the driver’s cone mass shown at normal and plus and minus 5 and 10% of the normal values.

Select *Iteration* again. Select “*Driver*” and “*BL*” from the drop down selection boxes. Select “down only” and change the “value” to 20. Press “*OK*”. A 20% reduction in driver BL product is now shown as the five curves on each plot. The labels in the upper right show the values (always in MKS units). Drag the mouse with the left button down while over the “*labels*” to move them out of the way. Double click on them to remove them.

Select the *plot parameters* form by right clicking the mouse on the plot area, change the values and press *OK*.

Re-open the *plot parameters* form and select the *X-axis* by clicking the *X-Axis tab*. Change the values and press *OK*. The plot colors can also be changed from this form by clicking on the *colors* tab. Click on a line number or the background to change the line colors or the line colors. The vertical line is the cursor color. The line data can also be changed (more on this can be found in the manual.)

Select “*Print*” from the “*File*” menu, or the *Main Toolbar*, to print this curve (a color printer is preferred).

View the *Polar* response by selecting the  button from the *Plot Toolbar*. This plot will not be very instructive to begin with. That is because the polar radiation pattern is omni-directional and there is nothing to show. Drag the slider at the side of the graph to some higher frequency where the driver is not radiating omni-directional. The *Polar* curves are plotted as pressure isobars. The circumferential circles represent angles away from the axis. The radial lines represent angles around the central axis. On the right side of this plot is the standard polar response diagram plotted as magnitude (normalized to the maximum value) versus the angle off the axis - theta.

Select the *Project Form* once again. In the box labeled “*Orient Theta*” change the value to 30 “degrees”. This will point the driver upward thirty degrees. Close the *Project Form* and “*update*” the project. Now select the *polar plot* and find a new frequency of interest. Note that the driver is pointed upward 30 degrees as evidenced by the new pressure contours.

Close *SPEAK* by selecting *Exit* in the *File* Menu.

Multiple Subsystems

Start *SPEAK* once again and open the project “*Dipole*”. Select the *Project Form* and note that there are two subsystems defined. They have the same drivers in the same enclosures (very large box) but one has reversed phase (the “*Reverse Polarity*” button is selected). Note that the two drivers are at different locations on the baffle separated vertically by 16 cm.

Open the *Plot Parameters* form (Left click on plot) and make sure that the frequency response has three decades with the lowest at one. Note that the toolbar shows that multiple subsystem plots are now available. This is evidenced by the symbol . When this symbol appears the project has multiple subsystems declared. Normally each subsystem is plotted individually on each plot (SPL, displacement, voltage, etc.). In the present case the two subsystems are identical, except that they are at a slightly different position and out of phase from one another - the two curves overlap each other.

At this point set the *Summation Mode* by pressing . The subsystems are now summed together at the field point location resulting a classic dipole response.

Examine the polar plot by pressing the *polar* button. Note the pattern as the frequency slider is moved to higher frequencies. For this example the polar resolution is probably too low. Select *Constants* from the menu to bring up the *Constants* form. Select *Numerics* and increase the *Theta* and *Psi* resolutions (smaller numbers). The data will update once you close the Constants Form. Note the improved resolution but *much* slower execution times.

Complete Systems

Open the project “*Two-Way*” and view the *Project* form (). Note the use of a woofer and a tweeter defined with different driver locations as subsystems one and two. You can view the data that make up this system by selecting the *Driver*, *Enclosure* or *Crossover* buttons on this form. Remember that the databases are in *Database mode* when viewed from the project form and that changing the data OR selecting another record will change this data in the project itself.

Initially the *Plot* form shows the woofer and tweeter responses shown as two separate responses – axial only. Selecting *Single Sub-System* mode by pressing the  button on the *Plot toolbar* allows you to looking at the responses of a single sub-system. You can select a Sub-system with the right and left arrow keys on either side of the *Single Sub-System* button. (Note that iterations are only possible in the *Single Sub-System* mode.) Compare the voltage seen by the two systems. It is identical. This is because the system is not “*bi-amplified*” as defined in the *project* form. When in *non-bi-amplified* mode the voltage supplied to subsystem one is used for all subsystems. This is, of course, logical since in reality if a single amplifier drove the system then all subsystems would have to see the same voltage.

Turn off the *Single Sub-System Mode* by pressing the *Single Sub-System Mode* button once again. Now sum the two Sub-Systems by selecting the *System Summation* mode . Examine the polar response by selecting the *Polar Response* plot button. Find the crossover frequency (700 - 800 Hz.) on the slider bar. You can single step through the frequencies with the left and right keyboard keys. Now select the *impedance Plot* and note that the impedance is now the combined impedance of the system.

Select the *project* form and change the *polarity* of the tweeter to normal. Update the data and re-examine the polar response at crossover.

There are many other features in *SPEAK* which are highlighted in the following sections of the manual, but at this point you have an understanding of the basics of *SPEAK*. Enjoy!

Projects

A project is composed of *Subsystem Definition*, *Array Definition*, *Constants*, and *Plot* specifications. This data is contained in *Project* files on the users hard drive with the suffix ".prj". There is no limit to the number of project files that a user can maintain.

New projects are created with the *File -New Project* menu item. A project can be saved with the *File - Save As ...* menu item. *Projects* are opened with the *File - Open* menu item. The last several user projects are available in the menu item *File - Recent Project*.

Each new project is based on the "Default.prj" file which was installed in the *SPEAK* folder when the program was setup. If this file becomes damaged or corrupted the user can no longer create new projects. To correct these situations replace this file with one downloaded from the web site.

Open

Projects can be opened from mass storage, or created new. From the *File* menu topic select *Open Project* to pick from existing projects or *New Project* to create a new project based on the "Default" project.

Creating

Selecting the menu *File - New Project*, creates a new project. Projects are created with the attributes of the "*Default.Prj*" file in the users *SPEAK* directory. In other words the project "*Default.prj*" is the template for all other projects.

Save Project As ...

Projects can be saved with any name by selecting the menu item *File* and the selection *Save Project As ...*.

Export Project Parameters

The item will create an **.spd* file which contains all of the parameters for a project – unlike the project file itself. This is useful for sharing projects with others who may not have all of the records in a given project in there databases. When a project is loaded which is cannot find specified records in its parameter databases it looks for an **.spd* file of the same name. If found it will create this missing records from the data in the *spd* file.

Import/Export Data

Data can be imported into *SPEAK* in numerous formats;

- User format – an ASCII, comma delimited file that should be identical to one written with the "Export Data" option. Export a project to view this format. There is a 13-line header followed by one line per frequency data point with complex pressure, displacement, voltage and current on each line. The header will be ignored if the lines are blank, but there must be 13 lines. Any data can be input into *SPEAK* if put in this form.
- Laud – frequency data and impedance data can be read in. Remember that this pressure data is un-scaled.
- Audio Precision – standard data written in this format can be read into *SPEAK*.
- Particular subsystem data can be exported with the *Export Data* selection. A message box will appear which asks which subsystem to export followed by the name

of the file to write. The file is an ASCII file which can be read into any other program, i.e. Excel, etc.

Print

The menu item *File - Print* prints the current plot on the windows selected printer. Color printers are strongly recommended as it can be hard to decode the lines in black and white.

Databases

General

A fundamental part of *SPEAK* are its parameter databases. *Project* files contain *Systems Definition*, *Array Definition*, etc.- i.e. systems level data. In addition there are three parameter databases, *Driver*, *Enclosure* and *Crossover* – i.e. the components of the system. These databases can exist on a central server or in one or more local directories and there can be many different versions. The databases are all Jet 4.0 compatible and can be administered at a higher level with Microsoft Database tools such as *Access* or *Visdata*.

Until a project has not been selected (no *Plot* form is showing) a menu item *Databases / Database Location* is available. Selecting this menu item will bring up a form which will show the locations/ directory of the currently selected parameter databases. They can be changed by clicking on the database location label that will bring up a file locator box to allow for the selection of a new database location.

Database records can be transferred from one database to another by selecting the *Databases / Download External Data / other* menu item. A form will come up asking you to find the source database from which to download the data. The source database will then be connected allowing you to find the record that you want to download. Select it and it will be downloaded to the current database.

Driver data can be downloaded from the *Global_Driver* database by selecting *Databases / Download External Data / Driver*. The *Global_Driver* database contains over 1400 drivers by many manufacturers. No claim is made to the authenticity of this data – it is public domain information. Drivers can be added to the *Global_Drivers* database with **Visdata**, which can be downloaded from the web site.

Access to each of these databases has a common interface functionality. Different ordering of the records can be selected with the menu item *Databases / Sort Method*. The methods are by *date last modified* and by *name*, both can be *ascending* or *descending*.

There is a database form for each of the parameter sets. When opened from the *Project* form or the Menu item *Databases* the database forms operates in *database mode*. In this mode each database form has an *Add Record*, *Delete Current Record* and an *OK* command button. Selecting the *Add Record* button creates new records, which can only be done in database mode. The new record is created as a clone of the current record. The user is required to specify a unique name for the new record.

If the *Delete* button is enabled then the current record can be deleted by pressing it. If a record is locked then it cannot be deleted or modified without first clearing the lock, which cannot be done by the user from within *SPEAK*. If modifying a locked record is desired then simply create a new record based on the locked record. New records are never locked. To lock a record right click on the database form and select “*Lock Record*”. If required a record can be unlocked by using an external database tool like *Visdata*.

In the database mode the record shown on the database form when it is closed becomes the new specification for that parameter set in the *Subsystem Definition*.

If the database is called from the *Toolbar* then it is always in non-database mode. In this mode only that record specified for the Current Subsystem is shown.

When the *OK* command is selected the form will close. If the data has changed the program will ask if you want it updated or not. If the record is *Locked*, then a message box will inform you that the data cannot be changed and is returned to its previous settings.

Sort options

The data can be sorted by “owner” or “type”. This option is selected by right clicking on a parameter form. The type differs with database, but the owner designation is the same. The owner can be declared at any time, but is initially set by the owner of the application (set at first startup and stored in the registry). Fictitious owners are possible to sort a database by a limited number of records.

- Drivers can be sorted by “Full, Woofer, Midrange, Tweeter”.
- Enclosures can be sorted by type “Closed, Closed Free, Ported, etc.”
- Crossovers can be sorted by type “Active Low pass, High pass, and Passive Low Pass, High Pass”.

Records are not always “tagged” when created (only Enclosures).

Name

The current record name is located at the top of the form between the record selection buttons (in database mode). This should be a unique name within the database. Most characters are allowed in names except @, \$, *, ", ', since these have special meanings to the database.

Record Select

The record selection buttons at the top of the form can be used to change the current record, whose name is in the center. A list box can also be selected for a direct selection of the record. These buttons are not available in the non-database mode of operation. The current record can be advanced to the first record "<<", the previous record "<", the next record ">", and the last record ">>".

Add New

New records can be added by pressing the *Add New* button. The new record will have the same parameters as the current record. The easiest way to modify a driver without losing the current driver data is to simply create a new driver based on the current one. This can be done for any of the databases.

Delete Current

The current database record can be deleted by pressing the *Delete* button. This button is only visible if the current record is not locked.

OK

The *Ok* button is used to accept the current data as shown. The record is recorded (as long as it is not locked) once the *OK* button has been pressed. If the record has been modified the system asks if you want to save the changed data to the database.

Description

The description field is a text string of length 256 characters. It is used to describe the record.

Date Created

The date created label show the date on which the current record was first created.

Date Modified

The date modified label shows the date on which the record was last modified.

System Definition

Project Definition form is the window into the *Project* file. A complete *Project* definition consists of the *System* description including the *System Array* definitions, the *Amplifier* type and output level, and *Subsystems* specifications including the *Driver Location*, the *Array Definition*, and the *Acoustic Loading* configuration.

The following picture shows the *Project* or *Subsystem Definition* form:

Project Definition - two_way

System
 Two Way demo system
 Created: 5/20/97 Modified: 1/26/01 System Array

Amplifier
 Bi-amped Voltage Current Source 10 Volts

Subsystems
 Sub1 Sub2 Add Delete
 woofer
 Driver JW Audio 12 JL Audio 12"
 Enclosure 22 liter box 22 liter closed box with
 Crossover Series L Series inductance
 Subsystem Array reverse polarity

Location
 Radius 10 cm.
 Psi angle 0 °
 Offset distance 0 cm.
 Orient. Theta 0 °
 Orient. Psi 0 °

Acoustic Loading
 Normal
 Baffle Admit 0 < 1.0
 Flat
 Enclosure Vol. 0 m³
 Leakage 0 Ohms

Close

Subsystem

Subsystems are defined on the *Project* form. There can be up to five subsystems. Each subsystem has its own driver, enclosure and crossover. Each subsystem also has its own specification of the *acoustic loading* condition, *amplifier source* level (if in bi-amped mode), *driver location* and pointing *Orientation*. It is up to the user to make sure that the location and acoustic loading specification is consistent

with the system being modeled.

Typical projects can include woofers, midranges and tweeters as subsystems, or numerous drivers acting as an array. Different subsystems can also be used to compare different enclosure, crossover, etc. designs in a single project.

Add

The *Add* button on the *Subsystem tab* is used to add a new subsystem.

Delete

Like the *Add* button the *Delete* button on the *Subsystem tab* deletes the current subsystem.

Driver

Pressing the button with the *Driver* record name on it brings up the *Driver* database in database mode. In this mode the data can be changed and the record can be changed. If the record is changed then this becomes the new "*Driver*" the current subsystem.

Enclosures

Pressing the button with the *Enclosure* record name on it brings up the *Enclosure* database in database mode. In this mode the data can be changed and the record can be changed. If the record is changed then this becomes the new "*Enclosure*" the current subsystem.

Crossovers

Pressing the button with the *Crossover* record name on it brings up the *Crossover* database in database mode. In this mode the data can be changed and the record can be changed. If the record is changed then this becomes the new "*Crossover*" the current subsystem.

Acoustic Load

The response of a loudspeaker system is dependent on the acoustic loading that the system sees. For example, in a very small enclosure like a car, the low frequency response will be accentuated by the static or volumetric mode of the enclosure (see Blind and Geddes "The Equalized Sound Power Automotive Sound System" AES Preprint #3601, presented at the AES Convention in October 1984).

There are Five loading conditions available:

Free Field

The Free Field Loading condition is an unrealistic condition since it is virtually impossible to encounter. Only very large anechoic rooms would meet this criteria. The more common condition is an *Half Space* (infinite baffle) ,*Finite Enclosure*, or *Un-baffled* loads.

Normal

Normal loading is the most common. It uses the radiation impedance of a piston in an infinite baffle. The *Normal* option also has the capability to calculate the effect of a baffle impedance. Input as a specific acoustic admittance (inverse of acoustic impedance) there are several options for its frequency variation. A value of zero represents a rigid baffle. A value of 1.0 represents complete sound absorption at the baffle, an unrealistic condition. In fact, seldom can the admittance exceed about .5. The admittance can be "flat", independent of frequency. It can "rise to 1kHz." which represents a likely scenario since no absorption can occur at very low frequencies. In this later case there is no absorption at

low frequencies and it increase to be about one half its final value at 1 kHz. after which it asymptotically approaches the specified value. For example, a value of admittance of about .4 "rising to 1 kHz." would represent a sound absorption blanket about 6 inches deep on the baffle face. The other options work likewise. They are "fall to 1 kHz.", which is unrealistic, "rise to 100 Hz." which would be a very thick blanket, and "rise to 10 kHz." which would represent a typical thickness blanket.

Corner Loading

The *Corner Load* option represents the loading of a loudspeaker placed in a corner of a room (ignoring of course the other seven room corners!). The load factor is eight times the sound radiation, falling to no effect as the beam width of the system collapses. This choice is an approximation of the corner load effect that, in reality, can be quite complex.

Small Room

Small Room loading can be quite substantial at very low frequencies for small tightly sealed rooms. Virtually the only place that this effect occurs is in vehicles (with the windows up!), but can sometimes occur in specially sealed small listening rooms. The volume of the room must be specified as well as the *Room Leakage*. Small values of leakage will negate the effect very quickly. Only the lowest (static pressure) mode of the room is represented in this calculation.

Enclosure Volume

The *Enclosure Volume* is measure in m³. It represents either the volume of the finite enclosure or the volume of the small room depending on the loading option selected.

Room Leakage

The *Room Leakage* is measure in Acoustical Ohms. It needs to be large for the small room effect to occur. Nearly all small rooms have some leakage (or we would suffocate!)

Finite Enclosure

Closely associated with the Acoustic Loading specification is the choice of enclosure as "Baffled" or "Free". Only in an anechoic room is the infinite baffle response even close to being correct for a real situation. A real enclosure in a real room is very complex. However *SPEAK* does do a calculation of the effect that a spherical enclosure has on the system response. This is useful since all enclosures have diffraction effects, but the real diffracted field can be difficult to predict. One should always try and minimize cabinet diffraction since there is no such thing as "good" diffraction. Thus the logic is this: If one minimizes enclosure diffraction by rounding the enclosure edges (know to reduce diffraction) then the limit of this process is a sphere. The sphere then represents that enclosure for which the diffraction has been reduced to its lowest feasible amount. The spherical enclosure option then gives one an idea of the response of the system in "free space" with all correctable diffraction reduced as low as possible.

It is important to know the dimensions of this sphere. Its radius is determined from it volume. Its volume is the sum of all of the volumes specified in the enclosure record OR the value of *Enclosure Volume* whichever is larger. This allows for any size sphere greater than the minima volume that can contain the specified enclosure volumes

The radiation load is calculated as that of a vibrating cap in this sphere and the polar radiation pattern is that of this same cap. In this configuration the loading transition from free field to half space, which occurs in reality, is accounted for. The theory behind these calculations can be found in Morse *Vibration and Sound* as well as several other texts. It is rather complex involving an infinite series to perform

the calculations. The number of terms used in this calculation can be set in the *Constants* form.

Dipole and Bipole

Now that the spherical enclosure series summation has been explained it is possible to define the use of *Dipole* and *Bipole* radiation. In the spherical model odd terms represent non-symmetric output while even terms represent symmetric output. Real sources generally contain all terms. However, if one deletes alternate terms, all odd or all even, and doubles the sum of the rest then in effect a symmetric or anti-symmetric enclosure is created. A *Dipole* enclosure has identical sources on the front and back but vibrating “out-of-phase” – anti-symmetric, while a *Bipole* enclosure has identical drivers on the front and back “in-phase” - symmetric. This is accomplished quite simply by considering only alternate terms in the summation. The total number of terms has a pronounced effect on the results and can be very unrealistic for small numbers of terms. Keep the number of terms above ten unless you are comfortable with spherical radiation problems.

Naturally, the above capability can only be used for spherical enclosure types.

Source

The *System* Definition includes the type of amplifier driving each transducer. The options are:

Bi-amp

Bi-Amplification means that each driver has a separate amplifier. This option is necessary to use different electrical filters for each driver since this is the only way to achieve this feat in practice. When not chosen all subsystems see the active filter of the first subsystem.

Source Type

This option selects a voltage source or current source amplifier for the current subsystem. A voltage source has a drive voltage that is independent of the load. A current source has a drive current that is independent of load.

Source Level

The *Source Level* is in Volts for a voltage source and in Amps for a current source specification. It is the RMS level of the input to the system.

Driver Location

Each driver in the system must have a specified location. There are three parameters that define this location. They are:

Radius

The radius is specified in meters. It is the radial distance in the plane of the baffle from some convenient point called the origin, i.e. the center of one of the drivers or the port. Since the port location is not specified it is assumed to be at the origin. Thus for the best accuracy, if a port exists use it as the origin and specify all of the drivers relative to this port. This is not a requirement however.

Psi angle

This parameter is measured in degrees and is an angle in the plane of the baffle. Since the location of the driver must be specified and the radius specifies how far out from the origin the driver is, this param-

eter specifies the angle that a vector of length *Driver Location Radius* makes from vertical. For example a driver located directly above the origin would have a Psi of 0 degrees, directly below would be 180 degrees, and to the right of the origin would be 90 degrees.

Orientation Angle Theta

A driver in *SPEAK* can be pointed in a non-axial direction. The amount that the driver is tilted away from the axis is set by the value of *Orient Theta* in degrees. *Orient Theta* sets the angle off the normal of the baffle in which the tilt takes place.

The algorithm used for this simulation is simple and accurate only for relatively small angle of less than about 30 degrees.

Orientation Angle Psi

A driver in *SPEAK* can be pointed in a non-axial direction. The direction that the driver is tilted away from the axis is set by the value of *Orient Psi* in degrees. *Orient Psi* sets the angle away from the vertical in which the tilt takes place.

The algorithm used for this simulation is simple and accurate only for relatively small angle of less than about 30 degrees.

Driver Parameters

A driver is basically composed of three parts, a resonant diaphragm and acoustic coupling volume in front of the diaphragm and a motor structure. These two coupled systems act under the influence of two basic influences on their parameters, heat and cone movement. There are numerous driver parameters that can be used to describe the performance of a driver. They are broken down into five main areas with each area a tab on the following form:

Parameter	Value	Unit
Motor BL	14.8	N / Amp
Resistance	8	Ohms
Inductance	1	mH
Semi-Induct.	0.1	Semi-H.

The tabs are listed below

Mechanical

There are six mechanical parameters:

Mass

Cone *mass* is specified in grams. It represents the entire moving mass of the cone excluding the radiation mass load. This distinction is important since the standard Thiele-Small parameters are defined with the radiation mass included in the moving mass of the cone. In *SPEAK* this can be a problem since this radiation mass is calculated for the specific system involved and added to the cone mass. Thus the electro-mechanical specification of cone mass should not include the radiation mass meaning that the radiation mass must be removed from the mass calculated from the standard Thiele-Small values.

This situation can be remedied by inputting Thiele-Small parameters into the Thiele-Small input option box. When this is done the radiation mass of a monopole is subtracted off of the cone mass when the electro-mechanical parameters are calculated. The Thiele-Small parameters should thus be determined in a closed box configuration. Consistent with the usual definitions the Thiele-Small values shown in *SPEAK* include the radiation mass – for a monopole.

Compliance

The cone *Compliance* is specified in mm. / Nt. In other words, the number of mm. the cone would move if a one Newton force were placed on it. The compliance is a dynamic quantity and the compliance measured with a static force can be very different than the dynamic compliance. This situation is not well handled in practice when measuring compliance. In *SPEAK* the dynamic compliance should be used. The compliance is the inverse of the stiffness.

Resistance

The cone *Resistance* is measured in kg. / s. which is also known as Mechanical Ohms. It is the ratio of the force on the cone to the cone's velocity. This value seldom exceeds ten or much less than one in MKS units. It represents the mechanical portion of the cone damping.

Area

The cone *Area* is measured in cm^2 . It is the actual radiating area of the driver. It is usually somewhat larger than the cone itself since the outside suspension (surround) radiates some sound. A rule of thumb is to add one half of the surround area to the actual cone area. It should also be noted that this area is the area normal to the cone motion **NOT** the actual area of the cone.

Depth

The cone *Depth* is used to simulate the effect on the radiation of the finite volume in front of the cone. This effect can be seen in the work of Geddes and Porter in "*A Boundary Element Approach to Finite Element Radiation Problems*" in JAES Vol. 35, #4 (April, 1987). An approximation of this effect is put into place if the *depth* > 0.0. This approximation is accurate for depths less than about one half of the radius of the cone.

Leakage

The cone *Leakage* is measured in $\text{kg} / (\text{sec m}^4)$ also known as MKS Acoustical Ohms. In a good driver it is a very large number. It represents the leakage past the driver components from the front to the back. Small values (high leakage) can have a profound effect on many types of enclosures.

Electrical

There are four driver electrical parameters:

Bl

The driver *Bl* product is measured in N. / A. It represents the coupling of the input current to the force on the cone. In a linear analysis it is assumed to be constant. It never is. In *SPEAK* the *Bl* product can be made non-linear.

Resistance

The coils electrical *Resistance* is measured in Ohms. It is the total resistance of the coil and lead-in wires. This value is the DC resistance and not the resistance measured at some higher frequency. These numbers differ because of inductive effects in the voice coil as noted in Semi-inductance.

Inductance

The coil *Inductance* is measured in mH. It is the total linear inductance of the coil. This inductor has the normal 90-degree phase rotation of a simple inductor. A real voice coil has both a simple inductance and a semi-inductance in series.

Semi-inductance

The Semi-inductance is measured in semi-Henry's. It is an inductor that represents the eddy current losses in the pole piece caused by the oscillating magnetic fields of the voice coil. This concept is described in detail in a *JAES* paper by John Vanderkooy. A real voice coil will have both a real inductance and a Semi-inductance value in parallel. A Semi-inductor impedance goes as $z(\omega) = I_s \sqrt{i \cdot \omega}$ where I_s is the semi inductance value in Semi-Henry's. If this value is not know it is necessary to make

it very large to remove its effect (since it is in parallel with the standard inductor).

Thermal

There are three thermal parameters:

Coil Coefficient

The *Coil Thermal Coefficient* is measured in Ohms / Degree (Centigrade). It represents the change in the resistance of the coil as it heats.

Magnetic

The *Magnetic Thermal Coefficient* is measured in % BL (reduction) / ° (Centigrade). It represents the loss of flux density as the magnet heats. For a ceramic magnet structure this value is about .1 % / °C. An Alnico magnet structure has almost no loss with temperature. No published data is known for Neodymium Iron Boron.

Resistance

The driver *Thermal Resistance* is measured in ° (Centigrade) / sec. It represents the thermal inertia of the system, i.e. how long it takes to heat up the motor structure with a given thermal input.

Geometrical

There is only one driver geometrical parameter.

Aspect ratio

The drivers *Aspect ratio* is dimension less. It represents the ratio of the height to the width of the driver. If it is one then the driver is round. It can be seen that numbers greater than one are equivalent to their inverse with the driver turned by 90 degrees. If the aspect ratio is close to 1.0 (<.95, > 1.05), then 1.0 is assumed. Also it should not be too small (<.1) or too large (> 10.0) as numerical instabilities can result (which will be obvious).

Nonlinear

Nearly all of the parameters that are used to specify the driver change with cone motion. The most important are the changes of *Bl* product and cone stiffness with cone excursion. If these changes are specified by (or best fit by) an equation of the form:

$$Bl(x) = Bl + bl_1 * x + bl_2 * x^2$$

then the harmonic distortion and power compression can be predicted. The linear value of *Bl* is that value specified above. Note that the value of *bl*₂ must always be negative.

The parameters *bl*₁ and *bl*₂ are the parameters that specify the changes of *Bl*(*x*) with the cone excursion *x*.

Likewise the stiffness of the cone can be represented as:

$$k(x) = 1/C_{ms} + k_1 * x + k_2 * x^2$$

The parameters *k*₁ and *k*₂ represent the symmetrical and non-symmetrical increase in cone stiffness with displacement.

In real drivers there are more than just two terms required in the above expansions. However, these higher order terms should always be minimized in a good loudspeaker. There must always be some third order term as these values must change with displacement. Because of this the non-linear models

gets better as the driver gets better. The nonlinear analysis in *SPEAK* is probably a poor approximation for a highly nonlinear unit or large excursion creating 50% distortion or greater.

Stiffness quadratic

The driver *Stiffness Quadratic* parameter is measured in Nt. / m². It represents the non-symmetric change of stiffness with cone displacement.

Stiffness cubic

The driver *Stiffness Cubic* parameter is measured in Nt. / m³. It represents the symmetric change of stiffness with cone displacement.

BL quadratic

The *BL Quadratic* term is measured in Nt. / (Amp m) and represents the anti-symmetrical fall off of the *BL* product with cone displacement.

BL cubic

The *BL Cubic* term is measured in Nt. / (Amp m²) and represents the symmetrical fall off of the *BL* product with cone displacement.

Thiele-Small Parameters

The Thiele-Small form shows the Thiele-Small style of data presentation.

Thiele-Small Data		
f_s	40.0	Hz
V_{as}	200.	liters
R_e	8.00	Ohm
S_d	600.	cm. ²
Q_m	5.83	
Q_e	1.37	
OK		

Upon closing the form (press OK) the data in the current record is updated to reflect changes made to the Thiele-Small parameters. This includes the subtraction of the radiation mass load from the driver cone mass (based on a monopole load) as described in *Mechanical/Mass*.

The driver data for the current record is translated into the six Thiele-Small Parameters. They are:

f_s

Specified in Hertz, f_s is the free air resonance of the driver. The Thiele-Small data has the radiation load of the driver added to the unloaded mechanical cone mass. The mass load is that of a baffled loud-speaker in free air, not an un-baffled one.

V_{as}

V_{as} is measured in liters. It is the volume of air that has the same compliance as the cone.

R_e

R_e is measured in Ohms. It is the DC value of coil resistance.

S_d

S_d is measured in cm^2 . It is the cone area and is identical to the *cone area* for the mechanical set.

 Q_m

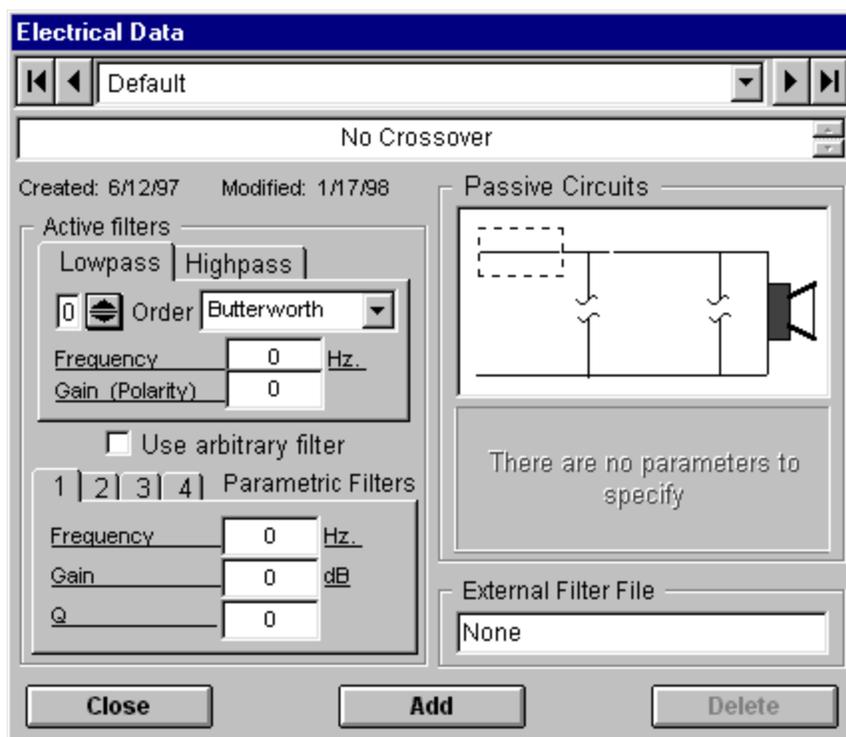
Q_m is dimensionless. It is the transducers Q due to mechanical damping only.

 Q_e

Q_e is dimensionless. It is the transducer Q due to electrical damping only.

Electrical Parameters

Crossovers can be active or passive and a set of parametric filters is also available

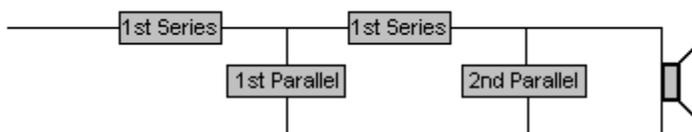


Passive Circuits

A wide variety of passive filters can be modeled with *SPEAK*. Nearly any conceivable configuration can be created as a combination of the four available legs. Each leg can be one of several specific leg types.

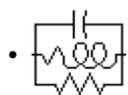
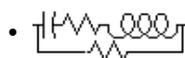
Passive leg selection

Each one of the individual legs in the passive network is selected by clicking on that particular leg. There are four legs, as shown above. Once selected the leg will toggle between the various leg types available.



Leg type

There are four leg types:

-  A capacitor with a parallel resistance (leakage)
-  An inductor with a series resistance
-  An inductor and capacitor in parallel (with resistors)
-  An inductor and capacitor in series with a parallel resistance

Passive Components

The four passive components that make up each leg are:

Inductor

The *Inductor* is specified in mH. It is always associated with an inductor resistance.

Inductor Resistance

The *Inductor Resistance* is measured in Ohms. This resistor represents the resistance always found in series with inductors.

Capacitor

The *Capacitor* value is in μF . The capacitor always has a parallel resistor associated with it.

Parallel Resistor

The *Parallel Resistor* is measured in Ohms. It is the resistance that is always in parallel with a capacitor.

Active Filters - Specified

Active filters are specified by setting the filter type, the filter order and the filter frequency. It is also possible to specify an arbitrary transfer function.

Filter type

There are six different types of active filters available, each with an order from one to five. Order zero is used to turn off the active filter. The filter can be high or low pass.

The filter types are:

- **Bessel** - A Bessel filter is also known as minimum delay filter since it has (not surprisingly) the least propagation delay of all the filter types. It also has a very gradual transition to its final slope.
- **Paynter** - The Paynter filter is also known as a compromise or Thompson-Butterworth. It is a compromise between the minimum delay and the maximally flat filter structures.
- **Butterworth** - The Butterworth filter is known for its maximally flat response. It is a very common filter structure.
- **Chebycheff** (1dB, 2dB and 3dB ripple) - The Chebycheff filters are actually an entire class of filters with ripple in them. They are specified by the amount of ripple allowed (1dB, 2 dB, etc.). Up to 3 dB of ripple can be used. These filters have very steep initial slopes but suffer from a large amount of delay and ringing.
- **Linkwitz-Riley** - The Linkwitz-Riley filter is composed of two second order Butterworth filters in series. Thus it is always of fourth order.

Order

Each of the six filter types can be from order one to order five. (Note: The filter type is meaningless for 1st order filter as they are all the same.) (Except the Linkwitz-Riley filter) Setting the order equal to zero turns off the filter calculation.

Gain

The filter gain is dimensionless. It is not measured in dB. It can be used to adjust the amplifier gain for bi-amped systems.

Frequency

The Filter Frequency is measured in Hertz. It specifies the frequency at which the filter has lost 3 dB of its passband level.

Active Filters - Arbitrary

Selecting the *Use arbitrary filter* button changes the filter specification portion of the form to:

The screenshot shows a form titled "Active filters" with several input fields and a checkbox. The fields are arranged in a grid-like structure:

- Row 1: Three buttons labeled "1 - LP 2nd", "2 - LP 2nd", and "LP 1st".
- Row 2: Three buttons labeled "1 - HP 2nd", "2 - HP 2nd", and "HP 1st".
- Row 3: A "Frequency" label followed by a text input field containing "0" and a "Hz." label.
- Row 4: A "Gain (Polarity)" label followed by a text input field containing "1".
- Row 5: A checked checkbox followed by the text "Use arbitrary filter".

This form of data input creates arbitrary filters from two low-pass and high-pass second order filters, one low-pass and high-pass first order filters. Any of the "specified" filters can be created with this form of specification, but virtually any filter shape is possible with this type of specification.

Gain

The first order filters have gain, which is dimensionless. It is not measured in dB and is used to adjust the amplifier gain for bi-amped systems.

Frequency

The Filter Frequency is measured in Hertz. All of the filters have this parameter. It specifies the frequency at which the filter is reduced by 3 dB from its pass-band level.

Q

The Q specifies the standard terminology for the filter stages cutoff shape. It is dimensionless. A Q of .7 represents a Butterworth filter shape.

Parametric Filters

A set of four parametric filters are available. Each filter has a gain, center frequency and Q .

Gain

The parametric filter *gain* specifies the gain of the filter at the center frequency. It is measured in dB and can be positive or negative.

Q

The Parametric Filter Q specifies the width of the filter stage. It is dimensionless. A Q of one is about a half octave wide, but varies slightly with gain.

Frequency

The Parametric Filter *Frequency* is measured in Hertz. It is the center of the bandpass filter.

External Filters

External filters can be “read” into *SPEAK* by selecting this box and finding the desired filter file from the Open File control menu. At the present two basic external filter files are allowed.

Audio Precision files

Audio Precision files which are composed of magnitude followed by phase, or magnitude only are supported. If magnitude only is used then ONLY a single number can be present as the next number will be interpreted as a phase. Thus in magnitude only there are two numbers, the frequency and magnitude and in magnitude and phase there are three numbers frequency, magnitude and phase.

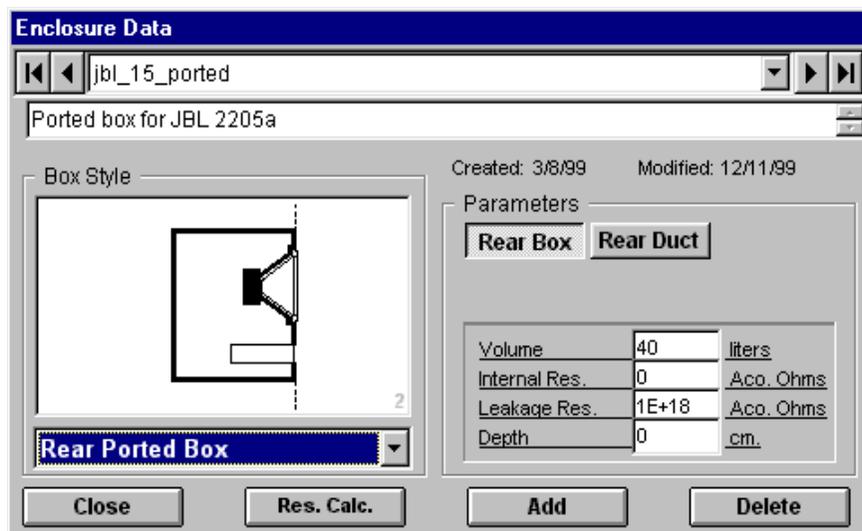
User Files

User files (*.usr extension) are very useful as they can be created from *MATHCAD* or any other numerical program. They consist of a single first line, which is ignored (use for a title), followed by a listing of frequencies and real and imaginary parts of the desired function. *SPEAK* will only use 256 points for this filter so more points will simply be truncated (actually every other point is used if the number exceeds 256 - the end points are not truncated per see.)

There can be as few as three points in this external file and the data will be interpolated to the correct number of points. Zero slopes are assumed at the end points, which can cause some unusual fits outside of the region of specification. Points can be manually inserted into a file to correct for these aberrations. The first and last points are always used.

Enclosure Parameters

The enclosure database contains the information necessary to specify the manner in which the driver is housed. An enclosure is defined by selecting the enclosure type from a drop down list box. There are nineteen box types.



Box type

SPEAK has nineteen defined box types. These boxes are of two environment types; infinite baffle and spherical mounting with four basic enclosure types. The enclosure types are:

- **Closed** - a box with no ports of any kind. It is specified simply by a volume
- **Rear Ported** - a single box with a rear port on it. This is a standard bass reflex enclosure
- **Front Ported** - a closed rear box with a front ported box. This design is commonly known as “bandpass” (for more info see Geddes *"An Introduction to Bandpass loudspeaker Enclosures"*, JAES, Vol. 37, #7 (August 1989)
- **Dual Ported** – a ported box is placed on both the front and the back. This design is commonly known as a “dual bandpass” and is also discussed in the above referenced paper.

In these basic designs a “port” (as defined by *Webster* - an opening in an enclosure), can contain a Duct, a Passive Radiator, an Acoustic Lever™ or a waveguide (horn) of four types; Exponential, Conical, or Oblate Spheroidal, or Bi-Spheroidal.

For each enclosure type selected tabs will appear which specify the parameter sets required to define

the chosen enclosure. A listing of all available enclosures is:

Table 1:

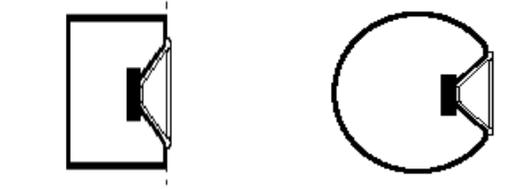
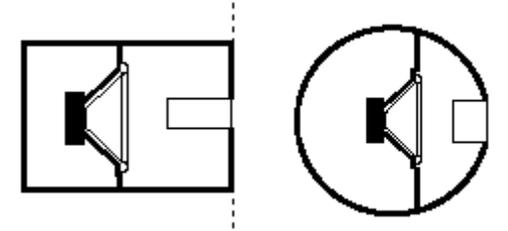
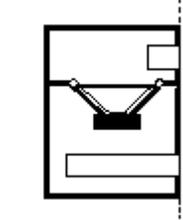
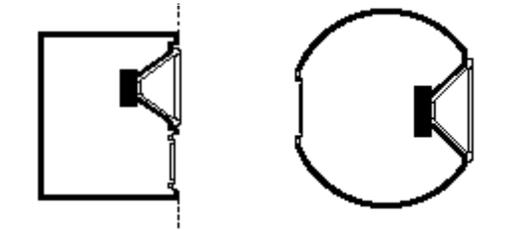
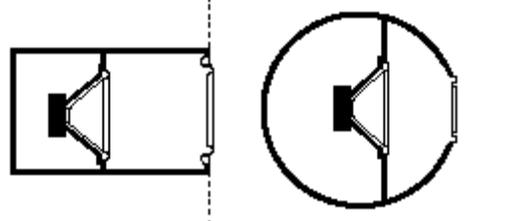
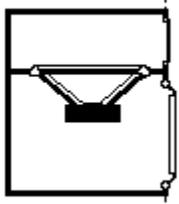
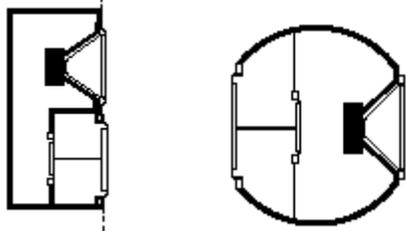
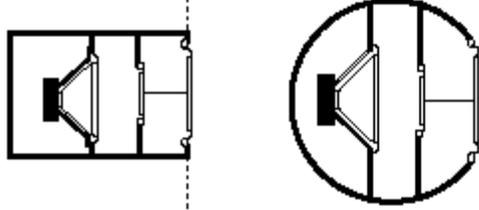
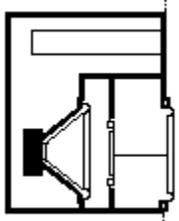
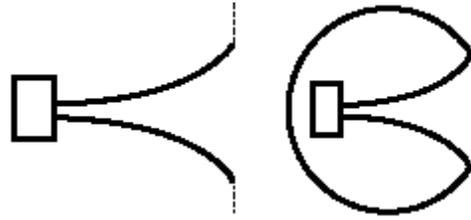
Type	Picture	parameter sets:
Closed Box Sphere		Rear box
Rear Ducted Box Rear ducted sphere		Rear Box Rear Duct
Front Ducted Box Front Ducted Sphere		Rear Box Front box Front Duct
Dual Ducted Box		Rear Box Front Box Front Duct Rear Duct
Rear Passive Radiator Box Rear Passive Radiator Sphere		Rear Box Rear Passive Radiator
Front Passive Radiator Box Front Passive Radiator Sphere		Rear Box Front Box Front Passive Radiator

Table 1:

Type	Picture	parameter sets:
Dual Passive Radiator Box		Rear Box Front Box Front Passive Radiator Rear Passive Radiator
Rear Lever Box / Sphere		Rear Box Rear Lever Rear Lever Box
Box Front Lever Sphere		Rear Box Front Box Front Lever Front Lever Box
Lever and Ducted Box		Rear Box Front Box Front Lever Front Lever Box Rear Duct
Waveguide in Baffle Waveguide in Sphere		Rear Box Front Box Front Duct Horn

Box

There are three box parameters for each box:

Volume

The Box *Volume* is measured in cubic centimeters. It is the actual usable volume. Port, and driver volumes must be excluded from this number.

The increased apparent volume due to the use of internal damping material is not calculated automat-

ically. Loosely stuffing, but filling, a box will increase the apparent volume by about 30%. More or less material than this will give a lesser increase. For more info see Dorfstatter, Blind and Geddes "In Pursuit of Elusive Loudspeaker Parameters" AES Preprint # 4019 presented at the AES Convention in May 1985.

Internal resistance

The *Internal Resistance* is measured in Acoustical Ohms. This parameter will be small (Hundreds of Ohms) for an unfilled box, and can become large (several kiloOhms) if the box is stuffed with damping material. The increased apparent volume due to this damping material is not calculated automatically. (See Box Volume)

Leakage Resistance

The *Leakage Resistance* is measured in Acoustical Ohms. It represents the leakage of a box to the outside. The volume flow from this leakage is calculated and added (subtracted) into the radiation response. In a dual box system the Box Leakage is different from the Driver Leakage since the later leakage will be between the two boxes. In any single box system Driver Leakage and Box Leakage are the same thing.

Depth

The Box *Depth* parameter is measured in cm. This parameter is seldom used and is included to account for some unique effects that can occur if an enclosure has one dimension much larger than the others. In this case the enclosure appears more like a tube than a box. The depth is used along with the volume parameter to calculate the dimension of this tube. The program may decline to use the depth parameter if it detects a numerical instability, i.e. a very short tube, etc.

Ports

Boxes can be connected to the outside through ports (holes in the cabinet walls) by one of three type of connections, ducts, passive radiators or acoustic levers (types cannot be mixed). One of three types of connection must be chosen for the ports in the enclosure. They are:

Duct

A duct has three principle parameters. They are:

Length

The Duct *Length* is measured in cm. It is the actual physical length of the duct. The apparent increase in length (end correction) due to the radiation mass is calculated and added automatically, thus no end correction need be used.

Area

The Duct *Area* is measured in cm². It is the radiating area of the duct. The actual cross sectional shape of the duct is, for the most part, irrelevant. Only for duct cross sections very different from circular will there be an error. The duct is assumed to be round and the end correction (mass load) is calculated for this shape. The end correction will be a little too high for non-round ports.

Internal Resistance

The Port *Internal Resistance* is measured in Acoustical Ohms. It represents the added internal resistance of a port, usually due to port stuffing of acoustical damping material. The inherent port losses due

to flow resistance are calculated automatically and should not be re-added. The port damping is assumed to be continuously distributed and a large amount of damping concentrated at either end will be incorrectly represented.

Passive Radiator

Passive Radiators are an alternate form of port whereby the required acoustical mass is obtained by the actual mass of the moving diaphragm. A passive radiator can act differently from a port if its compliance is not high enough. It also tends to have higher internal damping than a port. Unlike a port, the passive radiator does not take up box space.

There are four parameters to define a passive radiator:

Mass

The Passive Radiator *Mass* is measured in grams. It is the total moving mass of the device. This mass should not include the radiation mass effect as this is calculated automatically.

Compliance

The Passive Radiator *Compliance* is measured in mm. / Newton. The passive radiator must be supported somehow and the device will thus have some compliance associated with this support. If the compliance is high enough then the passive radiator will behave nearly identical to a port. Low compliance will complicate the situation.

Resistance

The Passive Radiator *Resistance* is measured in kg / sec, Mechanical Ohms. This parameters represents the internal losses in the passive radiator itself. It will typically be a small value around 1.0. It should be small as it is undesirable to loss energy in the passive radiator itself.

Area

The Passive Radiator *Area* is measured in cm^2 . It is the actual radiating area of the radiator. Similar to the Driver Cone Area this area must include some of the surround area to be accurate.

Acoustic Lever TM

An Acoustic Lever is a patent pending design consisting of two pistons of differing area rigidly connected together. Details of this invention are published in JAES February 1999 Vol 48, #2. Reading this article is critical to the understanding of the Acoustic lever. Acoustic Levers are an alternative to Passive Radiators whereby the volume velocity is transformed by the ration of the outer to inner areas. An Acoustic Lever has compliance, resistance, mass, two areas and an inner volume between the two diaphragms, which also has acoustical resistance.

The parameters to define an Acoustic Lever are:

Acoustic Lever Mass

Measured in grams this is the total *mass* of the Acoustic lever.

Acoustic Lever Compliance

Measured in N./ mm. this is the value of the *compliance* of both surrounds of an Acoustic Lever.

Acoustic Lever Inner Area

An Acoustic lever has two pistons of unequal area. The areas are measured in cm^2 . This parameter

represents the value of the *inner*, driven, area.

Acoustic Lever Outer Area

An Acoustic Lever has two pistons of unequal area. The areas are measured in cm². This parameter specifies the value of the *outer*, radiating, area.

When Acoustic Levers are used two more parameters are required to define the Acoustic Lever Volume, the air space between the two diaphragms. The definitions of these parameters is consistent with the Volume definitions given above.

NOTE: Acoustic Lever enclosures are a patented technology. It is unlawful to make, have made, or sell an enclosure containing an Acoustic Lever without explicit permission. Licensing information can be obtained from:

Earl R. Geddes
43516 Scenic Lane
Northville, Michigan 48167
(248) 305-9328
Earl@GedLee.com

Array

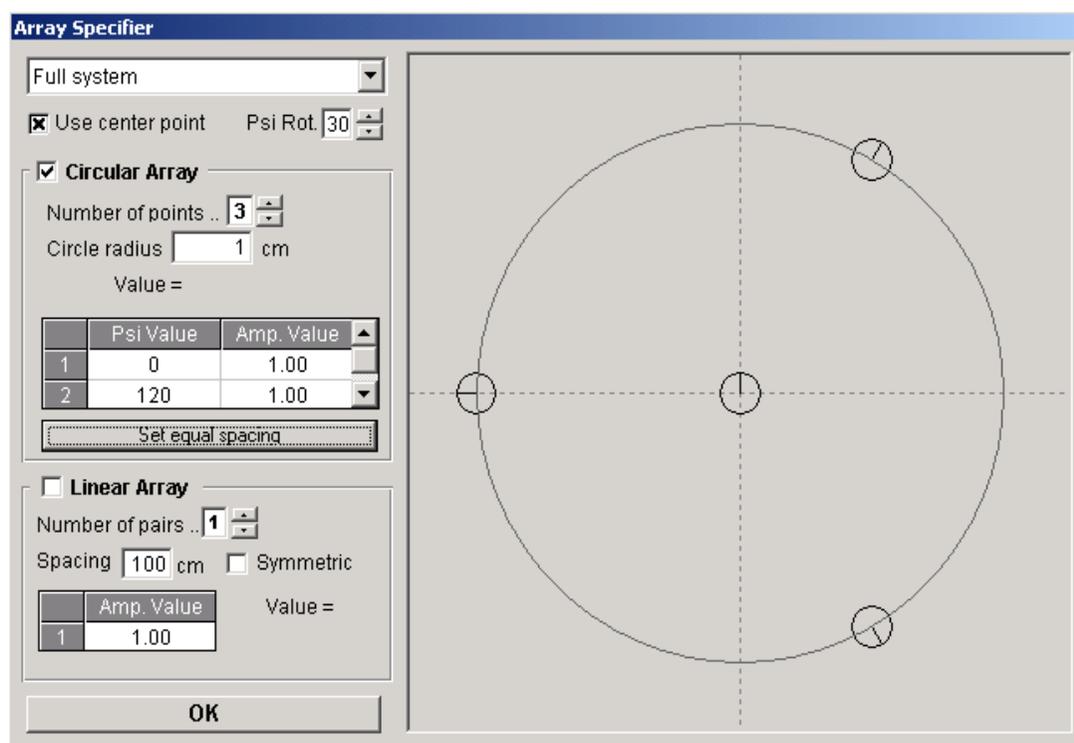
Background

Arrays of transducers are a very important consideration. This is because it is becoming more and more common to see them and it can be very difficult to analyze the performance of arrays. Speak has always been able to analyze arrays in a limited way but this new capability takes things to a new level. Arrays can be hard to understand and tricky to implement, but the simulations can be quite enlightening.

Arrays can be simulated by using several of the same driver as different subsystems located at different places. This creates a rudimentary form of an array. Using this technique only about four elements could be in any array. By using the Array Tool one can simulate nearly any array possible. It is even possible to simulate arrays of entire systems, as is common in Pro sound applications.

The Form

The Array form is shown below:



System selection

The first drop down selection box allows you to select the full system of subsystem that you want to array. For example, there may be dual woofer in which case you would array the woofer subsystem. Or there could be dual or a line of tweeter in which case you would select the tweeter subsystem. You could also stack many full systems on top of one another in which case you would design a system and then array that system

Circular and Linear arrays

The next option to consider is which type of array would work best for the array under study.

The Circular array places the units around a circle of variable radius. There can be up to five elements on this circle as well as an element in the center. The amplitude and location on the circle can be set individually, except for the center point, which is always 1.0.

The linear array places the units along a line, which itself can be rotated. The distance between the elements is variable, but constant, and each element pair can have different amplitude.

Each circle has a small line that indicates the direction of $\psi = 0^\circ$ (vertical) in the coordinate system of the subsystem. For a single driver this has no meaning, but for a system it is very important to note this direction as it can have a profound impact on the radiated response.

Plots

SPEAK has an extensive plotting capability. There are five basic plot types. They are:

Sound Pressure

The Sound *Pressure* plot shows the sound pressure in N / m^2 in the linear mode and dB SPL relative to $.00002 N / m^2$ in the log mode. There can also be shown on and off axis responses if in the Single Subsystem mode. The sound pressure is determined at the Field Location specified for the system.

Cone Displacement

The Cone *Displacement* plot shows the RMS cone displacement curve for the selected data sets. The Cone displacement is in mm. in linear mode and in dB relative to 1 mm. in the log mode. The cone displacement plot is not available in the system summation mode. (A summed cone displacement has no meaning.)

Impedance

The *Impedance* Plot shows the system or subsystem impedance depending on the System Summation mode and the bi-amp mode. In bi-amped, System Summation mode the impedance of each driver in the system is shown.

In the non bi-amped System Summation mode the impedance shown is that of the entire system with all drivers connected to the same common amplifier.

In all other modes the Impedance plot shows the subsystems impedance.

Voltage

The *Voltage* plot shows the drive voltage at the system or subsystem input terminals. This voltage, therefor, drives all of the passive components is such exist. In bi-amped mode the individual voltages to each subsystem are plotted. In non bi-amped mode only one curve is shown in System Summation mode, namely that of the system input voltage.

Current

The *Current* plot shows the input current in milliAmps in linear mode and dB relative to 1 milliAmp in log mode. It is the current into the system in System Summation / bi-amped mode and the current into each subsystem in all other modes.

Port Data

The *Port* plot can be configured to show several things. It can plot the displacement of the port (not meaningful for Ducts) or the volume velocity. Front or rear ports can be shown.

Harmonic Distortion

The Harmonic distortion plot is only available in single subsystem mode. When clicked it generates a simulation of the second and third harmonics based on the data for the driver entered into the nonlinear tab of the driver database. The simulation is accurate only for small amounts of distortion, not highly nonlinear devices. The algorithm becomes unstable for very high levels of distortion and the calculation will exit with a comment to that effect. The distortion in this case will be well over 50% THD.

When several subsystems are in the project, each subsystem has its harmonic distortion calculated. Clicking on the next subsystem toolbar tool (<,>) will show the response of another subsystem and its distortion. Several drivers or enclosure types can then be compared with one another in a very straightforward manner.

Polar Plot

Polar plot shows the Sound Pressure Isobars for the current system or subsystem. The colored lines represent the contour curves for the off axis positions with $\theta = 0^\circ$ at the center and $\theta = 90^\circ$ at the outermost circle. The vertical line represents a Psi value of 0° and psi varies in a clockwise manner. Thus the entire front half plane is represented in this figure. By clicking on one of the color bars one can change the contour level increments.

Each plot type has arbitrary Y Axis scaling and selectable X Axis scaling (except Polar). All Plot functions are available on the Plot Toolbar.

The actual curves that are plotted will depend on several factors. First is the selection of Single Subsystem mode. In this mode the response of a single subsystem is shown. The on axis and off axis curves are shown only in this mode. The single subsystem mode is selected by pressing the button . The buttons on either side of the single subsystem button will select the current subsystem being plotted (as well as the current subsystem for the Parameter Forms).

The summation button  sets the program into System Summation mode. In this mode all of the subsystem SPL's are summed at the specified field points. Being in the far field is particularly important in the summation mode (see Constants, Field Location). The summation mode can be used to calculate the response of multiple drivers in a single box, or a system (woofer, midrange, tweeter, etc.). The summation mode can also be used to transition loading conditions by setting different loads for a driver and using crossovers and summing to perform the transition.

The plotting form allows for data to be saved to buffers or on disk as a file. By double clicking on a curve the Copy Curve Form is displayed.

Iteration of nearly any variable is possible by selecting the iteration mode when in single subsystem mode.

Plots can be printed at any time.

Plot Toolbar

There is a Toolbar at the bottom of the Plot Window. This Toolbar contains the following commands:

-  - Sound Pressure
-  - Displacement
-  - Impedance
-  - Voltage
-  - Current
-  - Port data
-  - Polar Plots
-  - Harmonic Distortion

 - Update Data

The Update Data button is enabled whenever the data has changed and the plot can be updated. It must be pressed to update the plots.

 - System Summation

The System Summation option sums the field pressure at the specified points. It also sums the input currents in the non Bi-Amped mode of operation. It allows for the investigation of the combination of multiple drivers in a single system.



- Single Subsystem mode

The Single Subsystem mode will plot only one subsystem at a time. It is not available in the System Summation mode. In Single Subsystem mode the on and off axis curves are plotted for the Current Subsystem which is shown on this button. The Single Subsystem mode is also useful for setting the Current Subsystem in order to change a parameter in one of the Parameter Databases.



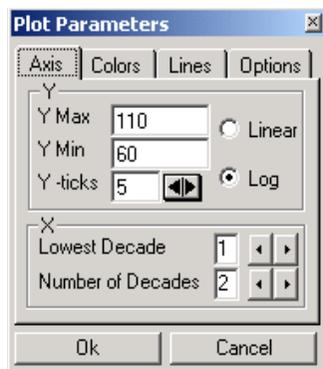
- Iteration Mode

Iteration mode allows for iteration of any parameter. This is very useful when trying to optimize a design.

View

Plot Parameters form

The *View/Plot Parameters* menu selection item will show the *Plot Parameters* form. This form allows the setting of the following components: (the *Plot Parameters* form can also be opened by right clicking the mouse when placed over the plot area).



Axis tab

The X axis has several options. They are

Lowest Decade

The lowest decade specifies the lowest frequency for which analysis is performed. This integer is the power to which 10 is raised times 2, i.e. 0 means 2 Hz., 1 means 20 Hz. etc.

Number of Decades

The number of Decades specifies the total number of decades over which the system is simulated. The Number of Decades plus the Lowest Decade cannot exceed 4, i.e. 20 kHz. This restriction is automatic.

The Y Axis parameters are:

Log - Lin

This selection sets the type of vertical axis.

Max

The Maximum value specifies the value at the top of the plot. The entered value may sometimes be rounded depending on the plot type. Any value can be input.

Min

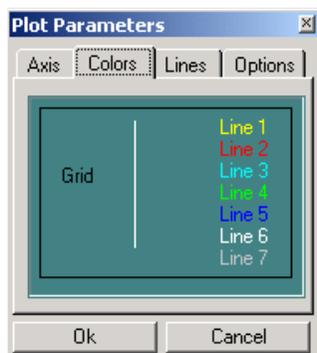
The Minimum Y Value specifies the value to be used at the bottom of the plot. Logarithmic and linear values are saved separately.

Ticks

The number of tick marks on the vertical axis. The difference between the MAX and MIN values should be divisible by Ticks to yield a smooth increment, i.e. $6 (\text{Max}) - 1 (\text{Min}) / 5 \text{ Ticks} = 1.0 \text{ unit per tick}$

The Log Axis option plots the Y Axis in logarithmic units. The dB reference is different for each plot as follows:

- Pressure - .00002 Nt / m² (Standard dB SPL)
- Displacement - 1.0 mm
- Impedance - 1.0 Ohm
- Voltage - 1 Volt (Standard dBV)
- Current - 1.0 mAmp



Colors tab

The plot colors can be changed by clicking on the item. The available items are:

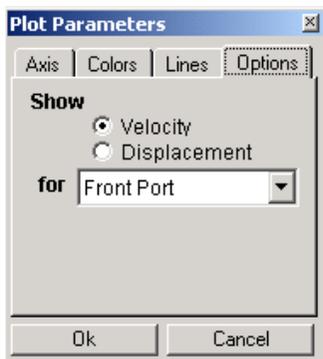
- Plot Background** - The color of the background.
- Grid Color** - The color of the grid lines - inside of the main plot area
- Cursor Color** - The color for the cursor color. Click on the vertical line.
- Line Color** - Each line has a different color.



Lines tab

The lines that are drawn on the SPL plot are selectable. The available items are:

- **0 - 90° values.**
- **0 - x° averages** – an average SPL value for that range
- **x° - y° averages** – an average SPL from x to y
- **No Line** – no line is plotted after this line



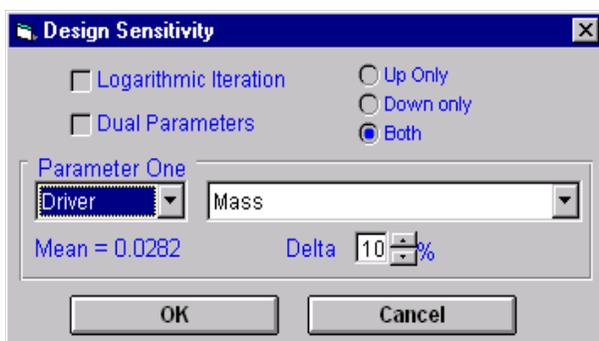
Options tab

Sets the options for plotting on the Ports plot.

Iteration Mode

The iteration mode in *SPEAK* is a very useful function. Only available in single subsystem mode it allows for the iteration of nearly any parameters in the simulation. Activating this button brings up the Iteration form.

Iteration Form



The first drop down box selects the database of interest. The second list box selects the specific parameter of interest.

There are several options:

Iteration Form, logarithmic Iterations

In logarithmic iteration the sequential data runs are multiples of the base value. The multiplicative factor is input in the Factor box.

Iteration Form, Dual Parameters

Dual parameters allows for the simultaneous iteration of two different parameters. The iterations are done as follows:

The base value and the following iterations (either log or linear)

Curve	Increase Parameter	Decrease Parameter
1	one, two	
2	one	two
3	two	one
4		one, two

Up Only, Down Only, Both

In linear iteration mode this selection allows for the parameter to increase only (Up Only), decrease only (Down Only), or iterate two increments up and two increments down.

Constants

The constants parameters are available from the menu "*Constants*". They are global in the sense that these constants are used for all subsystems. There are two basic areas defined on the constants form:

Environment

The environment variables set the physical environment of the project. The variables are:

Speed of Sound

The Speed of Sound in air is measured in m./sec. It is the speed at which sound travels, usually 341 m./sec. at room temperature.

Ambient Temperature

The Ambient Temperature is measured in degrees Celsius. Room temperature is usually about 20 degrees Celsius.

Air mass density

The Air Mass Density is measured in kg / m³. It is the mass of a cubic meter of air, usually 1.19 kg / m³ at room temperature.

Field Point Location

The location of the field point is an essential part of the use of *SPEAK*. It is important to understand that *SPEAK* does not assume a far field location in any of its calculations. This is a powerful but sometimes confusing capability. First there are two parameters which define the field point(s) (since there are five field locations run for each subsystem). They are:

Radius to field point

The Radius to Field Point is measured in meters. It is the distance from the acoustic center to the simulated microphone position. The sound pressure level is strongly dependent on this value.

Psi Angle to Field Points

The Angle Psi around the center axis is measured in degrees in the plane of the baffle. It represents the angle through which the off-axis points are placed. A value of 0 degrees corresponds to vertical, while a value of 90 degrees corresponds to the horizontal plane - to the right.

The radius value must be large enough to get into the far field of the system if the results are to be what one expects. This is because transducers act very differently in the near field than the far field. It can also occur that as the field point moves away from the axial point the field location is getting relatively closer to one driver than another. This will give unexpected results if one is close enough to the system. The far field is what we are used to working in. For many systems the standard 1 meter distance is not yet in the far field. It is recommended that a distance of 10 meters be used as this is nearly always sufficient.

The Psi angle is irrelevant for the axial response since all Psi values have the same "zero degree's off axis" location. The need for a Psi specification is that as the field point moves away from the axial or normal (perpendicular to the baffle) location there are 360 single degree possibilities. Each one is likely to be different! Specifying Psi allows one to examine the non-axi-symmetric characteristics of the use of multiple drivers.

Numerics

There are several numerical constants which can be set. These are:

Number of Points

The Number of Points specifies the total number of data points in the analysis. The higher the number of points the higher the resolution of the analysis, but a longer analysis time will result. Generally 200 points is sufficient. Sometimes more points are required to resolve fine details.

Number of Theta Points

The polar Theta (angle away from the forward axis) resolution is determined by taking 90 degrees and dividing by this number. The frequency plot form will only plot the first seven polar curves, so these curves may not go to 90 degrees.

Number of Psi Points

The angular resolution in the psi direction (around the central axis) is determined by dividing 360 degrees by the value of this constant. This value only has an effect on the polar response graph.

Spherical Modes

The calculation of the radiation from a sphere, as done for a finite enclosure, involves the calculation of the spherical modes. High frequencies require more modes to be accurate, while the calculation of very large numbers of modes that are not significant is inefficient. The correct number can be found by increasing this value until no little change is noted in the results. The calculations can become unstable for very high numbers.

Import and Export

Import Data

Selecting the menu item File/Import Data will bring up a form allowing you to select a file for importing measured or simulated data. There are two types of data files which can be read:

Audio Precision files

Audio Precision files which are composed of magnitude followed by phase, or magnitude only are supported. If magnitude only is used then ONLY a single number can be present as the next number will be interpreted as a phase. Thus in magnitude only there are two numbers, the frequency and magnitude and in magnitude and phase there are three numbers frequency, magnitude and phase.

User Files

User files (*.usr extension) are very useful as they can be created from MATHCAD or any other numerical program. They consist of a single first line which is ignored (use for a title) followed by the data in ASCII form. *SPEAK* will only use 256 points for this filter so more points will simply be truncated (actually every other point is used if the number exceeds 256 - the end points are not truncated per see.)

The imported data can be of four types:

- **dB SPL** - this form reads the data in as a log sound pressure value followed by an optional phase angle.
- **Linear Sound Pressure** - a linear sound pressure value can be read, but must be input as a real and imaginary part.
- **Impedance** - the impedance is read in as a magnitude value followed by an optional phase value.
- **Log Impedance** - the logarithmic value for the impedance curve can be read in followed by an optional phase value

Once read in this data can be plotted at any time by pressing the “EXT” button on the *Plot Toolbar*. If this button is not enabled then external data is not currently available and must be loaded from the menu item.

Other file types for importing will be added in the future (Contact GedLee for requests)

Export Data

Data can be exported in ASCII form for use by other Windows programs by selecting the menu item *File/Export Data*. A form will come up and ask you to select a curve for exporting and then it will request a file name. This file will be saved in the applications directory.

Glossary

acoustic mass

The mass of a port in kg. / m⁴. It is the inertia or mass referred to an acoustic area (squared).

acoustical Ohms

Acoustical Ohms in *SPEAK* are in MKS units of kg / sec m⁴ or $\text{Rho} * \text{Speed} / \text{Area}$.

administrator

The person or persons who have access to the database manager.

axis

A vector perpendicular to the driver face emanating from the origin.

baffle

The baffle is a plane, on which the driver is mounted. The driver locations are measured in this plane. In the case of an un-baffled driver there is no baffle.

base value

In iteration the base value is the value that the parameter under investigation has in the data base.

crossover

A crossover is an electrical circuit which divides the frequency spectrum in parts for distribution to different transducers optimized for the particular frequency range (Woofer, tweeter, etc.)

current subsystem

The number of the subsystem currently available for parameter changes as well as data plotting.

database

A database is an ordered array of records which can be accessed in random order. In *SPEAK* there are three global databases and numerous local ones. The databases are in MS Access format.

driver

A Driver is an electro-acoustic transducer.

driver axis

An imaginary axis, perpendicular to the baffle and centered on the driver.

enclosure

An enclosure is a box (or two or three) which houses a driver. It can have ports or passive radiators as well as the driver itself.

legs

There are four branches in the passive network. Each branch is called a leg.

local

In the context of *SPEAK* local means in the same directory as the executable.

mechanical Ohms

A Mechanical Ohm in *SPEAK* is in MKS units of kg / sec. It is the ratio of force to velocity.

Morse

The author of the leading texts in acoustical physics (in this authors opinion), including "Vibration and Sound", ASA reprint, "Theoretical Acoustics" (With Uno Ingard), Mcgraw-Hill and the two volume set "Methods of Theoretical Physics" (with Herman Feshbach) Mcgraw-Hill. These texts played a key role in many of the numerical methods used in *SPEAK*. Morse is the only author to correctly identify the limiting assumptions built into the Webster's Equation as one example of his thoroughness.

origin

The origin in *SPEAK* can be set at will but should be located at the port if one exists.

parameters

Numbers which represent a particular value of some fundamental variable, such as cone mass or duct length.

port

In *SPEAK* a port is a hole in an enclosure to the outside of the enclosure. This hole must have a device connected to it - a duct, a passive radiator or an Acoustic Lever.

project file

The project file is an *Access* database with a suffix of ".prj". it contains the data for the plots (colors, scales, etc.), the subsystems (names, locations, orientations) and the run constants.

Specific acoustic admittance

The specific acoustic impedance is the inverse of Specific Acoustic Impedance.

Specific acoustic impedance

Specific Acoustic Impedance is the impedance of the medium in units of $\text{Rho} * \text{speed}$ or $\text{kg} / (\text{m}^2 \text{ sec})$

subsystem

A Subsystem is the *SPEAK* name for a driver in a box with a crossover.