APPENDIX

Other Selected Spreadsheet Programs

APPENDIX

[(Section Index)]

SPEAKER WORKSHOP IMBEDDED HELP FILES

Spkrwork.hlp

Note: If you lose the left sided Browser Bar:

1. Look at the View Menu and make certain that the Project tree is marked
2. If not, mark it.
3. If so, close Speaker Workshop
4. Start/Run/Type “regedit”
5. Click Ok
6. In regedit’s tree pane on the left, click the following:
    a. + Hkey_Current_User
    b. Software
    c. Audua
    d. Speaker Workshop
    e. Structures
7. Highlight Tree Info in the detail pane on the right
8. Click “Delete”
9. Restart Speaker Workshop.

SELECTED MENUS

(Section Index)

Note that for any operation performed in SW, there is an undo/redo function, which is available by right clicking to bring up the pop-up menu.

Window menu commands

The Window menu offers the following commands, which enable you to arrange multiple views of multiple documents in the application window:

New Window	Creates a new window that views the same document.
Cascade	Arranges windows in an overlapped fashion.
Tile	Arranges windows in non-overlapped tiles.
Arrange Icons	Arranges icons of closed windows.
Split	Split the active window into panes.
Window 1, 2, .	Goes to specified window.

Help menu commands

(Section Index)

The Help menu offers the following commands, which provide you assistance with this application:

Help Topics	Offers you an index to topics on which you can get help.
About	Displays the version number of this application.

Chart menu commands

(Section Index)

The Chart menu offers the following commands:

Add data...	Lets you select a dataset to add to the chart.
Note...	Lets you add a note to the chart.

Calculate menu commands

(Section Index)

The Calculate menu offers the following commands:

FFT...	Calculates the FFT of time data.
IFFT...	Calculates the inverse FFT of frequency data.
Group delay...	Calculates the group delay of frequency data.
Waterfall...	Calculates the waterfall plot of a pulse (time data).
Frequency Response...	Calculates the frequency response based on time data.
Impedance...	Calculates the impedance based on frequency data.

Combine...	Combine two datasets arithmetically.
Splice...	Splice two datasets together at a frequency.

Transform menu commands

(Section Index)

The Transform menu offers the following commands:

Delay...	Delay the data by a specific time delay, or try to remove excess delay caused by measurement offset.
Limit...	Limit the minimum and maximum data swings.
Invert...	Invert the data (and/or phase) of a dataset.
Scale...	Scale the data arithmetically by a constant.
Truncate...	Truncate the time scale or frequency range of a dataset.
Filter...	Execute a low pass, high pass, or bandpass filter on the time or frequency data.
Smooth	Smooth the data per octave increments.
Make Chart	Create a new chart with this dataset as the base.

Driver menu commands

(Section Index)

The Driver menu offers the following commands:

Estimate Parameters...	Calculate the estimated Thiele/Small parameters for a driver based on impedance measurements.
Set Equivalence…	Manually set the electrical equivalent circuit of a driver (if you are unhappy with the automatic estimation)

Measure menu commands

(Section Index)

The Measure menu offers the following commands:

Frequency response	Popup a menu with various frequency responses to be measured. See below.

The Frequency response popup submenu offsets the following commands:

On Axis	Measure the frequency response on-axis.
30 degrees	Measure the frequency response 30 degrees off-axis.
60 degrees	Measure the frequency response 60 degrees off-axis.
Nearfield	Measure the nearfield frequency response.
Farfield	Measure the farfield frequency response.
Port	Measure the frequency response of the port.
Gated	Measure the gated frequency response of the driver.

Microphone Response	
Impedance	Measure the final driver impedance.
Time response	Measure time response.
Harmonic distortion...	Measure harmonic distortion.
Intermodulation Distortion...	Measure Intermodulation distortion.

Passive Component	Measure a passive (R,L,C) component.

Calculation menu commands

(Section Index)

The Calculation menu offers the following switches and commands:

Switches

Cone excursion	Check to calculate cone excursion when parameters change.
Frequency response	Check to calculate frequency response when parameters change.
Group Delay	Check to calculate group delay when parameters change.
Impedance	Check to calculate impedance when parameters change.
Transient response	Check to calculate transient response when parameters change.

Commands

Sealed...	Tune a sealed enclosure to the desired response.
Vented...	Tune a vented enclosure to the desired response
Merge Port Response...	To merge a port and driver nearfield responses appropriately for a vented (ported) enclosure.

Network menu commands

(Section Index)

The Network menu offers the following commands:

Insert	Pop up a menu allowing you to insert components in the network.
Grid	Pop up a menu allowing you to pick various grid sizes. These are used when you move components around to allow for easier alignment of components.
Calculate response...	Calculate the response of the network.
Create Goal…	Create a goal dataset. That is, synthesize the response of a perfect driver using a given crossover style.
Optimize Network…	Select this to have Speaker Workshop find the best component values to match to a given target (goal) response.

Perturb Components...Vary a component through a range and see the resultant response curves.

The Insert submenu offers the following commands

Resistor	Insert a resistor in the network.
Capacitor	Insert a capacitor in the network.
Inductor	Insert a inductor in the network.
Impedance compensation.	Insert impedance compensation in the network. This can be the inductive rise at high frequencies or the resonant peak.

LPad...	Insert an L-Pad into the circuit
Stock crossover...	Insert a stock crossover into the network.

Sound menu commands

(Section Index)

The Sound menu offers the following commands:

Play...	Play the signal.
Loop play	Play the signal repeatedly.
Play again	Play the signal again (no options).
Record...	Record while playing the signal.
Loop record	Record repeatedly (like an oscilloscope).
Record again	Record the data again (no options)
Stop everything	Stop all play/record activities (especially required for loop play

and loop record).

General tab (Options Preferences dialog)

(Section Index)

This page contains miscellaneous parameters used throughout Speaker Workshop. Currently this contains:

Unit of Measure	Select English (inches, feet, cu. Ft.) or Metric (cm., Meters, liters). You can change this on a field basis by just clicking the unit displayed. The unit of measure will cycle through the choices as you click it
Country Code	Enter a 1 to use your local date, time, and number formats. Enter a 0 to use ANSI-English formats.
Single click opens resources	Click this on so that when you click a resource in the tree with the mouse it is opened. If this is off you need to double-click a resource to open it.
Open last document at startup	This will automatically open whatever file you last had open when you start up Speaker Workshop. Otherwise Speaker Workshop starts up with a blank document.
Verbose Status Display	Click this on to have the status display in the lower right corner of the window show the unit of measure (Hz, Ohms, etc.). Otherwise the status display is more concise and smaller.
48kHz Sample Rate Checking	Click this off if you are having problems with 48kHz sampling rates. This checkbox is specifically for the Creative Labs SB16 and AWE64 cards.
AutoSave every xxx minutes	Check this on to have auto save enabled. Fill in the field with the number of minutes between auto saves. A

document named myfile.swd will then be saved to myfile.$.swd every xxx minutes once the document is modified.

Driver and Enclosure Info Font Select the font to be used in the tabular information presented in the Driver and Enclosure resource views. A nice small font fits lots of data whereas a larger font is perhaps more readable. This font will also be used when printing the view (File / Print).

Number of Past Measurements to Keep Set this to a number greater than 0 to have Speaker Workshop create trails. A trail is created when you execute a measurement (such as On-Axis response) more than once. A file mydriver.OnAxis.trl1 is created containing the last measurements and a chart is created showing the current and last measurement. This lets you compare measurements as you change component values or move a microphone around or change gating. Trails are temporary and are not saved (in fact, when you save the trails are deleted). To make a trial dataset permanent right-click it in the tree and select the Make Permanent option. It will still be overwritten by new trails so renaming it is a good idea.

Tuned Calculator Dialog (Options Calculator Tuned)

(Section Index)

Use this dialog to calculate resonant frequencies and Q's of tuned circuits. A tuned circuit is an RLC (resistor, inductor, capacitor) circuit with either all three branches in series or parallel.

Fields

Series / Parallel Select series to analyze a series circuit, parallel to analyze a parallel tuned circuit.

L Enter the inductance of the tuned circuit.

R Enter the resistance of the tuned circuit.

C Enter the capacitance of the tuned circuit.

Results Frequency Resonant frequency of the tuned circuit

Q Q of the tuned circuit

View menu commands

(Section Index)

The View menu offers the following commands:

Show   Only available while viewing a signal. Popup a submenu to change the signal display. See below.
Zoom   Popup a submenu with various zoom options. See below.

Locations	Shows or hides the location display.
Project Tree	Shows or hides the project tree window.
Toolbar	Shows or hides the toolbar.
Vu Meter	Shows or hides the vu meter display.
Status Bar	Shows or hides the status bar.

The Zoom submenu offers the following options:

Zoom in	Increase magnification by a factor of two, based at the point last clicked by the mouse.
Zoom out	Go back to viewing the entire dataset.

Note that you can also zoom by drawing a rectangle with the mouse (in data display) and by using the right-mouse menu options.

The Show submenu (available only while viewing a signal) offers the following options:

Output	Only available while viewing a signal. Select this to display the output data created while playing or recording a signal.
Left input	Only available while viewing a signal. Select this to display the left input data created while recording a signal.
Right input	Only available while viewing a signal. Select this to display the right input data created while recording a signal.
Both inputs	Only available while viewing a signal. Select this to display both channels of input data created while recording a signal.

If you request frequency response output when recording, then left/right/both will show frequency response data. If you request impedance output when recording then Right Input will show the impedance data.

Resource menu commands

(Section Index)

The Resource menu offers the following commands:

New	Pop up a submenu to add new resources to the project. See below.
Open	Open a window showing the selected resource.
Rename	Rename the selected resource.
Import	Import a new resource from another application.
Export	Export a resource to another application.

The New popup submenu offers the following commands:

Folder Create a new Folder to hold other resources.
Chart Create a new Chart to display datasets.
Driver Create a new Driver.
Enclosure Create a new Enclosure.
Network Create a new Network.
Signal Create a new Signal generator.

Options menu commands

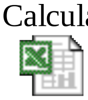
(Section Index)

The Options menu offers the following commands:

Calculator Popup a submenu containing commands to bring up various calculators.
Calibrate Change and create calibration curves for the system.
Preferences Change program preferences.

Wizard Change program preferences.
The Calculator popup submenu offers the following commands:

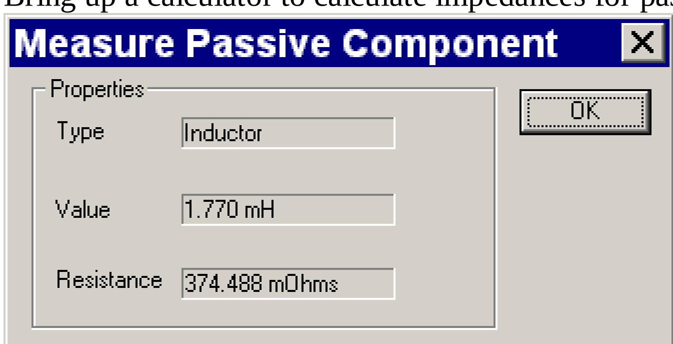
Distance Bring up a calculator to convert between time and distance and frequency (using the speed of sound in air at a temperature). You can also use the Speed of Sound Calculator Spreadsheet that also accounts for humidity



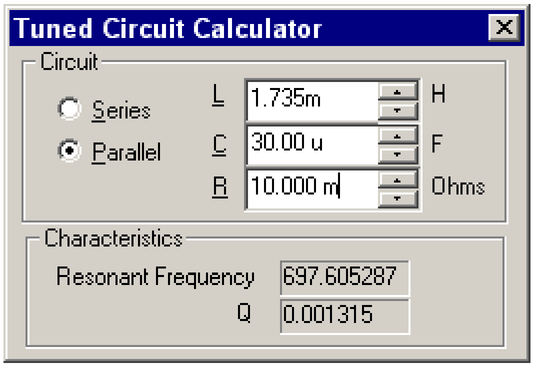
rho.xls

and altitude

Passive Bring up a calculator to calculate impedances for passive objects.



Tuned Bring up a calculator to calculate resonant frequencies and Q's of tuned circuits.



The Wizard pop-up submenu offers the following commands:

Check Sound Card Bring up a dialog box showing how compatible your sound card is with Speaker Workshop.

Note Properties Dialog

(Section Index)

The note properties dialog lets you change where and how a note is displayed on a chart. The options are:

Frequency The X Value (usually frequency) the note is attached to. This value is used to draw a line from the data point to the note box. It also determines value fill-in (see below).
Dataset Which data set the note is attached to
Text The text to show in the note. This includes fill-in fields.
Font The font to display the note in.
Line Draw Line Turn this off if you don't want to see the connecting line (good for headings).
Weight The connecting line weight
Style The connecting line style
Box Draw Outline Turn this off to not draw an outline around the text box
Fill box Turn this off to draw the box transparently
Fill color Select a fill color for the box (if not transparent)

To get a note that has more than one line in it, just press Ctrl+Enter at the end of the line when typing in the Text box.

Speaker Workshop allows you to use some data fields when defining a note. These fields are all started with a % sign:

%f The frequency the note is attached to
%yd The data value. The d is optional and signifies a specific dataset (the default is to use the attached dataset). E.g. %1y uses the first dataset, %y2 uses the second dataset.
%pd The phase value. The d is optional and signifies a specific dataset.

For example, assume the note is attached to a dataset at 950 Hz with a data value of 12.3dB at that point. The string

Using %fHz with value %y1dB will show on-screen as “Using 950Hz with value 12.3dB”

If you change the frequency the note is attached to, the field values shown will change in keeping with the frequency shift.

> NOTE: Although you can add notes to dataset charts (and other charts), only resources called “Chart” have the notes saved when you do a save. If you want to keep your notes around, create a chart (Resource / New / Chart) and use it to display your datasets and notes.

GLOSSARY

(from Parts Express @ http://www.partsexpress.com/resources/spterms.html)

(Section Index)

Acousta-Stuf™ : Acousta-Stuf™ is a sound absorption and dampening fiber typically used in sealed box enclosures. Acousta-Stuf™ is a crimped polymer fiber that was designed to offer a similar performance to long hair wool. This non-volatile synthetic fiber is superior to other materials because it is safe to handle and will not decay with age.

Acoustic Suspension: A type of loudspeaker enclosure that uses a sealed rear chamber to contain the back wave and provide damping of the cone motion. To qualify as an acoustic suspension system, the enclosure must literally be airtight.

Active Crossover: An electronic high or low-pass filter that is placed between the pre-amplifier or source and the amplifier. The benefits of an active filter include removing components from within the speaker, removing the complexities of driver impedances from the equation, and greater flexibility with regard to crossover slopes and points.

Air, Airy, Airiness: A subjective term often used to describe a speaker's ability to reproduce very high frequencies with detail and low distortion. Airiness helps provide an ambient sound field and is very important to producing a "live" sound.

Amplifier: An electronic device responsible for increasing signal levels. A power amplifier produces the high currents necessary for driving speakers. A pre-amplifier is responsible for increasing the low voltages associated with turntables, microphones, or other low-voltage devices.

Anechoic Chamber: A room that is designed such that the walls absorb all incoming sound waves and reflect nothing back. An anechoic chamber is useful for measuring speakers without the negative influences of the typical listening room. Using an anechoic chamber can provide a superior picture of the theoretical output of a system, however real-world factors such as room gain and floor bounce cannot be measured.

Anechoic Response: The frequency response of a driver or system measured in an anechoic environment. As above, this response does not include any room effects such as room gain, floor, or wall reflections.

Attenuate (attenuation): The reduction in output of a signal. In speakers, a tweeter is oftentimes attenuated to match the level of a woofer. This attenuation can be achieved with series or parallel resistors, but often an L-pad is used to maintain a constant impedance load to the crossover.

Back plate: A steel plate that is on the back of a loudspeaker driver's magnet structure that transmits the negative magnetic pole into the pole piece. A bumped back plate has a raised central portion that helps prevent the voice coil from hitting the plate on the down stroke.

Baffle: The front panel of a speaker where the drivers are mounted. A baffle can either be the front wall of an enclosure, or a two-dimensional plane where a driver is mounted. A baffle is used to separate the radiated front and back waves of a driver.

Baffle Step: An increase in the high frequency output of a loudspeaker as the radiation pattern changes from 4-pi space to 2-pi space. At wavelengths shorter than half the width of a baffle, the waves "bounce" off the front baffle and are reinforced due to reduced acoustic impedance. At wavelengths longer than half the baffle width, the waves no longer are reinforced off of the front baffle and radiate in all directions. The result is a 6dB increase (step) in the output above the baffle step frequency.

Baffle Step Compensation: A circuit that is used in a speaker crossover to "compensate" for the increase in output at higher frequencies due to the baffle step. Typically the change in output across the baffle step is 6 dB. Baffle step compensation can be achieved by using a low-pass filter at or near the baffle step frequency to counter the natural rise. However, this will only be successful in speakers that have relatively low crossover points where excessive attenuation above the baffle step is not a problem. Baffle step compensation can also be achieved by using an inductor and resistor in parallel with a second resistor shunting to ground. Resistor values are generally on the order of the nominal impedance of the driver and the inductor is generally in the .5-1.0 mH range.

Bandpass: A combination of high-pass and low-pass filters that yield a section of flat response with a roll-off on either end. In the acoustic realm, a bandpass can be achieved by using a single driver within a front and back enclosure tuned to different frequencies. In the electronic realm, a bandpass filter is usually used on the midrange of a three-or-more-way speaker to allow only a narrow band to be reproduced.

Bandpass gain: A phenomenon that occurs in electrical and acoustic systems when the high-pass and low-pass sections of a bandpass filter interact with each other. As the pass-band region of the filter narrows, the amount of bandpass gain also increases.

Bass: The lowest portion of the audio frequency spectrum, generally from 20 Hz to 160 Hz.

Bessel Filter: A type of crossover filter that has a small peak in the response at the crossover frequency. The Q of the filter is slightly higher than average, and phase characteristics are average.

Bi-amp(ing): The ability of a single speaker to be driven by two separate amplifiers. Generally this is accomplished by having two sets of inputs on the back of the speaker, one going to the tweeter high-pass filter and one going to the woofer low-pass filter. It is also possible in 3 or more-way systems by combining the tweeter and midrange into one section, etc. This method can be used to allow separately adjustable levels for the treble and bass, but is not guaranteed to produce positive results.

Bi-pole: A speaker using two drivers facing opposite directions and operating in phase with each other. In home theater setups, bipolar speakers produce a somewhat diffuse sound field, but there is still some direct radiation at the listener.

Biscuit: A small spline of wood that is used to help reinforce a joint. The biscuit is placed into a slot and glued, where it absorbs moisture and swells up. The swelling action along with the increased gluing surface area yields a very secure joint.

Bi-wiring: Bi-wiring uses the same internal layout as bi-amping, but is accomplished by using one amplifier channel with two separate runs of wire to the speaker. There are many claims about the sonic improvements of this technique, but very little scientific evidence to back them up.

Binding Post: The most widely used method of accepting speaker-level connections on mid to high-end speakers. A binding post consists of a metal shoulder with a protruding threaded rod on which a nut tightens down.

Bondo®: An epoxy-based filler traditionally used in autobody repair. Makes an excellent wood filler for speaker building because of its great adhesion to MDF, fast curing time, and ease of sand-ability.

Bucking Magnet: A charged ring-type magnet that can be used to help shield a driver. The bucking magnet is secured to the rear of the motor structure with the like magnetic poles together. This will reduce the stray magnetic field, but will also affect the T/S parameters of the driver.

Bumped Back Plate: A back plate that has a protruding central portion that helps prevent the voice coil from hitting it on the down stroke.

Butt Joint or Lap Joint: In woodworking, a type of joint that connects two pieces of wood by fastening the end-grain of one piece to the face of another. The weakest type of joint, due to the lack of lateral support and the limited gluing surface area.

Butterworth Filter: A crossover filter slope that yields a maximally flat frequency response in the pass-band with minimal phase shift. Drawback is a shallower slope than other filter topologies.

Capacitance: A measure of the ability of a device to store electric charge and resist changes in voltage. Capacitance is measured in Farads.

Capacitor: An electronic component composed of two metallic plates separated by a dielectric. Stores electric charge and opposes changes in voltage. In speaker building and all AC circuits, a capacitor acts as a high pass filter. Typical values in crossover networks are in uF, or 1/1,000,000th of a Farad.

Causal system: Is one for which the output response happens only after application of the input -which is obviously a requirement if the data represents any real system.

Center Channel: The speaker used in a surround-sound setup that is responsible for reproducing vocals and other centrally located sounds. The speaker is generally magnetically shielded to prevent interference with CRT based screens. While many will argue that a center channel speaker is not necessary, in cases with a lot of off-axis movie watching, it greatly helps keep dialogue centered on the screen. Care should be taken to keep the center channel timbre matched to that of the front speakers, which is important to provide a smooth transition as sounds move from one speaker to the next.

Clipping: A type of distortion that occurs when the tops of the sine wave are cut-off or "clipped". This generally occurs in amplifiers when they exceed output voltage, and can be very detrimental to a speaker due to the non-linear motion that is created.

Closed Box: A completely sealed loudspeaker enclosure. See Sealed Box

Coaxial: A type of loudspeaker transducer that has separate high-frequency and low-frequency drive units together in one driver. In most situations, a tweeter is suspended in front of the woofer cone, but it can also be located on top of the pole piece where a dust cap is normally found. Coaxial arrangements can create a "point-source" where the acoustic centers of both drivers are on the same axis.

Comb Filtering: An artifact seen in multi-driver systems that is the result of constructive and destructive interference from multiple point sources. The addition or subtraction of multiple sources will vary with location relative to the speaker. Comb filtering becomes more of a concern at higher frequencies due to the shorter wavelengths involved. Most often used when talking about line arrays where spacing between tweeters can be problematic.

Compliance: The overall stiffness of a speaker driver's suspension. Represented in the Thiel-Small parameters by the figure Vas, the equivalent air volume with the same springiness.

Compression (power): A condition in loudspeaker transducers that is a result of high temperatures in the voice coil, causing an increase in resistance and overall impedance. Symptoms of compression include a decrease in sound output, unpredictable spectral changes, and other audible distortions. This term is most commonly used when referring to a speaker's ability to remove heat from the voice coil at high powers.

Compression (driver): A type of driver that forces a larger radiating surface area through a small opening. This is then usually attached to a horn loading system that provides greater directivity and better acoustic coupling with the surrounding air mass. The advantage of compression drivers is their very high efficiency and ability to produce very high output.

Cone: The portion of a driver that is attached to the voice coil and excites the air as the coil moves. The main function of a cone is to increase the radiating area of the voice coil while maintaining a rigid form. The traditional conical shape is most often used because it yields excellent strength to weight ratio when force is applied from the vertex.

Conjugate Network: Another name for impedance compensation or Zobel networks. A circuit consisting of a capacitor and resistor in series, in turn paralleled to the driver. Is used to counter the rising impedance found in most drivers above their resonance point. This enables more ideal functioning of a crossover.

Copper Cap: A copper ring that is placed on top of the pole piece in order to reduce eddy currents. Helps reduce distortion and improve high-frequency performance. Also known as a shorting ring.

Counter- EMF: A voltage generated in the opposite direction of the input signal as a result of the voice coil moving back through the magnetic gap. The harmful effects of counter-EMF can be reduced by having an amplifier with a high damping factor. Also called Back-EMF

Curvilinear Cone: A type of cone that is flatter towards the surround and curves progressively steeper towards the voice coil. Benefits of a curvilinear cone can include better high-frequency performance and better off-axis response.

Crossover: An electrical filter within a loudspeaker responsible for dividing up the frequency spectrum and sending portions to the appropriate drivers.

Coil: In crossover construction, a simplified name for an inductor. Used because most inductors look like a large coil of magnet wire.

Cutoff: The frequency where useful output can no longer be produced. Usually this is the F3 of a speaker, the frequency where the response is 3 dB down.

Dado: In woodworking, a groove that is machined into a piece of wood to accept another board for making T-shaped joints. A joint formed by using a dado is strong because of the increased gluing surface area, and the presence of some lateral stability.

D'Appolito Configuration: An arrangement of two woofers and a tweeter such that the tweeter is placed vertically between the two woofers. A D'Appolito configuration yields a narrower high-frequency vertical dispersion that in turn reduces floor and ceiling reflections. A superior vertical symmetry is achieved compared to traditional two-way speakers. One thing to consider when building D'Appolito style systems is the increased distance between the acoustic centers of the woofers. This may cause comb filtering problems when crossed at high frequencies.

Damping (or Dampening): The ability of a material to reduce vibrations. On a driver, a cone coating or surround material can minimize vibrations within the cone, yielding flatter frequency response. Overall movement of the cone can be damped electrically or mechanically through the voice coil and suspension. In speaker cabinets, damping materials can reduce wall vibrations when applied directly to the walls, or can absorb acoustic energy from within the enclosure itself.

Damping Factor: A measurement of an amplifier's ability to control the motion of a speaker at the stop of transient impulses. Technically defined as the ratio of the load impedance to the amplifier's output impedance. The high output impedance of the amplifier enables it to absorb the back EMF generated by the voice coil. A damping factor greater than 10 is usually adequate, however amplifiers with damping factors up to several thousand are available. Note that damping factor varies with frequency as the load's impedance changes.

DC resistance: A measure of the pure resistance of a driver's voice coil at rest. Is used to help calculate crossover networks and determine nominal impedances.

Decibel (dB): A logarithmic scale used to measure relative acoustic output levels. Zero Decibels is defined as the quietest sound the average human is capable of hearing. Traditionally, a difference of 3 dB is considered the smallest change in loudness that the average human can detect. A doubling of perceived loudness is equivalent to a 10dB change in acoustic output. A 3dB increase in acoustic output requires double the amplifier power, while a 10dB increase in acoustic output requires a one-hundred-fold increase in amplifier power.

Detail: A subjective term used to characterize a speaker's ability to reproduce and separate small variations in input. Usually used to describe a tweeter's ability to play intricate overtones and nuances.

Diffraction: A series of constructive and destructive interferences that occur as waves change directions or go around obstacles. Typically, main diffraction concerns are at the edges of the front baffle and around the frames of drivers. A typical diffraction will look like a series of dips followed by peaks in the response.

Dipole: A loudspeaker type that features one or two drivers that emanate sound in opposite directions out of phase with each other. A benefit of a dipole speaker is the cancellation of the sound at 90 degrees to the listening axis, which helps reduce side-wall room interactions. In home theater, dipoles are often used to create a "diffuse" sound field where there is minimal sound radiated directly at the listener.

Directivity: The tendency of a loudspeaker transducer to radiate sound in a particular direction. Typically used to describe the dispersion patterns in horn-loaded drivers. A very directive driver will project sound only to a small portion of three-dimensional space. This can be very helpful in sound reinforcement where coverage needs to be tightly controlled.

Dispersion: The characteristic pattern of how a loudspeaker radiates sound in a three-dimensional space. Horizontal dispersion describes the amount of sound output at various angles side-to-side from the listening axis. Vertical dispersion describes sound output at various angles up and down from the listening axis. Many times controlled dispersion is used to reduce unwanted reflections from floors, ceilings or other obstructions in the sound field.

Distortion: Any type of error in the reproduction of an audio signal. Distortions can be produced at any point in the music chain, and can be caused by analog, digital, or mechanical errors.

Dome: A type of loudspeaker driver that uses a convex, dome-shaped diaphragm. Traditionally only used for mid-to-high-frequency drivers due to the limited structural integrity of this shape. It does have the advantage of generally improved dispersion with less diffraction compared to cone-shaped drivers.

Doping: A thin layer of viscous material that is added to the surface of drivers to dampen resonances within the diaphragm.

Driver: A term for a loudspeaker transducer in its raw state without an enclosure. Driver types are woofers, tweeters, midranges, compression drivers, domes, etc.

Dual Voice Coil: A speaker driver, usually a woofer or subwoofer, that has two voice coil windings on one former. There are actually two sets of terminals on the woofer for hooking an amplifier to each coil. A dual voice coil woofer allows stereo signals to be summed and produced from one driver. Other benefits include additional wiring flexibility (series or parallel combinations) and the ability to use one coil to change the electrical damping characteristics of the woofer.

Dynamic Range or Dynamics: A measure of a system's abilities to produce very quiet and very loud sounds. In digital devices, dynamic range measures the difference between the largest and smallest possible signals produced. In loudspeakers, dynamic range is a somewhat subjective term used to describe a speaker's ability to produce quiet sounds and very loud sounds with good intelligibility and low distortion.

Early reflections: The first reverberated sounds to reach the listening position generated by direct reflections from floors, ceilings, and walls. These are the most harmful to sound reproduction because they arrive very soon after the original signal and at fairly large magnitudes.

EBP: Efficiency Bandwidth Product, equal to the Fs divided by Qes. The EBP is used to help determine what type of enclosure a woofer is suitable for. The general rule of thumb is that EBP's less than 50 are better for sealed enclosures and EBP's >50 are better for vented enclosures. However, this is only a general rule of thumb, successful designs can be achieved that do not follow it.

Efficiency: A rating of how much acoustic output a driver or system will produce with a given amount of input power.

Electrostatic Speaker: A loudspeaker type that uses a thin dielectric film suspended between two electrically charged panels. The motion of the diaphragm is the result of electrostatic charges pulling or pushing on it. This arrangement is unique because there is no voice coil; the signal is applied by changing the voltage on the electric panels.

Enclosure: A cabinet that entraps the rear wave from a loudspeaker transducer to keep the front and back waves separate. Also serves to enhance bass response due to the physical properties of the air enclosed.

Even Order: Any of the crossover slopes that are of an even order, usually 2nd or 4th.

Excursion: The distance a driver's diaphragm is capable of moving from the at-rest position. Maximum linear excursion (Xmax) refers to how far a driver cone can move while the still under control of the motor. Mechanical excursion is how far the cone can physically move including portions where the voice coil is out of the magnetic gap. Typically, excursion figures represent the amount of movement in one direction from the at-rest position. However, movement in both directions is sometimes given with a peak-to-peak rating.

F3 (6,8, etc.): The point in an acoustic roll-off where the output is 3 Decibels down from the baseline level. The 3dB figure is used because this is the point where a decrease in output will be noticeable to the average human.

F10: The point in an acoustic roll-off where the output is 10 Decibels down from the baseline level. The 10dB figure is important because this is where the average human will perceive a loudness of one-half of the baseline level.

Farad: The unit of measure of capacitance. The ability to store one coulomb of energy at one volt is equivalent to 1 Farad. In most loudspeaker applications, the values commonly used are in the uF range, or 1/1,000,000th of a Farad.

Far-Field: The far-field can be defined as any distance from a loudspeaker at which inter-driver integration is complete. Typically set at 1m. Far-field measurements are useful because they usually take baffle step and driver-to-driver spacing into account. In studio monitoring situations, near-field and far-field listening techniques are used to evaluate a mix.

Fc: The resonant frequency of a closed box system. Sometimes called Fcb

Ferro fluid: A fluid that has magnetic properties that allow it to be attracted to magnetic fields. Often used in tweeter magnetic gaps to provide mechanical damping and to help conduct heat away from the voice coil.

Fiberglass: Fiberglass is a material often used in speakers as a damping material. Fiberglass has excellent thermodynamic characteristics useful for speaker building, but is considered less safe than polymer based damping materials. Fiberglass can be used as general stuffing in sealed enclosures, or used to damp walls in vented cabinets.

Film and Foil Capacitor: A type of capacitor that uses two separate layers of a solid metal and dielectric film. This type of capacitor generally has superior audio characteristics than other types of capacitors.

First-Order Crossover Network: A crossover network that uses a single component as a filter, yielding a cutoff slope of 6 dB per octave.

Five-way Binding Post: A type of connection usually found on speakers and amplifiers used for connecting speaker wire. The term five-way comes from its ability to connect to multiple wire termination methods: bare wire (compressed under the nut), bare wire (through-hole), speaker pin, spade plug, banana plug, and banana plug (through-hole). Used very often on high-end speakers for its flexibility and ease of use.

Floor Bounce: Typically, the first of the early reflections to reach the listening position from a loudspeaker. The negative effects of this reflection are the greatest due to the close relative lengths of the original signal and reflected signal. The usual result of floor bounce is a large dip and hump in the frequency response between 100 Hz and 200 Hz.

Fourth Order Bandpass: A type of bandpass enclosure that uses a sealed rear chamber and a vented front chamber.

Fourth Order Crossover Network: A filter type that uses four components to produce a roughly 24 dB per octave roll off.

Flush Mounting: A process of recessing a driver in its baffle so that the faceplate of the driver is even with the surrounding baffle. Flush mounting will prevent diffraction effects that occur as the waves go around the edge of a mounting flange.

Frequency: The number of cycles of a wave that pass a given point in a given time. Most often measured in Hz (cycles/second).

Frequency Response: A measurement of a loudspeaker driver or system's output over a large range of frequencies. A typical frequency response curve plots loudness in dB vs. frequency. This information is useful because the overall tonal characteristics of a speaker can be determined from this plot. Also, the useful operating range of a speaker can be measured.

Fs: The resonant frequency of a loudspeaker driver in free-air.

Full-range driver: A driver that is designed to produce a wide range of frequencies. There are no set limits to what frequencies must be covered in order to qualify as a full-range driver. It is very difficult to produce a driver that is capable of producing both ends of the frequency spectrum simultaneously, in most cases either top-end or bottom-end response will be sacrificed to a certain extent.

Fundamental: The lowest or primary tone produced in the spectrum of a given sound.

Golden Ratio: A ratio often used in calculating the internal dimensions of a speaker enclosure. The ratio is 0.62 :1.0 : 1.62 and is used because it spreads the internal resonances of the cabinet over the broadest frequency range.

Grill: A screen or mesh that covers the front of a speaker or driver to protect it from damage. Despite using acoustically transparent grill cloth, the grill frame may cause some negative effects because of diffraction problems.

Group Delay: A measurement of the amount of phase delay induced by a filter at various frequencies. Ideally, a filter would pass or attenuate all signals without any changes in the phase of the input signal. In the real world this is not the case, and differing amounts of delay will be induced at varying frequencies. By examination of a group delay plot we can see how phase has been affected, and we can detect problems that may cause phase distortion or "smearing" of the signal.

Harmonic: A multiple of the fundamental frequency that is found in many locations within the sound reproduction chain. Each higher harmonic is produced at a smaller output than the last. In speakers, harmonics are detrimental to the accurate reproduction of a signal and are one of the primary forms of distortion. Naturally occurring harmonics are responsible for the unique sound characteristics of varying instruments and voices.

Helmholtz Resonance: Resonances that occur when air or other fluids are excited and form standing waves within a fixed volume. The traditional examples of Helmholtz resonance are organ pipes and "blowing over the top of a bottle." As can be imagined, Helmholtz resonance produces extremely detrimental frequency response problems. When designing ports for vented enclosure, Helmholtz resonance must be considered in situations where port length is much greater than port diameter. Helmholtz resonance can also be found in long, narrow enclosures of any shape.

High-Pass Filter: A filter that allows high frequencies to pass, but cuts off lower frequencies. Used on tweeters and midranges to limit low frequency production, thus reducing excursion and distortion.

Hilbert Transform: Transforms a frequency response plot into its minimum phase equivalent.
It has no effect on the magnitude of the data, changing only the phase data.

Impedance: A complex calculation of the resistance of electron flow in alternating current circuits. Impedance is calculated from a combination of resistive, capacitive, and inductive elements in a circuit. Measured in ohms.

Impedance Curve: A plot of the impedance of a loudspeaker across the entire frequency spectrum. An impedance plot is useful in determining many key parameters in loudspeaker design. Most driver T/S parameters can be derived from the impedance plot, as well as most in-box performance parameters.

Inductance: A device's ability to resist changes in current, measured in Henries.

Inductor: An electronic device that resists changes in current due to the production of a magnetic field around itself. In crossovers, an inductor is a coil of insulated magnet wire that acts as a low-pass filter. Inductors come in several types, air-core, iron-core, and ferrite core. Air-core inductors are considered the best for audio applications, followed by iron core, and then ferrite core. When large inductance values are needed, iron or air core inductors may be used. An inductor should have the lowest possible DC resistance to allow maximum power throughput and minimal negative effects on the crossover. Values typically seen in crossover networks are in mH, or 1/1000th of a Henry.

Infinite Baffle: A loudspeaker enclosure type that theoretically uses an infinitely large rear chamber to contain the back wave. Since an infinitely large volume is not possible in many situations, traditionally any volume roughly 5 times the Vas of the driver is considered infinite. Woofers to be used in infinite baffle situations must have a high Qts, giving them adequate damping in a free-air situation, and allowing them to operate effectively. Benefits of an infinite baffle enclosure are extremely clean and uncolored sound with very low bass output capabilities. In home construction, infinite baffle woofers are often installed in ceilings or floors, using an attic or basement to contain the back wave.

Isobaric: A loudspeaker configuration in which two woofers are sealed together with a very small airspace in between. The two drivers can be facing each other in a "clamshell" arrangement, or placed very close together with the magnet of one woofer near the cone of the other. An isobaric configuration yields an overall Vas which is half that of a single woofer. When the two woofers are wired in parallel an increase in efficiency results, but the maximum SPL is not increased since it is still excursion-limited. In a clamshell arrangement, some distortion can be reduced due to the cancellation of odd-order non-linearity's.

Jasper Circle Jig: A jig that mounts to the base of a router to allow easy machining of circular holes and recesses. Pre-drilled with all of the holes and labels necessary to create any diameter circle.

Kevlar®: A synthetic fiber produced by DuPont® that is sometimes used in loudspeaker driver cones because of its high strength to weight ratio.

L-Pad: A means of attenuating the output of a tweeter or midrange using a combination of series and parallel resistors. An L-pad can provide variable levels of attenuation without changing the impedance that the crossover sees.

Labyrinth: A type of loudspeaker enclosure that is similar to a transmission line. A labyrinth features a constant cross-sectional area that has damping material lining the walls only. The typical length is 1/4 wavelength.

Le: Abbreviation for voice coil inductance. This is the voice coil inductance measured in millihenries (mH). The industry standard is to measure inductance at 1,000 Hz. As frequencies get higher there will be a rise in impedance above Re. This is because the voice coil is acting as an inductor; consequently, the impedance of a speaker is not a fixed resistance, but can be represented as a curve that changes as the input frequency changes. Maximum impedance (Zmax) occurs at Fs.

Linkwitz-Riley Crossover: A type of crossover popularized by Siegfried Linkwitz, of a second or fourth order classification. Both drivers are 6dB down at the crossover frequency and sum to zero to yield a flat frequency response on axis. L-R crossovers also produce a main lobe that is perpendicular to the drivers' central axis.

Listening Room: A dedicated room designed specifically for listening to hi-fidelity sound reproduction. A listening room typically has dimensions conducive to good listening and is well damped to reduce room reflections.

Lobing: The three-dimensional shape of how sound radiates from a multiple-point-source speaker. The sound will vary with different angles relative to the listening axis due to the separation of the acoustic centers. At some angles, there will be cancellations at certain frequencies. Lobing occurs vertically in vertically aligned speakers and horizontally in horizontally aligned speakers.

Low-pass: A filter type that allows low frequencies to pass, while rolling off higher frequencies. A low-pass filter is used on woofers to reduce their output at frequencies where they experience cone breakup or poor off-axis response.

Magnetic Gap: The round opening in the top of the motor between the pole piece and the top plate. The magnet's entire field is concentrated into this small gap where the voice coil sits. Having a narrow magnetic gap contributes to a high-efficiency speaker, but care must be taken to ensure that the voice coil will not rub on either side.

MDF: Medium Density Fiberboard. A type of engineered wood product that is used extensively in the loudspeaker industry. It is used because of its relatively high mass and good damping characteristics. MDF is made from glued and pressed wood pulp fibers; the process is very similar to the paper making process. MDF can be either a very light brownish-yellow color or a darker brown, depending on what type of sawdust is used in its making. This mainly varies by what part of the country the MDF is coming from. MDF machines very well, it will hold sharp edges and complex forms very well. Care should be taken in securing MDF due its layered nature that tends to separate when inserting screws. Most glues will work with MDF, though the suggested types are standard "yellow" carpenters glue or polyurethane-based glue.

Metallized Film: A method of constructing capacitors using a non-conductive dielectric with a thin layer of metal deposited on one side. The very thin layers of film are then rolled up to produce a large surface area of alternating layers of metal and dielectric. Most relatively inexpensive capacitors are constructed using this method.

Midrange: The central portion of the audible frequency spectrum. The midrange is considered one of the most critical areas in speaker performance due to the location of human vocals and many instruments in this area. Midrange frequencies can range from as low as 200 Hz up to 4000 Hz, though the traditional range does not extend quite as high.

Minimum phase: Refers to data for which the phase portion (if unwrapped) is the closest it can be, at each point, to zero, but which still allows the system (that the frequency response represents) to be "causal". For instance, merely setting the phase values all to 0 degrees for a response curve would give the lowest phase shift, but if the data were then to be IFFT'ed to the time domain (to generate an Impulse Response or IR), the resulting IR would likely be seen to be active in its "negative time". Negative time is represented in the far right portion of an impulse response - that part can be interpreted as being a combination of the decayed portion of the response (which should be nearly zero) combined with the response before the input was applied (which must be zero in a causal system).

Miter Joint: A joint type in which both pieces of wood are beveled and glued together. Miter joints exhibit improved strength compared to butt joints because of the greater gluing surface area and the securing of end-grain to end-grain. Miter joints are most often used in situations where odd angles are being joined, or the builder does not want any exposed end grain. In MDF, miter joints are far superior to butt joints, because they join end-grain to end-grain and will not de-laminate the material when stressed.

Motor: The motor consists of the top plate, back plate, magnet, and pole piece. The motor contains the parts that are responsible for the motion of a loudspeaker diaphragm. The strength of the control over the voice coil and cone are determined by the motor design.

Mono-pole: The traditional type of loudspeaker in which sound is radiated in one direction. As opposed to a di-pole or bi-pole type speaker that radiates sound in multiple directions.

Mylar Capacitor: A capacitor type that uses Mylar as a dielectric. Considered superior to electrolytic capacitors, but not as good as polypropylene or film-and-foil capacitor types.

Near-Field: The near-field can be defined as any distance relative to a speaker at which driver integration is not fully complete. Typically the near-field of a speaker is distances less than 1 meter. Near-field measurements are useful because they can measure the response of a driver without the effects of the room. Near-field measurements are only good up to several hundred Hz however, since interactions with cabinet edges and across the driver itself are not taken into account. In studio monitoring situations, near-field and far-field listening techniques are used to evaluate a mix.

Non-Polar Electrolytic Capacitor: A type of capacitor that uses a thin layer of oxidized metal as the dielectric between layers. Non-polar refers to its ability to be used in either direction in a circuit.

Notch Filter - Parallel: A filter used in crossover construction that attenuates the signal only at a specific frequency. The "notch" can be adjusted to a specific frequency, depth and width. The most versatile type of notch filter is the "parallel notch filter" or "parallel trap circuit"; these are two different names for a combination of a resistor, inductor, and capacitor in parallel. By adjusting the values of these components, the location, width, and depth of the notch can be manipulated. These filters are very difficult to design, due to the complex interactions of the non-ideal portions of each component. A notch filter like this is usually designed using formulae to calculate approximate values and then trial-and-error to get the exact desired result. The advantages of a parallel notch filter are that they work independently from the impedance of the driver, and can be added to an existing crossover network.

Notch Filter - Series: The series notch filter is used primarily on tweeters and dome midranges to reduce the magnitude of the impedance peak at the resonant frequency. The large impedance peak on non-Ferro fluid enhanced domes can cause erratic performance of the crossover near the resonant frequency. An inductor, capacitor, and resistor are connected in series to each other, all of which are connected in across the terminals of the driver.

Nyquist Frequency - One-half the sampling frequency of the digital system. Signals applied to the input to such a system are subject to aliasing. In systems that sample at a frequency higher than that ultimately used, the sampling frequency to be considered is the lowest sampling frequency that occurs in the signal path.

Octave: An interval in the audio frequency spectrum equal to one-half or double of the starting value. One octave above 400 Hz is 800 Hz, one octave below 400 Hz is 200 Hz.

Odd-order: Any of the crossover slopes that are of an odd order, usually 1st or 3rd order.

Padding: A term used synonymously with attenuation, usually referring to reducing the output of a tweeter, i.e. padding down a tweeter.

Parallel: A method of connecting electrical components such that the voltage drop across each component is the same. In speaker building, this is accomplished by connecting the terminals such that "positive is to positive" and "negative is to negative." In a speaker, using two drivers in parallel causes an arguable 6dB increase in output. This is due to the halving of the system impedance and a doubling of radiating area. Two drivers in parallel in the same enclosure require a doubling of cabinet space. It is important to note that the total system impedance of two speakers in parallel will be half that of one driver.

Passive Crossover: An electrical filter within a loudspeaker responsible for dividing up the frequency spectrum and sending portions to the appropriate drivers. Most passive crossovers consist of a combination of resistors, capacitors, and inductors. They do not require any external power and are performed to the signal at the speaker-level.

Passive Radiator: A moveable piston often constructed like a woofer without a motor structure, which is used to tune a box to a certain frequency. The frequency at which a passive radiator (of a fixed diameter and compliance) is tuned is controlled by the moving mass of the diaphragm. Some benefits of a passive radiator enclosure are the elimination of extremely long ports, the non-existence of port noise, and higher frequencies will not leak out through the port.

Phase: Phase is a relative measurement of the difference between "where" periodic waveforms are at a given time. Two waves are "out of phase" (180 degrees) when the crest and trough of both waves occur at the same time. This will cause a cancellation of the two waves. When two separate speakers are connected "out of phase" a de-localized sound field will be created with a dramatic reduction of bass output. In crossover design, phase is used to measure the relative output of multiple drivers at a certain frequency. Some crossovers exhibit excellent phase response, meaning that inter-driver destructive interference is kept to a minimum. Phase can vary anywhere from 0 to 360 degrees and varies with frequency as well.

Piezo Tweeter: A type of tweeter that used a simple piezoelectric crystal as the diaphragm. These piezoelectric crystals mechanically vibrate as current passes through them. The advantage of this type of tweeter is that no crossover is needed due to their high internal impedance.

Planar Transducer: A loudspeaker drive unit that uses a thin film suspended between two magnets as the diaphragm. The voice coil itself is etched onto the diaphragm, and as current flow through it, the diaphragm moves back and forth. In a planar transducer, the diaphragm is attached along the length of the driver. Most planar transducers are dipolar by nature, though a rear chamber may be used to contain the back wave. Due to the limited excursion capabilities of most planar transducers, it is difficult to produce low bass frequencies.

Pole Piece: The portion of a loudspeaker transducer that provides the negative magnetic pole on the inside of a voice coil. In the most general sense, the pole piece is a cylinder of metal that is on the inside of the voice coil. A T-shaped pole piece has a smaller diameter towards the bottom, and is wider at the top, yielding a T-shaped profile. The widest portion of the T is directly across from the top plate of the magnet structure. A vented pole piece has a hole through it to allow air to enter the magnetic gap to increase cooling abilities.

Port: A cutout or tube in a vented box that "tunes" the box to a certain frequency. The port provides an additional air mass that is excited at its own frequency, enabling extended bass response. A port can be anything from a hole in a cabinet to a 4" diameter by 20" long piece of PVC. In a given box, a longer port corresponds to a lower tuning frequency.

Port Noise: An occurrence in vented box systems where erratic "wind noise" is created by the movement of air through the port. Port noise is generally described as a "chuffing" sound that will occur at maximum excursions of the driver. By increasing the diameter of a port, the speed at which air moves through the port will be reduced, in turn reducing port noise. Port noise can also be reduced by using flared ends on the port tubes, which provide a superior airflow across the transition from the port to the outside air.

Power Handling: The amount of electrical power that can flow through a loudspeaker driver before functioning ceases. Most often, the thermal power handling is quoted. This is a measure of how much power a driver can take before the voice coil insulation melts or other joints come apart and the driver fails. In reality, most drivers will reach a point of maximum excursion before their thermal power-handling limit is reached.

Push-Pull: A method of using two woofers in the same box. A push-pull enclosure uses two woofers on opposite sides of the box operating out of phase with each other. Overall performance will be roughly the same as in a traditional two-woofer format, except for a potential decrease in some forms of distortion. This occurs because some distortions may now be present out of phase thus canceling out.

PVC Pipe: The typical plastic pipe that is used for plumbing in homes. It is very useful for making ports in loudspeakers due to its availability in a variety of lengths and diameters.

Quasi-Anechoic: A method of making loudspeaker measurements that can produce results similar to those found in a full-blown anechoic chamber, but in a standard room. These measurements can be taken by "gating" out the reflections from the boundaries within the room. Quasi-anechoic responses will be very accurate above a certain frequency, but will have invalid results below that point.

Q: The losses or relative damping (ratio of stored to dissipated energy or ratio of reactive to resistive energy) of a system. In an impedance plot, a driver Q can be determined by how high and narrow the resonance peak is. A high, narrow peak indicates a high Q, while a lower, wide peak indicates a low Q.

Q Parameters: Qms, Qes, and Qts are measurements related to the control of a transducer's suspension when it reaches the resonant frequency (Fs). The suspension must prevent any lateral motion that might allow the voice coil and pole to touch (this could damage the loudspeaker). The suspension must also act like a shock absorber. Opposing forces from the mechanical and electrical suspensions act to absorb shock.

Q't: The Q of a loudspeaker's suspension plus the load of the rear chamber in a 4th-order bandpass box.

Qes: The losses or relative damping (ratio of stored to dissipated energy or ratio of reactive to resistive energy) of a driver at Fs, considering only its electrical (non-mechanical) resistances. This is a measurement of the control coming from the speaker's electrical suspension system (the voice coil and magnet).

QL: The Q of a vented speaker cabinet resulting from all of the box losses (acoustic weaknesses).

Qms: The losses or relative damping (ratio of stored to dissipated energy or ratio of reactive to resistive energy) of a driver at Fs, considering only its mechanical (non-electrical) resistances. Qms is a measurement of the control coming from the speaker's mechanical suspension system (the surround and spider).

Qtc: The Q of a sealed loudspeaker considering both mechanical and electrical resistances.

Qts: The losses or relative damping (ratio of stored to dissipated energy or ratio of reactive to resistive energy) of a driver, considering both mechanical and electrical resistances.

Rabbet: In woodworking, a groove or recess that is placed along the edge of a board to improve joint quality. Using rabbets to make a joint provides increased strength because of the increased gluing surface area, and the bonding of both end-grain and side-grain fibers.

Re: The DC resistance of a loudspeaker transducer, measured in ohms and it is often referred to as the 'DCR'. This measurement will almost always be less than the driver's nominal impedance. Consumers sometimes get concerned the Re is less than the published impedance and fear that amplifiers will be overloaded. Due to the fact that the inductance of a speaker rises with a rise in frequency, it is unlikely that the amplifier will often see the DC resistance as its load.

Resistance: A material's ability to resist the flow of electrons, measured in ohms.

Resistor: An electrical component that provides resistance to current flow. In crossovers, resistors are used in conjunction with other elements to produce various filters, and to attenuate output.

Resonant (Resonance) Frequency: The frequency at which an object will naturally mechanically vibrate.

Ribbon Driver: A type of loudspeaker transducer that uses a thin metallic film suspended between two magnets as the diaphragm. A true ribbon driver differs from a standard planar transducer in that the diaphragm is attached only at the ends. Thus, a true ribbon driver is very delicate, but is able to move very quickly and can produce high frequencies very accurately.

Ringing: A type of distortion found in loudspeakers that is usually caused by the natural resonances within a driver's cone material. It can also be used to refer to the poor transient performance of a driver or system.

Satellite Speaker: A typically small speaker designed to reproduce frequencies above a certain point. A satellite speaker is used in conjunction with a subwoofer to fill in the lower octaves. Advantages of satellites are the ability to use small enclosures that can be more discreet in a room. The disadvantage of them is the difficulty integrating the lower and higher frequencies.

Sd: The radiating surface area of a loudspeaker driver.

Sealed Enclosure: A type of loudspeaker enclosure in which the rear of the woofer fires into a tightly sealed chamber that is completely separate from the surrounding air space. Sealed boxes have a higher F3 than a vented box with the same woofer, but the low-end response rolls off at a shallower rate. Sealed boxes exhibit superior transient response and group delay characteristics than ported boxes.

Second Order Filter: A filter that uses a combination of two components to yield an approximate 12 dB per octave roll-off.

Sensitivity: A measure of the acoustic output of a loudspeaker resulting from the application of a fixed input power. Most often, sensitivity is measured in decibels at a distance of 1 meter from the source with 1 watt of input power (dB 1W/1m). It is also seen as dB at 2.83V/1m, which allows a better comparison of sensitivities regardless of load impedance. 2.83V is the amount of voltage necessary to deliver 1 watt of power into an 8-ohm load.

Series: A method of connecting multiple electrical components such that there is the same current through each. In loudspeakers, connecting woofers in series is accomplished by connecting the woofers "positive to negative." Putting two woofers in series will double the system impedance and system radiating area. The net result of using two woofers in series is no gain in efficiency. However, the increase in surface area will reduce the amount of excursion required to produce a given SPL, and a higher overall SPL can be produced.

Series Notch Filter: The series notch filter is used primarily on tweeters and dome midranges to reduce the magnitude of the impedance peak at the resonant frequency. The large impedance peak on non-Ferro fluid enhanced domes can cause erratic performance of the crossover near the resonant frequency. An inductor, capacitor, and resistor are connected in series to each other, all of which are connected in parallel across the driver.

Shielded: A loudspeaker driver or system that has a very small stray magnetic field making it acceptable for use near CRT-based screens. To shield a driver, a magnet of the opposite polarity is attached to the rear of the motor structure. This reduces the stray magnetic field considerably, but to further improve shielding a ferrous cup is placed over the entire magnet assembly. When done properly, the combination of the second magnet and the metal cup will reduce the stray field to almost nothing. Also, many drivers that use neodymium magnets are inherently shielded very well due to the smaller stray magnetic field.

Shorting Ring: See Copper Cap.

Snake Oil: A term used in the audio community to describe ideas or technologies that claim to give large improvements in performance, but there is little or no scientific evidence to back these claims.

Speaker: A driver or combination of drivers that is used as a system to convert electrical signals into acoustic output.

Spider: The corrugated cloth ring that attaches the voice coil to the frame of a speaker. The typical accordion spider is used because it allows forward and backward motion without allowing the voice coil to move side-to-side.

SPL: Sound Pressure Level. A measurement of acoustic output, made in decibels.

Step Response: A measurement of how a speaker responds to a theoretically infinitely fast transition from zero output to a finite output. An ideal step response will look like a triangle without any extra zigs in it.

Subwoofer: A loudspeaker transducer specifically designed to produce extremely low frequencies. A subwoofer will generally only be capable of output up to several hundred Hz. In home theater, a separate track is recorded for the subwoofer. These are typically powered by their own amplifier, separate from the main speaker amplifiers.

Surface Mounting: The installation of loudspeaker drivers such that the frame of them mounts on top of the baffle. This should generally be avoided to reduce diffraction around the driver frame, and is more important at higher frequencies.

Thiele-Small parameters: Thiele-Small parameters are a set of data characterizing the electrical and mechanical properties of a loudspeaker transducer. This data can be used to help design enclosures and predict the driver performance within them.

Third Order Crossover Network: A filter that uses three components to produce a roughly 18 dB per octave roll off.

Three-Way Speaker: A speaker system that uses three separate drivers to cover the entire audio spectrum. Usually a large woofer, a smaller midrange and a tweeter are used together.

Toe(in): A slight angle placed on a pair of speakers so that the drivers are facing at an angle other than perpendicular to the wall. This is helpful to make sure that the listening seat is at the proper axis relative to the speakers.

Top Plate: A steel plate that is on top of the magnet that transmits the positive magnetic pole to the outside of the voice coil. The frame is also attached to the motor structure through the top plate.

TQWT or TQWVP: Tapered Quarter Wave Tube or Voight Pipe. A type of transmission line enclosure where the driver is placed at the 1/3 point of the length of the line. The tapering of the line produces a slight horn-loading effect on the bass frequencies, which can help boost bass response.

Transient Response: A transient is a sudden change in signal amplitude, experienced in fast transitions from quiet to loud. In a speaker system, the transient response describes a speaker's ability to tightly control its cone motion. The cone will start and stop quickly to match the input signal with a minimal amount of distortion and time smearing.

Transmission Line: A type of loudspeaker enclosure that routs the back-wave of a speaker through a long tunnel that eventually exits via the front of the speaker. The theory of the transmission line is to make a line length equal to 1/4 or 3/4 of a wavelength, therefore the front and rear waves will add to each other upon exiting the front of the line. Because of the long, narrow rear enclosure, Helmholtz resonances tend to be a problem and produce significant ripples in the low frequency response. Line stuffing is generally used to increase the apparent length of the tube and distribute the Helmholtz resonance frequencies. However, getting exact line lengths tends to be difficult, and will often result in poor performance. There have been several very successful transmission line designs, however they are much more difficult to perfect than the traditional sealed and ported enclosures.

Treble: The highest frequencies in the audio spectrum. Generally from 4 kHz up to beyond audibility.

Tweeter: A type of loudspeaker transducer that is responsible for producing the top of the frequency range. There are several traditional styles of tweeters: ribbons, domes, and cones. The key feature of a tweeter is its low moving mass, enabling it to produce the fast vibrations necessary for high frequency reproduction. Another advantage of a tweeter is its ability to produce high frequencies well off-axis.

Two-Way Speaker: A speaker system that uses two drivers to cover the entire audio range. In most standard speakers, the crossover occurs in the 2 kHz to 4 kHz range.

Vas: The Thiele-Small parameter that measures the overall compliance of a loudspeaker transducer. The Vas is defined as the volume of air that has the same compliance as the driver.

Vented Enclosure: A type of loudspeaker enclosure where the woofer is mounted in an enclosed box except for one vent or port connecting to the outside air space. Vented enclosures feature a lower F3 point than a sealed box for the same woofer. The lower F3 comes at the expense of a steeper low-frequency roll-off. In a vented enclosure, the driver excursion is at a minimum at the box tuning frequency, and increases dramatically below this. Care must be taken to prevent over-excursion (unloading) of the woofer at frequencies below the tuning frequency.

Voice Coil: The coil of magnetic wire that moves within the magnetic gap of a loudspeaker transducer. The voice coil is the portion of a driver that directly transforms electrical energy into mechanical energy.

Voice Coil Former: A cylindrical tube of paper, aluminum, or Kapton® that the voice coil is wound on. A former must be very strong and have excellent thermal properties to prevent deformation of the voice coil at high power levels. A vented voice coil former has holes in it, through which hot air can escape to help cool the motor structure.

Woofer: A loudspeaker transducer that is responsible for producing sound in the 40 Hz to 200 Hz range. To produce a given SPL at lower frequencies, increased radiating area and increased excursion capabilities are needed. These increases are necessary for low frequency production, however they inhibit the driver's ability to produce high frequencies.

Xmax: The measurement of how far a diaphragm can move while still maintaining linear behavior. Traditionally it was defined as the voice coil length minus the air gap height. With the current high-strength motors, significant control over the cone is still possible even without the entire voice coil in the gap.

Z: Abbreviation for impedance. In T/S parameters, Z represents the nominal impedance of a speaker.

Zobel Network: Another name for a conjugate network, which is used to flatten the impedance rise found in woofers at increasing frequencies.

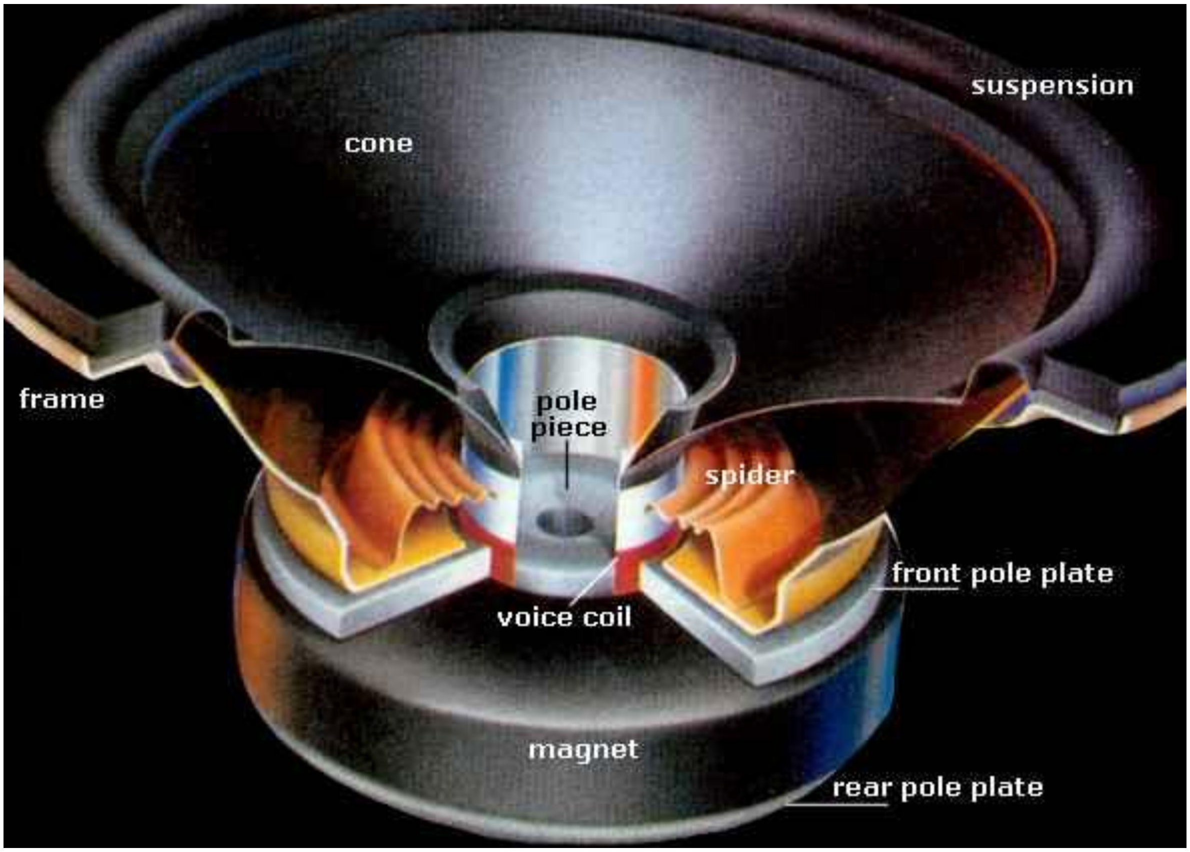
CHOOSING COMPONENTS

(Section Index)

Components should be chosen based upon the project that you are doing.

CHOOSING DRIVERS

Parts of a Driver



DRIVER CHOICE FACTORS

(Section Index)

When choosing drivers, the most important features will be choosing drivers that have proven themselves to be capable of good integration (in proven designs) and drivers that have a frequency and power range necessary for your project. Imaging will be enhanced if the drivers are closely matched in frequency response and impedance and if the driver's distortion is low. The wider the frequency demand on a single driver, the greater the Doppler distortion. A driver with a lower Qts or with a higher VAS will tend towards a better transient response. Guidance from experienced speaker builders can be of tremendous benefit.

The voice coil drive current along with the reactive back current caused by the current induced motion of the voice coil (EMF) modulates the magnetic gap field. Since this tends to be asymmetric, it can cause 2^{nd} harmonic distortion. A shorting ring at the base of the pole piece can decrease this distortion.

Non conductive voice coil formers (Fiberglass, Kapton) yield Qms's that are 2-4x higher than conductive voice coil formers (Aluminum). Non conductive formers do not introduce Eddy currents so they introduce less distortion. Non conductive formers typically have a Qms of over 4. Non conductive formers tend to have a higher output response, particularly at higher frequencies and this is most prominent off axis.

Larger voice coils (longer length) have more inductance due to the greater number of turns and more layers. This causes more of an upper frequency roll off. This also causes less efficiency due to the greater weight on the coil.

To achieve a higher frequency extension, the ratio of voice coil mass/cone mass must be as small as possible.

Conical diaphragms tend to have a high peak at the higher frequency end but they have a wider bandwidth. Convex diaphragms have a smoother frequency response with less of a peak.

Solid dust caps have their own air chamber resonance that can be addressed by venting the pole piece or the voice coil. Porous dust caps don't build up the resonance but tend to radiate sound out of phase with the diaphragm.

Concave cones are higher in efficiency in the upper frequency range but they have a more narrow directivity than convex diaphragms.

Compliance is as a result of the suspension system and is controlled by the spider (about 80% of the effect) and by the surround (about 20% of the effect).

Rubber surrounds tend to smooth the response and provide better damping but they are more costly than foam. Santoprene looks like rubber, is inexpensive like foam, and provides poor damping ability.

The following are general rules of thumb taken from Lynn Olson (http://www.aloha-audio.com/library/speaker-design2.html):

		Strength	Weakness
Tweeters			With dome tweeters, there is a path length difference between the center of the dome and the rim due to the dome height with a phase impact on frequency response such that there is a high frequency drop off due to phase nulling. The higher the dome, the lower the frequency of the first dip. A phase shield adds additional propagation delay to break up the phase difference and "fill in" the null. Soft domes are better able to compensate for phase losses than are harder domes. Lower profile dome tweeters are less subject to this problem.
	Soft Dome (fabric)	Self damping, fairly flat response, natural open sound, non fatiguing	Can occasionally be a dull sound
	Metal Dome (Aluminum, Titanium, Beryllium)	Uniform piston action, high resolution sound, good transparency, excellent dispersion,	Ultrasonic peaks which can cause resonance in sound causing IM distortion, average to poor damping
	Ribbon	Excellent transient response, low distortion	Generally requires a high frequency crossover or high order crossover
Mid/Woofer			
	Paper	Excellent self damping, resolution, good detail, flat response, and gradual cone break up	Less detail than rigid types, medium efficiency, properties may change over time due to wear
	Soft Dome	Measures well but sound had been challenged	Limited bandwidth and power handling, side by side rocking
	Bextrene	Excellent imaging, excellent consistency	Low efficiency, coloration of sound, rapid break up at high levels, some high frequency resonance
	Polypropylene (often combined with fillers such as mica, talc, carbon black, acrylic)	Flat response, low coloration, good impulse response with good internal damping	Poor transparency, Doesn't integrate well with metal
	Rigid-Carbon Fiber	True piston action, outstanding bass and midrange	Characteristic double peak region at the top end of the working range that generally requires 2 notch filters to remove
	Rigid-Kevlar	Good frequency response, smooth roll off, low IM distortion	Could have a chaotic break up, ringing could be a problem, may require a notch filter
	Rigid Overall	Best transparency, imaging, depth, high efficiency, high peak levels, low IM distortion	Some have severe upper end peaking, have chaotic break up, may have a fatiguing sound, may require a notch filter

CHOOSING CROSSOVER COMPONENTS

(Section Index)

1. Sound of ***Capacitors*** (in ascending order of general quality):

NP electrolytics	Muddy, Veiled sound
Mylar	Detailed, Transparent, More grainy sound
Polypropylene	Detailed, smooth
Metalized film	Best after burn in, double better than single
Film and Foil	The more layers the better, may be more difficult to manually work with.

2. For Capacitors, use only Bipolar (non polar) types in crossovers. Note that non-electrolytic or non-ceramic caps are non polar. If you require a large value, you could wire electrolytic capacitors in series with opposite polarities.
3. As a rule of thumb, use caps rated with a voltage of at least half that of the wattage at which you will drive the speaker.
4. In a low pass filter, the parallel capacitor's job is to show a decreasing impedance to the series inductor with the rise in frequency. Putting the capacitor on the amplifier side is a BAD idea; it will defeat the purpose, and show a very low impedance to the amplifier at high frequencies.
5. Look for a low Series Resistance and Low Dissipation Factor.

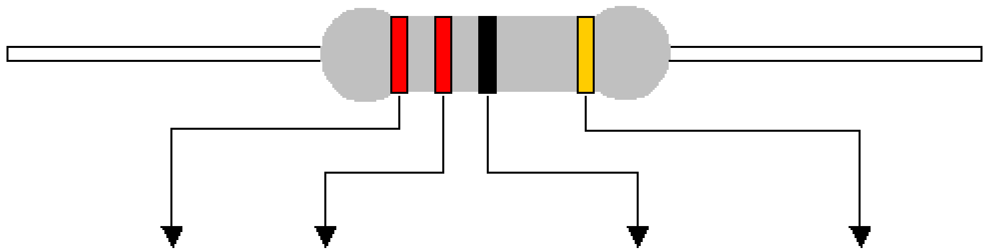
For further reading see the following links:

http://www.capacitors.com/picking_capacitors/pickcap.htm
http://members.aol.com/sbench102/caps.html
http://www.geocities.com/kreskovs/CapTests.html

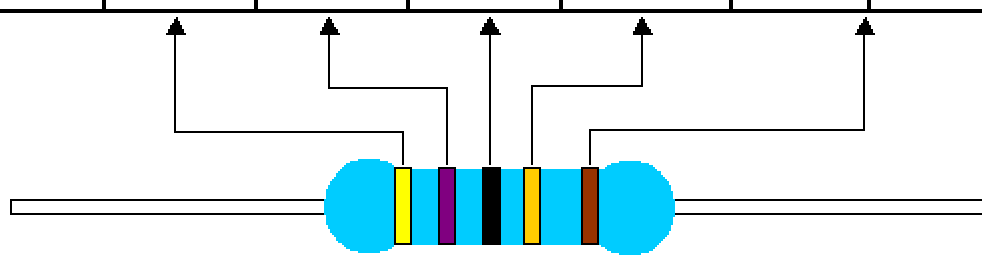
Resistors:

1. Generally, non-inductive resistors may be better in crossovers than other types.
2. Do make sure that the resistors that you use are rated highly enough, especially if you are likely to drive the speakers hard. You will likely be safe with resistors rated at 10 watts if you are unsure.
3. The following is a chart regarding how to read resistor values from http://www.token.com.tw/resistor/resistor-color-code.htm

TOKEN RESISTOR COLOR CODE



COLOR	1ST BAND	2ND BAND	3TH BAND	MULTIPLIER	TOLERANCE	
BLACK	0	0	0	1		
BROWN	1	1	1	10	± 1%	F
RED	2	2	2	100	± 2%	G
ORANGE	3	3	3	1K		
YELLOW	4	4	4	10K		
GREEN	5	5	5	100K	± 0.5%	D
BLUE	6	6	6	1M	± 0.25%	C
VIOLET	7	7	7	10M	± 0.10%	B
GREY	8	8	8		± 0.05%	A
WHITE	9	9	9			
GOLD				0.1	± 5%	J
SILVER				0.01	± 10%	K
PLAIN					± 20%	M



Inductors:

1. Air core: less likely to saturate but harder to get small size and higher inductance.
2. Metal core: used where high values are necessary.
3. Inductors should be air core with the lowest DCR possible for the Woofer Circuit, as iron core type will more easily saturate. This is less critical in the tweeter circuit as saturation is less likely.
4. Vance Dickason suggests using Inductors that have a DCR of 1/10th that of the voice coil or less.

CALCULATING NETWORK ELECTRICAL RELATIONSHIPS

(Section Index)

	Series	*Parallel*
L	Adds	Adds Reciprocally
Q	Same	Adds
I	Same	Adds
V	Adds	Same
R	Adds	Adds Reciprocally
C	Adds Reciprocally	Adds

Spreadsheet to Calculate Capacitors and Resistors in Series and In Parallel

"caps and resistor values.xls"

Note: One method of measuring inductance is the following:

1. connect the output of the sound card to the amplifier
2. connect the output of the amplifier to a resistor (accurately measured) in series with an inductor.
3. Generate a 1 kHz signal (via the soundcard) and measure the AC voltage with a DMM across the resistor.
4. Next measure the AC voltage across the inductor while playing the 1 kHz signal.
5. The inductance is equal to (Resistor*voltage across the inductor)/(Voltage across the resistor*2*pi*1000).

WIRING

(Section Index)

Internal Wiring

1. Internal wiring is often argued. Consider the use of Cat 5e for the tweeter and woofer signals and 14 g stranded for the grounding circuit.

External Wiring

1. A rough formula (from www.pcavtech.com/techtalk/wire_size/index.htm) to choose loudspeaker cable is as follows (see Appendix for a more complete AWG/Metric Conversion chart):
    a. Desired resistance per foot=DCR of the speaker/(wire length [ft] *200) or to use the chart below for metric it would be DCR/(wire length in centimeters * .61)
    b. Choose a gauge that multiplies out to a lower number than below:

American Wire Gauge	8	10	12	14	16	18	20	22	24
	.00067	.0010	.0017	.0026	.0042	.0066	.011	.017	.027

    c. Attenuation = 20 * log (Impedance at the frequency tested/total impedance of driver at freq tested + total impedance of the cable) in dB
    d. Targeting less than a 0.12 or 0.13 dB loss (0.1 or lower is ideal) is another reasonable approach. Here is a spreadsheet to assist with that:

"Wire Resistance Calc1.xls"

    e. At a given frequency, the impedance of that inductance(Z) = 2 *PI*f*L(in Henries (not micro henries)
    f. Inductance(L) = 0.281*log [(conductor spacing center to center in inches)/(conductor radius in inches)]micro henries per foot

Note that speaker cables can act as antennas in the AM frequency band and may cause distortion in the output stage of a solid-state amplifier, if strong radio frequency signals are present. In particular, the cable capacitance in conjunction with the inductance of a driver voice coil may form a resonant circuit for these frequencies. The resonance can be suppressed by placing a series R-C circuit of 10 ohm/2 W and 0.33 uF/100 V across the cable terminals at the speaker end.

AWG to Metric Conversion Chart with Resistance

AWG Number	Ø [Inch]	Ø [mm]	Ø [mm²]	Resistance [Ohm/m]	Resistance [Ohm/ft]
1	0.289	7.35	42.4	0.000407	0.000124
2	0.258	6.54	33.6	0.000513	0.000156
3	0.229	5.83	26.7	0.000647	0.000197
4	0.204	5.19	21.1	0.000815	0.000248
5	0.182	4.62	16.8	0.00103	0.000314
6	0.162	4.11	13.3	0.00130	0.000396
7	0.144	3.66	10.5	0.00163	0.000497
8	0.128	3.26	8.36	0.00206	0.000628
9	0.114	2.91	6.63	0.00260	0.000792
10	0.102	2.59	5.26	0.00328	0.001000
11	0.0907	2.30	4.17	0.00413	0.001259
12	0.0808	2.05	3.31	0.00521	0.001588
13	0.0720	1.83	2.62	0.00657	0.002002
14	0.0641	1.63	2.08	0.00829	0.002527
15	0.0571	1.45	1.65	0.0104	0.003170
16	0.0508	1.29	1.31	0.0132	0.004023
17	0.0453	1.15	1.04	0.0166	0.005059
18	0.0403	1.02	0.823	0.0210	0.006400
19	0.0359	0.912	0.653	0.0264	0.008046
20	0.0320	0.812	0.518	0.0333	0.010149
21	0.0285	0.723	0.410	0.0420	0.012801
22	0.0253	0.644	0.326	0.0530	0.016154
23	0.0226	0.573	0.258	0.0668	0.020360
24	0.0201	0.511	0.205	0.0842	0.025663
25	0.0179	0.455	0.162	0.106	0.032307
26	0.0159	0.405	0.129	0.134	0.040841
27	0.0142	0.361	0.102	0.169	0.051509
28	0.0126	0.321	0.0810	0.213	0.064919
29	0.0113	0.286	0.0642	0.268	0.081682
30	0.0100	0.255	0.0509	0.339	0.103322

Resistance:

- AWG 15 copper is about 10 milli-Ohm per meter.
- Adding 3 to the AWG number doubles the resistance; Subtracting 3 halves.
- Adding 10 to the AWG number tenfolds the resistance; Subtracting 10 reduces by a factor 10.

Diameter:

- AWG 18 has a solid core diameter of about 1.0 mm.
- Adding 6 to the AWG number halves the diameter; Subtracting 6 doubles.
- Adding 20 to the AWG number reduces the diameter by a factor of 10; Subtracting 20 tenfolds.

DAMPING FACTOR CALCULATIONS

(Section Index)

To calculate the Damping Factor of an amplifier,

1. Use a 10 Watt Non Inductive Wire Wound resistor
2. Keep the voltage below the resistors rating.
3. Select a frequency and measure the voltage. Call this V1.
4. Add a resistor in series and measure the voltage again. Do this quickly as the resistor will get hot.
5. Call this V2 and it should be less than V1.
6. The amplifier's Impedance (I) is V2/R.
7. The Output Impedance=(V1-V2)/I.
8. The Damping Factor = (the Speaker Impedance at that frequency)/(The amplifier output Impedance).
9. Note that if the voltage increases with a load, it is indicative of bad internal wiring and these amplifiers should generally be avoided. This suggests a negative impedance and the voltage will rise with an increasing load.

Amplifiers have a source impedance or an internal output impedance. Loudspeakers present a load impedance to the amplifier as well. The ratio of the load impedance to the source impedance is the Damping Factor of the Amplifier. The typical amplifier source impedance might be around .02 ohms. The damping Factor will vary based upon the load with an 8 ohm speaker yielding a DF of 400 while a 4 ohm speaker yields a DF of 200. The DF describes the amplifier's ability to control or regulate the loudspeaker load.

One way to measure this is to measure the output voltage of an amplifier with no load and then place a load and measure the value again. The (No Load Voltage-Load Voltage)/Load Voltage equals the amplifier's percentage of regulation. The reciprocal of this number is the damping factor into that particular load. Note that DF varies with load. So if we use a 4 ohm resistor and the No Load Voltage was 10 Volts, and the Load Voltage was 9.95 Volts, the regulation would be (10-9.95)/9.95 or .0050 or .5%. The reciprocal of .005 is 199 which would be the Damping Factor at 4 ohms. The source impedance can now be calculated remembering that the Load Impedance/Source Impedance = DF so Load Impedance/DF=Source Impedance or 4/199 = .02 ohms.

This becomes important when thinking about the amount of load impedance added by cables. A 22-gauge wire will add 1 ohm for every 60.75 feet while a 16 gauge wire will add 1 ohm for every 153.6 feet. You would double this for AC current used in speakers as two wires are leading to the loudspeaker. At 30 feet of wire you would have to add roughly one ohm to the load resistance for the 22 gauge wire or .4 ohms for the 16 gauge wire. The difference in load of 5 vs 4.4 ohms results in a decrease in power output as Power = Voltage^2/Resistance so there would be (for our 10 Volt example) 100/5 vs 100/4.4 or 20 vs 22.7 watts. Of the wattage produced, the speaker is seeing 4/5 * 20 or 16 watts (20% power loss) in the first example and 4/4.4 * 20 or 18.2 watts in the second example (vs a 10% power loss). In addition, the damping factor difference is 4/(.02+1) or 3.3 in the first case and 4/(.02+.4) or 9.5 in the second case. In essence, there will be a significant difference in amplifier control and in power output. It is generally recommended to target a Source to Load Impedance Ratio of 7-10:1 with higher being better.

COMMONLY USED FORMULAS AND DEFINITIONS (NOT NECESSARILY THE ONES USED IN SPEAKER WORKSHOP)

Adapted from Bill McFadden @DIYsubwoofer.com 1993 with some formula additions from other sources.

(Section Index)

Definitions

\Gno	Driver efficiency (\Gn is Greek character eta)
Amax	Maximum amplitude of loudspeaker frequency response
ap	effective radius of port in m
aw	piston radius of the driver
Bl	Force factor
c = 344.8 speed of sound	
Cab	acoustic compliance of the box
Cas	Acoustic Compliance
Cms	mechanical compliance
Cms	Suspension Compliance
d	box depth
dBpeak	Maximum peak or dip of loudspeaker system response
Dia	Diameter of driver
Dmin	Minimum diameter of tubular vent to prevent excessive vent noise
Dv	Diameter of tubular vent
eg = 2.83 V generator output	
F3dB	Half-power (-3 dB) frequency of loudspeaker system
Fb	Resonance frequency of enclosure (box tuning frequency)
Fmax	Upper frequency limit of driver's piston range
Fs	Free air resonant frequency of the driver in Hz
K1	Power rating constant
K2	SPL rating constant
Le	Voice Coil Inductance (the industry standard is to measure at 1kHz; higher frequency leads to higher impedance
Lea	VC inductance transformed to acoustical side
Lv	Length of tubular vent
Ma1	radiation mass of the front of the driver
Ma2	acoustic mass of outer end of port
Mab	air load mass at the back of the driver in the box
Mad	acoustic mass of the voice coil assembly
Map	acoustic mass of the air in port
M_{AS}	Acoustic mass
Mm1	mass of air on one side of diaphragm
Mmd	mass of the voice coil assembly
M_{MS}	Moving mass
Par	Estimated displacement-limited acoustic power rating
pax	pressure at distance x
PeakSPL	Thermally-limited RMS sound pressure level in passband
PEmax	Thermally-limited maximum RMS input power
Per	Estimated displacement-limited electrical power rating
pi = 3.14159265359	

Qb	Total Q of system at Fb
Qb	absorption loss of the box
Qbt	total box loss
Qe	electrical Q of the driver
Qes	electrical Q value of the driver
Ql	leakage loss of the box
Qm	mechanical Q of the driver
Qp	total Q of the port
Qr	Ratio of Qb to Qts and Fb to Fs
Qt	total Q of the driver
Qts	Total driver Q at Fs
Rab	real part of box impedance
Ral	enclosure leakage loss
Rap	acoustic resistance of the port
Rar1	resistance of the front side of the driver
Rar2	resistance of the front side of port
Ras	acoustical resistance of suspension
Re	voice coil DC resistance in ohm
Rea	electrical DC resistance transformed to acoustical side response

rho0 = density of air =1.20997 kg/m^3

Rm	Real part of air load
Rms	mechanical resistance of suspension

ro = 1.18 density of air 1.18kg/m3

Ro = density of air (1.18 kg/m^3)

Sb	area of panel on which the driver is mounted
Sd	Estimated effective projected surface area of driver diaphragm (Equivalent piston area in m^2)
Sp	effective area of port in m2
SPL	Efficiency of driver in dB SPL at 1W/1m
Sqrt	meaning the square root of ...
Sw	piston area of the driver
t	length of port

u = 1.56/100000.0 kinematic coefficient of viscosity

Vas	Volume of air having same acoustic compliance as driver suspension
Vb	Inside volume of enclosure
Vd	Peak displacement volume of driver diaphragm
Vr	Ratio of Vas to Vb
Xm	Imaginary part of air load
xmax	Peak displacement limit of driver diaphragm (.5*throw)

Closed-Box Systems

(Section Index)

\Gno = 10^((SPL-112)/10)
Amax = Qb^2/(Qb^2-0.25)^0.5 for Qb >(1/2)^0.5, 1 otherwise
dBpeak = 20*LOG(Amax)
F3dB = Qr*Fs*((1/Qb^2-2+((1/Qb^2-2)^2+4)^0.5)/2)^0.5
Fb = Qr*Fs
Fmax = c/(pi*0.83*Dia)
K1 = (4*pi^3*Ro/c)*Fb^4*Vd^2
K2 = 112+10*LOG(K1)
Par = K1/Amax^2
PeakSPL = SPL+10*LOG(PEmax)
Per = Par/(\Gno)
Qr = (1/Qts)/(1/Qb-0.1)
Sd = pi*(Dia*0.83)^2/4
Vb = Vas/Vr
Vd = Sd*xmax
Vr = Qr^2-1

Frequency-dependent equations:
dBmag = 10*LOG(Fr^2/((Fr-1)^2+Fr/Qb^2))
Fr = (F/Fb)^2
ws = 2 *pi * fs
Pmax = K1*((Fr-1)^2+Fr/Qb^2))/(\Gno)
SPLmax = K2+40*LOG(F/Fb)
Thermally-limited RMS SPL = PeakSPL+dBmag
Frequency of Xmax in a closed box =((1-(1/(2*Qtc^2)))^.5)*Fc

Ported Box Systems

(Section Index)

Fmax = c/(pi*0.83*Dia)
Par = 3*F3dB^4*Vd^2
Per = Par/(\Gno)
\Gno = 10^((SPL-112)/10)
PeakSPL = SPL+10*LOG(PEmax)
Lv = 2362*Dv^2/(Fb^2*Vb)-0.73*Dv
Vd = Sd*xmax
K1 = (4*pi^3*Ro/c)*Fs^4*Vd^2
K2 = 112+10*LOG(K1)
Mat Mat = Ma2 + Map
Qp total Q of the port
Ma Ma = Mad + Ma1 + Mab
Ra1 Ra1 = Rea + Ras + Rar1
Rat Rat = Rap + Rar2
Sw = pi*aw*aw
Qt = Qm*Qe/(Qm+Qe)
Cms = Vas/(Sw*Sw*ro*c*c)
Rms = 1/(2*pi*Fs*Cms*Qm)
Mmd = 1/(2*pi*Fs*2*pi*Fs*Cms)
Mad = Mmd/(Sw*Sw)
Ras = Rms/(Sw*Sw)
Cas = Cms*(Sw*Sw)
Vas = Cas*(ro*c*c)
Fs = 1/(2*pi*sqrt(Cas*Mad))
Rea = (Bl*Bl)/(Re*Sw*Sw)
Lea = (Le/(Bl*Bl))*Sw*Sw
Rar1= (pi*Fs*Fs*ro)/c
Ma = Mad + Ma1
Ra1 = Rea + Ras + 2*Rar1
Qm = (2*pi*Fs)*Mmd/Rms; also Qm = 1/(2*pi*Fs*Cas*Ras)
Qe = 2*pi*Fs*Re*Ma*Sw*Sw/(Bl*Bl)
Qt = Qm*Qe/(Qm+Qe); also Qt = (2*pi*Fs)*Ma/Ra1
Rm = (Bl*Bl)/Re + Rms + 2*(2*pi*Fs*2*pi*Fs*Sw*Sw*ro)/(2*pi*c)
Xm = 2*pi*Fs*(Mmd + 2*Mm1) - 1/(2*pi*Fs*Cms)
SPL = 20*log10(pax/0.00002); (referenced to Po)
Sp = pi*ap*ap
Mat = Ma2 + Map
Mab = (ro*d*Sw)/(3*Sb*Sb) + 8*ro*(1 - Sw/Sb)/(3*pi*sqrt(pi*Sw))
Cab = Vb/(ro*c*c)
Fb = 1/(2*pi*sqrt(Cab*(Mat)))
Qb = 1/(2*pi*Fb*Cab*Rab)
Rab = 1/(2*pi*Fb*Cab*Qb)
Rap = (ro/Sp)*sqrt(2*2*pi*Fb*u)*(t/ap + 1)
Rar2= (pi*Fb*Fb*ro)/c
Rat = Rap + Rar2
Qp = (2*pi*Fb)*Mat/Rat
Ql = 2*pi*Fb*Cab*Ral
Qt = 1/(1/Qp+1/Qb+1/Ql); also Qt = 1/(1/Qp+1/Qb)

Mm1 = 2.67*aw*aw*aw*ro
Sd = pi*(Dia*0.83)^2/4
dBpeak = 20*LOG(Qts*(Vas/Vb)^0.3/0.4)
Dmin = (Fb*Vd)^0.5
Ma2 = (0.6*ap*ro)/Sp
Map = (t + 0.6*ap)*ro/(pi*ap*ap); (includes inside loading)
Ma1 = 0.27*ro/aw
Fb = (Vas/Vb)^0.31*Fs
F3dB = (Vas/Vb)^0.44*Fs
pax = 2*(eg*Bl*ro)/(Re*Ma*4*pi*Sw*1.0); (in half-space)
pax = (eg*Bl*Sw*Fs*ro)/(1.0*Re*sqrt(Rm*Rm +Xm*Xm))
Vb = 20*Qts^3.3*Vas

Frequency-dependent equations:
A = (Fb/Fs)^2
B = A/Qts+Fb/(7*Fs)
C = 1+A+(Vas/Vb)+Fb/(7*Fs*Qts)
D = 1/Qts+Fb/(7*Fs)
dBmag = 10*LOG(Fn4^2/((Fn4-C*Fn2+A)^2+Fn2*(D*Fn2-B)^2))
E = (97/49)*A
Fn2 = (F/Fs)^2
Fn4 = Fn2^2
Pmax = (K1/\Gno)*((Fn4-C*Fn2+A)^2+Fn2*(D*Fn2-B)^2)/(Fn4-E*Fn2+A^2)
SPLmax = K2+10*LOG(Fn4^2/(Fn4-E*Fn2+A^2))
Thermally-limited RMS SPL = PeakSPL+dBmag
ws = 2 *pi * fs

Electrical Impedance

(Section Index)

Bl=sqrt((rho0*c^2*Sd^2*Re)/(ws*Vas*Qes))
Cmeb = Mab*(Sw*Sw)/(Bl*Bl)
Cmep = Map*(Sw*Sw)/(Bl*Bl)
Cmep2= Ma2*(Sw*Sw)/(Bl*Bl)
Cmer = Ma1*(Sw*Sw)/(Bl*Bl)
Cmes = Mad*(Sw*Sw)/(Bl*Bl)
Cms=Vas/(rho0*c^2 *Sd^2)
Lces = Cas*(Bl*Bl)/(Sw*Sw)
Lecb = Cab*(Bl*Bl)/(Sw*Sw)
Qes=1/(ws*Cms*Res)
Reb = (Bl*Bl)/(Sw*Sw*Rab)
Rel = (Bl*Bl)/(Sw*Sw*Ral)
Rep = (Bl*Bl)/(Sw*Sw*Rap)
Rer = (Bl*Bl)/(Sw*Sw*Rar1)
Rer2 = (Bl*Bl)/(Sw*Sw*Rar2)
Res = (Bl*Bl)/(Sw*Sw*Ras)
Res=(Bl)^2 / Re

Efficiency

Reference Efficiency (n_0) = K*(Fs^3*Vas)/Qes where K=9.64*10^-10 (liters); 9.64*10^-7 (m^3); 2.7*10^-8 (ft^3). This will be a fraction. Multiply by 100 for percentage. In SpL per 1 watt at 1 meter=112/(10*Log(n_0)). Ranges of .35% to 1.5% are typical. This is most accurate if measured on a baffle or in an enclosure that is approximately the same size as the speaker cabinet that the driver will be in. In a sealed box, (n_0)=(K*Fc^3*Vas*Vb)/(Qec*(Vas/Vb)).

General Formulae

- The phase at any point from the source is equal to sin(2*pi*frequency*time) + original phase angle.
- 2pi radians equals 360 degrees and 1 radian equals 57.2958 degrees
- omega or the radian frequency = 2*pi*f
- 1 degree = .01745 radians
- One period of a sine wave = time it takes to pass through one cycle = 1/frequency
- RMS voltage = Peak voltage/SQRT(2)
- Wavelength = c/f
- Inverse square law: For each doubling of distance away from the source, the dB reading will change as (Reading at distance 1) – (20*Log(distance 1/distance 2))

ABSORPTION AND REFLECTIVE PROPERTIES

(Section Index)RT-60 to Sabins. RT-60 is room echo time. Depends on the amount of absorption in the room, and is an INVERSELY LOGARITHMIC relationship. Doubling the amount of physical Sabins in a room will reduce room time by more than half - often much more. Acoustical Loss of Consonants (AlCons) expressed as a percentage at a given distance, amount to clarity of sound. The relationship is simply INVERSE to the effective Sabin count. Double the Sabins, double the distance for a given percentage of AlCons. Full absorption is 1; Full reflection is 0

Chris Whealy has developed a spreadsheet that is primarily for use with the development of a control room for a recording studio. These principles can be applied to box design as well. It is a very comprehensive spreadsheet that is from http://www.whealy.com/acoustics/ and is © Chris Whealy, 2005. Following it is another spreadsheet with additional values that may be plugged into the page called Absorption coefficients.

"Control Room Calculator V2.62 XL97

abscoeff.xls

Note that when evaluating modal behavior, these are always calculated based upon the rigid boundaries so carpet, porous absorbers, suspended ceilings, etc. do not count and you would use room dimensions to the rigid boundaries behind them.

EFFECT OF STUFFING

(Section Index)

Tom Nousaine (www.integracaraudio.com/caraudio/resources/fiberfill/) reports that a sealed enclosure will be affected in the following way from the use of Dacron-polyester fiberfill:

SEALED ENCLOSURE 1.4-ft³ Box			
Stuffing Density (lb/ft³)	System Resonance (Fsb)	Effective Size	Percentage Gain
0	56.6	1.4	--
0.70	53.0	1.6	14%
0.75	52.7	1.7	21%
1.50	51.7	1.8	29%
1.75	50.8	1.9	36%
2.60	50.4	1.6	14%
3.10	52.6	1.2	-14%

SEALED ENCLOSURE 5.1-ft³ Box			
Stuffing Density (lb/ft³)	System Resonance (Fsb)	Effective Size	Percentage Gain
0	42.0	5.1	--
0.25	42.0	5.1	0%
0.50	41.2	5.8	14%
0.75	40.3	6.2	22%
1.00	39.4	6.5	27%
1.25	38.6	6.5	27%
1.50	40.2	5.6	9%

PORTED ENCLOSURE 1.4-ft³ Box			
Stuffing Density (lb/ft³)	System Resonance (Fsb)	Effective Size	Percentage Gain
0	42.0	1.4	--
0.40	39.1	1.6	14%
0.85	37.2	1.8	29%
1.25	35.2	1.9	36%
1.40	34.2	2.0	43%
1.75	35.2	1.9	36%

WINDOWING AND FFT'S

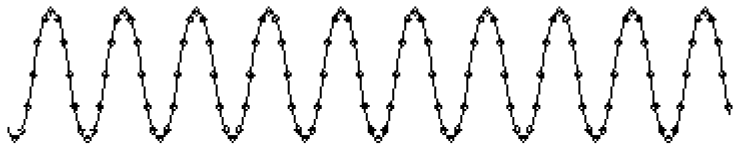
(Section Index)

When doing sine wave measurements, a sine wave passing through zero amplitude at both the beginning and end of a time window will result in an FFT spectrum that consists of a single line with the correct amplitude and frequency. If the initiation of the sine wave does not coincide with both the start and end points of the time window, the waveform will be truncated and the signal sample will then be discontinuous. This leads to something called leakage. Leakage amounts to spectral information from an FFT showing up at the wrong frequencies. Spectral leakage is the result of an assumption in the FFT algorithm that the time record is exactly repetitive and that the signals are perfectly periodic at intervals that correspond to the length of the time sample observed. If the time record has a non-integer number of cycles, this does not occur. The continuous spectrum of the window gets shifted from the center of the main lobe to a fraction of the frequency that corresponds to the difference between the frequency component and the FFT line frequencies (smearing the spectrum). This shift causes side lobes to appear in the spectrum and it causes some amplitude error as well at the frequency peak because the main lobe is now sampled off center. There would be no leakage if one could perfectly coincide the testing window zero time points with the sampling times but this is nearly impossible to do in practice. The use of windowing attempts to address this issue.

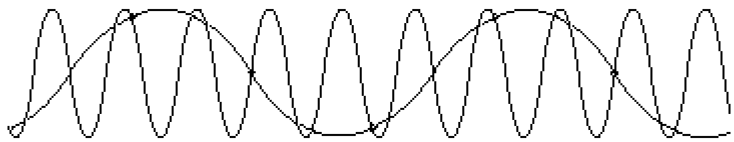
Windowing forces the signal to be periodic in time by multiplying it by a data window. The data windows smooth and taper the signal to zero at each end of the sample period. Windowing functions act on raw data to reduce the effects of the leakage that occurs during an FFT of the data. Each of the windows does this a little differently so the results will vary. While smoothing windows do a good job of forcing the ends to zero, they also add distortion to the time series, which results in side lobes. These lobes effectively reduce both the frequency resolution of the analyzer (the appearance is as if the spectral lines are wider) as well as the amplitude resolution (because a portion of the sample is removed during the windowing process). Windows attempt to compensate for the removed amplitude by adding to the values near the center of the sequence.

If you collect an exact integer number of cycles, all windows yield the same peak amplitude and have excellent amplitude accuracy. If you use Uniform (Rectangular) windowing (which is essentially no windowing) and use entire range, the signal will not be altered. At a non-integer number of cycles, the Rectangular Window introduces the greatest amount of spectral leakage.

According to the Nyquist criterion, the sampling frequency must be at least twice the maximum frequency component in the signal. If this is not adhered to, frequency components above half of the sampling rate are introduced below half of the sampling rate resulting in erroneous representations of the signal called aliasing. You can't avoid leakage, but by windowing functions minimize the effects of performing an FFT over a non-integer number of cycles when applied to data prior to performing an FFT. The diagrams below are from The Fundamentals of FFT-Based Signal Analysis and Measurement in LabVIEW and LabWindows/CVI at http://zone.ni.com/.

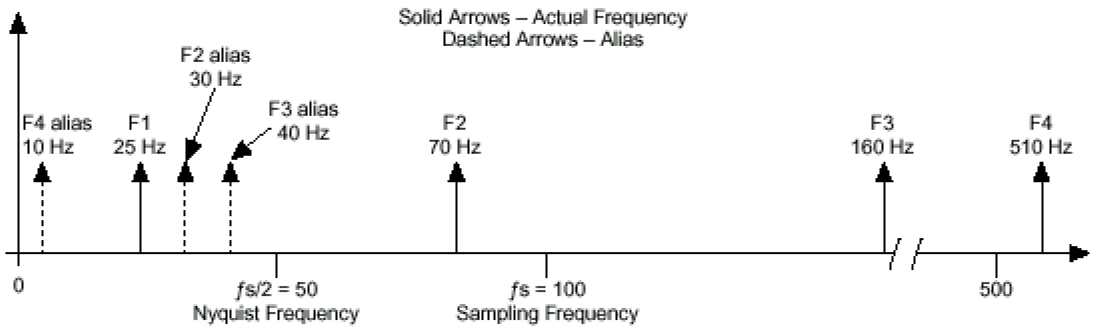


Adequately sampled signal



Aliased signal due to undersampling

The diagram below shows the alias frequencies that appear when the signal with real components at 25, 70, 160, and 510 Hz is sampled at 100 Hz. Alias frequencies appear at 10, 30, and 40 Hz.



Windows have a main lobe around the frequency of interest. The main lobe is the frequency domain characteristic of the window. The peak along with the – 3 dB (.707 of the peak gain) and – 6 dB (0.5 of the peak gain) points of the main lobe define the shape of the window response. Side-lobes represent leakage, i.e. energy appears outside the main lobe. The width of main lobe determines the resolution capabilities. All windows have various main/side lobe ratios and transition widths. A wider transition region causes □lower side lobes and a narrower transition region causes □higher side lobes.

The sampling frequency determines the frequency range (spectrum bandwidth). Frequency resolution is determined by the number of points acquired in the time domain signal record. To increase the frequency resolution for a given frequency range, one would increase the number of points sampled at the same sampling frequency.

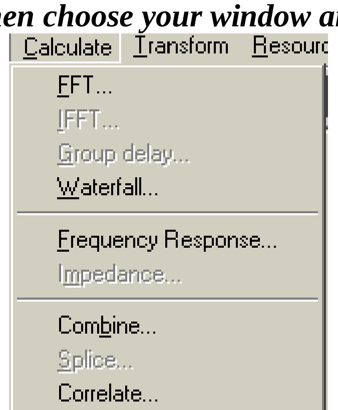
Different windows provide a trade-off between resolution and leakage. The Hanning Window is optimized for Impulse response data while the Blackman Window is optimum for power spectrum evaluation.

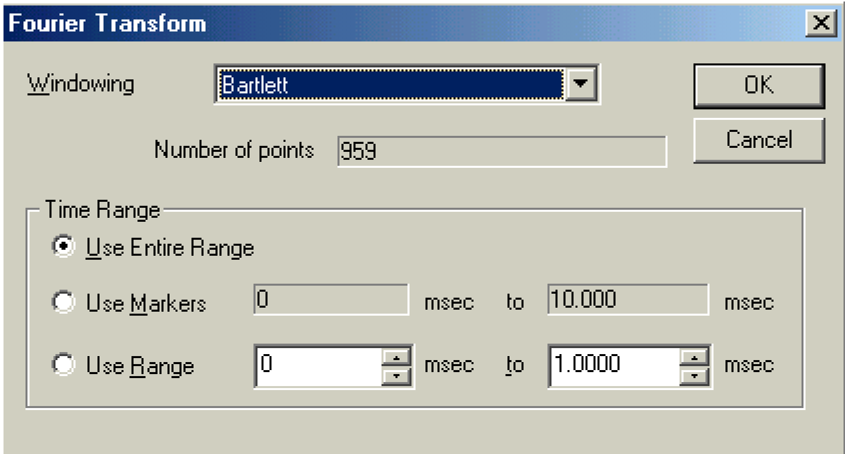
Window	-3 dB Main Lobe Width (bins)	-6 dB Main Lobe Width (bins)	Maximum Side Lobe Level (dB)	Side Lobe Roll-Off Rate (dB/decade)
Uniform (None)	0.88	1.21	-13	20
Hanning (Hann)	1.44	2.00	-32	60
Hamming	1.30	1.81	-43	20
Blackman-Harris	1.62	2.27	-71	20
Exact Blackman	1.61	2.25	-67	20
Blackman	1.64	2.30	-58	60
Flat Top	2.94	3.56	-44	20

The power of a given frequency peak is computed by adding the adjacent frequency bins around a peak and is inflated by the bandwidth of the window. You must take this inflation into account when you perform computations based on the spectrum. The worst case amplitude errors are demonstrated in the table below (from The Fundamentals of FFT-Based Signal Analysis and Measurement in LabVIEW and LabWindows/CVI at http://zone.ni.com/).

Window	Scaling Factor (Coherent Gain)	Noise Power Bandwidth	Worst-Case Amplitude Error (dB)
Uniform (none)	1.00	1.00	3.92
Hann	0.50	1.50	1.42
Hamming	0.54	1.36	1.75
Blackman-Harris	0.42	1.71	1.13
Exact Blackman	0.43	1.69	1.15
Blackman	0.42	1.73	1.10
Flat Top	0.22	3.77	< 0.01

Note: SW defaults to a usable window system. If you want to control this yourself, go to Menu/Calculate/FFT and then choose your window and your desired parameters.





Rectangular (Uniform) window
Rectangular window is the simplest. A Uniform (or Rectangular) window has the narrowest lobe. A Uniform window is actually no window but by the very nature of taking a slice in time picture of the input signal, the signal is convolved using a Sine Wave function so there is still a windowing effect.

Blackman, Hamming, Hann, and rectangular windows are all special cases of the generalized cosine window. The concept behind these windows is that by summing the individual terms to form the window, the low frequency peaks in the frequency domain combine in such a way as to decrease side lobe height. This has the side effect of increasing the main lobe width. The Hamming and Hann windows are two-term generalized cosine windows. The Blackman window is a popular three-term window while the Hamming, Hann, and Blackman may be the most commonly used for Impulse testing with loudspeakers.

A Flat Top Window widest main lobe and therefore has excellent amplitude accuracy however it has a poor frequency resolution and more spectral leakage. It has a lower maximum side lobe level than the Hann Window but a slower roll off rate.

The **Kaiser window** is an approximation to the prolate-spheroidal window, for which the ratio of the main lobe energy to the side lobe energy is maximized.

Chebyshev Window (also Dolph-Chebyshev) minimizes the main lobe width, given a particular side lobe height. It is characterized by an equi-ripple behavior, that is, its side lobes all have the same height.

Strategies for Choosing Windows:

To choose a window, you must guess the approximate signal frequency content. Distant frequency interference from the central frequency would warrant a high side lobe roll off rate. Strong interfering signals nearby the frequency of interest would warrant a window with a low maximum side lobe level. Spectral resolution becomes important if there are two or more signals very near each other; this would call for a window with a very narrow main lobe (higher resolution). If the amplitude component of the frequency or interest is more important than exact frequency location, a wide main lobe will better identify this (better amplitude accuracy). If a signal spectrum is fairly flat or broadband in frequency content, a Rectangular (Uniform) window is best.

It is best to use a Uniform (Rectangular) window when analyzing frequency response. If you generate a Swept Sine Wave (Chirp), you are generating a signal that is sinusoid swept from a start frequency to a stop frequency. If you match the acquisition frame size to the length of the Chirp, accuracy will be maximized.

Below is a list of recommendations from The Fundamentals of FFT-Based Signal Analysis and Measurement in LabVIEW and LabWindows/CVI at http://zone.ni.com/:

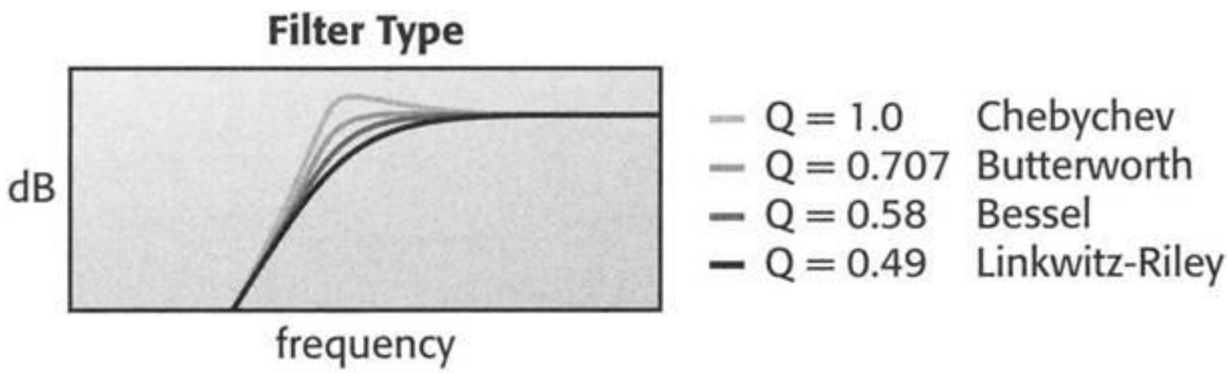
Signal Content	Window
Sine wave or combination of sine waves	Hann
Sine wave (amplitude accuracy is important)	Flat Top
Narrowband random signal (vibration data)	Hann
Broadband random (white noise)	Uniform
Closely spaced sine waves	Uniform, Hamming
Excitation signals (Hammer blow)	Force
Response signals	Exponential
Unknown content	Hann

FILTER & CROSSOVER TYPES

(Section Index)

From http://www.audioholics.com/techtips/audioprinciples/Loudspeakers-Crossovers.html

The filter type can be described in several different ways. Low-pass and high-pass filters in two-way crossover networks are often identified by their "Q". The Q is the resonance magnification of the filter and it is recognized by the shape of the "knee" of the amplitude response. Filters with a high Q tend to "ring" and exhibit poor transient response. Unlike drivers and boxes, which use only numerical values for Q, filters are sometimes named after the engineer(s) who first described them. Some examples are shown in the amplitude response graph below.



Filter Types

The filters in three-way crossover networks (and some two-way networks) are often identified as either "APC" or "CPC" depending on the way they combine. APC stands for "All-Pass Crossover" and it refers to those crossover networks whose filters sum to create a flat voltage output. APC networks are generally considered the best choice because they make it possible for the speaker to have a flat on-axis amplitude response. Common APC networks include 1st- and 3rd-order Butterworth filters and 2nd- and 4th-order Linkwitz-Riley filters. CPC stands for "Constant-Power Crossover" and it refers to those crossovers whose filters sum to provide a flat power response. The power response of a speaker is the total of both its off-axis and on-axis amplitude response. In other words, it is the total acoustical power that is radiated into a space. CPC networks can be beneficial in reverberant environments where the off-axis response is important.

The difference between APC and CPC networks can be understood electrically by a comparison of their input to output voltages. APC networks satisfy the following expression:

[VI] = [VL + VM + VH]

This means the absolute value of the input voltage will equal the absolute value of the sum of the output voltages of each filter at all frequencies. CPC networks satisfy the following:

VI^2 = VL^2 + VM^2 + VH^2

This means that the square of the input voltage will equal the sum of the squares of the output voltages of each filter at all frequencies.

Filter Summary

These generalizations assume that the drivers are properly aligned at the crossover frequency. This means that they are mounted in such a way that the direct sound from each driver arrives at the listener's ear at the same time at the crossover frequency. Another important assumption is that the impedance response of each driver has been equalized so that it appears to be approximately resistive to the crossover network. Also, the sensitivity of the drivers is assumed to have been equalized with an appropriate L-pad.

Finally, the following descriptions assume that all filters in the crossover network are of the same type. If a two-way crossover network has a 4th-order Linkwitz-Riley low-pass filter, it is assumed that it also has a 4th-order Linkwitz-Riley high-pass filter. If you choose to use mismatched filters, you'll have to rely on the your own measurements and experience to determine the results.

1st-order Filters

Advantages: Can produce minimum phase response (Butterworth only) and a maximally flat amplitude response. Requires the fewest components.

Disadvantages: Its 6 dB/octave slope is often too shallow to prevent modulation distortion, especially at a tweeter's resonance frequency. Achieving minimum phase and a maximally flat amplitude response requires very careful driver alignment and only occurs when the listener is located at exactly the same distance from each driver. It has a 90-degree phase shift which can result in lobing and tilting of the coverage pattern.

Two-Way
1st-order Butterworth: Produces a -3 dB crossover point to achieve a maximally flat amplitude response, minimum phase response and flat power response that qualifies it as both an APC and CPC network. The 90 degree phase shift results in a -15degree tilt in the vertical coverage pattern if the tweeter and woofer are vertically separated by no more than one wavelength at the crossover frequency and if the acoustical depth of the tweeter and woofer are carefully aligned at the crossover frequency. The tilt will increase and lobing can become severe if the drivers are separated by a greater distance or are misaligned. These problems appear as a ripple in the amplitude response. Filter Q = 0.707.

Two-Way & Three-Way
1st-order Solen Split -6 dB: A custom version of the 1st-order Butterworth filter (two-way crossovers) or 1st-order APC filter (three-way crossovers) that uses a -6 dB crossover point to minimize the disadvantages of a crossover network with standard 1st-order Butterworth or APC filters.

Three-Way

Note. 1st-order filters are usually not recommended for three-way crossover networks because their shallow 6 dB/octave slopes do not provide adequate separation. 1st-order APC: Produces -3 dB crossover points to achieve a flat amplitude response.
1st-order CPC: (Seldom used.) Produces -3 dB crossover points to achieve a flat power response.

2nd-order Filters

Advantages: Can produce a maximally flat amplitude response. Requires relatively few components. Has a 180-degree phase shift which can often be accommodated by reversing the polarity of the tweeter and which produces minimal or no lobing or tilt in the coverage pattern. Is less sensitive to driver misalignment than 1st-order filters.

Disadvantages: Although the 12 dB/octave slope is better than a 1st-order filter, it may still be too shallow to minimize the modulation distortion of many drivers.

Two-Way
2nd-order Bessel: Produces a -5 dB crossover point to achieve a nearly flat (+1 dB) amplitude response. The summed group delay is flat. It has a low sensitivity to driver misalignment and resonance peaks. Filter Q = 0.58.

2nd-order Butterworth: Produces a -3 dB crossover point that sums to a +3 dB amplitude response and a flat power response that qualifies it as a CPC network. It has a medium sensitivity to driver misalignment and resonance peaks. Filter Q = 0.707.

2nd-order Chebyshev: (Seldom used.) Produces a 0 dB crossover point to achieve a +6 dB amplitude response with about ±2 dB of ripple. The summed group delay has a significant peak just below the crossover frequency. It has a medium sensitivity to driver misalignment and resonance peaks. Filter Q = 1 .0.

2nd-order Linkwitz-Riley: (Very popular.) Produces a -6 dB crossover point to achieve a maximally flat amplitude response that qualifies it as an APC network. It has a -3 dB dip in the power response. The summed group delay is flat. It has a medium sensitivity to driver misalignment and resonance peaks. Filter Q = 0.49.

Three-Way
2nd-order APC: Produces -6 dB crossover points to achieve a flat amplitude response but the power response will have approximately 3 dB of ripple.

2nd-order CPC: (Seldom used.) Produces -3 dB crossover points to achieve a flat power response but the amplitude response will have approximately 3 dB of ripple.

3rd-order Filters

Advantages: Can produce nearly flat amplitude response. With an 18 dB/octave slope, it is better able to minimize modulation distortion. Less sensitive to driver misalignment.

Disadvantages: Requires more components. Has a 270-degree phase shift which can result in lobing and tilting of the coverage pattern.

Two-Way
3rd-order Butterworth: (Popular for some D'Appolito mid-tweeter-mid designs.) Produces a -3 dB crossover point to achieve a maximally flat amplitude response and flat power response that qualifies it as both an APC and CPC network. A 270 degree phase shift results in a + 15 degree tilt in the vertical coverage pattern if the tweeter is wired with normal polarity and a -15 degree tilt if the tweeter is wired with reverse polarity. (D'Appolito mid-tweeter-mid designs overcome much of this tilt problem and produce a more symmetrical coverage pattern.) It has better group delay than a 1st- and 2nd-order Butterworth network. Filter Q = 0.707.

Three-Way
3rd-order APC: Produces -3 dB crossover points to achieve a flat amplitude response but the power response will have a modest ripple (usually less then 1 dB) that increases slowly as the spread between the two crossover frequencies increases.
3rd-order CPC: (Seldom used.) Produces -3 dB crossover points to achieve a flat power response but the amplitude response will have a varying amount of ripple (typically 1 to 3 dB) depending on the spread between the two crossover frequencies.

4th-order Filters

Advantages: Can produce a maximally flat amplitude response. With a 24 dB/octave slope it provides the best isolation between drivers resulting in the least modulation distortion. Has a 360 degree phase shift which results in "in-phase" response and which promotes minimal or no lobing or tilt in the coverage pattern. Is the least sensitive to driver misalignment.

Disadvantages: Requires the most components. The increased number of inductors can result in substantial insertion loss because of inductor DCR.

Two-Way
4th-order Bessel: Produces a -7 ½ dB crossover point to achieve a nearly flat (-1 ½ dB) amplitude response. The summed group delay produces a moderate bump just below the crossover frequency. Filter Q = 0.58.

4th-order Butterworth: Produces a -3 dB crossover point that sums to a +3 dB amplitude response and flat power response that qualifies it as a CPC network. The summed group delay has a significant peak just below the crossover frequency. Filter Q = 0.707.
4th-order Gaussian: (A seldom used filter that is constructed with an asymmetrical filter topology.) Produces a -6 dB crossover point to achieve a nearly flat amplitude response with moderate ripple. The summed group delay produces a moderate bump just below the crossover frequency.

4th-order Legendre: (A seldom used filter that is constructed with an asymmetrical filter topology.) Produces a -1 dB crossover point that sums to a +5 dB amplitude response with minor ripple. The summed group delay has a significant peak just below the crossover frequency.

4th-order Linear-Phase: (A seldom used filter that is constructed with an asymmetrical filter topology.) Produces a -6 dB crossover point to achieve a nearly flat amplitude response with moderate ripple. The summed group delay produces a moderate bump just below the crossover frequency.

4th-order Linkwitz-Riley: (Very popular. Sometimes called a "squared Butterworth" filter. Also used for some D'Appolito mid-tweeter-mid designs.) Produces a -6 dB crossover point to achieve a maximally flat amplitude response that qualifies it as an APC network. It has a -3 dB dip in the power response. The summed group delay produces a moderate bump just below the crossover frequency. Filter Q = 0.49.

Three-Way

4th-order APC: Produces -6 dB crossover points to achieve a flat amplitude response but the power response will have approximately 3 dB of ripple.

4th-order CPC: (Seldom used.) Produces -3 dB crossover points to achieve a flat power response but the amplitude response will have approximately 3 dB of ripple.

Source : Xover Pro Harris Technologies

Acknowledgements: I would like to thank Shane Rich (Technical Director of RBH Sound, Inc) for helping with the compilation of this information to serve as a tool in forthcoming technical articles and reviews of loudspeakers.

ESTABLISHING A CROSSOVER POINT

(Section Index)

In determining a crossover point, you would want to first compensate for any impedance peak, and then look at the excursion demands of the maximum desired SpL at the crossover point and at all points surrounding this with the filter in place. This can be modeled (see my spreadsheet). You must also recognize the drivers thermal limitations and at what point you start to realize unacceptable distortion (again at the maximum desired listening levels). For off axis response, most target 3dB down at 30 degrees off axis though it would vary based upon room reflectivity. Certainly a smooth off axis response is desirable. A cardiode response may have an ideal response in general. One must be cautious in realizing that manufacturer specifications of Xmax may refer to either linear Xmax or Xmax that does not take into account the linear range of the excursion. Xmax is a good indication of the point at which distortion is probably rising quickly. The problem is that Xmax is hardly ever done in the way DUMAX defines Xmax. Some basic rules of thumb from Dave Dal Farra (DDF) are:

1. every 6 dB more o/p doubles excursion at a given frequency
2. above resonance, doubling frequency at a given voltage halves excursion (approx, depends upon driver Q)
3. below resonance, equal voltage in = equal excursion (approx, Q depending).

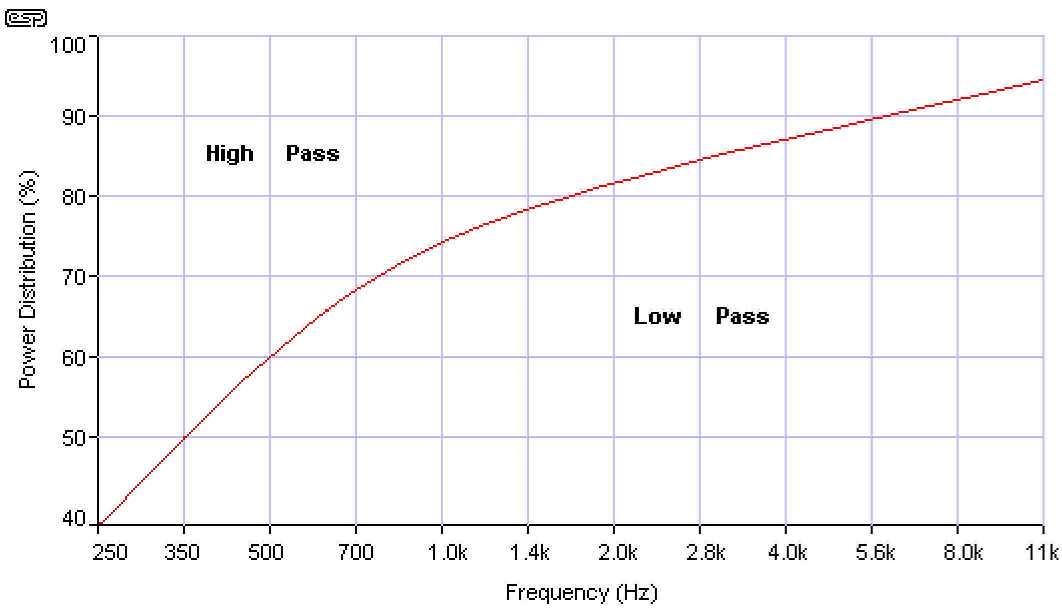
A way of approaching things more methodically would be (per DDF) as follows:

Determine max acceptable distortion you'll tolerate from the tweeter. It depends upon the harmonic signature. I'd recommend two targets: x% 3rd order and higher, and total%. Find max V that still satisfies both constraints.

1. Measure and determine the tweeter input voltage vs. frequency, with sines, that lead to this distortion. Excursion is the main consideration for this test. Don't let the sines run too long, or thermal issues may dominate (low crest factor of sines).
2. Knowing your amp's o/p capability, sim your xovers with the amp at its max voltage o/p and look at the voltage at the tweeter. If you exceed the voltage determined by the tweeter characterization, at any frequency, the tweeter may strain on transients.
3. Once you have an xover candidate worked out, calculate the power dissipated by the tweeter:
   a. assume a "worst case realistic" (for the tweeter) music spectral density (long term). These curves are available on the web.
   b. Based on this curve, calculate the amp o/p. Assume approx 15 dB peak/rms ratio for the music: ie total amp power o/p is 15 dBV (not dBm) below its transient capability. This is for well recorded classics and jazz. For pop, assume about 6 to 8 dB (approx).
4. Knowing the xover function, and the tweeter impedance, you can calculate the power dissipated by the tweeter. If it exceeds manufacturer's recommendation, back to step 3.

This may be more work than most DIY will want to do so looking at Xmax and Voltage inputs seem to be a reasonable compromise.

From http://sound.westhost.com/lr-passive.htm
The power loss is naturally proportional to the input power, and for our example, I shall assume a maximum amplifier power of 100W. Use the chart below to determine how much power will go to the tweeter, using a crossover frequency of 3.0kHz.



Power Distribution Chart

Working along the frequency axis, we see that at 3kHz, the power in the low pass section will be about 85% of the maximum (85W), so the high pass power level is about 15%, or 15W for our 100W system.

Other issues to consider: the peak of the baffle step will make the tweeter work less for a given acoustic crossover curve until that point while it will have to work harder below that step to make up for it. Generally, the baffle step occurs at a point where there is significant tweeter attenuation but this becomes more of a factor with lower order filters or lower frequency crossovers. One additional consideration is amplifier clipping. If volume demands are high compared to amplifier capabilities, it will lead to amplifier clipping; this introduces high frequency (non linear) distortion and additional demands upon the tweeter. To shape distortion test signals for drivers, one would best be served by using peak values in music content to test for distortion due to excursion and average values to test for long term power handling issues such as thermal limitations and creep. Suspension creep factor is defined as a continued slow displacement under sustained force in the dynamic behaviour of the driver. The actual curve of the test signal envelope depends upon the type of music listened to in order to establish a spectral density in the test signal.

Some might recommend the use of a series conjugate filter on the tweeter to lower the impedance and thereby lower the fc of the tweeter which will decrease distortion and increase power handling capability at those frequencies affected.

BAFFLE DIFFRACTION: A REAL BEGINNER'S INTRODUCTION

Paul Verdone from FRD Consortium

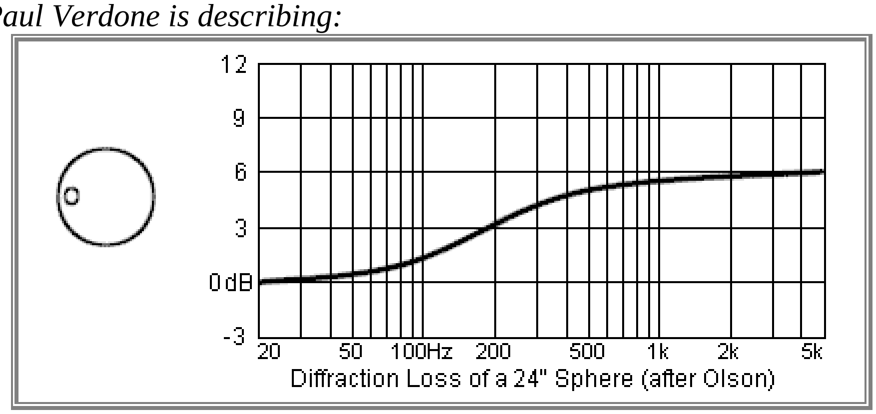
(Section Index)

A baffle is a reflector. It divides the space in front of it from the space behind it. The larger its size, the longer it takes to get around it. In acoustics, longer distances, means longer time delays, which means lower frequencies. So, the larger the baffle, the lower the lowest frequency that it reflects (why this is so should become clearer a little further down). If the baffle were infinitely long, so would be the time delay traveling across it, and the lowest frequency reflected by the baffle would be near 0 Hertz. Smaller baffles only reflect higher frequencies. Larger baffles do that as well. As the baffle size is enlarged, so the cut-off frequency, where the baffle stops reflecting effectively, is reduced. This cut-off frequency is a key element of loudspeaker design and is known as the Baffle Step Frequency. (My note: Baffle Step is most evident on axis and decreases in effect off axis. Room reverberations tend to decrease the impact of the baffle step somewhat)

Over the frequency range that a baffle is reflecting, all of the sound is kept in front of the baffle, so the space that is filled by the sound is only half the "total space". The "total space" is usually described by an acoustic geometry term known as 4PI Space. The Area in front of the baffle is also usually described with the term 2PI space. This is a very straightforward once you get used to it. As long as the sound is only kept in 2PI Space by the baffle, its concentration or field strength is twice as great as when the sound is permitted to (go around the baffle and) fill the total 4PI space. This difference in strength or gain is another key element of loudspeaker design and is known as the Baffle Step Gain.

The Baffle Step Gain is not a fixed amount, but depends on the surrounding environment (your room), how far your speakers and baffle are away from the room boundaries, and how far you are away from the speakers and baffle. But in general, the Baffle Step Gain always tends to approach + 6db. The Baffle Step Frequency is not a fixed frequency either, but is really a frequency region or transition zone over which the change in gain happens, and can span several octaves.

*Note: From John Murphy in his article **Baffle Step and Baffle Diffraction** at* (http://www.trueaudio.com/st_diff1.htm) *the following is a graphic representation of what Paul Verdone is describing:*



What is interesting is that the effective size of the baffle and the frequencies that it reflects, from the perspective of the speaker, depends not on how large the baffle is, but rather how far away the edge of the baffle is from the speaker. Why? Because the frequencies reflected by the baffle are based on the distance that the sound must travel before bending around the edge of the baffle into 4PI space: In acoustics, longer distances, means longer time delays, which means lower frequencies. So it is not the absolute size of the baffle that effects reflection and Baffle Step Frequency, but the distance the speaker is from the edge of the baffle.

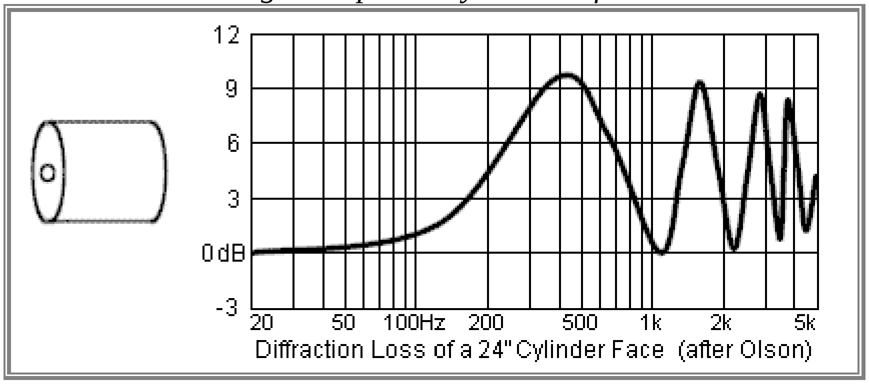
Well, if the distance to the baffle edge is the determining factor to the reflected frequencies, then the placement of the speaker on the baffle is a major part of establishing that distance. If the speaker is closer to an edge, then more of the lower frequencies will wrap around the edge and not get reflected forward. If further away, then the distances traveled along the baffle are longer, so the time delays are longer so more of the lower frequencies are reflected.

The other thing to note is that, not only the placement of the speaker matters, but the shape of the baffle as well. If the shape of the baffle is circular and the speaker is centered then all the distances to the baffle edge are the same. If the baffle shape were square, and the speaker still centered, that there would be a series of different distances from the center of the speaker to the edge of the baffle. If the baffle were triangular shaped this would also be true, and the ratio of the shortest distance to the longest distance from the center of the speaker to the edge of the baffle edge would be greater. In other words, the variance in these distances is greater.

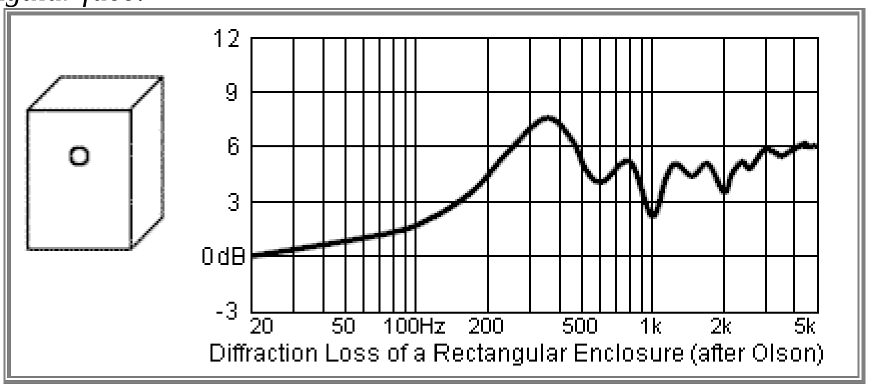
This variance of distance to a baffle edge is another key element of diffraction. But why? Well, diffraction as mentioned above, is about the distance that the baffle edge is away, and thus the time delay that the sound of the speaker experiences in its route along the baffle edge, trying to get around the baffle into 4 PI space. The reason that the delay is important is based not on what is happening along the path across the baffle, but on what happens when the sound reaches the edge. More specifically, as the sound wave tries to turn around the edge, the wave now finally sees a larger 4 PI space. Since the space is larger than the 2PI space the sound wave was restricted to before reaching the edge, then sound wave has to expand. This expansion is a change of pressure of the sound as it makes the turn around the baffle edge. This change in pressure is an acoustic event and creates a reflection sent off in all directions visible from the edge where the change in direction around the baffle corner happens. The magnitude of the reflection is related to the pressure change, which is determined by the angle turned around the edge, or the angular space seen at the boundary.

This reflection is the basis of baffle diffraction. This reflection arrives at the listening position and affects the sound that got to the listening position directly (not by way of the baffle). The two wave mix and affect each other. Since the baffle edge reflection took a longer path to get to the listening position, there is a time delay involved. In acoustics, time delays create reinforcement or cancellation depending on the frequency involved. This change in gain versus loss cycles up and down with rising frequency because the time delay has a different phase angle at each frequency. This cycling of the response up and down is sometimes referred to as a comb filtering response, because the periodicity of the peaks and nulls in the response are equally spaced and resemble a comb.

Note: Again from John Murphy in his article Baffle Step and Baffle Diffraction at (http://www.trueaudio.com/st_diff1.htm), the following is a graphic representation of what Paul Verdone is describing: compare a cylindrical face



To a rectangular face:



So for each path that the sound wave takes on its travels across the baffle, hitting the edge and wrapping around it, there is also a path that goes the other directions and arrives at the listening position. How far that distance across the baffle to the edge and back to the listener is the key delay that establishes the peaking and the nulling effect. Frequencies lower than this delay are unaffected. Frequencies above that delay warble up and down averaging twice as strong because of the reflection's contribution. This reflection's contribution is the energy or strength difference between 2PI and 4PI space and accounts for the Baffle Step Gain or what the baffle cumulatively reflects. The corner frequency is established by the time delay and is the basis of the Baffle Step Frequency.

But each path from the speaker to the edge to the listener may be different. If the speaker is not centered on the baffle, then some reflections will arrive at the listening position early and some even later. If the baffle shape is non-circular, then the paths taken are again different. If the listening position is further away then the path delays may be a little larger or smaller (triangulation with the direct wave). If the listening position is off to the side, or above then the paths from the baffle edge are again different.

Note: John Murphy reports that the following are estimated equations for calculating the midpoint (or –3dB) for the baffle loss at any given point on the baffle edge

f(3) = 115/W(B)

(where W(B) is the baffle width (side to side) in **meters**)

or

f(3) = 380/W(B)

(where W(B) is the baffle width in **feet**).

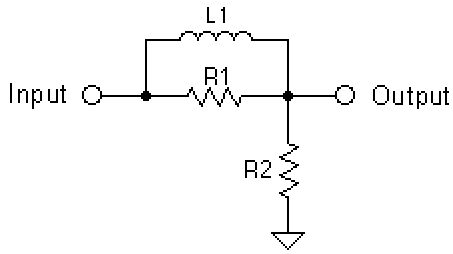
The edge size and edge shape also affects these delays. Sharp edges provide a precise point where these reflections begin. Beveled edges provide more than one point, displaced in space a thus time. Rounded edges provide many points along the curvature. An extremely large curved surface, like a curved or spherical baffle represents one very large continuous edge or a near infinite set of reflective points. The size and shape of the speaker itself also affects these delays. Smaller drivers tend to start the journey of the sound waves aimed at the baffle from a near single central point. Larger diameter speakers actually launch many such waves each from a different point on the surface of the driver. So each of these points on the driver contribute a part that may get to that edge a little earlier or a little later.

It is the sum of millions of these individual paths that come together that collectively change the direct wave and create the sound of the baffle. If these delays are all very similar, then the reinforcements and cancellations will be as well and the response will look rather bumpy. If there are hundreds of minute differences in the delays then the overall effects in phase will tend to cancel and the response will look flatter. The transition zone, know as the Baffle Step Region will also be effected by these variances. If the variances are very small then there will be a rather large bump followed by a smaller dip, before the responses starts to even out a bit.

In addition to the time delay with each of the diffracted wave components, there are also magnitude differences. Longer paths mean weaker signals and less contribution. The size of the area of the driver radiating and the region used to describe it have an associated strength by virtue of its area. Speakers also have directivity, which means they tend to become more directional with rising frequency, so the magnitude of the energy fired sideways at the baffle edge will tend to be smaller with rising frequency. Even the edge shape effects magnitude. All the reflections from a sharp edge happen or emanate from that single edge. A Beveled or Chamfered edge has two such event locations with different lesser strengths because the angle involved in each part of the turn is different. On a rounded edge the center on the activity is somewhere in the middle of the rounding, but has no precise spot. The individual contributions from the each piece of the rounded edge are very small, but collectively they total the same reflection as if the reflection were sharp. In this case, both the time delay and the magnitudes of each of these sub-reflections are different.

So Diffraction Simulation is about tracking all of these of these delays and magnitudes and summing them at the listening position to predict the effective frequency response contributed by the baffle. The baffle shape, driver size, driver position and edge treatment each effects this contribution. This contribution is the called the Baffle Signature. It is not a separate response. It is an affecter or change in response by the baffle, much like the change in the response from the crossover, the box model or the room itself. The Baffle signature can be viewed as something to optimize. If can also be viewed as something that requires compensation elsewhere. The Signature may be subtracted or removed from a measured response of a speaker in a test baffle to correct the measurements to reflect what the speaker really sounds like by itself. The signature can also be added to predict how that speaker would sound on a particular target baffle.
-- Paul Verdone

Addendum: John Murphy has calculated that the following circuit will act as baffle step compensation:



where R2 represents the loudspeaker nominal load impedance and R1 = R2
The inductor **L1 =W_BR/1.021**
Where W(B) is the baffle width in meters
R is in Ohms
L1 in millihenries.

OR

L1=3.213*W_BR
Where W(B) is the baffle width in meters
R is in Ohms
L1 in millihenries.

SOUND CARDS

(Section Index)

SOUND_CARD_DATA
BASE.xls

SB Live!
Freq Response: 10-44kHz
S/N: 96dB
Noise floor –120dB
Sampling rate for playback and record: 8kHz-48kHz
Microphone impedance 600 Ohms
Line In Impedance 47 kOhms
CD Audio In Impedance 50 kOhms
Mic Sensitivity 10-200 mVpp
Line In Sensitivity 0-2 Vpp
CD Audio In Sensitivity 0-2Vpp
AD/DA Resolution 16 bits

<u>MICROPHONE CALIBRATION FILES</u>

(Section Index)

1. **To import an a calibration file not in .FRD format, open the file in WordPad, Rename the file with a .frd extension, then use the SW import command.**

ecm8000cal.zip

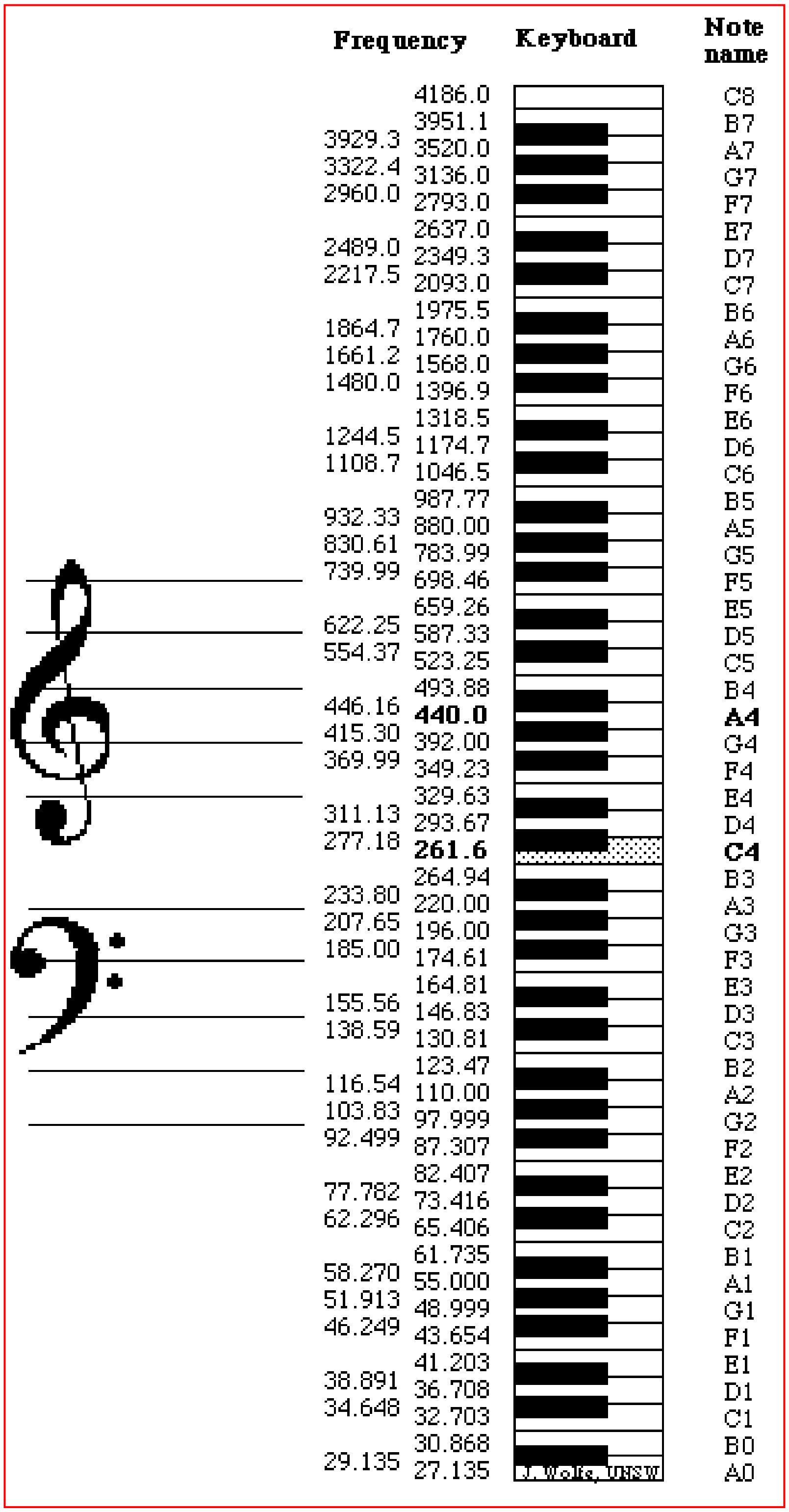
WM61A.zip

"WM60a mic-Calib-File.FRD"

THE FREQUENCY OF NOTES

from: http://www.phys.unsw.edu.au/~jw/notes.html

(Section Index)



FREQUENCY OF INSTRUMENTS

From: http://www.listenhear.co.uk/general_acoustics.htm

(Section Index)

WOODWIND	Top
Piccolo	630Hz - 5K
Flute	250Hz - 2.5K
Oboe	250Hz - 1.5K
Clarinet (B flat or A)	125Hz - 2K
Clarinet (E flat)	200Hz - 2K
Bass Clarinet	75Hz - 800Hz
Basset Horn	90Hz - 1K
Cor Anglais	160Hz - 1K
Bassoon	55Hz - 575Hz
Double Bassoon	25Hz - 200Hz
BRASS	Top
Soprano Saxophone	225Hz - 1K
Alto Saxophone	125Hz - 900Hz
Tenor Saxophone	110Hz - 630Hz
Baritone Saxophone	70Hz - 450Hz
Bass Saxophone	55Hz - 315Hz
Trumpet (C)	170Hz - 1K
Trumpet (F)	300Hz - 1K
Alto Trombone	110Hz - 630Hz
Tenor Trombone	80Hz - 600Hz
Bass Trombone	63Hz - 400Hz
Tuba	45Hz - 375Hz
Valve Horn	63Hz - 700Hz
STRINGS	Top
Violin	200Hz - 3.5K
Viola	125Hz -1K
Cello	63Hz - 630Hz
Double Bass	40Hz - 200Hz
Guitar	80Hz - 630Hz
KEYBOARDS	Top
Piano	28Hz - 4.1K
Organ	20Hz - 7K
PERCUSSION	Top
Celeste	260Hz - 3.5K
Timpani	90Hz - 180Hz
Glockenspiel	63Hz - 180Hz
Xylophone	700Hz - 3.5K

From Geoff Husband - http://www.tnt-audio.com Copyright © 1999

Type of voice	Frequency range Hz
Bass	87.31 - 349.23
Baritone	98.00 - 392.00
Tenor	130 - 493.88
Contralto	130.81 - 698.46
Soprano	246.94 - 1,174.70

SOUND PRESSURE POWER OUTPUT MEASUREMENTS

(Section Index)

N/m^2=(10^(((dB/20)+(20*LOG(2*10^-7))/20)))*100

Sound Pressure Level (dB)	Sound Pressure (N/m^2)	Environment
140	200	Military Jet Taking Off
130	63.245553	Pneumatic Drill at 2 feet
120	20	Engine Room in a Ship, Small Aircraft Taking Off, This level hurts your ears!
110	6.3245553	Factory, Shouting in the Ear, This level is usually quite uncomfortable.
100	2	As loud as acoustical instruments (except drums and trombones) can play. Extremely Loud.
90	0.6324555	Food Mixer, Tractor, Passing Underground Railroad, Fairly Loud.
80	0.2	Busy Road, A typical loud listening level that doesn't make lthe neighbors complain.
70	.06324555	Sound level of easy listening music - a little loud for talking over in intimate circumstances; sound level in a quiet car at 60 mph with the windows rolled up.
60	0.02	Loud Conversation, Restaurant
50	0.0063246	Conversational Speech, Rainfall
40	0.002	Quiet Residential Area, Office, or Library
30	0.0006325	Soft Whisper or a quiet conversation
20	0.0002	Whispering at five feet
10	6.325E-05	Normal Breathing, Rustle of leaves in the breeze
0	0.00002	Normal Threshold of Hearing

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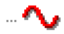
Instrument	Range measured in dB
Bass drum	35 - 115
Cymbal	40 - 110
Organ (orchestral)	35 - 110
Piano	60 - 100
Trumpet	55 - 95
Violin	42 - 95

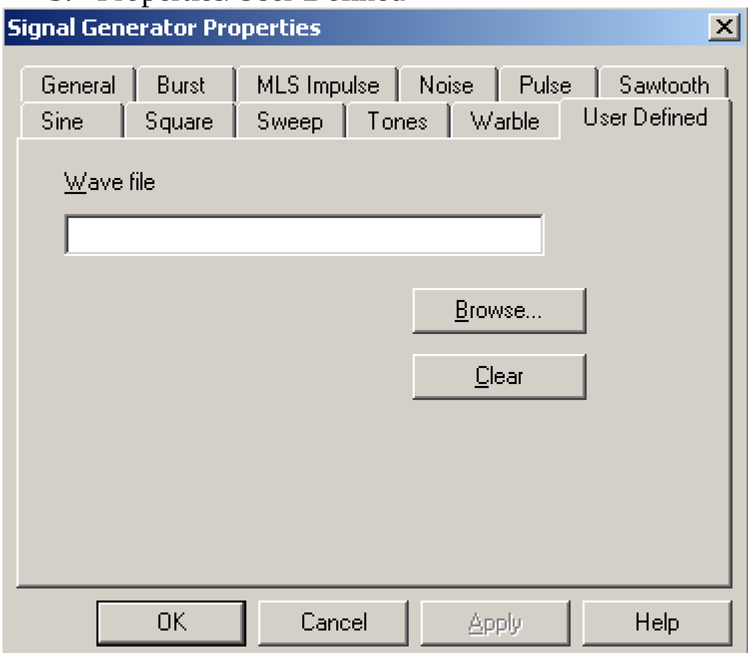
MAKING YOUR OWN TEST TONE CD

(Section Index)

Speaker Workshop allows you to import your own test tones to do testing.

To import your own test tones:

1. Click on  on your tree
2. Right Click on the Chart
3. Properties/User Defined



4. You can name and import any signal that you have saved.
5. Signals may be produced with any programs that can save signals in a .wav format.

The following are tone generators that are shareware/freeware programs that allow you to generate your own .Wav files so that you can make your own test CD. The first is a shareware program, Test Tone Generator copyrighted by NCH Swift Sound.

Next is a shareware program DaqGen 1.4 (http://www.daqarta.com/DGINTRO.HTM) copyrighted by Interstellar Research whose current version does not allow for recording but the next edition (to be released in early 2006) will allow recording of signals. This program is somewhat more extensive than NCH:

Using the following spreadsheet, you can calculate octave measurements to measure various octave fractionals for testing which will help in developing your own test tone CD (1/3, 1/6, etc.):

OTHER SELECTED SPREADSHEETS/PROGRAMS

(SECTION INDEX)

Two Conversion tools are available in the Appendix for converting different units to one another. The first is written and copyrighted by Joshua F. Madison:

Convert.exe

There is an updated version that installs on the computer as well (no more accurate, just more conversions are available at http://www.joshmadison.com/software/convert/

Another conversion application was written by Isaac MCN and is used with his permission:

Fct.exe .

DECIBEL CONVERSION SPREADSHEET

db_calc(1).xls

WORKSHEETS BY DICK VAN NIEROP TO INCLUDE: LPAD, AVERAGE INDUCTANCE AND CAPACITANCE, ZOBEL AND NOTCH FILTERS, DETERMINATION OF OFFSET, OFFSET CALCULATIONS 1&2, COMB FILTERING

"Nierop calc parametes & Q.xls"

FOR SOUND ATTENUATION CALCULATIONS

the following is from Chris Whealy at http://www.whealy.com/acoustics/Porous.html :

"Porous Absorber Calculator V1.5 XL97.

FOR CALCULATING IN ROOM EFFECTS:

Thorsten Ezee Loesch has written the following spreadsheets:

Xlbaffle.xls

CALCULATING SD FROM SURROUND OUTSIDE EDGE TO CONE OUTSIDE EDGE

"Sd and Vd estimates1.xls"

WAVELENGTH VS. FREQUENCY AT VARIOUS TEMPERATURES AND HUMIDITIES WITH MINIMUM GATE CALCULATIONS

"Frequency vs wavelength1.xls"

ROOM MODE CALCULATOR

"Room Mode Calculator1.xls"

Floyd Toole, Vice President Acoustical Engineering,
Harman International Industries, 8500 Balboa Blvd.
Northridge, CA 91329. 818 893 8411 ftoole@harman.com

courtesy of

LINKS AND PROGRAMS

(Section Index)

AUDIO DIY CENTRAL
Jim Holtz, Paul Verdone, Peter Smith (Pjay), Jim Salk, Dan Wesnor and Jeff Bagby act as Steering Committee for this DIY site.
http://www.audiodiycentral.com
Description: A DIY site with links, projects, white papers, tutorials and reviews. Even a neat little program called BoxyCAD (used in this manual).

AUDIO KARMA DISCUSSION FORUM
http://www.audiokarma.org/forums/
Description: an audio lovers forum.

AUDIOHOLICS
http://www.audioholics.com/techtips/
Description: another audio lovers forum.

CLAUDIO NEGRO'S HOME PAGE
http://www.claudionegro.com/
Description: a great SW tutorial from which much information in this manual was taken.

CROSSOVER SIMULATION IN SPEAKER WORKSHOP – TUTORIAL (JK ANDREASEN) http://www.angelfire.com/electronic/loudspeaker/
Description: another SW tutorial from which some information for this manual was taken.

DIY Audio.com
http://www.diyaudio.com/
Description: Lots of informative forums and good information.

DIY LOUDSPEAKER DESIGNER'S SELECTION GUIDE (THE LDSG)
http://ldsg.snippets.org
Description: a compendium of recommended drivers, components, and useful information

ELECTRICAL ENGINEERING CALCULATORS
http://www.ifigure.com/engineer/electric/electric.htm
Description: lots of electrical calculators and resources

ELLIOTT SOUND PRODUCTS
http://sound.westhost.com
Description: Lots of good projects and educational articles

ERIC WALLIN
http://home1.gte.net/tammie_eric/ericindex.html
Description: Here is a good tutorial SW using and gives the instructions on how to build the JIG and a mic preamplifier.

JAN'S SPEAKER PROJECT (DICK VAN NIEROP)
http://home.hccnet.nl/ine.dick/
Description: Project and measurement of a two way speaker using SW.

JOHN DUNLAVY DISCUSSIONS
http://home.austin.rr.com/tnulla/
Description: A highly respected Speaker Designer/Builder who strives for "accuracy" in speaker building

JOHN KRESKOVSKY'S HOME PAGE (HOME OF THE TRANSIENT PERFECT DESIGN)
Music and Design
Description: A lot of useful theory from a highly regarded source

LEO'S HOME PAGE
http://bellsouthpwp.net/1/j/1jfrank/A_Speaker_Workshop_Tutorial.html
Description: Page about speaker building and Projects including a SW tutorial on taking measurements and using the crossover portion of the program.

MADISOUND SPEAKER BUILDING FORUM
http://www.madisound.com/cgi-bin/discuss.cgi
Description: Vendor with reliable service and a great forum

MARK Krawiec HOME PAGE
Mark K's speaker pages
Especially interesting studies and discussion of Distortion

MATHEMATICS HELP CENTRAL
http://www.mathematicshelpcentral.com/graph_paper.htm
Description: A free Graph Paper generating program that can do regular, logarithmic, and other types.

PARTS EXPRESS SPEAKER BUILDING FORUM
http://www.pesupport.com/cgi-bin/config.pl
Description: Vendor with reliable service and a great forum

QUICKMIX
http://www.prodtechpart.co.uk/quickmix/index.htm
Description: Program that allows to save and recall the Windows mixer settings.

RANE CORPORATION
http://www.rane.com/library.html
Description: a commercial site with several good articles and spreadsheets

RIGHT MARK AUDIO ANALIZER
http://audio.rightmark.org/download.shtml
Description: Program that tests your sound card, useful in choosing the right volume in the calibration portion. Can also test distortion.

ROMAN BEDNAREK HOME PAGE
http://www.rjbaudio.com/

Home page of Roman Bednarek, which has several tutorials regarding Speaker Workshop and some interesting observations and studies. He also has some speaker projects.

SPEAKER WORKSHOP FORUM
http://www.speakerworkshop.com/forum/index.php
Description: The Ultimate forum for answers to your Speaker Workshop questions

SIEGFRIED LINKWITZ
http://www.linkwitzlab.com
Description: A page from one of the Masters in the industry. Lots of good information and a few spreadsheets.

STEREOPHILE MAGAZINE
http://www.stereophile.com/reference/
Description: Interesting reviews and several educational articles

THE EDGE SOFTWARE BY SVANTE GRANQVIST
The Edge program is a baffle diffraction simulator.
http://www.tolvan.com/edge/
Description: An excellent program that assists with circuit design for baffle compensation

THE FRD CONSORTIUM
http://www.pvconsultants.com/audio/frdgroup.htm
Description: Several useful freeware programs and papers that include some of the best freeware available for speaker building.

THE SOUNDS OF DISTORTION
By David Carlstrom
http://www.pcavtech.com/techtalk/dist_sound/index.htm
Description: You can actually hear samples of various types of distortion here

THE SUBWOOFER DIY PAGE
(Brian Steele)
http://www.diysubwoofers.org/ with a forum at:
http://www.diysubwoofers.org/talkshop/
Description: Formulas and theory regarding Subwoofers and Woofers

THE THIELE-SMALL LOUDSPEAKER DATABASE
http://www.thielesmall.com/
Description: The database contains all sorts of data on over 5128 drivers for those who want bypass T/S measurements.

TRUE AUDIO (INFORMATION FROM JOHN MURPHY)
http://www.trueaudio.com/index.htm
Description: Software, charts, and excellent information

UNIFIED BOX MODEL FOR LOUDSPEAKER DESIGN
http://home20.inet.tele.dk/kou/ubmodel.html
Description: Probably the best Excel spreadsheet program for calculation and simulation of closed, ported, passive radiator and bandpass systems.

FINAL COMMENTS

(Section Index)

Putting this together has been an educational and fun experience for me. I want to thank all of those who have contributed to my learning curve. This would include the many people on the bulletin boards of Madisound, Parts Express, DIY audio and, of course, all of those DIY'ers who have sites and information that they freely share. Most of all, I would like to thank Nelson Wood and Claudio Negro for the tremendous efforts that they put forth in reading and editing this project.

I have made every effort to assure that all information is accurate but be advised that the use of this information is at your own risk. I don't believe that it is much of a risk (though I am not fully certain of the use of Speaker Workshop with Distortion testing) that you take, but I stress again, I am a novice!

I hope that you enjoy reading this and get as much out of it as I did in putting it together. Please feel free to contact me through the Speaker Workshop Private Message System if you have any questions, concerns, comments, or corrections. I hope to make this a living manual with updates as indicated.

Enjoy!!

Jay