ACD-L The Active Crossover Designer

FOR USE WITH LADSPA PLUGINS AND ECASOUND

A SET OF VERSATILE EXCEL-BASED LOUDSPEAKER CROSSOVER DESIGN TOOLS

AUTHOR: CHARLIE LAUB
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A Brief Introduction

What are the ACD Tools?

The Active Crossover Designer (ACD) tools, comprise two types of spreadsheets for Microsoft Excel or OpenOffice Calc that allow the user to design high-performance active loudspeaker crossovers using software based DSP (IIR filters only). The spreadsheets are free, and open-source. All worksheets, formulas, and cells can be viewed. The ACD tools come ready for loudspeaker with up to 4 drivers. The functionality of the tools is augmented via “extensions” for the core tools.

A Short History of the ACD Tools

ACD originally grew out of my own need for an active loudspeaker crossover filter designer. A few years later the existing ACD tools were extended for modeling the loudspeaker on multiple axes, and then re-written for use with the Linux audio program ecasound with the filters implemented as LADSPA plugins.

The DIY community in which I participate is filled with people who are willing to share their knowledge with others, and I want to continue on with this theme. By keeping the tools “open source” I am hoping that others will learn the inner workings of the tools, make their own modifications and additions, and in turn share these with the community.

Capabilities of the ACD Tools

ACD gives the user the tools to carry out the steps that are needed to develop a high-performance loudspeaker crossover from measurements of the drivers installed in the loudspeaker cabinet. Sets of measurements are used to determine the relative acoustic offset (and phase relation) between of each driver. A wide variety of filters can be used to create the crossover. Several types of plots as a function of frequency (e.g. the magnitude and phase response) are provided. When the crossover design phase is completed, the designer can use data presented in the ACD-L tools to implement the crossover on a computer running a linux based operating system such as the Raspberry Pi. The design approach used in ACD results in excellent agreement between the model of the loudspeaker crossover (in these tools) and the frequency response of the completed loudspeaker with the crossover connected.

The Purpose of This Manual

This technical manual is not a tutorial and will not show you how to use ACD to develop a crossover for your loudspeaker. There is a separate tutorial for that, however, this manual provides a technical review of the ACD tools that you will find useful when you do actually begin to develop a crossover for your loudspeaker.

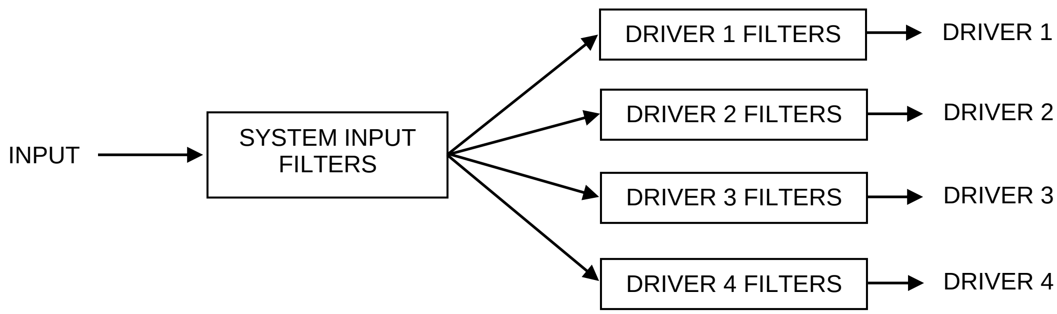
An Overview of the ACD Tools

The Active Crossover Designer (ACD) Tools create a model of the loudspeaker as seen at the listening or measurement location (e.g. by the microphone) in terms of amplitude (SPL) and phase. The model predicts the frequency response of the loudspeaker system, including the crossover, as a whole. Since the model is based on measurements of the drivers in the finished loudspeaker cabinet it is quite accurate as long as the data supplied to ACD is also accurate.

The ACD loudspeaker model is build using two types of spreadsheets: driver response spreadsheets and system response spreadsheets. Each of these types can be further subdivided into “on axis” and “off axis” flavors. The on-axis spreadsheets contain the tools for importing data, describing the crossover filters and other functions. The off-axis spreadsheets primarily allow off-axis driver responses to be included in the loudspeaker model and controlled by the filters specified in the on-axis spreadsheets. For each driver in the loudspeaker, and for each-axis of the loudspeaker model, a separate driver response spreadsheet is used. For each axis that the loudspeaker will be modeled on, a system response spreadsheet sums all of the driver response spreadsheets for that axis into the system response. This mimics the “in air summing” of each drivers’ output that occurs with the loudspeaker.

Crossover Routing – the Signal Flow

The signal flow in ACD uses the following routing scheme:



The system input filters are found in the System On-Axis Response spreadsheet. These might include things like shelving filters or EQ that spans the operating range of multiple drivers.

Each Driver On-Axis Response spreadsheet has a set of filters that only operate on the signal that is sent to that driver, independent of the other drivers.

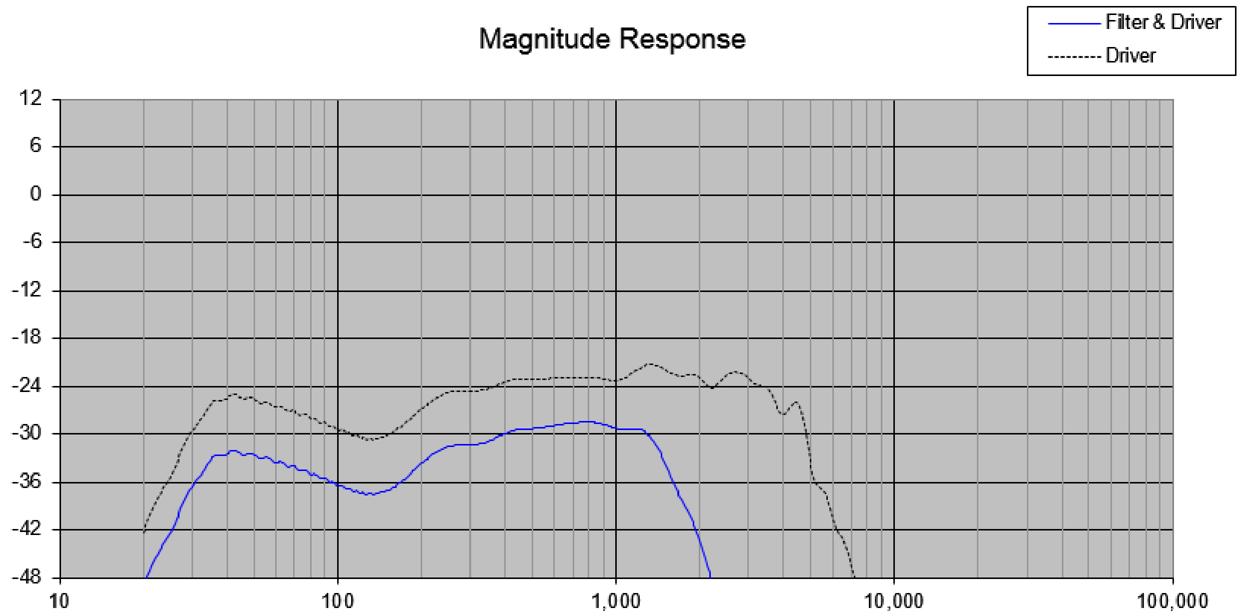
The system input filters act on all drivers in the same way. It’s helpful to use these filters when you do not want to disturb the relative phase angles between drivers but want to shape the overall frequency response.

As a whole these filters describe the complete crossover for the loudspeaker

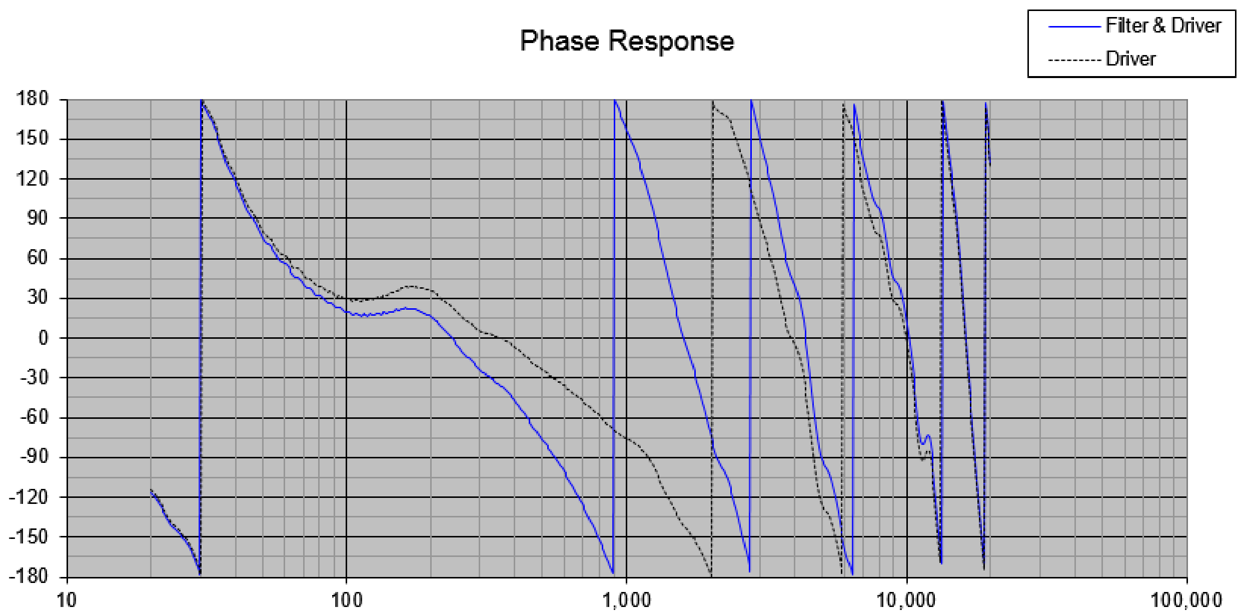
Plots of the Magnitude and Phase of Filters, Drivers, and the System

Plots of magnitude and phase are provided in the On-Axis Response spreadsheets. These show the frequency and phase response of the drivers, the system, and their filters. In addition to these plots, an extension to the ACD tools allows the response for multiple axes to simultaneously be plotted so that the designer can see the effect of the crossover filters on the full loudspeaker response (not just the on-axis response). The plots help to guide the designer to achieve both smooth on axis response, but smooth off axis response as well. The multi-axis plotting extension is not directly covered by this manual.

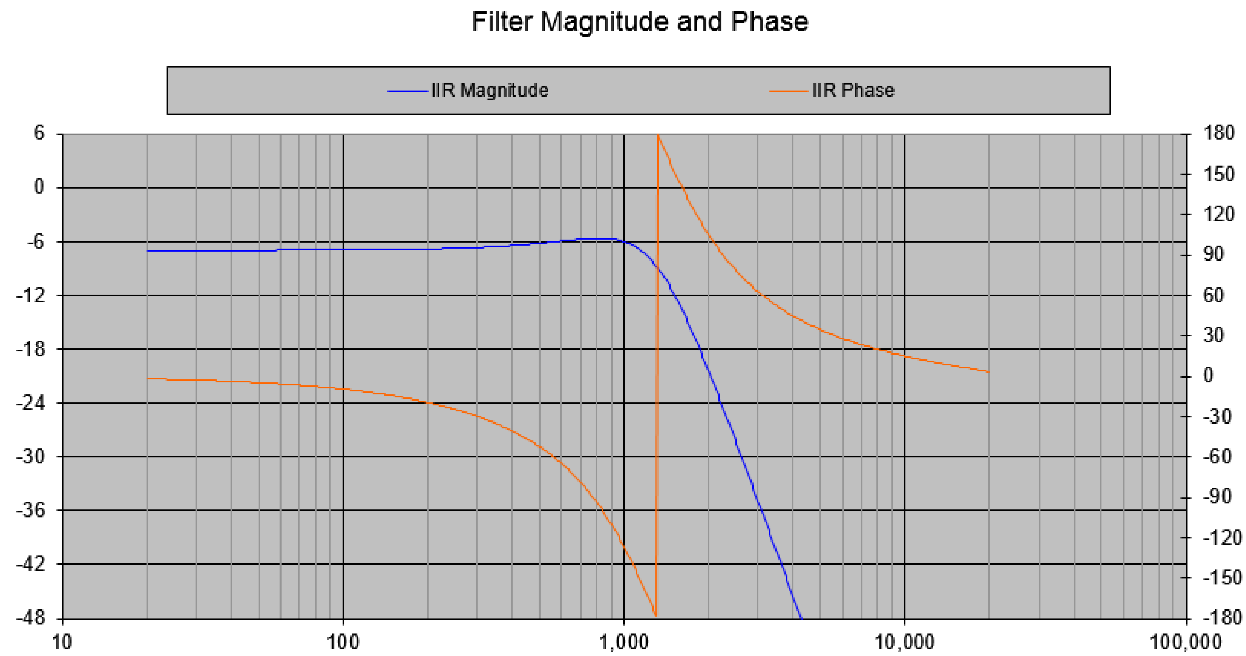
Some examples of ACD plots are shown below:



Above: An example of the driver response magnitude plot for a woofer, showing the raw (dashed) and filtered (solid blue) frequency responses.



Above: The driver phase response for the woofer above, showing the raw (dashed) and filtered (solid blue) frequency responses. The data presented in the phase plot includes the acoustic delay resulting from the relative acoustic offset of this driver from the reference plane.

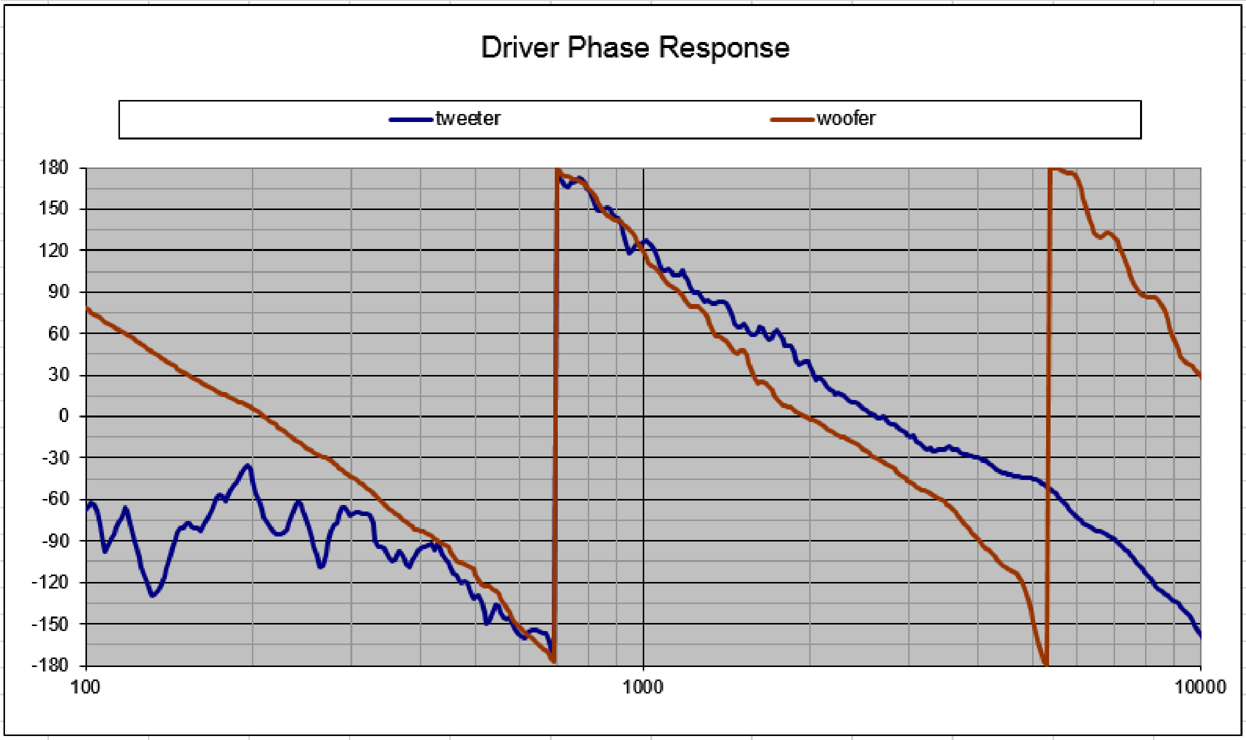


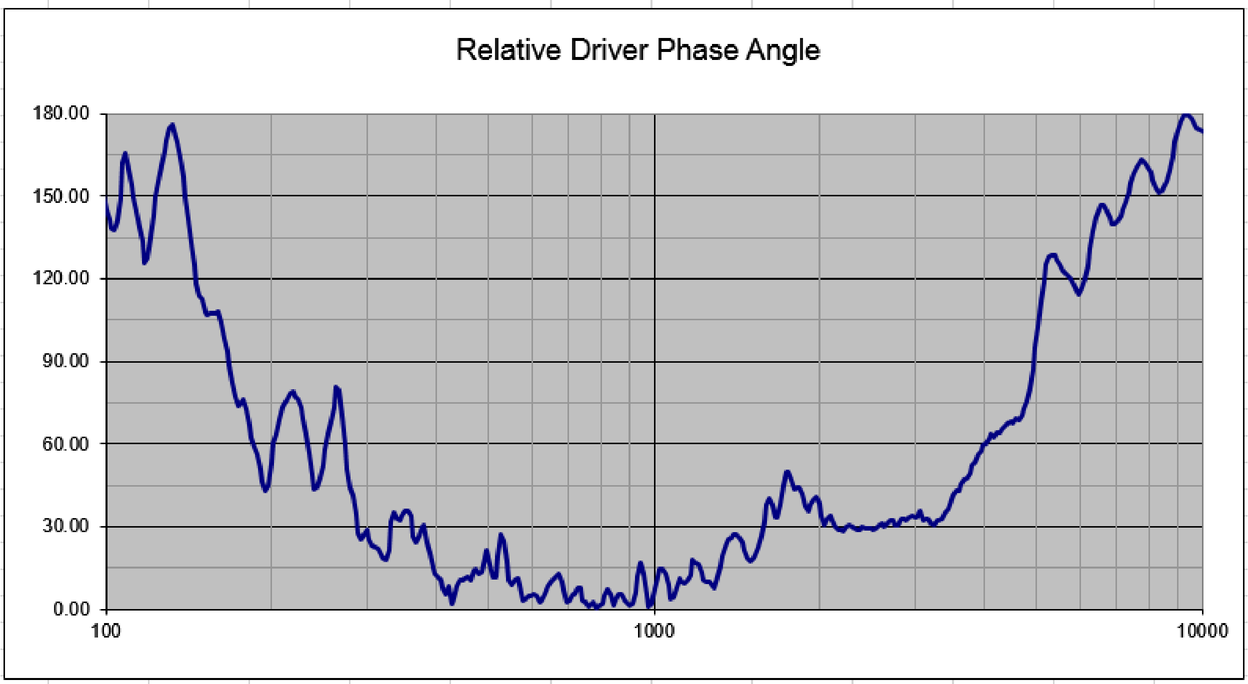
Above: An example of the filter response magnitude plot, showing analog (blue) and IIR digital (red) frequency responses, and phase responses.

The Relative Phase Response between Two Drivers

A multi-way loudspeaker crossover consists of two or more drivers. For portions of the frequency response more than one driver will be producing sound. This is called the “crossover region” and it is important to know the phase relationship between the drivers that are operating at the same time. When the sound produced by the drivers arrives at the listening position in phase they combine by adding together. If the sound produced by the driver is out of phase or close to it, there will be a null or cancellation in this frequency band. Typically one wants the phase angle between drivers operating within the same frequency band to no more than 45 degrees. Less than 30 degrees is ideal.

In order to make it easy to see the relative phase angle on one plot, an extension to ACD is available that can be linked into the driver response spreadsheets. It produces a plot of the difference in phase angle, or the relative phase angle, between two drivers. This takes into account the phase changes produced by the crossover filters AND the driver’s own phase response. It’s very important for the designer to keep the relative phase angle under control when developing the crossover, and this plot is very helpful in that regard. An example is provided below.





Above: An example of the information provided in the two-driver phase tracking extension. The lower plot shows the relative phase angle between the drivers.

The Filter Table

Both the System Response and Driver response Spreadsheets contain tables for describing filters and EQ that will make up the crossover for the loudspeaker. An example of the filter table is shown below:

	Filter 1	Filter 2	Filter 3	Filter 4	Filter 5	Filter 6	Filter 7	Filter 8
type:	4	21	21	0	0	0	0	0
gain [dB]:	0	0	0	0	0	0	0	0
polarity*:	1	1	1	1	1	1	1	1
Fop:	500	3000	3000	1	1	1	1	1
Qp:	1	0.707	0.707	1	1	1	1	1
Foz:	150	1	1	1	1	1	1	1
Qz:	1	1	1	1	1	1	1	1
* NOTE: for polarity enter 1 for normal polarity and -1 (or anything except 1) for reversed polarity								

Each filter is described in a column of the table. Above the filter table is a key that shows which number to enter in to the "type" field to indicate the type of filter to implement. Other filter parameters include: gain [dB], polarity, Fop (pole frequency), Qp (pole Q), Foz (zero frequency), and Qz (zero Q). For an explanation of each of these parameters, please see the section Poles and Zeros. The values of Q for each stage of some filters commonly used in loudspeaker crossovers is given in the section A Quick Review of some Well-Known Loudspeaker Crossovers

Many loudspeaker crossovers are named after a famous person in audio history and most of these have one or more favorable aspects, however, it's far more important to achieve a balanced tonal response from the loudspeaker than it is to restrict the crossover in any way. It's often a good idea to begin your design with one of these well-known crossover types, so we quickly review these in the following section.

Loudspeaker Crossovers using Butterworth or Linkwitz-Riley Filters.

Not all parameters are required for some types of filters. For first order low-pass, high-pass filters, and all-pass (filter types 1, 2, and 3) only the Fop parameter is used. For the first order shelving filters (types 4 and 5), Fop and Foz are needed to describe the filter. For the second order low-pass, high-pass filters, notch, and all-pass (filter types 21, 22, 23 and 24) use Fop and Qp. For the second order biquadratic filter (type 25), enter Fop, Foz, Qp and Qz. For more detail about each of these filter types, see the section Types of Filters Available in ACD. The values for parameters that are not required by the type of filter selected are ignored.

System Frequency Grid

The response of the drivers is calculated for a large number of points between some minimum and maximum frequencies – these points are called the "frequency grid". The settings in the frequency grid table in the System Response workbook represent a "master grid" that describes the frequency regime within which all the data will be modeled and summed. All the driver frequency grids are synchronized to the master grid in the system response spreadsheet.

The grid is described by a lower limit and an upper limit. The data points used for calculation are spread out evenly on a log scale over this interval. An example of the input area for the start and end frequencies is shown below:

	SYSTEM INFO:
20	[Hz] start frequency for driver response
20000	[Hz] stop frequency for driver response
48000	[Hz] = sampling frequency

The Filter Sampling Rate

The sampling rate used to calculate the filter responses in each spreadsheet is linked back to the "master rate" that is set in the on-axis system response workbook (cell B11) shown above. Enter the sampling rate that will be used by your audio stream and DAC. This is only for plotting purposes, and does not influence the LADSPA filter descriptors.

Driver Response Spreadsheet

In addition to filter and EQ, the driver response spreadsheet contains several driver specific parameters that are used to generate the model of the driver's response at the measurement position. These cells are found near the top of the DriverInfo worksheet, as shown in the following image:

DRIVER CHARACTERISTICS

ACOUSTIC DELAY WITH RESPECT TO REFERENCE PLANE:	
0.125	driver acoustic delay in milliseconds
344	speed of sound in meters per second
4.30	equivalent driver physical offset in centimeters
1.69	equivalent driver physical offset in inches
POLARITY OF DRIVER VOICE COIL WIRING:	
1	**1** = normal polarity, **–1** = reversed polarity

Driver Polarity

A worksheet cell describes the polarity of the driver's electrical connection. Enter 1 for normal polarity and -1 for reversed polarity. Any value other than 1 will result in reversed polarity. This is a characteristic of the driver's motor construction.

NOTE: In addition to the polarity setting in the DRIVER CHARACTERISTICS section, each filter can be assigned normal or reversed polarity. To keep track of the overall polarity of the driver's acoustic output with respect to the polarity of the input signal, below the Filter and EQ section there is a highlighted box that indicates the polarity for the current set of filter and driver polarities, as shown below:

THE ACOUSTIC POLARITY IS THE SAME AS THE INPUT SIGNAL POLARITY

When the polarity is reversed, the message changes to:

THE ACOUSTIC POLARITY IS REVERSED WITH RESPECT TO THE INPUT SIGNAL POLARITY

Driver Acoustic Delay

The acoustic delay is specified in milliseconds. The further behind the reference plane a driver is located, the greater the acoustic delay of any signal coming from it. The location of the reference plane is arbitrary, and the acoustic center of the driver that is closest to the microphone (usually the tweeter) is often taken as the reference plane.

It is extremely important to get a very accurate estimation each driver's acoustic delay at the measurement position. Errors in this value will cause the system sum to be in error, especially around the crossover points. A simple yet excellent method uses three measurements: driver 1 alone, driver 2 alone, and drivers 1&2 together. The two driver measurement contains an interference pattern that is a sensitive probe for the delay. There is an example of how to determine the acoustic delay using this approach in the ACD tutorial, and links to two other tutorials on this important topic are provided under "Tutorials: How to Find the Acoustic Delay Using the Three Measurement Method" in the References and Further Reading section.

Note that, apart from modeling the acoustic delay of a driver in the loudspeaker, the acoustic delay parameter may be used to model digital delay.

The delay is translated into a physical offset distance, for reference only. The calculation depends on the speed of sound in air, which is density dependent. The user is able to enter a value for the speed of sound in air (343.3 m/sec at sea level at 20C), since it varies slightly with altitude and temperature.

System Response Spreadsheet

The purpose of the system response spreadsheet is to sum all of the driver responses together into the system response. Like in the DriverInfo worksheet in the driver response spreadsheet, a set of system EQ and filters is provided is provided in the SystemInfo sheet. Note that these filters are applied to the "input" to the system, e.g. before the signal is routed to the filters in the driver response spreadsheets, and so the system filters effect all drivers in the same way. This useful when your crossover has the drivers summing together nicely, but you want to tweak the system response without effecting the relative phase of each driver.

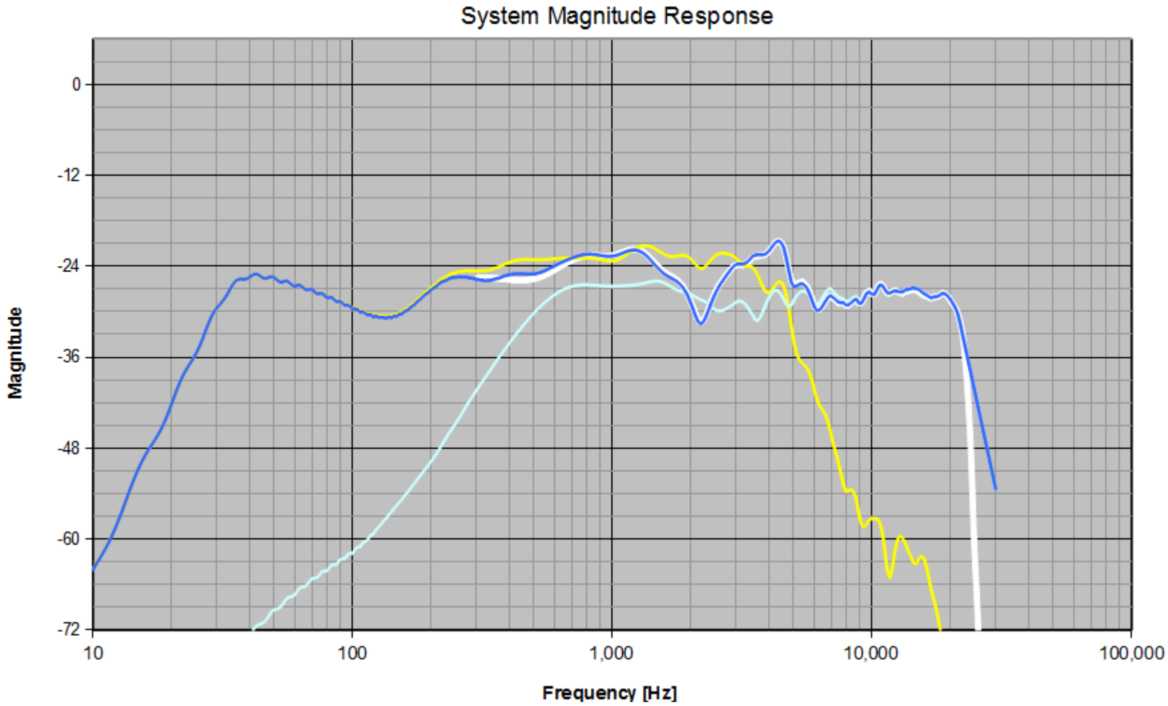
In addition to filter and EQ, the system response spreadsheet contains other specific items to help with some aspects of the crossover design process:

The Reference Response

The ReferenceResponse worksheet in the System Response spreadsheet is where an FRD file can be provided. This data, known as the "Reference Response", is shown in the System Magnitude plot on the SystemInfo worksheet, and can be used in various ways: as a target for developing the crossover, to compare the system response calculated by the ACD tools with a measured system response, or during the determination of the acoustic delay.

System Magnitude Response Plot and Magnitude Scaling Factors

The system magnitude response plot provides information on the summed system response, as well as for each driver in the system. If Reference Response data has been provided, this will also be shown in the plot. An example is provided below: An example of the System Magnitude Response plot is shown below:



Above: An example of a system response magnitude plot, showing the woofer (yellow), tweeter (light blue), ACD calculated (system response, dark blue) and measured (reference response, white) frequency responses.

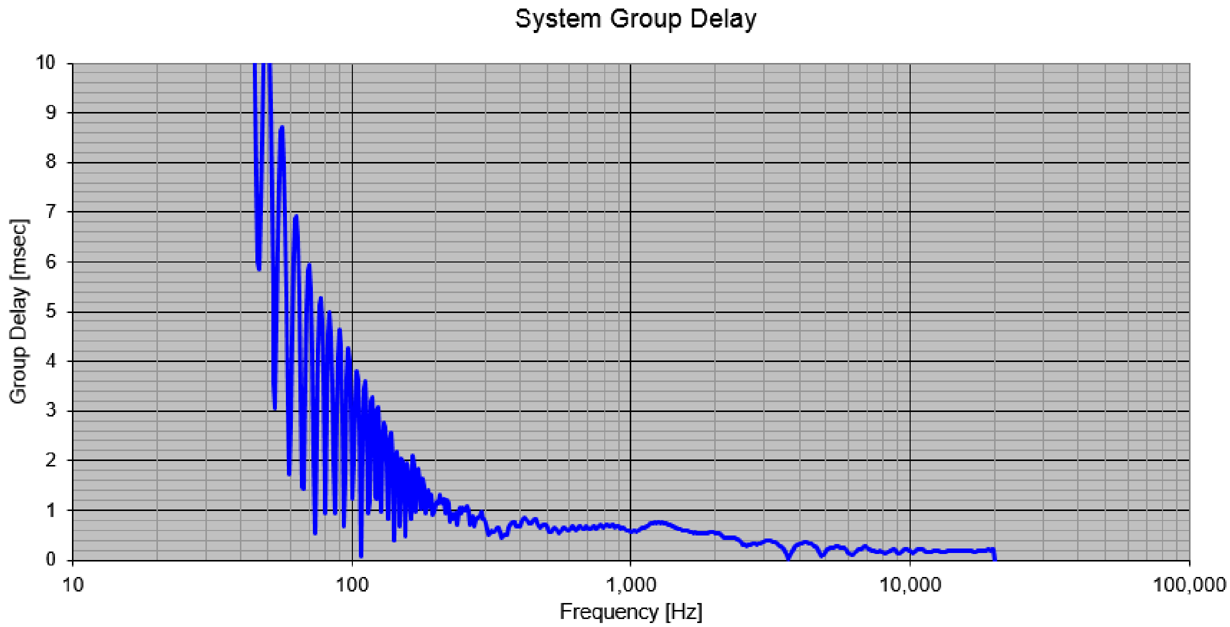
Sometimes the measurements obtained by the user have levels that are much higher, or much lower, than 0 dB. It is often convenient to show the system response sum with an average level of 0dB. To allow for this type of level correction two scaling factors have been included in the system response spreadsheet of the ACD tools, as shown below.

SYSTEM RESPONSE MAGNITUDE SCALING FACTORS	
0	dB offset to add to system response
0	dB offset to add to reference response

The value entered for the system response scaling factor is added to the responses of the drivers and the system sum in the Magnitude Response plot. To move the levels “up” on the plot, enter a positive value. The reference response level can be scaled separately, using the reference response scaling factor in the table.

System Phase and Group Delay Plot

In addition to the plots of system SPL, system EQ and system filters, plots of the system phase and group delay are presented. An example of these plot is shown below:



Above: An example the system group delay plot in the system response workbook. The low frequency noise is a result of small amplitude oscillations in the woofer data set.

The group delay is often calculated as $-d\phi/d\omega$, however, this can result in negative valves for the group delay, which violates causality. The calculations in ACD follow the formulation established in "*Suggestion for a New Formula to Calculate Group Delay from Frequency Domain Measurements*" by Itzhak Shapir (see reference 1 below), which are free of this behavior.

In ACD, the calculated group delay is influenced by both the phase and the amplitude of the system sum, and when it contains dips, valleys, or ripples these will show up in the group delay. In the plot above, you can see the oscillations at low frequency in both the phase and group delay. The source of these ripples is the diffraction response that was used to model the driver's behavior at low frequencies.

A Primer on Filters

Filters are combined into crossovers, so learning about filters can help you design your crossover appropriately. In order to become more familiar with the way filters are described in ACD, it's a good idea to review some terminology and mathematics.

What is a filter?

In the most general sense, a filter is a signal processing block that alters the magnitude and phase response of the input signal. A filter is commonly used to remove, change, or shape some components of the response. Active filters can cut and/or boost the magnitude, and/or can shape the phase response. Filter types such as low-pass and high-pass are well understood in terms of their amplitude and phase behavior.

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[1] "Suggestion for a New Formula to Calculate Group Delay from Frequency Domain Measurements", I. Shapir. Presented at the 36th European Microwave Conference, 10-15 Sept 2006.

Filters come in different levels of complexity and capability. Filters with higher complexity are typically built up from several stages of first and second order filters in what is termed a "cascade", meaning that filter stages are connected in series. Filters having some desirable characteristic or set of characteristics such as a "maximally flat passband" or "no overshoot in the step response" have been cataloged and named for famous people such as Butterworth. Most filters were not initially designed for loudspeaker crossovers, but were later adapted for that purpose.

Describing Filters – the Transfer Function

As mentioned above, filters are generally created as a cascade of first and second order filters connected in series. So what are these first and second order stages and how do we describe their filtering action? Mathematically, a filter's behavior is described by its transfer function. For background information and to learn more about transfer functions, please read the entries in the References and Further Reading section under the heading "Transfer Function". Transfer functions describe the relationship between the input to the filter, X(s), and the output of the filter, Y(s), where s is the Laplace operator. For our purposes, $s = j\omega$, where ω is the radian frequency ($\omega = 2\pi f$, where f is the frequency in Hertz) and j is the imaginary number equal to $(-1)^{0.5}$. See "Waves, Complex Numbers, and Phasors" in the section References and Further Reading to learn about this concept. The variable s can be thought of as being a kind of frequency variable, and anywhere you see "s" just replace it with "frequency" and you will likely be able to understand the overall behavior of the filter.

The Transfer Function describes how the magnitude and the phase of the input signal are changed by the action of the filter on the input signal to result in the output signal. This input-output relationship is represented by H(s). Since it is the ratio of output over input, and each of these can be described by a polynomial in s, any filter transfer function can be described by:

$$H(s) = \frac{Y(s)}{X(s)} = \frac{\sum_{i=0}^{i=M} b_i s^i}{\sum_{j=0}^{j=N} a_j s^j}$$

The highest power of s is equal to the order of the filter, and for stability $M \leq N$. The b_i and a_j are the transfer function coefficients. This formulation provides a very general way to describe the response of filters of any order, however, filters used in loudspeaker crossovers are usually first or second order, and higher order filters are created by cascading these two basic types.

When M = N = 1 we get an equation that can describe all first order filter transfer functions:

$$H(s) = \frac{b_1 s + b_0}{a_1 s + a_0}$$

Similarly, when M = N = 2, we get an equation that can describe all second order filter transfer functions:

$$H(s) = \frac{b_2 s^2 + b_1 s + b_0}{a_2 s^2 + a_1 s + a_0}$$

The b_i and a_j (the transfer function coefficients) are sometimes referred to as the "biquad coefficients" since, for a general second order transfer function, both the numerator and denominator are quadratic functions in s.

We can re-write these general first and second order transfer functions in terms of more familiar filter parameters, specifically the filter corner frequency, ω_0, the filter Q, and the filter gain, G. As an example, the transfer functions of high-pass (below at left) and low-pass (below at right) first order filters looks like this:

$$H(s) = G\frac{s}{s+\omega_0} \qquad\qquad H(s) = G\frac{\omega_0}{s+\omega_0}$$

By taking the general first order form and setting b_1=0, we get the low-pass transfer function. Likewise, by taking the general first order form and setting b_0=0, we get the high-pass transfer function. If we match powers of s, one can see that $b_0 = \omega_0 G$, $b_1 = G$, $a_1 = 1$ and $a_0 = \omega_0$ for the first order functions.

If we look at the first order HP filter transfer function (above at left) and replace “s” with “frequency” we can see that as “frequency” becomes much greater than ω_0, the ratio of the numerator and denominator approaches 1. When the “frequency” (the radian frequency) is about equal to ω_0, the fraction is equal to 1/2, which is approximately the corner frequency of the filter. As “frequency” gets lower and lower than ω_0, the fraction becomes smaller and smaller, meaning there is increasing attenuation. This is indeed the general behavior of a high-pass filter as a function of frequency.

Likewise, for second order high-pass (below, left), and low-pass (below, right) filters we have:

$$H(s) = G\frac{s^2}{s^2+\frac{\omega_0}{Q}s+\omega_0^2} \qquad\qquad H(s) = G\frac{\omega_0^2}{s^2+\frac{\omega_0}{Q}s+\omega_0^2}$$

Note that the denominator is identical for both the high- and low-pass functions, and only the numerator changes. We can again evaluate these phenomenologically by replacing “s” with “frequency”. There is an additional, intermediate, regime of frequency, when “Frequency” is about equal to ω_0 (near resonance) where the Q term becomes important. The value of the Q influences the response around ω_0 (e.g. “near resonance”). All of these regimes are typical of the response for a second order filter.

Poles and Zeros

Let’s rewrite the general second order transfer function using the parameters ω_0, Q, and G:

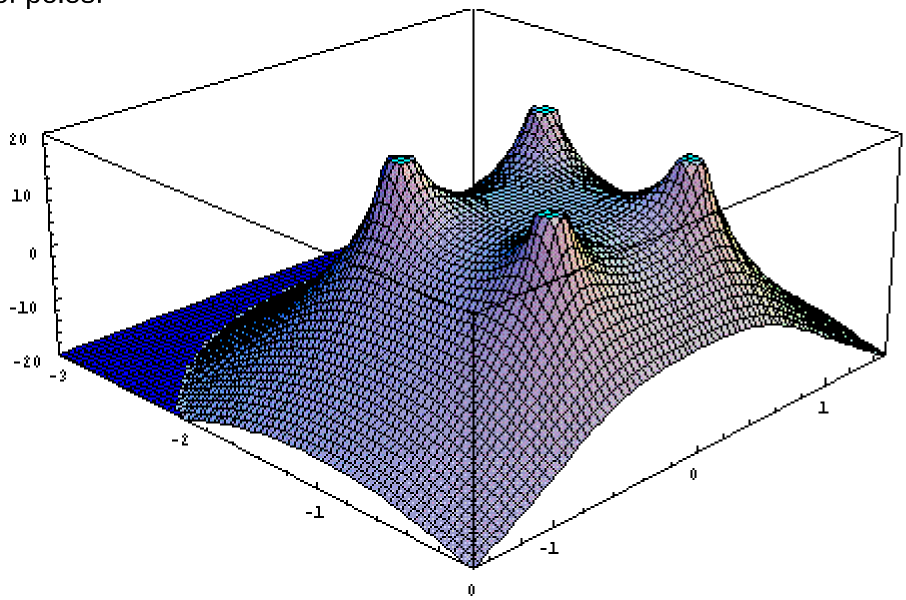
$$H(s) = G\frac{s^2+\frac{\omega_{0z}}{Q_z}s+\omega_{0z}^2}{s^2+\frac{\omega_{0p}}{Q_p}s+\omega_{0p}^2}$$

Since both the numerator and denominator have the same general form, so we need to use an additional subscript to refer to the resonant frequency, ω_0, and Q in the numerator (z) and the denominator (p). These subscripts arise from the terms pole (p) and zero (z).

Recall from math that when the denominator of a fraction approaches zero, the quotient approaches infinity. The quantities in the denominator are related to the poles of the transfer function, since when they sum to zero the transfer function value quickly increases to infinity. When you plot the transfer function in the complex plane, what you get is a surface. The “poles” look somewhat like what you would get if you draped some loose fabric over a few poles stuck in the ground.

On the other hand, if the numerator of a fraction approaches zero, the quotient becomes zero and we talk about zeros in the numerator. Rather than shooting up to infinity, zeros are like “holes” where the fabric is pulled down to zero, as if it were pinned to the ground. By using combinations of poles and holes in the complex plane, the surface of the transfer function can be made to take on the familiar shapes of high-pass, low-pass, and other filter types.

For an excellent visual tutorial about filter poles, see the link provided under the heading “Poles and Zeros in the Complex Plane” in the References and Further Reading section. It’s a highly recommended read! A figure, taken from the linked web page is shown below to illustrate the concept of poles:



Above: A 3-dimensional plot of a low-pass transfer function in the complex plane (left half only), showing four poles. Credit: Tim Stinchcombe (http://www.timstinchcombe.co.uk).

To recap, filters can be described by a transfer function. By using this mathematical representation of filters, we can completely describe any filter using just a couple of parameters. First order filters are characterized by a single variable, the corner frequency, ω_0. Second order filters are characterized by two variables, the corner frequency ω_0 ($\omega = 2\pi f$, where f is the frequency in Hertz), and the Q or "filter bandwidth". High-pass, low-pass, and other types of filters are formed when one or more of the terms in the general form of the transfer function is removed by setting the coefficient equal to zero. The variable s completely describes how the filter's response (magnitude and phase) changes with frequency, so using this nomenclature simplifies and parameterized the representation of filters in to a convenient form. As a result, it is this representation of filters, with poles and zeros, is used in the ACD tools.

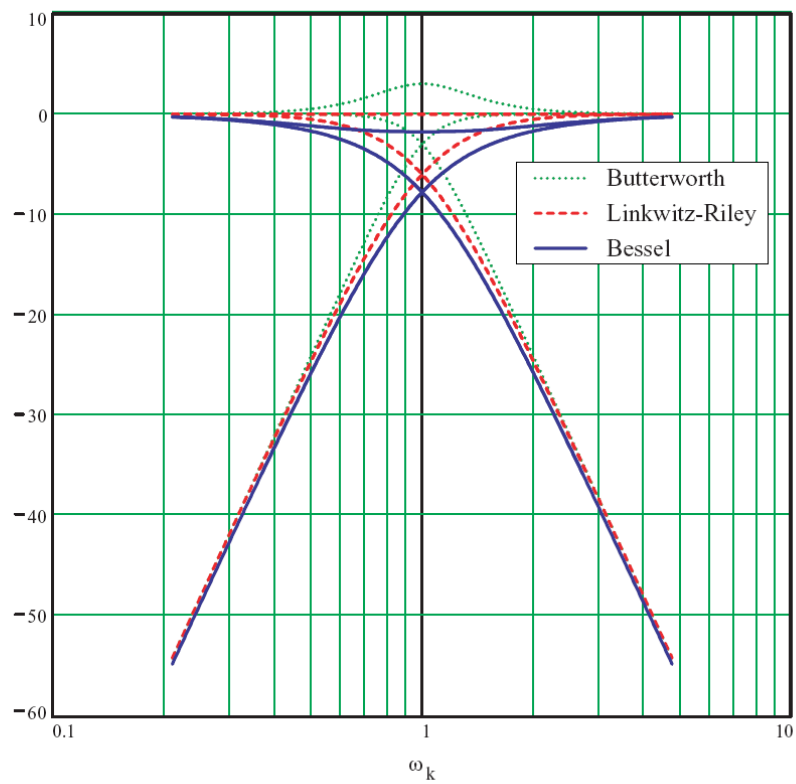
Using Filters to Construct Crossovers

What is the difference between a filter and a crossover?

A filter has a single function (a high-pass function for instance) and there is one input and one output. In its basic form, a crossover uses two or more complementary filters, (e.g. a high-pass AND a low-pass, together), so that the two outputs sum to relatively flat amplitude. This requires that the amplitude and phase response of the filters behaves in a certain, and complementary, way. Effective and high fidelity loudspeaker crossovers do not require that the filter order for HP and LP functions be the same, in fact asymmetric crossovers are common and can help to tailor the phase response and shape the amplitude response as needed to achieve the designer's frequency response goals.

At Right: A comparison of three types of crossovers, showing the high-pass filter response, the low-pass filter response, and the summed response.

Source: "A Bessel Filter Crossover, and Its Relation to Others", RaneNote 147.



A loudspeaker crossover has an additional level of complexity because multiple sound sources (the drivers in the loudspeaker) are located at different positions in space so that the wavefront from one driver that reaches the listening position has additional phase rotation with respect to the other drivers. Not only must a crossover serve a signal processing function, for instance to low pass and high pass the signal, but it must also compensate for the phase abnormalities in the driver wavefronts so that, within the band of frequencies in which one driver crosses over to another, the sound waves can sum in a predictable way. Often the driver responses display significant deviation from "flat response" as a result of a variety of acoustic phenomena, and the crossover must also compensate for these disturbances.

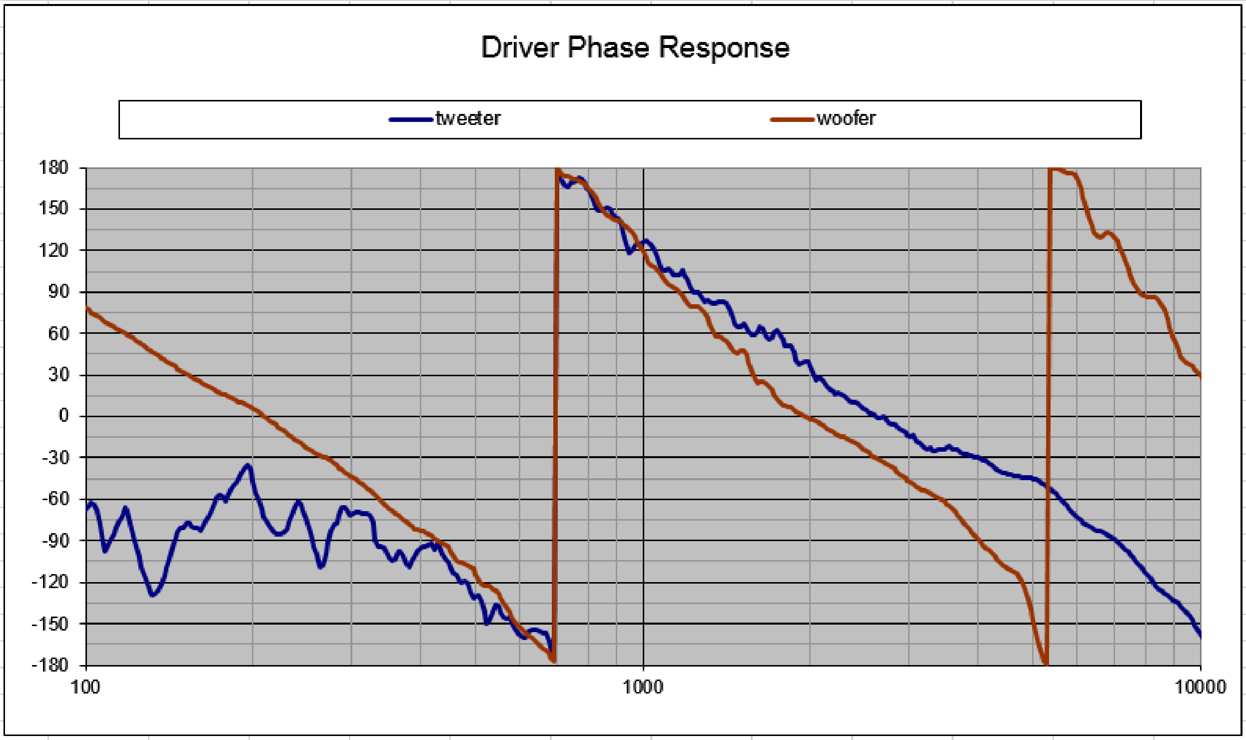
In general, the overall goal is to create a system of filters for each driver that result in the loudspeaker system wavefront having coherence and flat frequency response at the listening position, and a well-behaved response on other axes as well. In order to accomplish this goal, the crossover filter functions must be able to take on any shape, and this means that the filters must be able to be of arbitrary Q, usually in the range $0.5 < Q < 2$. As a result, crossover that rely only on "standard" filter types such as Butterworth, Linkwitz-Riley, Bessel, and so on can be very limiting for loudspeaker crossovers.

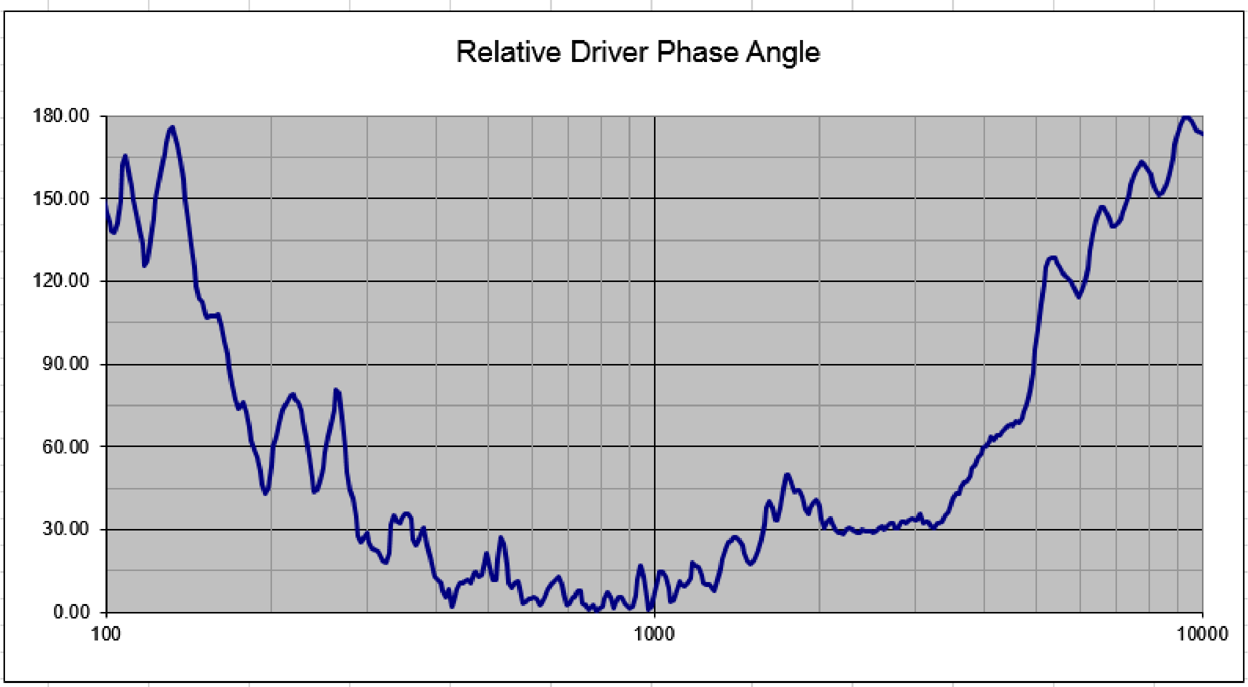
Types of Filters Available in ACD

This section lists some basic information on how to describe filters in ACD in terms of the filter parameters that are entered in the filter design table. For more information on the filter table, see the section The Relative Phase Response between Two Drivers

A multi-way loudspeaker crossover consists of two or more drivers. For portions of the frequency response more than one driver will be producing sound. This is called the "crossover region" and it is important to know the phase relationship between the drivers that are operating at the same time. When the sound produced by the drivers arrives at the listening position in phase they combine by adding together. If the sound produced by the driver is out of phase or close to it, there will be a null or cancellation in this frequency band. Typically one wants the phase angle between drivers operating within the same frequency band to no more than 45 degrees. Less than 30 degrees is ideal.

In order to make it easy to see the relative phase angle on one plot, an extension to ACD is available that can be linked into the driver response spreadsheets. It produces a plot of the difference in phase angle, or the relative phase angle, between two drivers. This takes into account the phase changes produced by the crossover filters AND the driver's own phase response. It's very important for the designer to keep the relative phase angle under control when developing the crossover, and this plot is very helpful in that regard. An example is provided below.





Above: An example of the information provided in the two-driver phase tracking extension. The lower plot shows the relative phase angle between the drivers.

The Filter Table. The types of filters available in ACD include: gain block, 1st order low-pass, 1st order high-pass, 1st order all-pass, 1st order bass and treble shelving, 2nd order low-pass, 2nd order high-pass, 2nd order all-pass, 2nd order notch, 2nd order biquadratic filter, parametric EQ, and 2nd order bass and treble shelving.

Filters in ACD-L are modeled as IIR digital filters. For background information about each filter type, see the references listed under "Filter Transfer Functions:" in the References and Further Reading section. The remainder of this section describes the available filter types in more detail.

Gain Block

The gain block has no filtering function associated with it. Its sole function is to provide gain adjustment and phase inversion independent of other filters.

High-Pass and Low-Pass Filters

First and second order high-pass and low-pass filter responses are well known. The first order HP and LP are described by a single parameter – the corner frequency, entered in the filter table as Fp. The second order HP and LP filters are described by two parameters, the filter corner frequency and the Q, entered in the filter table as Fop and Qp respectively.

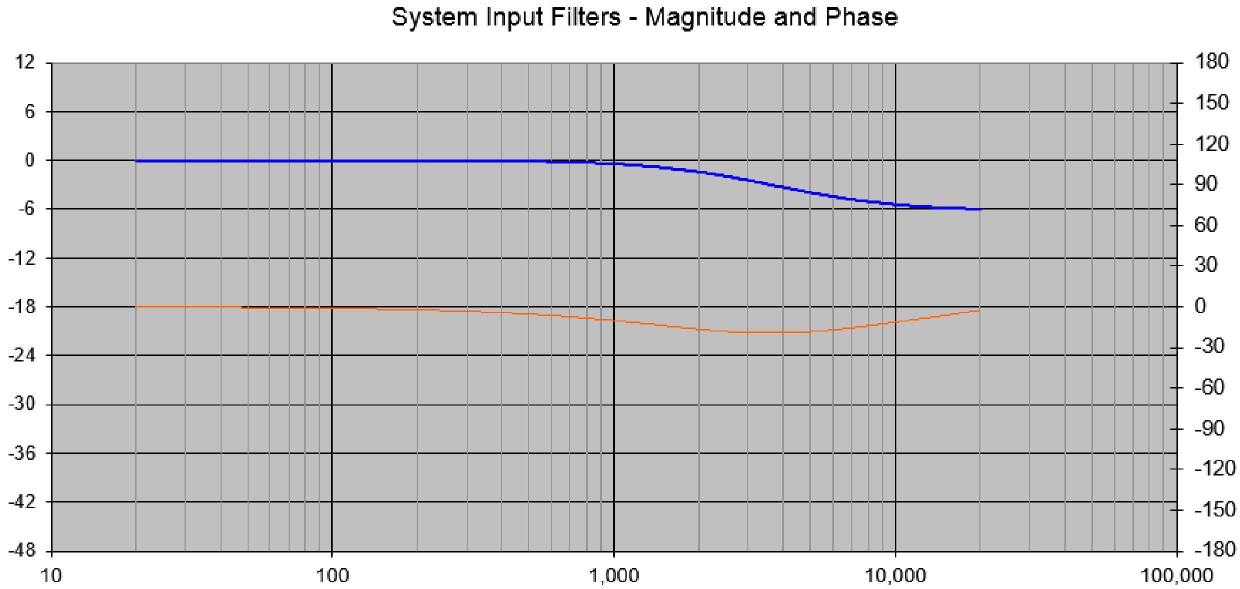
All-Pass (Time Delay) Filters

First and second order analog all-pass filters have flat frequency response but display a frequency dependent delay (the group delay). The AP filters are described in a similar fashion to the HP and LP filters. The first order HP and LP are described by a single parameter – the corner frequency, entered in the filter table as Fp. The second order HP and LP filters are described by two parameters, the filter corner frequency and the Q, entered in the filter table as Fop and Qp respectively. The shape of the group delay is similar to that of the frequency response for a low-pass filter of the same order and Q, that is to say that at low frequency the delay is a constant value, and then near the corner frequency the delay begins to fall, reaching zero at high frequencies.

First order bass and treble shelving filters

The first order shelving filter is useful for compensating for gentle frequency response changes such as the "baffle step". The filter has a slope of 6 dB/oct in the transition region, and the amplitude gently transitions between 0dB (no change) at one frequency extreme, to a higher or lower level at the other frequency extreme. For instance, the bass shelving filter always has a response equal to 0dB in the treble regime but can boost or cut the bass level. Likewise the treble shelving filter always has a response of 0dB in the bass, but the highs can be boosted or lowered.

An example of the response of a bass shelving filter having a boost of 6dB and Fop = 2k Hz is shown below:

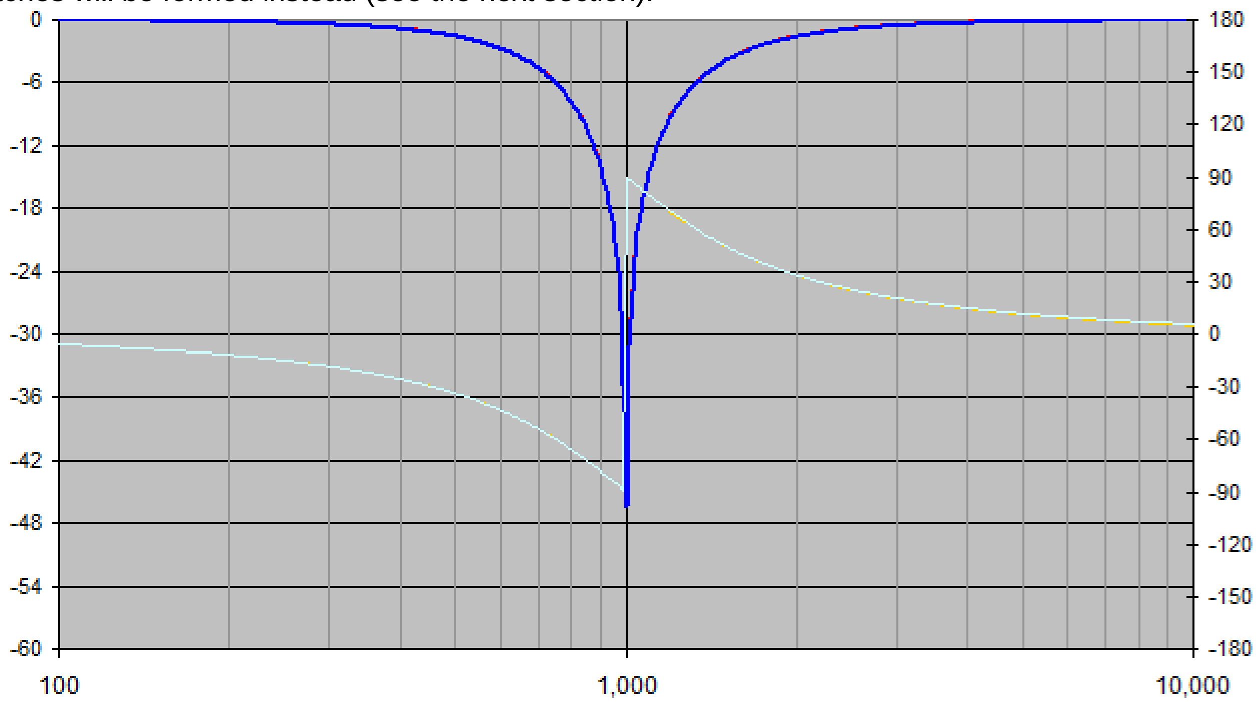


Above: An example of a 1st order treble shelving filter with Fop = 5000 Hz and gain = -6 dB.

Symmetric, High-Pass and Low-Pass Notch Filters:

Symmetric Notch:

A symmetric notch filter is described by setting both Fop and Foz equal to the notch frequency, and Qp equal to the notch Q. This produces a notch at the desired frequency with the desired Q, like the example shown below. If Fop ≠ Foz, one of the high-pass or low-pass notches will be formed instead (see the next section).



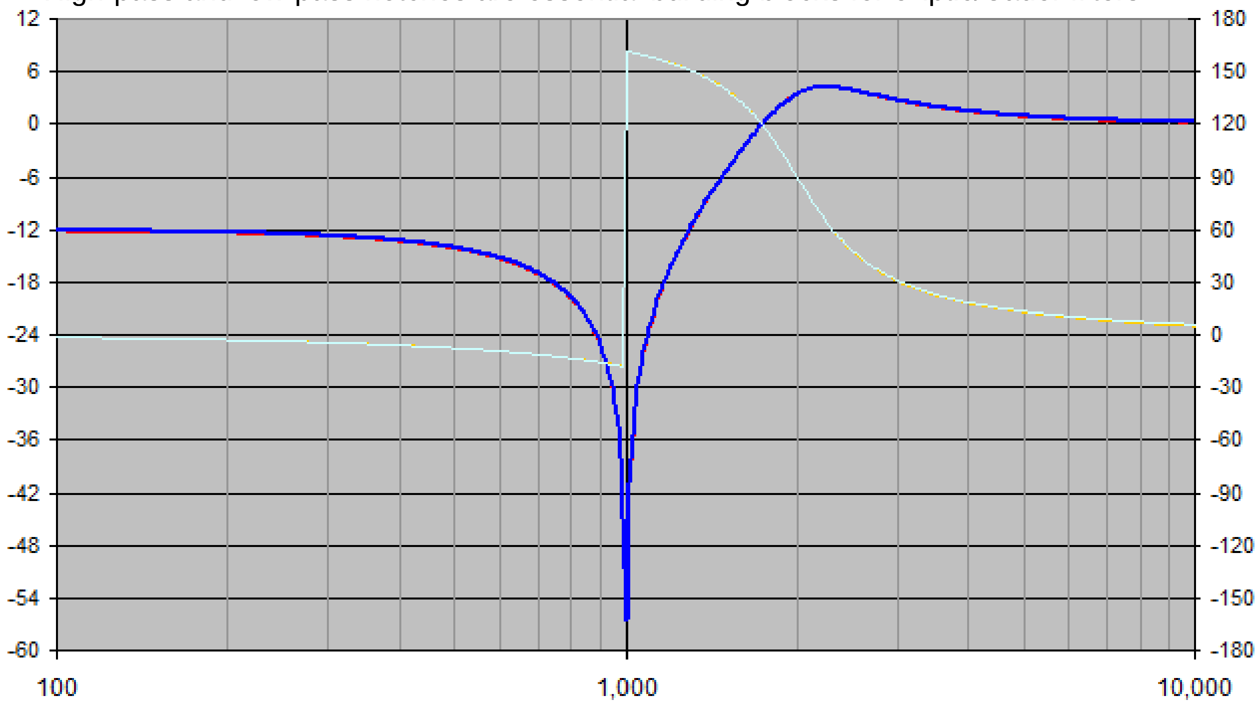
Above: Symmetric Notch Response. Fop = Foz = 1000 Hz and Q = 1.

High-Pass Notch and Low-Pass Notch

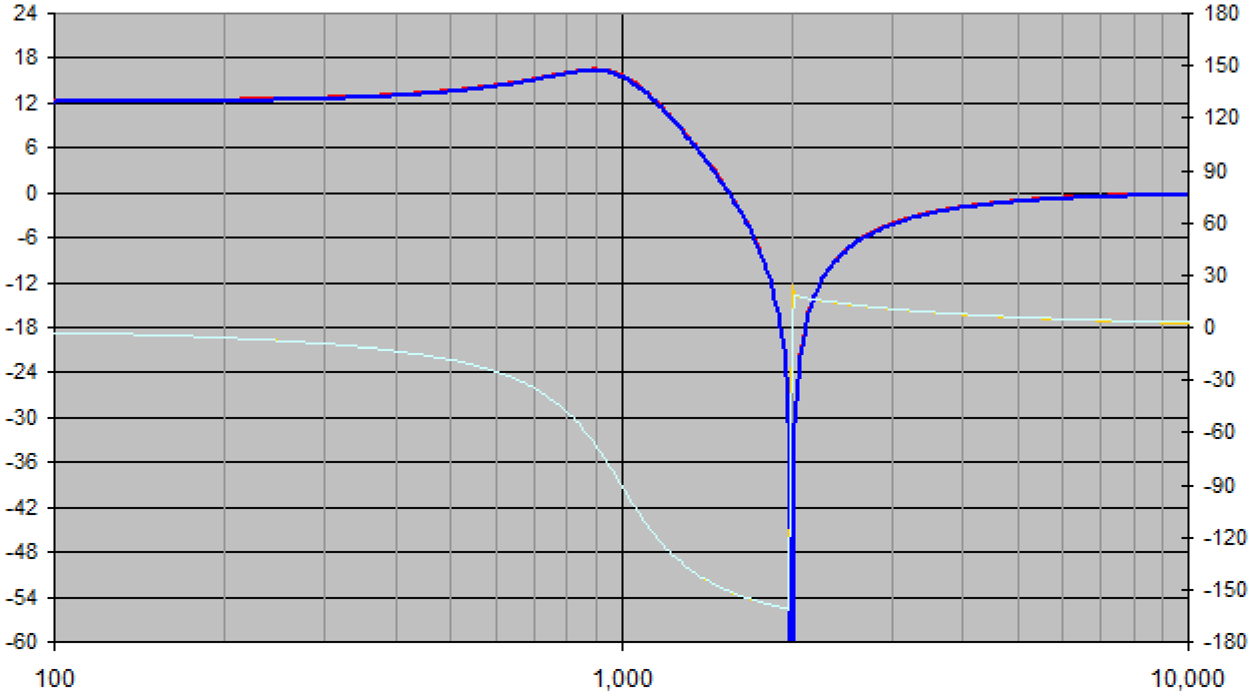
When the pole and zero frequencies are dissimilar, a high-pass notch or a low-pass notch is formed. If Fop > Foz the high-pass notch is formed. When Foz > Fop the low-pass notch is formed.

For these notch types, the "pass" function formed by the pole of the transfer function takes on the Q value supplied as Qp. Note that for both the high and low pass notches, the high-frequency response asymptotes to the 0 dB level. For the high-pass notch the low frequency level is reduced, and for the low-pass notch the low frequency level is elevated, compared to the high-frequency level. The difference in level is 12dB per octave of separation between Fop and Foz.

High-pass and low-pass notches are essential building blocks for elliptic/Cauer filters.



Above: High-Pass Notch Response. Q = 2, Fop = 2000 Hz and Foz = 1000 Hz.



Above: Low-Pass Notch Response. Q = 2, Fop = 1000 Hz and Foz = 2000 Hz.

Biquadratic Filters:

Biquadratic filters are the most general type of second order filter function, and are described by a full quadratic expression in s, both in the numerator and the denominator. This allows the Biquadratic filter to generate some more sophisticated filter shapes compared to the low-pass, high-pass, and notch filters, namely second order shelving filters, and EQ-like response shaping filters. The form of the biquadratic filter function is shown below:

$$H(s) = G\frac{s^2 + \frac{\omega_{0z}}{Q_z}s + \omega_{0z}^2}{s^2 + \frac{\omega_{0p}}{Q_p}s + \omega_{0p}^2}$$

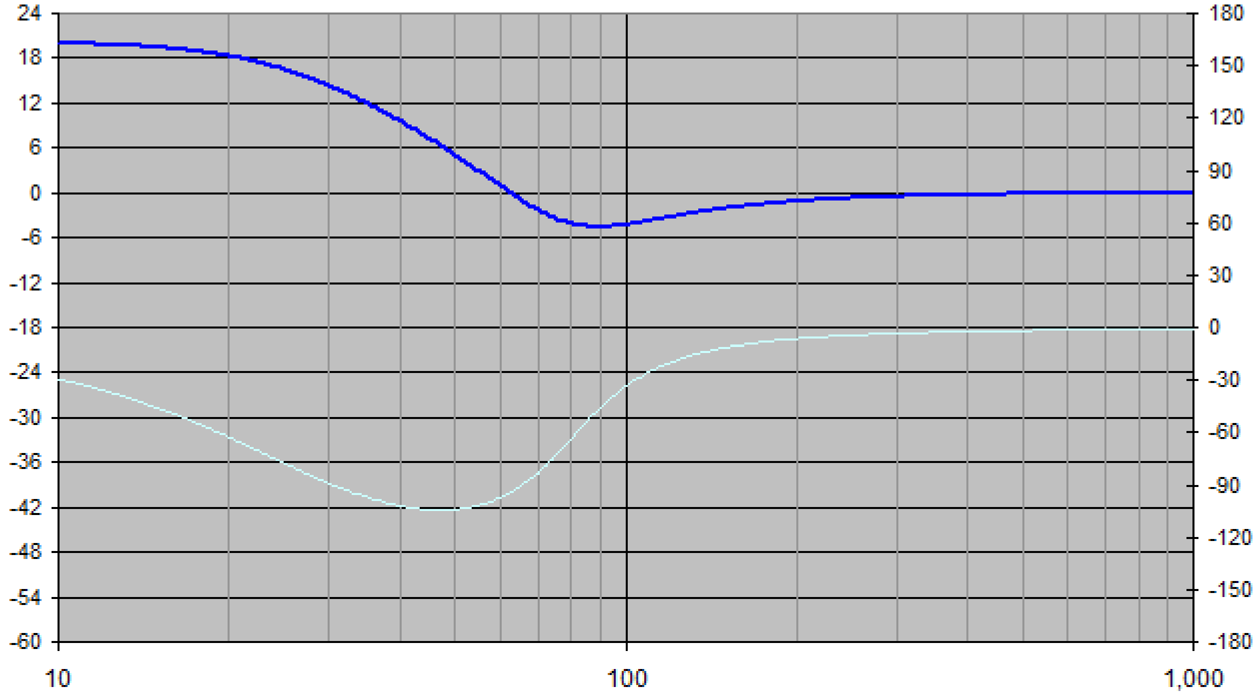
Both the numerator and denominator have the same general form, so we need to use an additional subscript to refer to the resonant frequency, ω_0, and Q in the numerator (z) and the denominator (p). These subscripts arise from the terms pole (p) and zero (z).

Linkwitz-Transform

Most DIYers are familiar with the “Linkwitz Transform”. The LT is just a biquadratic filter for which Foz and Qz are set equal to the box resonance frequency and Q of the driver in the box, and Fop, Qp are set equal to the desired “new” resonance frequency and Q. These values can be directly entered in the filter design table to create a LT filter. The LT got its fame as a closed-box subwoofer “equalizer” because a driver in a closed box rolls off a 12dB/oct below resonance and the LT increases the power at the same rate to flatten and extend the response.

A lesser-known application for the biquadratic filter is to change the low frequency behavior of a tweeter. The tweeter’s Fo and Q are modified by the filter such that together they result in a new Fo’ and Q’ (usually Fo’ > Fo). A tweeter is essentially a small driver in a small closed box and much the “Linkwitz Transform” the frequency response can be changed by the biquadratic filter at will. Rather than extending the response lower, the apparent resonance frequency is increased so that the tweeter's response can act as a stage in the crossover. This may be advantageous in some situations because the total phase lag will be different compared to implementing a HP filter stage.

An example of the Biquadratic Filter used as a Linkwitz-Transform type bass boost on a subwoofer is shown below:

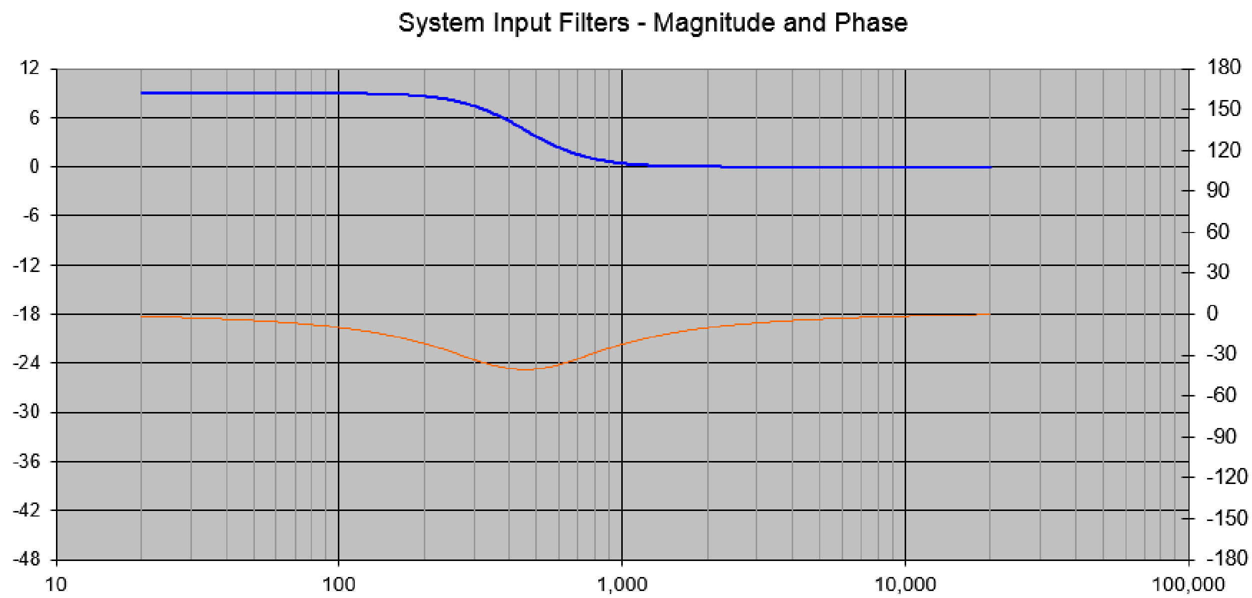


Above: "Linkwitz-Transform" Biquadratic Filter Response. Qp = 0.707, Qz = 1.6, Fop = 25 Hz and Foz = 80 Hz.

Second-Order Shelving Filters

Similar to the first order bass and treble shelving filters, second order bass and treble shelving filter boost or cut one part of the spectrum, however, the second order shelvers transition at 12dB per octave.

An example of a second order bass shelving filter is shown below:



Above: Second Order Bass Shelving Filter with Fop = 350 Hz and gain of 9dB.

Parametric EQ (analog and digital forms)

Parametric response equalization filters are useful for filling in dips or flattening peaks in the frequency response. These can be described in terms of the center frequency (Fo) the Q of the equalization (Qp) and the amount of boost or cut (dB gain). The digital EQ is symmetric, meaning that two filters having the same center frequency and Q but opposite boost (eg. +6dB and -6dB) will cancel each other out. In contrast the analog form mimics the shape of an analog EQ, which is derived from a bandpass filter. This will have a sharper band when used as a "cut" EQ than when it is used for "boost" just as the analog EQ circuit has when using the same parameters for Fo, Q, and boost/cut.

A Quick Review of some Well-Known Loudspeaker Crossovers

Many loudspeaker crossovers are named after a famous person in audio history and most of these have one or more favorable aspects, however, it's far more important to achieve a balanced tonal response from the loudspeaker than it is to restrict the crossover in any way. It's often a good idea to begin your design with one of these well-known crossover types, so we quickly review these in the following section.

Loudspeaker Crossovers using Butterworth or Linkwitz-Riley Filters

The well-known Butterworth and Linkwitz-Riley filter types can be set up by entering the desired crossover frequency in the ACD filter table, selecting the type of filters needed, and entering the Q for each stage of the filter. For instance, a, LR4 (4th order Linkwitz-Riley) filter has two second order stages, both having Q=0.707, and the corner frequencies of both filters are set equal to the desired crossover point. For all Butterworth and Linkwitz-Riley filters, the corner frequency of all filter stages is equal to the crossover frequency. The table below lists the value of Q for loudspeaker crossover filters or this type, up to 8^{th} order.

	1st order stage	second order stage #1 Q	second order stage #2 Q	second order stage #3 Q	second order stage #4 Q
LR2		0.5			
BUT2		0.707			
BUT3	✓	1			
LR4		0.707	0.707		
BUT4		0.54	1.31		
BUT5	✓	0.62	1.62		
LR6		0.50	1.00	1.00	
BUT6		0.52	0.707	1.93	
BUT7	✓	0.55	0.8	2.24	
LR8		0.54	0.54	1.31	1.31
BUT8		0.51	0.60	0.90	2.56

Other Types of Loudspeaker Crossovers

There are several other types of filters that can be used in loudspeaker crossover, including Bessel, Chebyshev Type I and II, Elliptic, Hardmann, NTM (Thiele), etc. These less common loudspeaker crossover types can all be modeled with the ACD tools because they are just combinations of one or more of the filters that ACD can describe. There is no magic going on!

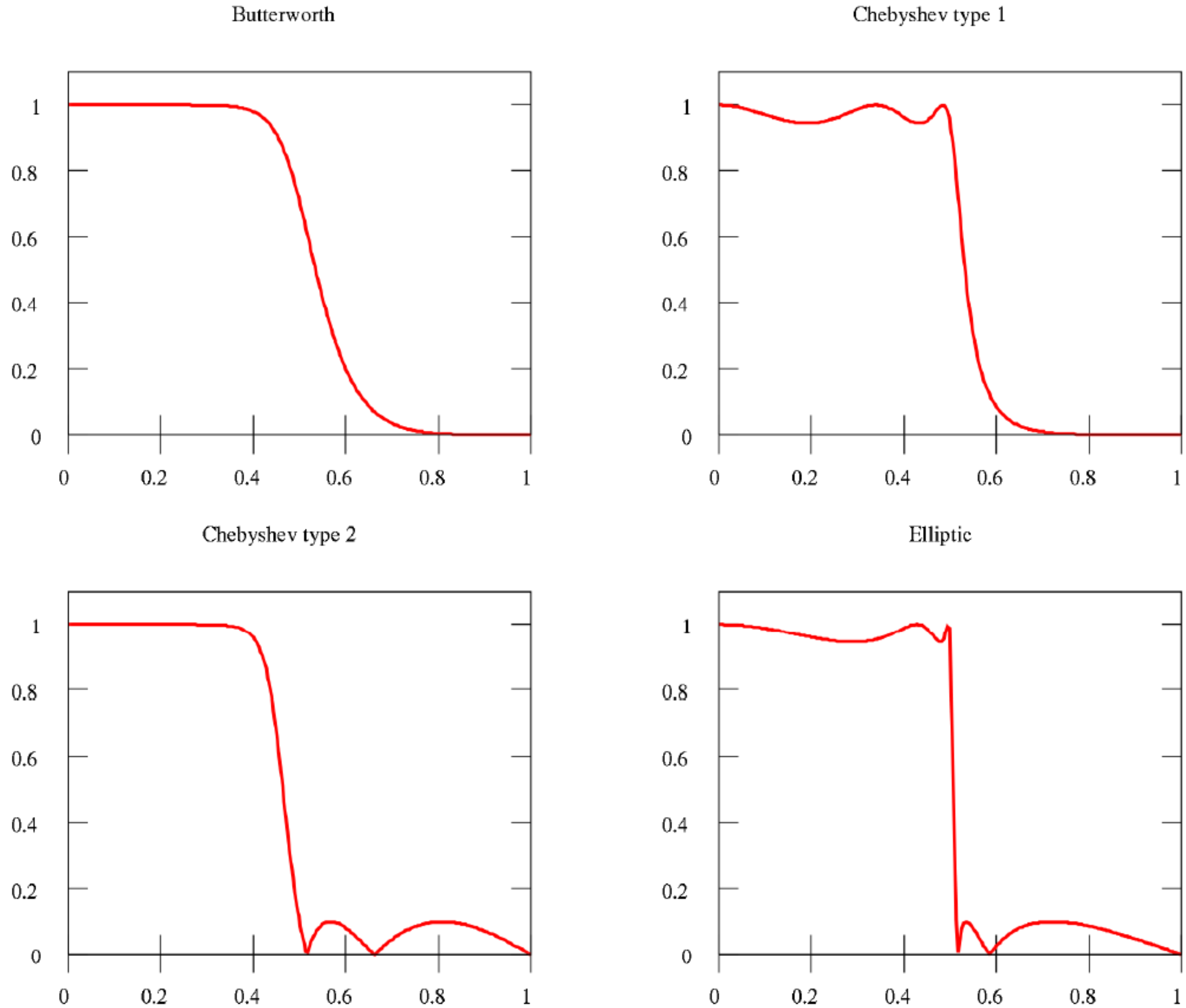
Bessel filters have little to no overshoot in the time domain (e.g. impulse) response and group delay. This gives them better transient response than other filters, however, the frequency selectivity is poor. This means that they have attenuation bands that are too gently sloped to be of much use in a loudspeaker crossover.

On the other end of the spectrum, Chebyshev all-pole filters (also known as type I Chebyshev) can have much faster roll-off than Butterworth filters for the same filter order, but have passband ripple. The amount of passband ripple is an additional parameter in the filter specification. If the passband ripple is low, e.g. less than 1 dB, then it is likely less than the fluctuations in the driver frequency response and is not of much concern. When passband ripple is reduced to zero, the filter becomes identical to the Butterworth filter.

Elliptic filters and Chebyshev (type II) filters are somewhat different than any of the types mentioned so far in that they do not always have continuous attenuation in the stop band. Chebyshev (type II) filters have a monotonic passband (without ripple) and enter the stop band like the Butterworth filter. The attenuation increase more rapidly, however, due to zeros in the stop band. The stopband response increases and falls back one or more times depending on the order of the filter. If the order of the filter is odd, there is a 6dB ultimate rolloff rate in the stopband. For even order the attenuation in the stopband levels off and stays constant. Elliptic filters have ripple in the passband like the Chebyshev type I filters and the stopband consists of one or more zeros like the Chebyshev type II filter. By allowing ripple in both the passband and stopband, the elliptic filter has the highest attenuation rate of all the filters mentioned here, for a given filter order.

Typically the Chebyshev Type II and Elliptic filters are more complex as a result of the extra parameters needed to describe the filters. More importantly, when Bessel, Chebyshev, and Elliptic filters are paired with the complimentary filter (e.g. HP with LP) having the same specifications, the HP and LP pair do not sum to a flat frequency response, however, by modifying the filters nearly flat frequency response can be obtained. Higher order Chebyshev and Elliptic filters can contain a very high Q (e.g. $Q > 5$) stage that will cause an undesirable peak in the group delay. This is why filters of modest order (typically 6th order or less) are used in loudspeaker crossover, regardless of the type of filter used.

A comparison of some of the filter types mentioned above is given in the following figure:



Above: A comparison of several different types of filters.

Frequency Response Information for ACD

Getting high quality and accurate measurement data for your loudspeaker project is crucial for getting an accurate prediction by the ACD model of the loudspeaker frequency response. This section reviews some concepts about loudspeaker measurements.

About the Need for Magnitude and Phase Information

While most people know that you want to control the frequency response amplitude (e.g. the SPL) with a loudspeaker crossover, fewer people understand how phase and phase alignment (the relative phase angle between drivers) influences the loudspeaker response. The phase response is influenced by the phase response of the driver, the phase response of the crossover filters, and the distance from the driver to the listening position. Once the physical layout of the drivers has been established, the crossover filters must be designed so that the phases of the drivers around the crossover point are sufficiently similar. As a result, accurately measuring the relative acoustic delay and the minimum phase response for each driver is very important for high-performance crossover development.

Without proper measurements and modeling of phase, the frequency response can be less than desirable. Poor phase alignment within the crossover region can lead to significant amplitude (frequency response) irregularities. It is important to carefully look at the relative phase angle between drivers around the crossover point (the "crossover region"). Maintaining the same rate of change of the phase for both drivers is also important (this is referred to as "phase tracking"). High Performance crossovers should have a relative phase angle that is no more than 90 degrees throughout the crossover region.

ACD allows for the user to see the phase of the driver, including the effects of acoustic delay and the crossover filters applied to the driver as part of the crossover. Phase is shown in the Phase Response plot in the driver's response spreadsheet. Using the "TWO-DRIVER PHASE TRACKING" extension for ACD, the user can easily see the relative phase angle and adjust the crossover accordingly.

About the FRD file format

There is a standard format for magnitude and phase information known as the FRD format, so named for the Frequency Response Data Consortium, the original developers of this specification. This file format is exported by several frequency measurement software programs such as ARTA.

The FRD file consists of three columns of data: frequency in Hertz, magnitude in dB, and phase in degrees. For more information, see: http://www.pvconsultants.com/audio/frdis.htm When importing FRD files in to the ACD tools, no header or comment lines should be included in the worksheet cells, only data.

What measurements should be used in the ACD tools?

For the Active Crossover Designer (ACD) to function properly a frequency response data file must be supplied for each driver, in the FRD format, preferably the minimum phase response. The driver frequency response can be obtained using one of a number of measurement packages, and saved in the FRD format. This is not the only source of FRD data, however. There are ways to estimate the driver response from manufacturer data or by using a box model (for lower frequency only), however, this is typically prone to error and the correct SPL level is not known. It's best to obtain a measurement system, and take actual measurements of the drivers installed in the loudspeaker cabinet, using gated or near field measurements.

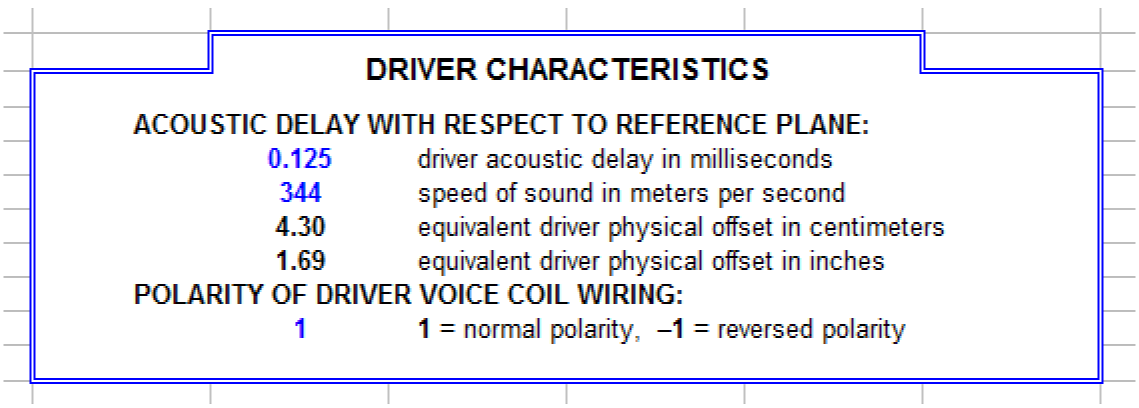
Non-minimum-phase measurements, or any frequency response data set, can be converted to the minimum phase response using the Hibert-Bode transform. Jeff Bagby's Response Modeler of the Frequency Response Blender can process FRD files into the minimum phase response. Some measurement software such as ARTA make the minimum phase form of the measured driver response available for export directly.

Determining the Acoustic Delay for a Driver

There is an extremely simple and elegant procedure for determining the relative acoustic offset/delay of each driver. This information is crucial for establishing the driver phase at the listening position for each driver. For each pair of drivers, take three separate measurements with the microphone in the same position and the gain unchanged: #1: the unfiltered frequency response of driver 1, #2: the unfiltered frequency response of driver 2, and #3: the frequency response with both driver 1 and driver 2 reproducing the same signal at the same time. For an example of how this process is carried out, see the ACD tutorial.

Once the three measurements have been taken, the acoustic delay can be accurately determined using the following method.

- Load the FRD files obtained in #1 and #2 above into their respective driver response spreadsheet.
- For each driver, make sure that all filters are turned off, the driver polarity is set to normal, and all delays are set to zero.
- In the Reference Response worksheet of the System response spreadsheet, paste the FRD file for the measurement taken when both drivers were simultaneously reproducing the same signal.
- When the spreadsheets are recalculated, the system response will be the sum of the individually measured driver responses. This will be overlaid with the reference response in the System Response spreadsheet frequency response plot.
- Return to the driver response spreadsheet for the driver that has a longer pathlength to the microphone (e.g. the woofer) and adjust the value for its acoustic offset until the system



response and the acoustic response overlap as much as possible. The delay should be entered as the parameter "driver acoustic delay in milliseconds" in the "DRIVER CHARACTERISTICS" section of the driver response spreadsheet, shown below:

- Once the acoustic delay is determined it should not be changed during subsequent crossover development.

The frequency response in the case where two or more drivers were operating concurrently typically has one of more notches where the wavefronts destructively interfered and regions where the responses added in phase. These interference patterns are sensitive probes of the exact acoustic offset. The value for the acoustic offset determined in this manner should be retained for the remainder of the crossover development.

For 3-way loudspeakers, it's possible to take three individual driver measurements and a single "all drivers on at the same time" measurement. Making pairwise "both drivers on" measurements would likely be more accurate, however.

Frequency Response Measurements for High and Low Frequency Regimes

Measurement techniques naturally fall into to two frequency regimes: those that are useful at "high" frequencies above about 500 Hz, and those that are useful for "low" frequencies. Because it is typically not practical to make full-range measurements using any single technique, separate measurements of the low and high frequency response (or sometimes approximation via modeling) are “stitched together” to obtain the full range, “wide-band” data. Since accurate measurements are needed to represent the loudspeaker, a short review follows.

High Frequency Regime: Gated Impulse Measurements

Measurements using gated MLS or swept sine signals are convenient for obtaining a quasi-anechoic frequency response measurement that is not influenced by the room in which the measurements are conducted. It is essential to have this capability in order to obtain accurate measurements in the high frequency regime. Measurement software such as ARTA and a calibrated microphone are reasonably inexpensive, and make such measurements routine. These techniques have more or less become the standard for measuring the response in the high frequency regime.

Low Frequency Regime: Combining Modeling and Measurement

At lower frequencies, one can measure the response or use modeling programs to obtain the response to reasonable accuracy.

For lower frequencies, the measurement task is complicated by several factors that make gated measurements impractical unless the space in which the measurement is made very is large (e.g. a gymnasium or the measurement is made outdoors), and the loudspeaker is positioned far away from buildings and walls. Without gating, reflections from nearby surfaces cause interference peaks and nulls that are dependent on where the measurement is taken. In very large spaces, “free-field” measurement to very low frequency can be carried out successfully because they only contain the reflection from the ground plane, which is often desirable to include in the low frequency measurement. Unfortunately, bringing both the speaker and all the measuring equipment to a large field to take measurement is not typically possible for most hobbyists. One way to get around this problem is to take “near field” measurements.

Nearfield Measurements

Nearfield measurements are taken with the microphone positioned as close to the driver surface as possible, usually only a few millimeters away and centered on the dustcap. Nearfield measurements have an upper frequency limit above which the nearfield response is no longer representative of the free-field response, however, this limit is several hundred Hertz – enough to overlap with the “high frequency” measurements. Unfortunately, nearfield measurements do not accurately capture the “baffle step” region of the response, which is critical for accurately describing the frequency response in the midbass region at the listening position. Near field measurements are dominated by the direct sound from the driver such that room reflections are so low in relative amplitude as to make them insignificant. This makes it possible to process the entire impulse response in to the frequency response, which will then be accurate to very low frequency. For some references on how to correctly obtain and extrapolate near field measurements to the far field, see “Frequency Response Measurements:” in the References and Further Reading section.

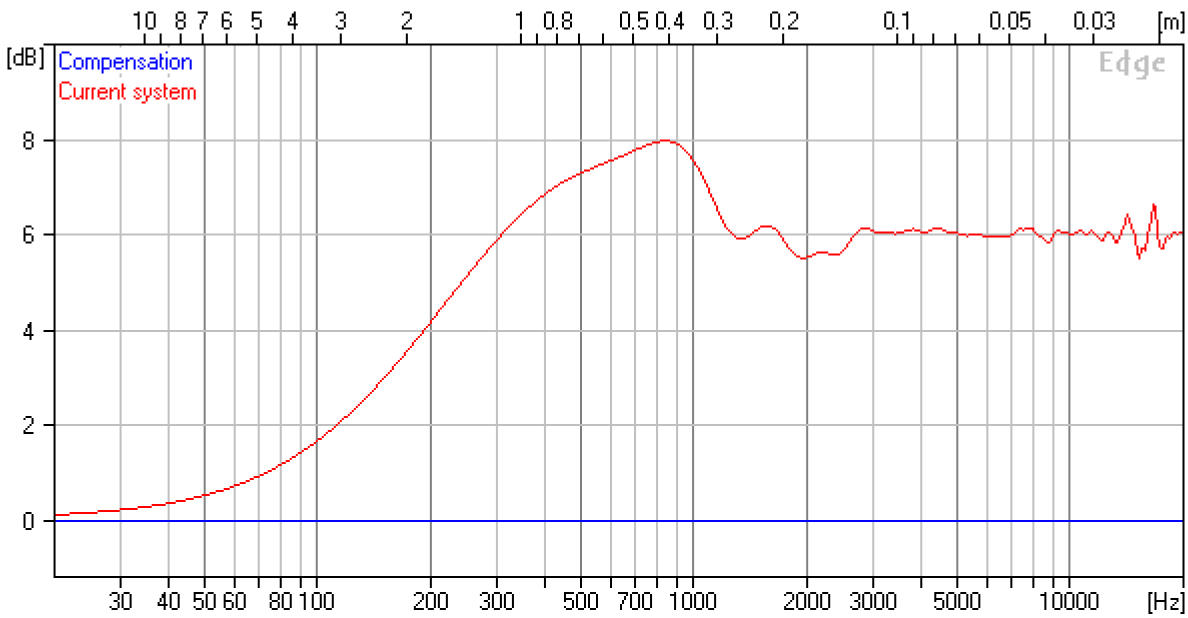
Modeling the Low Frequency Response

Rather than measuring the low frequency response, a good facsimile can often be obtained using “box model” software. These calculate the loudspeaker response from a set of parameters that are used to describe the mechanical and electrical behavior of low frequency drivers (Thiele-Small parameters). Box models can accurately represent the driver response at low frequencies where the driver is operating as a piston, when given accurate Thiele-Small parameter data. Since there can be significant differences from one driver to the next, it is best to measure the T-S parameters of the actual driver first, and use the measured parameters to generate the response with a box model.

Like nearfield measurements, the box model does not include the “baffle step” transition, and this must be added back into the low frequency data before proceeding.

Modeling the "Baffle Step" Transition

The baffle step transition can be characterized by a 3-6dB loss in output that occurs below 300 – 900Hz, often in conjunction with a "hump" of 1-3dB around 700 – 1500 Hz. An example of



the baffle step is shown below (modeled with The Edge):

The baffle step may be abrupt or smooth, and may include some small oscillations above 1000 Hz. The exact shape of this response magnitude transition depends on the size and shape of the front baffle of the loudspeaker and is due to interference from sound waves that diffract when they encounter the change in acoustic space at the edge of the front baffle.

Unfortunately, the lower portion of the baffle step transition occurs in about the same part of the frequency spectrum where the high frequency measurements are no longer accurate. Since low frequency measurements do not typically capture the transition well, modeling of the baffle step is a common way to include it in the overall "wide band" driver response. Luckily, several free software programs exist for this purpose. The Baffle Diffraction and Boundary Simulator by Jeff Bagby is an excellent tool for modeling the response of a loudspeaker driver including the reflections from the floor, rear wall, and a side wall. Other programs include the Baffle Diffraction and Boundary Simulator, The Edge, and Basta!. See the "Software" section for links to web pages for these programs.

Combining High and Low Frequency data with the Baffle Step Model

The SPL data for the high and low frequency regimes must be combined, and the baffle step transition added, to obtain wide-band frequency response data that accurately represents the driver's response in the loudspeaker. In addition, new phase data must be generated for this combined response. These tasks are performed with software tools designed for this purpose.

The typical approach is to add the baffle step transition response to the low frequency nearfield measurement or T-S model data. Next, this data is spliced to or blended with the high frequency gated impulse measurement data to obtain the wide-band SPL data. Finally, this data is mathematically processed to obtain the corresponding minimum phase data.

This can all sound very complicated, but luckily some handy software can make this a routine process that becomes second nature. The "FRD Response Blender and Minimum

Phase Extractor" by Charlie Laub", the "Frequency Response Combiner" from the Frequency Response Consortium, and the "Frequency Response Modeler" by Jeff Bagby, are software tools that allow the user to combine responses and process the resulting SPL data to obtain the minimum phase response. See the "Software" section for links to web pages for these programs.

It is beyond the scope of this text to describe how to use each software package. Please consult the manuals and instructions that accompany them for more information. There are several user forums dedicated to loudspeaker building where questions can be posted.

Measurement Gear and Tools – What Do I Need?

Some tools that are useful for measuring the frequency response of drivers include:

- A calibrated microphone such as the Dayton EMM-6 available from Parts-Express or a Behringer ECM-8000 calibrated by Cross Spectrum Labs of Springfield Massachusetts, USA.
- Microphone preamp with phantom power supply and computer interface (e.g. USB, Firewire)
- Measurement software such as ARTA, OmniMic, or Room EQ Wizard (REW)

How to Import FRD data into the ACD tools

The ACD tools are designed to work with measured response data, supplied as FRD files. By some means, the data must be entered into ACD worksheets designed to hold the FRD data for each driver, etc. The following is an overview of how to import these text files:

TAB DELIMITED DATA FILES

The import is easiest when the FRD data files consist of TAB delimited columns (e.g. FRD files generated by ARTA). The FRD file can then be opened in a text editor, all lines containing data copied, and then pasted directly into the worksheet starting in cell A1. Make sure that comment/header lines (if any) are omitted.

SPACE DELIMITED DATA FILES

If the FRD file is not TAB delimited, copy the data lines (omitting header lines), paste the data starting at cell A1 of the worksheet, and then use the Excel command "Data > Text to Columns" to covert the resulting single column to three columns. This is often the easiest procedure to use when the data file is delimited by spaces.

DATA FILES WITH OTHER FORMATS

The most robust way is to use the data import functionality built into Excel. Choose the menu command "Data > Import External Data" and follow the steps required to import the file correctly. For help on using this functionality correctly, check Excel's help or look up information online. This has the disadvantage that the data import query is saved along with the spreadsheet and may need to be re-run each time the spreadsheet is opened.

THE ACD FILE IMPORT/EXPORT EXTENSION

In addition to manual data import methods explained above, an extension for the ACD tools (that uses Macros, available for Excel only) is available that makes file import and export easy. See the ACD website for more information.

Tips for Avoiding Problems

ACD is fully open and editable by design. You can change and modify the look of plots and even customize calculations if you do desire. Because of this, you can also (accidentally) change the calculations or other behavior of the tools so that they no longer work properly. By following a few simple rules, these kind of accidents can be avoided.

The following is a list of things to avoid doing when using the ACD tools, and some explanation as to why these are “bad”:

Don’t Paste, Paste Values

NEVER use “cut” (control-x) when you want to copy information from one place to another, such as between cells in the Filter and EQ tables.

When moving data from one cell to another **AVOID** the standard paste operation. **INSTEAD DO THIS:** after the “copy” operation choose from the EDIT menu the item “Paste > Paste Special > Values” to paste only the value(s) of the copied cell(s) into the target cell(s). In the later versions of Excel this command is done thru the Home tab of the ribbon where the Paste icon is located.

WHY? Excel is a program that tries to be helpful by moving references for cells along with them when you use copy+paste or cut+paste. Moving data from one place to another can also move cell references around and this may completely re-arrange the calculations or corrupt them altogether. When you copy from other sources (other spreadsheets or other applications with formatting) it is possible to overwrite the intended formatting in the ACD tools. Get in the habit of never using paste (control+v) but instead always using “paste special > values”.

Overcoming Data Import and Export Problems

COMMON PROBLEMS IMPORTING FRD DATA FILES INTO Excel:

Excel has reasonably good built in text file importing functionality. Data can be pasted in from the clipboard or imported using Excel’s text file import commands. If you are not familiar with these, please use a search engine to get help from the web.

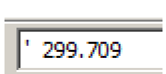
For Excel, an ACD the **FRD file import-export** extension uses VBA code to assist with data file importing and exporting when using ACD. IT can remove leading space and handle a wide range of delimiters. It makes routine and repeated importing of FRD data very easy.

Here are some things to check when there are still problems:

1. If the data in the FRD file has leading spaces before numbers, delete the spaces before importing. If this problem is occurring in a range of cells that has already been imported into Excel, use the menu item “Data > Text to Columns” to convert the contents of a column (process one column at a time) from explicit text back to a number.
2. Were header lines imported along with the data, causing problems? Delete them or consider using the ACD file import/export extension.

COMMON PROBLEMS IMPORTING FRD DATA FILES INTO Open Office Calc:

Unlike Excel, Open Office does not include built-in import capability that can import a text file into a range of cells within an existing worksheet. Also, the ACD extension for file import/export is not available for Open Office (the macros are not compatible with the OO basic language). The only remaining option for importing is to open the FRD file in a text editor, make any adjustments needed, then copy the data and paste it into the desired range of cells in OO Calc. When the data is separated by a tab or by a single space into columns, OO Calc is able to recognize that there are three columns of numbers to import. Additional spaces can cause Calc



At left: example of Calc adding an apostrophe to an imported number that had leading spaces in the FRD file

to think that text is being imported, and may add a leading apostrophe character before each number. This appears in the formula bar like this:
The cell contents are internally recognized as text instead of a number, some formulas will fail, and the data does not show up in various plots. Remove the extra spaces using a text editor with a Search/Replace function (search for the space character and replace it with nothing) or use the menu item “Data > Text to Columns” to convert the contents of a column (process one column at a time) back to a number. Tab separated data without additional spaces seems to be most reliable format to use when copying and pasting data into Calc.

References and Further Reading

Waves, Complex Numbers, and Phasors:

http://en.wikipedia.org/wiki/Sine_waves
http://en.wikipedia.org/wiki/Phasor_%28sine_waves%29#Definition
www.ece.rice.edu/~daniel/262/pdf/cxnums.pdf

Filter Transfer Functions:

"RC Active Filter Handbook" by F.W. Stephenson (Wiley). Chapters 1-3 are an excellent reference on transfer functions and filters, especially Table 3.1

"Analog Filter Design" by M.E. Van Valkenburg (Holt, Rinehart, and Winston, Inc.). Chapters 4-8, 10, 13 and 18 contain very useful information about filters and filter design.

A web site where you can enter transfer function coefficients and to see the resulting frequency and phase response. There is also a nice explanation of the transfer function in the first few paragraphs: http://www.williamsonic.com/BodeNyquist/index.html

General info on filters and transfer functions:
http://www.national.com/an/AN/AN-779.pdf
http://en.wikipedia.org/wiki/Transfer_function

Poles and Zeros in the Complex Plane:

This must read web page has great plots (many animated) showing what the complex plane looks like for several types of filters:
http://www.timstinchcombe.co.uk/index.php?pge=poles

Butterworth filter transfer functions, showing plots of the complex surface and the poles of the filter: http://www.crbond.com/filters.htm

2-D plots of the complex plane. Interface lets you place poles and zeros and then see the resulting magnitude and phase: http://web.mit.edu/6.302/www/pz/

Frequency Response Measurements:

"Low-Frequency Loudspeaker Assessment by Nearfield Sound-Pressure Measurement" D.B. Keele, J. Audio Eng. Soc., (April 1974):
http://www.xlrtechs.com/dbkeele.com/PDF/Keele%20(1974-04%20AES%20Published)%20-%20Nearfield%20Paper.pdf

"Nearfield Measurement with Multiple Drivers and Port"
http://www.klippel.de/fileadmin/klippel/Files/Know_How/Application_Notes/AN_38_Nearfield_Measurement_with_Multiple_Drivers_%20and_Ports%282%29.pdf

"Merging Near and Farfield Measurements"
http://www.klippel.de/fileadmin/klippel/Files/Know_How/Application_Notes/AN_39_Merging_Near_and_Farfield_Measurements.pdf

Merging Near and Farfield Measurements using ARTA:
http://www.artalabs.hr/AppNotes/AP4_FreeField-Rev03eng.pdf

How to Design Loudspeakers without Performing Measurements - "Simple Loudspeaker Design":
Presentation / Overview (PDF)
Zipped directory containing all the necessary software applications and demo files

Digital IIR Filters:

Digital (IIR) Filter Design: http://www.mikroe.com/eng/chapters/view/73/chapter-3-iir-filters/
Wikipedia entry for the bilinear transform: http://en.wikipedia.org/wiki/Bilinear_transform

Tutorials: How to Find the Acoustic Delay Using the Three Measurement Method

Tutorial by Jeff Bagby: https://www.box.com/s/ouxjjsx0m8bs00cil5iq
Tutorial by Dave Ralf: http://www.speakerdesign.net/sbarticle.html

Software

Frequency Response Measurement Software

Room EQ Wizard – free Java application for measuring room acoustics and analyzing room and loudspeaker responses from the Home Theater Shack.
ARTA - a free and full featured frequency response measurement, distortion testing, and loudspeaker impedance measurement suite from ARTALABS.
OmniMic - available from Parts-Express and includes a microphone and software.

Combining FRD Files and Minimum Phase Extraction

Frequency Response Modeler - Excel application for creating and manipulating FRD files in many ways. Can calculate the minimum phase from SPL data using the Hilbert transform.
Frequency Response Combiner - Excel utility for adding, subtracting, splicing, delaying, correcting, smoothing, adding filters, bending, changing sampling size, extending or reducing the bandwidth, and generally pre-processing FRD (Frequency Response Data) files. Includes a wide band Minimum Phase Extraction tool.
FRD Response Blender and Minimum Phase Extractor – Excel utility for combining FRD files, creating response tails, and extracting the minimum phase response.

Diffraction Modeling

Baffle Diffraction and Boundary Simulator - Excel application for modeling the effects of the baffle and room responses
Baffle Diffraction Simulator - tool for predicting Baffle Diffraction Response and Baffle Step Gain in Closed Box Systems
The Edge - a simulator for the "baffle step"

Thiele-Small “Box Modeler” Frequency Response Modeling Programs

Woofer Box and Circuit Designer - Excel application for designing subwoofers and active circuits (e.g. Linkwitz Transform)
Unibox - Excel Program for modeling the response of closed, vented, passive radiator and bandpass loudspeaker boxes
Basta! - a computer program for simulation of loudspeaker systems. Basta! can simulate open baffles, closed boxes, vented boxes (“bass reflex”) and 1- and 2-ported bandpass

systems. For ported enclosures, pipe resonances in the vent can be simulated. Instead of a vent, the ported enclosures can have a passive radiator. Basta! also includes simulation of baffle step, lossy voice coil inductance, multiple and isobaric drivers. Basta! can also simulate Active and passive crossover filters, most passive crossover networks and ACE-bass (developed by K.E. Stahl). Basta! also has a room gain approximation and can add a Linkwitz transform to the response. Commercial program, with a free version that has limited functionality.

Passive Crossover Design

Passive Crossover Designer – excellent and free Excel application for designing passive crossovers