“Port Duck with Test Sauce”

A Recipe for Great Tasting Audio

by
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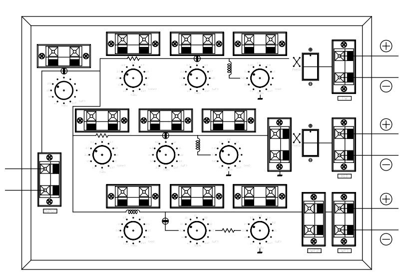
A COMPANION GUIDE
for

The Virtual CROSSOVER BOX™
MODEL VCB-100



Congratulations!

Congratulations on your purchase of the Vidsonix VCB-100 Virtual Crossover Box. Please read through this manual for some insight into speaker voicing and crossover design. Please read the VCB-100 white sheet for specifications - downloadable from www.vidsonix.com.

TABLE OF CONTENTS

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Introduction

Welcome to the world of designing sound. By purchasing the VCB-100 virtual crossover box and now reading this manual, you probably fall into one of the following categories: 1) You or your company is involved in designing or manufacturing audio speakers 2) You are involved in some sort of installation project that requires crossover tuning 3) You are new to the audio field and would like to learn more about it, maybe even considering a career in some sort of audio design or 4) You are an audio enthusiast looking to learn more or improve a speaker system.

If you are in the business and have been designing audio for a while, chances are you know exactly what the VCB-100 can do for you and how you want to use it. This manual may not yield much for you in the way of new knowledge, but it may be beneficial to you as a review of the basics.

If you are new to audio, you will most likely find this manual extremely helpful. However, beyond the basics, it is by no means very technical, and it does not intend to go over things that already have been satisfactorily described in other texts. All of the listings in the Reference section are recommended reading to further your audio knowledge.

In advance, I wish you all the luck and success in your audio field endeavors and thank you for purchasing Vidsonix products.

What's a Crossover?

Amazingly, the ways in which sound is reproduced by the typical speaker transducer, using a moving voice coil and piston induced by an alternating current, hasn't changed much over the last 60 years. Sure, there have been improvements and even some radical sound reproducing concepts, but none have proven themselves as being as cost effective and efficient as your typical speaker. Maybe it will be your breakthrough that changes that someday, but until then, "speakers is speakers!"

Let's start with the range of sound that our ears can hear. The range of human hearing is typically 20 Hertz (cycles per second) to 20,000 (20K) Hz. Because most speakers (drivers) cannot reproduce this entire range, multiple speakers must be used in a system, each intended to reproduce a specific range of frequencies to combine and give the complete spectrum.

Typically, a larger driver, a woofer, is used to create the lower frequencies, a mid-sized driver is used to create the mid-range frequencies, and the smallest driver, a tweeter, is used to create the higher frequencies.   For a specific driver to perform smoothly and efficiently, it should receive only the frequencies it is capable of reproducing. Frequencies above the intended range of the driver can cause cone break-up, causing distortion. That is why a 8" or 10" 2-way system must have a tweeter with a response that overlaps the woofer. If there is not there will either be a massive hole in the mid-range response or extensive distortion and wide tolerances in the upper response of the woofer to compensate. In addition, lower frequencies can damage or blow the mid-range or tweeter drivers. A crossover, sometimes referred to as a network, is needed for these reasons.

The crossover circuit divides the audio spectrum into two or more frequency bands, directing the frequencies to the appropriate driver. This is accomplished by using high-pass and low-pass filters (see Fig. 1). A high-pass filter lets frequencies higher than a certain cut-off frequency pass through, while blocking out the frequencies below that frequency. A low-pass filter does just the opposite. It lets frequencies lower than a certain cut-off frequency pass through the circuit while blocking frequencies above it. These filters may also be combined to create a band-pass filter.

The rate at which the frequencies are cut-off is called the slope of the crossover filter. You may obtain higher precision in your cut-off filter by increasing the slope (-6dB, -12dB ,-18dB, etc.), but this requires using more components (See Fig. 1). Note that they are called 1st Order, 2nd Order, etc. because of the complexity of the equation required to solve for the values.

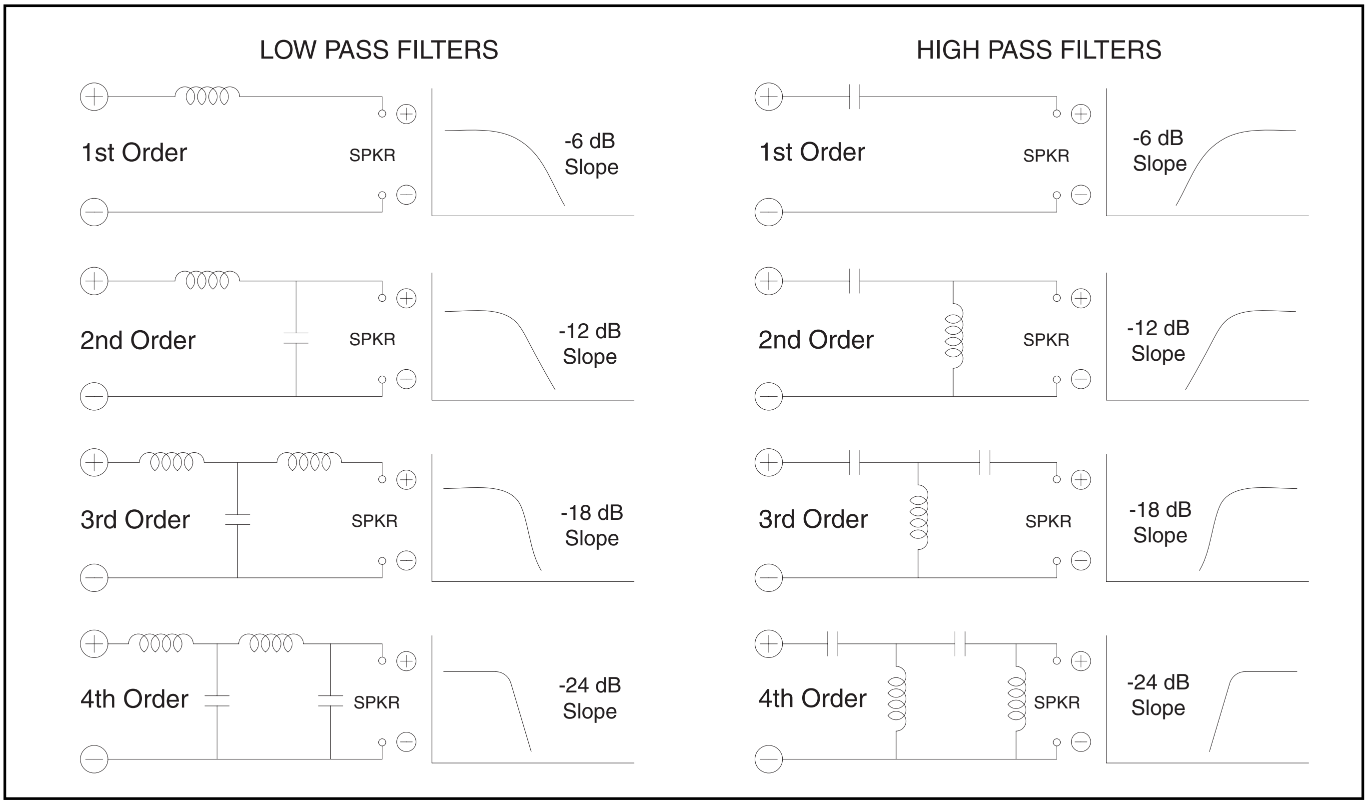


Figure 1: Different Filter Circuits

Types of Crossovers

First, we must make a distinction between the two types of crossovers: passive and active. An active crossover typically uses solid-state components such as transistors and integrated circuits, and thus is one that requires power to function. These are inherently much more expensive than passive crossovers. However, they do have some performance benefits such as better efficiency and distortion ratings, because the signal is processed before it is amplified. Bi- and tri-amplified systems typically will use active crossovers. One disadvantage with some active crossovers is that they use DSP, which creates an inherent 25-30ms delay. This is not good for live performances.

A passive crossover, on the other hand requires no power, using only passive components such as inductors, resistors, and capacitors. This is the typical type of crossover placed in a home stereo speaker where one amp is used to drive the speaker system. Passive crossovers are also used in many custom car speaker system installations. The Virtual Crossover Box is a passive crossover system.

Passive Crossover Component Types

An *inductor*, sometimes referred to as a "coil" or "choke", is a wound coil of wire, which may or may not have one of several types of iron or ferrite cores. Inductors without a core are called air core inductors. As the frequency of the signal passing though an inductor increases, the inductive reactance (a component of the impedance) also increases. This allows the inductor to be used as a filter that passes low frequencies but "chokes", or blocks, higher ones. Air core inductors have lower distortion and higher power handling capabilities, but tend to have a higher DC resistance. Iron cores use less wire, but they heat up easier and become non-linear, or "break up" - their impedance starts to change when over heated.

Capacitors perform exactly opposite to an inductor. As the frequency decreases, the capacitive reactance increases, allowing the capacitor to be used as a filter that passes high frequencies and filters out lower ones. As with all components, there are different types, tolerances, and quality factors of capacitors. One such factor is the DF factor, or dissipation factor, which indicates how much of what comes in is dissipated as heat (an efficiency factor). In addition, note that audio capacitors are non-polarized, since an audio signal is AC current; so you can't use a typical electronics-type polarized capacitor in a speaker crossover.

The best sounding (passes higher frequencies better) and most expensive capacitors (for series mid-range and tweeter use) are Mylar and polypropylene (best, very low DF factor) types. However, standard electrolytic capacitors are used in most everyday speaker systems, because of their generally acceptable performance and low cost.

Resistors are components that have nearly zero frequency dependence to them, and therefore their total impedance is pure resistance. They are good for manipulating the impedance curve of your system and matching impedances to your amp.

Be sure to use the proper power-handling (wattage) resistor in your crossover. However, just because you have 100W system does not mean you need to use 100W resistors (very expensive). For example, in your 100W 3-way system, in reality 35W may be going to the woofer, 25W may be going to the mid-range, and 10W to the tweeter. Where's the extra 30W? It is eaten up by the crossover, dissipated as heat or electromagnetic energy. Depending on the number of components in the crossover, using 10W resistors in each section should be just fine. Power testing and measuring with a multi-meter can help you double-check your estimates.

It must be noted here that all inductors and capacitors have inductance, capacitance, and resistance properties to them. For example, an inductor may have a resistance of up to 2 ohms due to the resistance of the copper wire (use heavier gauge wire to lower the resistance). You may factor these effects into your equations or compensate for them using other components. The equations used in our Pocket Calculator and most texts accept these effects as minimal and do not include them, so you must factor them in yourself if so inclined.

What Order or Slope Should I use?

Each text will give recommendations or opinions (see References) as to which order of network works best in certain cases or with certain drivers. However, there are no hard rules that say you cannot be creative.

Three items to keep in mind:

1) The crossover you design must be able to withstand the power rating of the system you are creating. This not only means using components that will handle the power, but cutting the frequencies off at the right point for the mid-range and tweeter. A second order network will generally provide better power protection than a first order because of it's steeper slope.

2) You must design or be able to check the phasing of your crossed over system to make sure it is correct. If it is not, there will be a deep notch in the frequency response curve, which you may or may not be able hear instinctively.

3) The higher order crossover you go, the more insertion loss (power dissipated or lost in the crossover circuit) and ringing in the impulse response you get. (This can be a benefit if you are trying to design a speaker that will meet a certain power handling criteria, and the drivers you have been dealt just don't cut it.) In addition, component tolerances tend to add up resulting in tough to maintain final tolerances in production. First (1st) order filters have the ideal transient response. Good transient response means that an impulse of sound arrives at your ears at a single instant in time (see Figure 2). Poor transient response means the sound is smeared, delayed, or repeated at slightly different times.

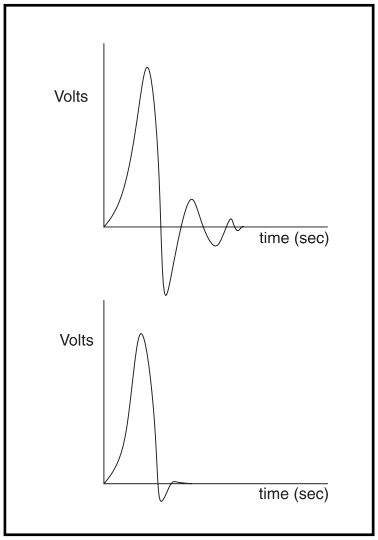


Figure 2: Transient Response

Selecting the crossover point is highly subjective. If you have a driver's specification or response curve printout, you can determine what frequencies are best suited for the driver. However in general, sub-woofers are crossed over at 150 Hz and below, woofers at 500 Hz and below, midranges at between 500 and 8000Hz, and tweeters crossed over at 4000 Hz and above. Remember, you are free to experiment and try what sounds best (what the VCB is best for -> voicing), but always keep in mind your power handling requirements. If you are designing an automotive system, keep in mind that peaks and dips in the response created in the auto enclosure need to be taken into account when choosing a crossover point.

Filter Alignments

Now that you have determined your crossover points, you need to calculate component values. With derived equations, you could theoretically just pick a capacitor value and find it's corresponding inductance value for a particular cut-off frequency (say, on a 2nd order crossover). However, the best corresponding values, or alignments, have been nicely researched for us by really smart guys, based on the "lobing" pattern of the filter. Each filter alignment is named after its originator:

Bessel - Best Time Delay
Butterworth - Flattest amplitude
Chebychev - Fast initial roll-off
Linkwitz-Riley - cascaded 2nd order Butterworth - phase identical everywhere

There are many more alignments than those listed here, although not as widely used in audio. Our Pocket Calculator is based on the Butterworth alignment, which is most widely used in audio. You may calculate the values by hand, by software, or by using our handy "Crossover Filter Design Pocket Reference Tool" (See Figure 3).

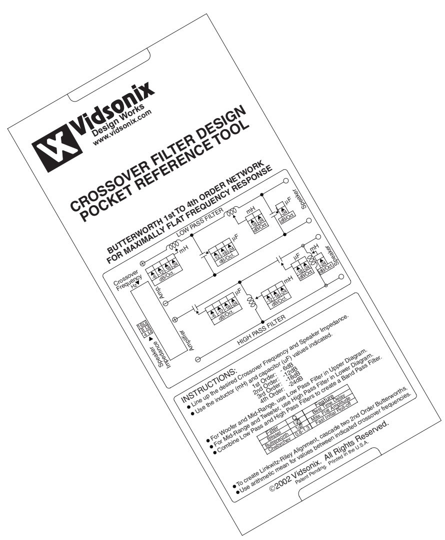


Figure 3: Vidsonix's Crossover Filter Design Pocket Reference Tool.

Correction Filters

There are several useful impedance correcting filters, such as a conjugate impedance network, a series notch filter, and a parallel notch filter. These are covered well in other texts, so we will not go into them here. They are also listed on our "Pocket Reference Tool".

One item to note is to use caution when using these filters. A notch filter may not be a good idea because if driver parameters change or vary in production, your notch filter will not be "notching" its intended frequency and thus causes more harm than good.

Design Goals

So what should your target design sound like or look like on a particular graph? Typically, a good crossover network should leave the frequency response curve relatively uniform to avoid artificial coloration of the music, and resistive losses should be minimized. However, depending on your end user, your desired frequency response may take the shape of one of many variations of a "flat" or uniform response (see Fig. 4). A speaker with an otherwise flat response curve may not sound as good as another with a less than flat due to the fact that there are many other subtle, yet audible, speaker characteristics besides frequency response SPL (sound pressure level).

A good impedance curve is generally smooth with an inductive rise and mechanical resonance, the peak of which should be fairly high (see Fig. 5). If there are bumps, that usually means there is some sort of mechanical anomaly.

Note: Where you place your test microphone with effect the frequency response curve. Some engineers test with the microphone on axis at tweeter level and some engineers test with the microphone on axis between the midrange and tweeter. Of course, the one tested at tweeter level will show higher SPL in the upper frequency spectrum. The standard distance for the microphone is one meter, unless you are purposely conducting some kind of near field measurement.

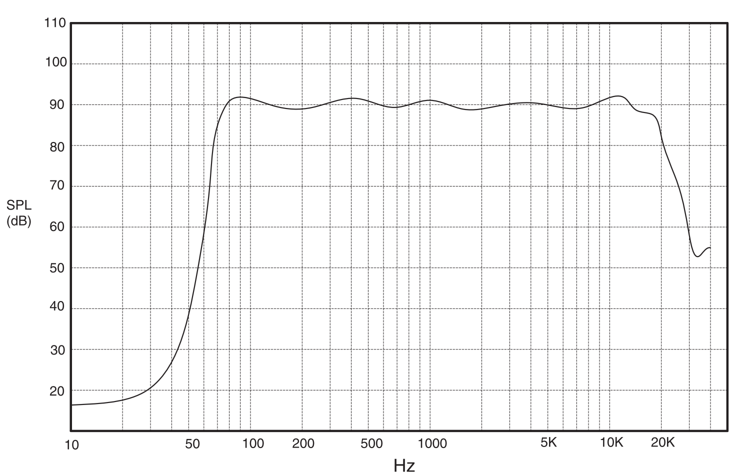


Figure 4: Example of a Flat Response Curve

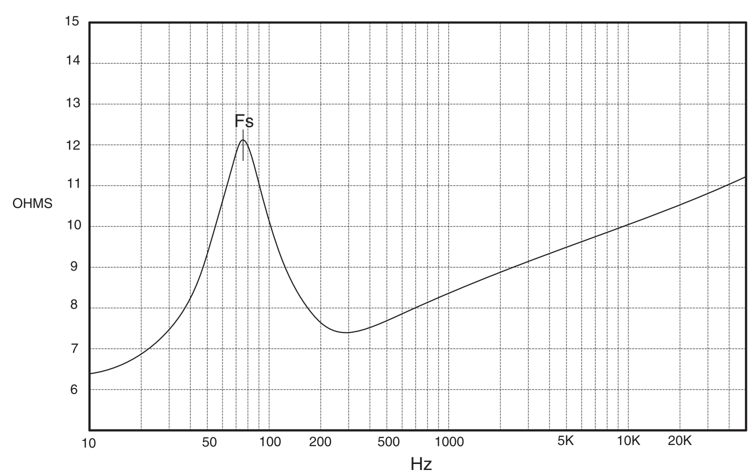


Figure 5: Example of a System Impedance Curve. Note that the mid-frequencies will typically have a different shape than the curve depending on what type of crossover has been added.

Testing Equipment

Good test equipment (hardware and software) is always a big help. Of course, if you are on a limited budget, you can try to do everything with a pencil and calculator (or use our Pocket Crossover Calculator) and listen to your results by ear. The only problem with not having any objective testing equipment is that you may have a substantial dip in the frequency response curve and not realize it. (Then, does it really matter?). If you have the resources, first buy frequency response capture hardware for your computer (see Reference section for a listing of several of these). With this hardware, you will be able to view the frequency response curve of your speaker graphically. With the better ones, you can view phase and any number of different types of data graphs, as well as being able to manipulate them. You will also need a good microphone with this set-up, and the better hardware/software combinations will allow you to import microphone correction files for your particular microphone. Stand alone hardware/testing equipment is also available, with no need for a computer.

You can also purchase software for designing enclosure sizes, port sizes, driver parameters, crossover, room sizes…anything that is time consuming by hand with a pencil and calculator can be done much faster, plus with visual aids, via software.

Design Methodology

There are two extremes to the methods (schools of thought) that audio engineers use. One is the proactive approach -- design it on paper, utilizing all known design data, theories, formula and their related software. This is the smart way to start a project. However, it assumes you have an infinite amount of time to collect all necessary data and work through all of the design calculations. Strictly using this method is not a good idea, since your design on paper does not guarantee you the same results in the real word.

The other approach is the trail and error method. This involves putting together a system and listening or measuring its frequency response output and adjusting it hands-on from there. This is called voicing the speaker. This method, with experience, will give you great insight into the design equations of the previous method. The engineer who uses this method solely by choice does so knowing that his success rate will be less that than that of the thoroughly designed approach. However, many times an engineer is forced into using this method, as in the design example that follows.

Of course, to yield optimum results you must use a balanced combination of these two methods, and in reality it is impossible to successfully complete a design without a degree of each. However, it is typical for an engineer to have a bias towards one or the other.

The Down and Dirty Method (Worst Case)

In a perfect world, you would have an infinite supply of resources and data to help you in your design process. That means having the right hardware and software, having an open budget on the final design pricing, and knowing all of your data up front (i.e. drivers made to your specifications or Thiele-Small parameters). Yet even in a perfect world, as stated before, your theory-produced speaker may not end up sounding like you had theorized!

In the real world of practicality, manufacturability, time constraints, cost constraints, and cosmetic constraints, the world is far less perfect. It is the mature audio engineer that comes to realize this, and embraces this without compromising his or her engineering principles.

Let's take a worst-case example. The factory has an overstock of several types of drivers from overproduction or customers not paying. You've been asked to design a system choosing from these drivers so that the factory can get rid of them and not have to create new tooling for this new system (to save money). Then, they tell you they are looking for a particular type of speaker, giving you the width, height and profile of the cabinet. And, finally, they want it to be the best sounding system they've ever heard (there's that perfect world again).

So what do you do? Start looking for a new job out of frustration? Well, why not make the best of a bad situation:

Let's say they've given you a width of 12" and a height of 40" for the speaker (to go with a television stand they are trying to market). O.K., this tells us that we can't use anything larger than a 10" woofer, unless we were going to design it side-firing, which is not the type of speaker they want. There are two 10" drivers that they have overstock on. Which one to choose? Why, the one with the best Thiele-Small parameters that fit our box volume, right? Well, yes, but not necessarily. Let's say the only midrange they have is a 3" and the only tweeter they have is a 1" soft dome (marketing already decided it needed to be a 3-way system), so these drivers have already been chosen for you. What about cosmetics? Which woofer better suits the midrange and tweeter combination? You may have to decide to go against the grain and choose the woofer that looks better with the other two (hopefully not).

As for the volume of the box, they didn't give you any firm constraints on the depth of the cabinet, so you have some play as the choice of volume. Again, though, your design must fall into "the sweet spot" of having your system look good, so you can't make the box too deep or too shallow so that it looks nasty or tips over. First, base the volume of the box on the woofer parameters you been given, and then make adjustments/compromises based on what is really possible.

Your next two choices may be based on experience, preference, or theory. Should we make the system ported or non-ported? Since we found that the box isn't as large as we would like for our woofer choice, and Marketing prefers "boom" compared to extended bass, we will make it ported. We decide to choose a 3" diameter port at first, since its relative size looks good with the system, and from experience, we shouldn't run into wind noise problems. Now, what about driver spacing (distance apart)? There are many suggested theoretical alignments, and most do not perform as expected on a consistent basis, so stick with what you've learned and have had good luck with. Plus, you always have the constraints of baffle size, construction type, and grille shape/position.

What about bracing, chambering the midrange, damping material, etc.? Marketing has let you know that there is no room in the budget for any of these. O.K. for now…

All right, we've got our initial cabinet/driver specs done, and you just happen to have a fine Sample-Making Department at your disposal. The cabinet's done in just two days!

PHASE I - Voicing Set-up

So, you hook up the drivers into the cabinet using extension wires (see Fig. 6) you've created to go with your Virtual Crossover Box and you've got your measurement system hooked-up, ready to take a frequency response curve of the monster you are about to create.

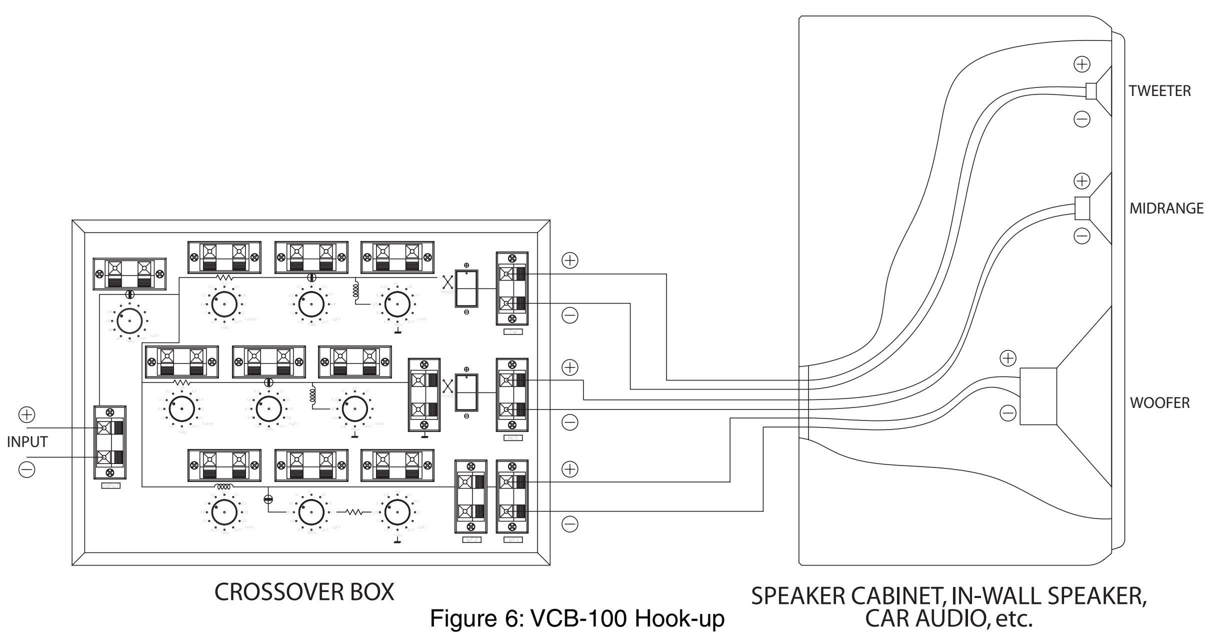


Figure 6: VCB-100 Hook-up

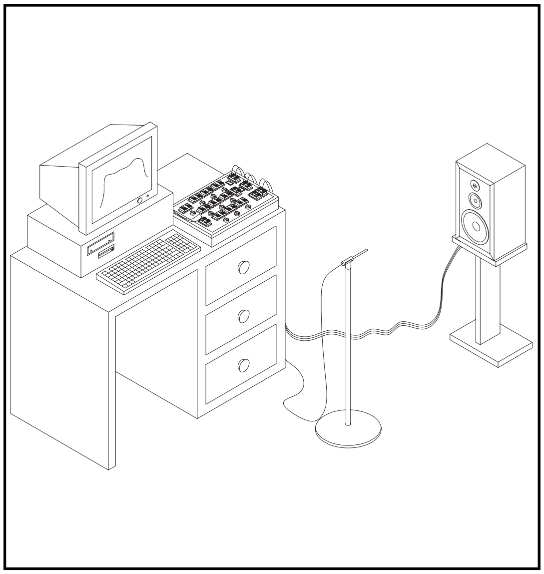


Figure 7a: Computer analyzer set-up.

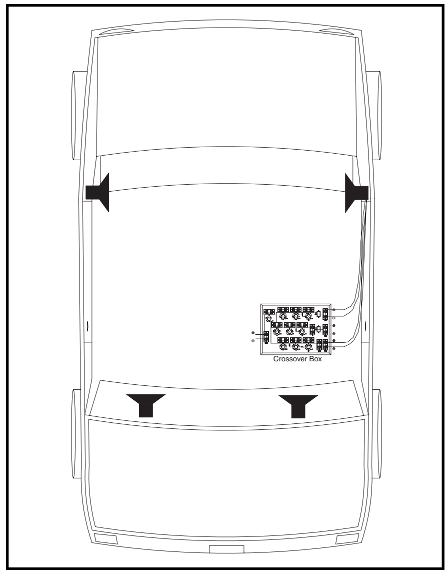


Figure 7b: Car Audio Applications

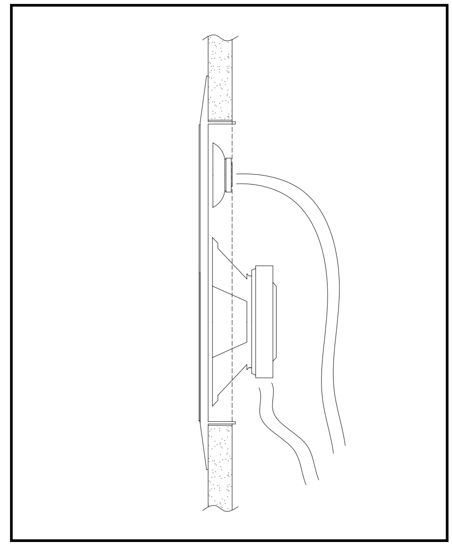


Figure 7c: In-Wall Speaker Voicing. Extension wires may be run inside or outside of wall for testing.

When designing a speaker, we voice the system using one speaker (we will test it in stereo later). The same goes for car audio and in-wall speakers (see Fig. 7). Of course, matching speakers (left and right) should end up using the same crossover. In car audio, the rears may need to be designed and crossed over at different point than the fronts. For this reason, multiple VCB-100 units may be desired for mobile audio testing.

What gauge wire to use? If you use thick cable for your system, you need just as heavy gauge in your crossover and vise versa. In addition, your resistive load will be less with heavier gauges (12 Gauge - very thick, 22 gauge - minimum thickness wire for any audio application of substantial power). For smaller crossovers the larger gauges are just not practical. Plus, the resistive load depends on the length of the wire, so running thick cables in your system (long lengths) and smaller gauges in your crossover actually isn't all that bad. For our budgeted crossover, we will probably end up specifying 20 gauge.

The first thing to do is take a response curve of each driver in the cabinet separately. That means taking the other two drivers out of the circuit either by disconnecting them or putting the corresponding switch to ext(ernal) or open. This will give you an indication of where you would like to place the crossover points (see Fig. 8).

Then, you'll need a starting point for the phasing of your drivers. Start with a network that has nothing on the woofer and just a capacitor (1st order) on the midrange and on the tweeter (say a 3.3 uF cap on each). Take a frequency response curve with the polarity switch of the midrange and tweeter both positive

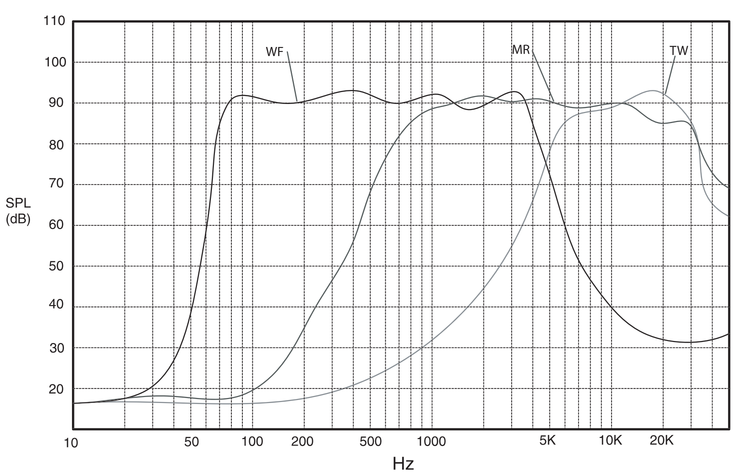


Figure 8: Response Curve of Each Driver on the same graph.

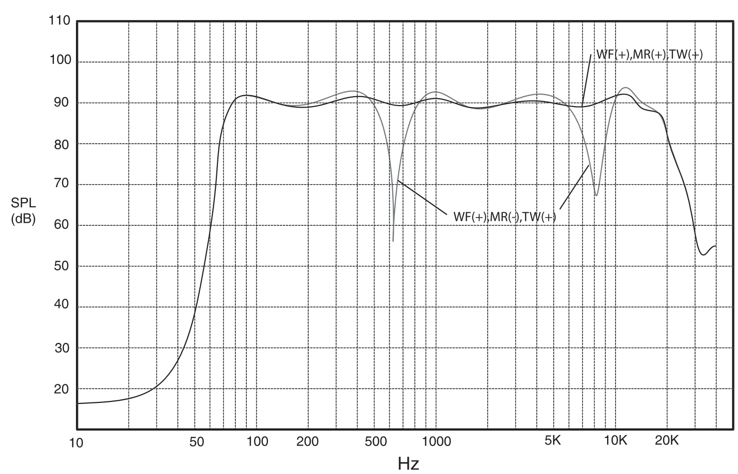


Figure 9: Two different phasing combinations (out of a possible 4) on a 3-way system.

(+, not reversed). Then, take response curves with the phasing as follows: TW(-);MR (+), TW (+);MR (-), and TW(-);MR (-), respectively. Keep in mind that these phasings are all in relation to the woofer, so you don't need to do anything with the woofer itself.

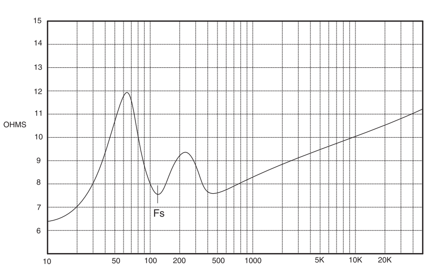
Now, compare the curves (see Fig. 9). Some will have major dips in them, and some will not. Choose the network that created the flattest curve as your starting point. Note that changing the order of the crossover sections (WF, MR, TW), i.e. adding capacitors or inductors, will change the phasing relationship. That is where good computer design programs come in handy (see References). For our quick and dirty method, we have just created a basis from which to start.

This is where the fun begins. From our individual driver curves we took before, we decide to crossover the woofer at 1000Hz, the midrange at 800Hz, and the tweeter at 4000Hz (4K). We could use a band pass on the midrange, giving it an upper crossover frequency, but due to cost constraints we decide not to do that for now. Also due to the cost constraints, we decide to use 1st order filters on each driver (one component each). We check our Pocket Calculator or other table/program for the component values that will give us the crossover frequencies above (using a Butterworth filter alignment for maximally flat response), and dial them in (using the appropriate rotary switch) as close as possible on the VCB. We then take a response curve of our current system.

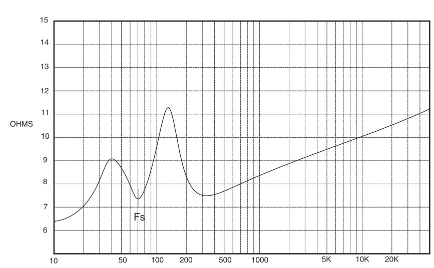
Port Tuning

There is extensive documentation and listed equations for determining the port size and length for a system. We can use these to determine your initial port length. We've only got a 5" length plastic port tube here with a 3" diameter. Let's put it in our system and give it a try.

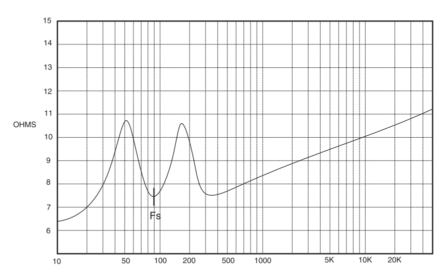
If you look at a typical impedance curve of a speaker, it will have a resonance peak at a particular frequency, as illustrated. You will notice that if you add a port to the system, you will get two resonance peaks. How does this happen? Actually, a good, simplistic way to look at it is that there is still only one peak, but the port effectively puts a dip into the resonance peak, making it look as though there are two. And, depending on the tuning frequency of the port, one "hump" my appear taller than the other or vice versa. One general rule of thumb is that the correct tuning places the port tuning at the box resonance frequency, resulting in equally tall humps. Of course, you can adjust your system, trading boomy-ness (frequency response peak, high Q, too short of port) for extended bass (long frequency response initial slope, low Q, too long of port). See Fig. 10.



Port too short or too large of diameter; tuning too high.



Port too long or too small of diameter; tuning too low.



Port about right; optimum tuning.

Figure 10: Impedance curves on speakers with varying port tube lengths.

If the response is relatively flat or close to what we determined we would like, we are ready to go to Phase II, Listening. If it is not quite to our liking (which is usually the case), we must manipulate the curve by changing crossover point (component) values, or even changing the order of each filter section (add components). Since you can change a value on the VCB in real-time and then take a response curve, you can see if you've gone in the right direction immediately. Learning how to manipulate the frequency curve is an art-form that comes first from reading enough to have the knowledge background and insight (see Reference), and then experiencing these ideas, theories, opinions, and comments yourself by experimenting with different crossovers with the VCB-100.

If you just can't seem to get the frequency response looking the way you want it, you'll have no choice but to change something besides the crossover. Whether that be tuning frequency (port length), adding damping material, making new samples with a new volume or driver spacing, using different drivers, etc., there will be times when you will be forced into situations in which you must let people know what is possible and what is not, and go a different route if necessary.

PHASE II - Listening

So, we think we've got a pretty good-looking response curve. It's going to sound awesome! Let's hook it up to a music source and give it a whirl. We change the input to the VCB from the testing equipment output to our receiver output with our CD player hooked up.

What type of music you should play is a very subjective and complicated topic that could take a whole book in itself. Four guidelines I personally use are listed here. First, always have a disc on hand that uses a good female voice and clean instrumentation (i.e. not hard rock). This is because you will be able to listen to the tonal qualities rather objectively and be able to decide if you hear any "coloration" of the sound or any distortions. Second, know what CDs the final evaluator is going to use for testing. This can be the final customer (ultimately), your Marketing Dept., the retailer, etc. This is just common sense. Third, have CDs that you like to play, no matter what kind of music it is. You've probably listened to a disc hundreds of times and you know how you like it to sound. Just keep an open mind here as to whether the sound you like is what the target listener will like or not. And, finally, have at least three different types or styles of music on hand to make sure the speaker does not have some sort of anomaly with a particular type, i.e. sounds good with all three.

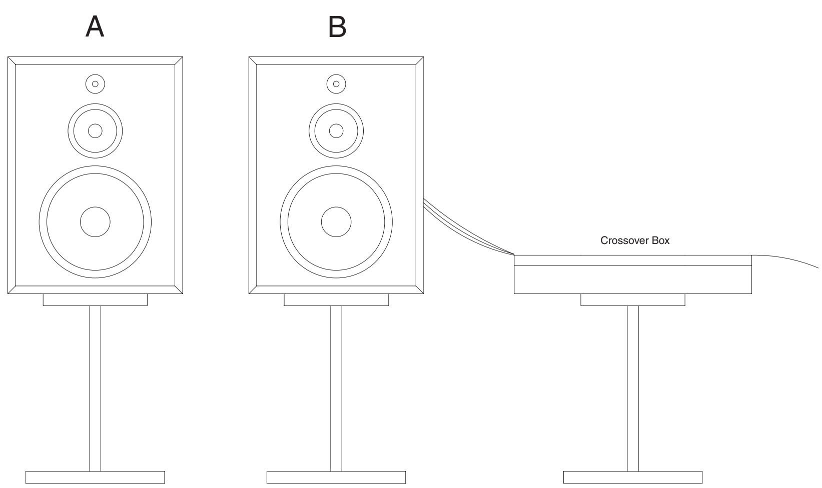


Figure 11: A/B testing.

So, you put your CD in and it sounds pretty good. Done, right? Just to be sure, let's A/B it (compare it) to a speaker of known quality (with a proven track record). For this, you will need some sort of speaker switch box to switch the signal (from the same source) between speakers. This can be a simple two channel A/B switch from your local electronics store or a wireless multi-channel switch box that is available from several vendors. Higher currents ultimately lead to non-linearity in the magnetic components of the driver. It is recommended to listen at different volume levels as well.

Ah-oh. Now the speaker doesn't sound as good as we thought. The speaker we are comparing it to sounds much better. Looks like we have some work to do.

How do we know if one speaker "sounds" better than another? We'll, we don't really. It is a very subjective matter, and everyone will have their own opinions or taste. However, as an audio enthusiast or engineer, you must learn to be as objective as you can. Of course, you need to start with a good reference point. That means you should have at least "normal" hearing (no detectable hearing loss), or at least know your limitations. Then, when you start working with sounds, sine waves, and music, over time you will be able to discern what frequencies are what just by listening. Yes, first you must be able to notice and differentiate the obvious, such as the kick drum is bass, the vocals are lower midrange, or that the hi-hat has a lot of high frequency timbre. As you gain more experience, and especially if you have some sort of musical instrument background, you will be able to precisely recognize sounds or instruments in the music that your otherwise average Joe doesn't even realize he is listening to. He is listening to the convoluted whole; you must be able to pick apart the whole into as many finite parts as you can. You will be able to discern which frequencies are too hot, sense that some frequencies may not be present (a dip in the response curve), that maybe there is a phasing problem between speakers, or even to what extent your speakers parameters are off, upsetting your system balance.

You may have heard the term "Golden Ear" used to describe someone who can sense all of these things and more out of music or an audio source. From my observations, anyone can learn to be a Golden Ear if they possess the following three characteristics: 1) A good ear, 2) Years of listening experience, and 3) A heightened vocabulary in describing what they hear to others, such as analogies like bass being described as warm, highs as tinny, etc. They can freely express and communicate what they are hearing so that others can easily sense and understand.

Okay, back on track. Our next step is to try to make our new speaker sound better than our reference. This may not be possible with our current configuration, but we're sure going to try.

From our listening test, we determine that our speaker has too much high-end compared to the reference. Let's change the tweeter crossover point a little higher so that less highs are going to the tweeter. We decide to change the cap value from 3.3uF to 2.7uF.

After listening again, sounds like it's still too high. Let's change the cap value from 2.7uF to 2.2uF.

Oops… that's too much. Let's go back to 2.7uF and add a small 1ohm resistor to the tweeter filter to decrease it's SPL slightly (See Fig. 12).

Hey…, that sounds about right! Since we added a resistor and we are very cost sensitive here, we also decide to try the woofer without the inductor on it (1st order inductive filter). Our speaker now sounds about as good as our reference!

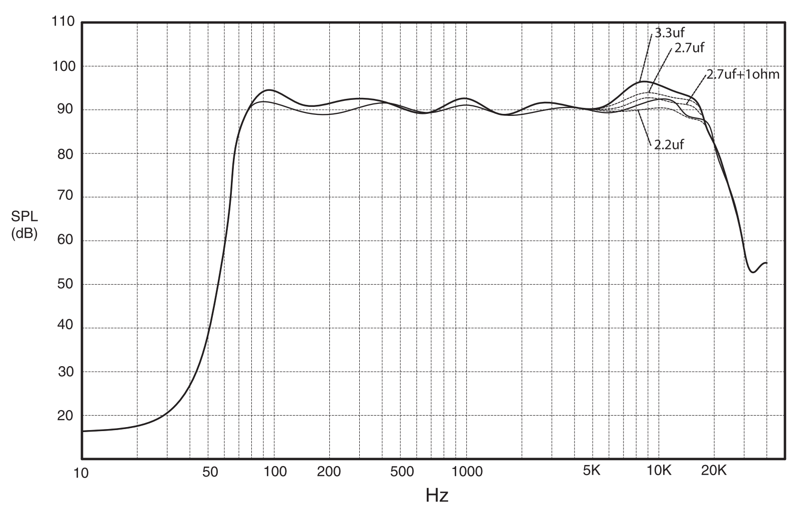


Figure 12: Changing the Tweeter Crossover Point.

Oh, if only everything was that easy. In reality, you'll probably end up trying different orders of filters and playing with phasing combinations just to get an idea of what is going on. In addition to listening to frequencies, you'll also be noticing and comparing timbre distortions, coloration, and overall ambience and "spatialization".

If you have two VCB-100s, you can always start with a reference set-up and compare it to a second identical set-up. Then, change the second set-up step by step if each change seams to be improving the sound. You'll always want to compare your results to a known reference to double check.

When listening, don't spend more than 2 or 3 hours at one sitting. Your ears and senses will become tired and the additional time spent will be worthless. Plus, to be subjective, you'll need to listen to your project on subsequent days to make sure 1) your ears have had a rest, 2) temperature/humidity weren't extreme on a particular day, effecting the sound or 3) you didn't have a cold or hearing issue on a certain day.

Of course, if things aren't working out and sounding as you hoped, you'll have no choice, as stated before, but to go back and change something besides the crossover.

PHASE III - Double Checking the Response Curve

Now it is time to go back and hook up the testing equipment as the input and take a response curve of your speaker. If you like the way it sounds and the frequency response comes out looking satisfactory, CONGRATULATIONS, you're done, FOR NOW.

At this point, you'll also want to check out your design in stereo (as a pair). You will find that some systems have better stereo imaging than others. In general, speaker systems that are less wide on the vertical axis tend to image better.

If it sounds good, but the frequency response is not acceptable, you'll have to determine if a trade-off is acceptable. Does changing the crossover enough to make the response acceptable affect the sound enough to make it unacceptable? You must make the call. If not, you have two choices: 1) start over from phase one taking a different crossover route, or 2) Change something besides the crossover.

PHASE IV - Gaining Approval of our Speaker

We think our system sounds great. Invariably, however, others will have different opinions about it. A good engineer takes all comments and opinions into account, large or small, and weighs them accordingly (by who, what, and why). Of course, Marketing or the Customer will have the final say, so anything they don't like, any opinions they have, or anything they want tweaked may as well be coming from God himself.

Apply the techniques and design principles you have learned, re-hook up the VCB and try to make little tweaks in the crossover until you feel their remarks are satisfied. Of course, you must gauge whether or not what they want changed can actually be done by just modifying the crossover.

This process may go back and forth, taking many iterations to achieve your final goal - that “SAMPLE APPROVED” status!

PHASE V - Power Testing

Once the design is done, you'll need to run an impedance curve to make sure there are no anomalies in it, such as the curve going below 4 ohms at any point (this is a requirement for many since many amplifiers will short out and turn off at loads under 4 ohms). If you had been using the proactive design and simulation software approach, you would most likely already know what your impedance curve was going to look like. But this must be double checked anyway. Of course, your customer may require a whole slew of data and curves to be supplied. (A common specification that also needs to be checked is a gauss reading on A/V/ shield speakers, especially center channels that may possibly sit on top of a television set).

Hopefully, we've designed our system with power issues in mind. These are discussed briefly in another section of this manual, and much more information can be learned from books in our Reference section. I will not mention any standard power test methods here, but different customers have different requirements, so make sure you keep up to date.

If you encounter problems, it's into trouble shooting mode we go, making trade-offs or repeating steps as necessary. Communication of problems to other parties is always the worst part of the job as an engineer. It will happen occasionally.

PHASE VI - Double Checking the Production Crossover

You've spec'ed the network to your crossover supplier. A few weeks go by, and you finally get production samples. You install it, and compare curves and listen.

If all is good, you'll probably want to power test it one more time and then you’re DONE.

However, sometimes samples will come in and be less than satisfactory. Issues that may arise include mutual inductance (how close and how the inductors are aligned on the PCB (printed circuit board)), component values and types not matching the ones used in designing the speaker, and power handling problems due to heat issues with component positions on the PCB.

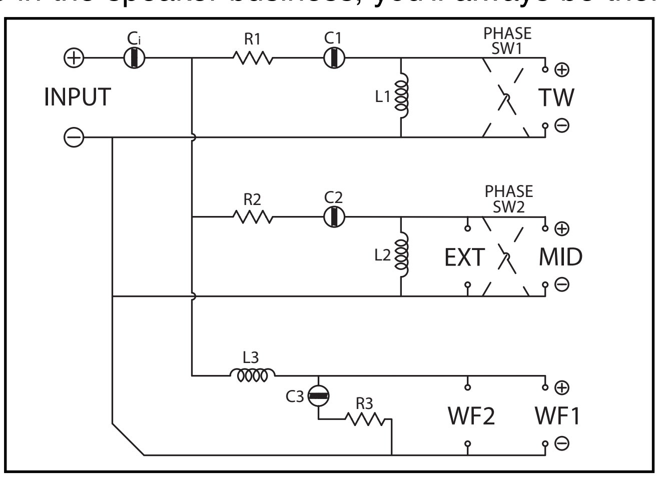
And last but not least…

"Port Duck with Test Sauce."

As an audio engineer, the way you communicate your projects and ideas will be the key to your success. While once working for a Japanese parent company, I would have to communicate back and forth to Japan or even China to give them specs, order samples, etc. Countless time can be wasted and mistakes can be made because of miscommunication (even so, I have observed miscommunication being used as an excuse for some mishaps or delays when it really wasn't. Inherently, the language barrier always has a built-in "whoops" card, to be played when needed, apparently). So don't be satisfied when you are describing something to another engineer, or anyone else for that matter, when he or she is nodding in agreement. If there are any doubts, ask a question to make sure they understand, whether they speak your language or not. And if they don't, it's not their fault, it's your fault. Present whatever you are saying in a way that the other party will understand.

So why the name "Port Duck with Test Sauce”? A few years back, I received a box of sample port tubes with speaker testing data from a China vendor. The box was labeled "Port Duck" instead of "Port Duct", and on the testing data inside were the scribbled words "Test Sauce" instead of "Test Source". True story. To this day, my taste buds still salivate. I believe this statement truly encompasses the audio field as it exists today in many ways. Plus, it makes me laugh.

They say that once you're in the speaker business, you'll always be there. If it is your passion, let us hope so.



REFERENCE: Schematic of VCB-100

REFERENCES - resources (books and websites) that you may find useful.

Speaker/Crossover Design
The Loudspeaker Design Cookbook by Vance Dickason - Available from Old Colony Sound Labs.
Advanced Speaker Designs for the Hobbyist & Technician by Ray Alden - Available at amazon.com.
Old Colony Sound Labs - http://www.audioxpress.com/bksprods/index.htm
AudioXpress - www.audioxpress.com - Publishers of Audio Xpress and Voice Coil Magazine.
Loudspeaker 101 - http://xsspl.tripod.com/Audio/
Art Ludwig's Sound Page - http://www.silcom.com/~aludwig/
Software Index Page - http://www.wssh.net/~wattsup/audio/
SpeakerBuilding.com

Speaker Testing Hardware/Software
SAMPLE CHAMPION - www.purebits.com
CLIO - www.cliowin.com- available from Old Colony Sound Labs.
MLSSA - http://www.mlssa.com/
LMS/LEAP - www.linearx.com
TERM-LAB - www.termpro.com
True Audio - www.trueaudio.com - Good loudspeaker simulation software and reference downloads.
HarrisTech - www.ht-audio.com - Enclosure and crossover design software.

Mobile Audio
Car Stereo Review's Mobile Entertainment Magazine- www.m-emag.com
MobileAudio.com - Internet Guide and FAQ
Car Audio & Electronics Magazine - http://www.caraudiomag.com/
Car Sound & Performance Magazine - www.carsound.com

Home/Industrial Installations
Sound & Video Contractor Magazine - www.svconline.com
CEDIA - www.cedia.org
NSCA - www.nsca.org

Miscellaneous
Audio Engineering Society - www.aes.org
Synergetic Audio Concepts - http://www.synaudcon.com/ - Audio Training Courses

Please email references@vidsonix.com if you would like to be added or removed from the above list.

About the Author

Charles Miltenberger has worked in the audio industry for over 12 years. He has extensive speaker system manufacturing experience in both Engineering and Quality Control Manager capacities working for a $50 million annual sales U.S. OEM manufacturer. He has worked with many industry professionals in Engineering and Sales capacities. Besides work experience training, he has attended audio seminars by A.E.S., Vance Dickason, and Syn-Aud-Con. Charlie has a B.S. in Electrical Engineering and a M.S. in Environmental Management.