

The Unofficial Speaker Workshop Manual *2nd Edition*

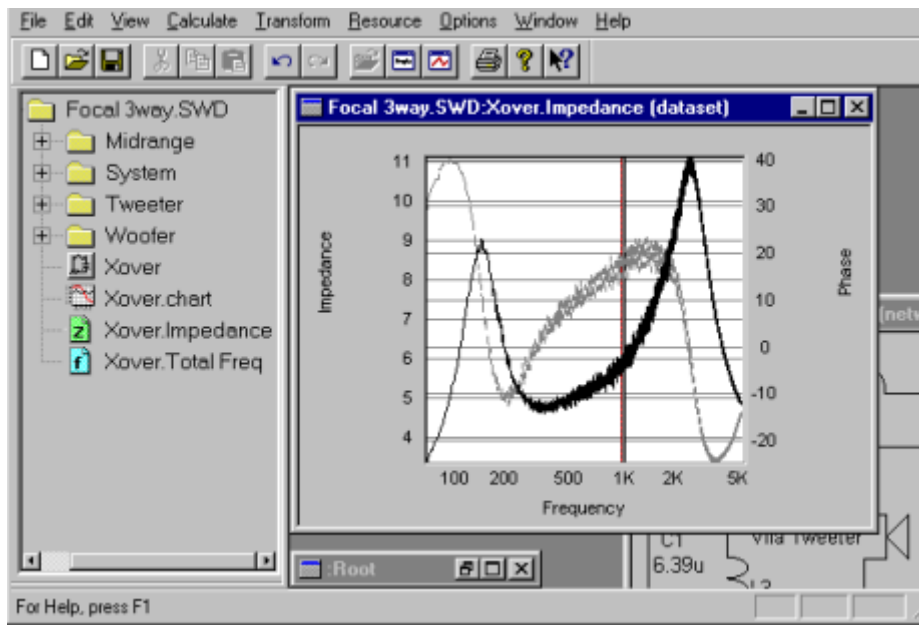
With assistance from other freeware programs

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INTRODUCTION

(TOC)

First the disclaimer: I am a novice with little experience in speaker building or design.

I started gathering information as I became interested in this hobby. I then discovered Speaker Workshop. It is a fairly powerful tool but the learning curve is steep and as a result, many people go to other commercial software. The issue with DIY is that people want a quality product that they can produce with the feeling that they did it themselves. Cost is always an issue. Many people have contributed much time and effort (and knowledge) in assisting all of us to enjoy this hobby more. I have gathered software, spreadsheets, and information in this one place and I hope that I have been able to present it in such a way as to make building a speaker easier and fun. The information that is Copyright is used with permission or is clearly indicated as being someone else's work. I have gathered information from many sources over the past couple of years and I have made every effort to provide appropriate credit; if I have inadvertently included information without providing credit, please let me know and I will certainly rectify this oversight. I have contributed a little information and a few spreadsheets to assist with clarification, calculation and understanding. *In order to use the spreadsheets, just double click on the icon and they will open up.* They were written in Excel 2000 but are likely to be compatible with most spreadsheet programs. I want to thank all of those who agreed to allow me to use their information and/or programs or spreadsheets in this Document. I hope that it is as helpful for all of those reading it as it has been for me in putting it together.

If you notice anything that you feel is in err, please let me know and I will research it and correct it as indicated.

I would like to thank Nelson Wood, Claudio Negro, Dick Van Nierop, George Soong, John Kreskovsky, and the entire FRD Consortium, and all of the other people from the Speaker Workshop Forum who have so graciously taken the time to explain things to others on the forum. I would also like to thank those who also have been so very helpful (to those of us with less experience and knowledge in this field) from other forums to include Parts Express, Madisound, DIYaudio, AudioKarma, Audio Asylum, and Pi forum. Finally, I would like to thank James Salk for so graciously posting this at the www.audiodycentral.com/ntutorials.shtml website.

One of the questions that is frequently asked is, which is better, a or b? Richard Hosch, a hobbyist and mechanical engineer from DIYaudio Forum stated the following, *"Speaker design is always a compromise. What is more audible to you... phase distortion, or harmonic distortion? Limited frequency response, or irregular polar pattern? ...Controlled behavior in the frequency domain (sharp cutoff, no or linear phase shift) can result in bizarre problems in the time domain... ringing, pre-echo...Pick your poison, and then do your best to optimize a design within that framework."* John Dunlavy furthers this with another perspective: *"there are the natural biases that many listeners have that favor their personal subjective preferences for such properties as "sweet sound, pretty sound, forward sound, rich sound, full-bass, etc.", that seldom relate to the*

accurate reproduction of complex musical transients, imaging, etc". The definition of audiophile is one who loves audio; I would argue that this love can be represented by a strive for accurate sound reproduction as championed by people like John Dunlavy or through a strive for preferential sound reproduction as described by Mr. Dunlavy above. My desire is to allow the reader to make his/her own choice with tools that will aid in this effort.

Finally, a note on how to use this Manual. I have divided it into sections with each section having an index or table of contents at the beginning. In this way, you can have each section open as you use it without undue use of computer resources. I have also compiled the overall table of contents to use as a reference in choosing which section to open. I have not removed the links so if you wanted to compile the entire Manual into one document with this Table of Contents at the beginning, I believe that the links would still function (though I have not attempted to verify this). The Manual is not locked so you would be able to do so yourself if it does not work.

I hope that this is helpful for all who read through it.

Sincerely,

Jay

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SPEAKER WORKSHOP MANUAL OVERVIEW **(FROM THE AUDUA WEBSITE)**

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Requirements:

Windows 95, 98, Me, XP, or 2000a
Full-duplex sound card (for testing)
PII, PIII, PIV or AMD similar processor

This program was written by Mark Zachmann
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The basic file type in Speaker Workshop (the document) is a project tree containing many different basic types (such as folders, drivers, enclosures) called resources. All charts, tests, alternate designs, etc. are saved in one single document - easily accessible with names that can be long and descriptive. The project tree window can be hidden, floating, or attached to a side of the program. The program has full multi-step undo/redo capability. Please note that it is beta software and it occasionally may quit a process in the middle. This is likely as a result of memory demand. If this happens, save your

settings, close the program, and re-open it again.

Measurements

Speaker Workshop can test and measure Driver Impedance with a resolution of 1/3 of a Hz. It uses a MLS signal of up to 256K samples at sample rates of 11K, 12K, 22.1K, 24K, 44.1K, 48K, 88.2K, and 96K. SW can find the value of a passive component to within approximately 1%. SW automatically detects Resistance, Impedance, and Capacitance and uses a full impedance curve to find a best fit for value and Q. The frequency response / transfer function of an amplifier can be tested from 20Hz to 18KHz with a resolution of 1/3 Hz. SW tests calibration channel and input channels simultaneously in one operation. With a suitable microphone, it can test the acoustic response of a driver over the same ranges as transfer function above. Total Harmonic Distortion can be measured in speakers and amplifiers to approximately .03%. Intermodulation Distortion can be measured down to approximately 0.03%.

Design

SW can design both sealed and vented enclosures for woofers and midranges using a range of Q and tuning values. Results include impedance, acoustic response, group delay, and cone excursion. Stock crossovers (1-4 order 2-way) and standard impedance corrections (inductive rise or resonant peak) are used. It can design impedance compensation circuits. SW includes full crossover design capabilities for designing

complex crossovers. It uses graphical user interface with zoom-in and zoom-out, placing components and drawing connections with calculated impedance and acoustic results.

Analytical Tools

SW includes a Signal Generator that can play 10 types of signals (incl. Sine, Square, Pulse, Sweep, Warble, Burst, MLS, Sawtooth, White and Pink Noise) as well as any custom signal that you load in a .WAV file format. It can record two input channels while playing (e.g. an audio oscilloscope). Transformations include FFT (Fast Fourier Transform), Inverse FFT, scaling, smoothing, splicing, and digital filtering. Each chart can include multiple datasets. Position information for each dataset in a chart is easily displayed. Zoom-in and out are supported and axes, titles, colors, etc. are programmable.

Speaker Workshop uses MLS (Maximum Length Sequence) rather than sine waves to perform most of its measurements. The MLS is used for all types of acoustic testing since it is noise-like and possesses a low peak to average power ratio (low crest factor). This produces a good SNR (Signal to Noise Ratio) for the recorded signal, and so subsequent calculations performed on the recorded signal can be more exact and informative. **The MLS impulse signal in the signal generator is the same type of signal as in Options/Preferences/Measurements.**

The Speaker Project is a detailed write up at the SW site about designing, building, and testing a subwoofer and 5 monitor speakers. It's a great tutorial about using Speaker Workshop and well worth the read (it can be found at the Speaker Workshop site).

Of note, Speaker Workshop is available in several different languages and in countries that use both Metric and Imperial Measurements.

SECTION I - EQUIPMENT AND BASIC SOFTWARE

DESIGN

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Though Speaker Workshop can work with external files that are imported into the program, in order to obtain your own measurements and then calculate and design speakers, you will have to have a measurement microphone with a preamplifier for the microphone as well as some type of jig set up. Most new soundcards will also require an external amplifier in order to do acoustic measurements. A good DMM (Digital Multimeter) is helpful for calibration, calibration confirmation, for checking voltage output and for measuring the DCR of speaker voice coils.

The Measurement Microphone

The Measurement Microphone must be an Omni Directional Condenser Type. This type of Microphone requires a pre-amplifier with phantom power. The most commonly used microphones are the Mighty Mike, the Behringer ECM 8000, and DIY microphones made out of either the Panasonic Capsule WM 60 or WM 61.

Jigs

Speaker Workshop requires a jig set up. This can be done with individual cables (as described by Claudio Negro below) or with a formal Jig. The most widely used Jig was designed by Eric Wallin and is demonstrated below. Various alterations of the Jig are available at both the Speaker Workshop Website and on the Web. This Manual will address the use of the Wallin Jig II and the use of a Cable based Jig (thanks to Claudio Negro) both of which are described in this Manual. Jig I could also be used, though it will require a slight modification to enable dual channel acoustic measurement. They can be found on the Web at:

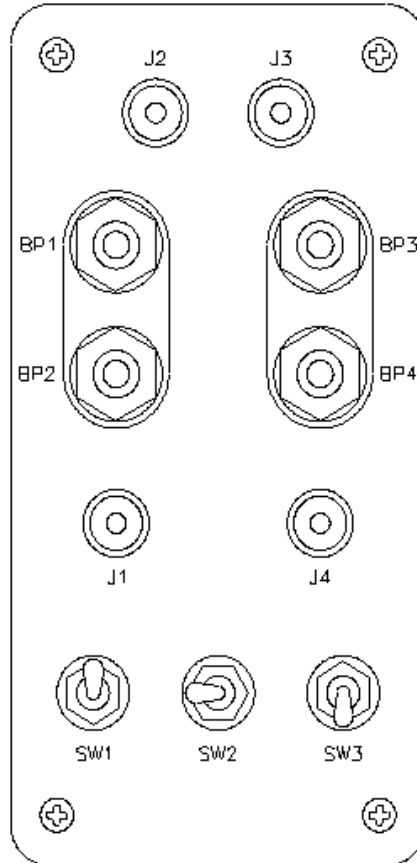
The Wallin Jigs: http://mysite.verizon.net/tammie_eric/ericindex.html

Cable Based Jig at: <http://www.claudionegro.com>

Eric Wallin's Jig II

Switches, Jacks, & Modes

Here is a look at the faceplate of the new jig:



The faceplate of the new jig.

Here is a list of the switches and jacks on the jig, and their function:

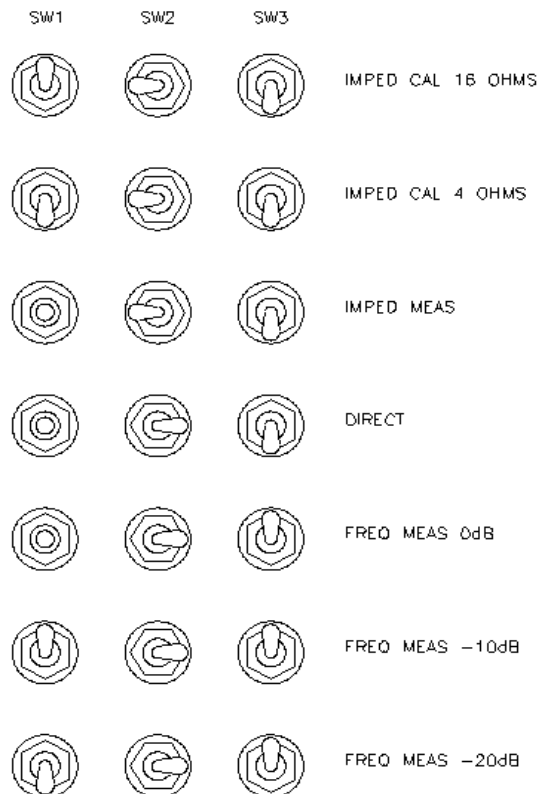
- J1 is a soundcard speaker level input to the jig in case you want to power everything with the speaker amplifier built-in to your soundcard. This is fine for any impedance measurements you might want to make, and also works for low level frequency response testing. Note that J1 and BP1 and BP2 are electrically connected in the jig, so disconnect your soundcard speaker level connection at J1 if you use an external amplifier connected to BP1 and BP2 or you could hurt your soundcard!
- BP1 and BP2 are amplifier level inputs to the jig in case you decide to use an external amplifier to power it. BP2 is at ground potential.
- BP3 and BP4 are the test points where you connect components like caps, resistors, inductors, and speakers to be tested. BP4 is at ground potential.
- J2 and J3 are left and right line out jacks respectively, that go to the line level inputs on your soundcard.
- J4 is an input that you connect your external microphone preamp line level output to.
- SW1 is a three-position switch and has a dual function. In the impedance-measuring mode it selects and switches the calibration resistors in and out of circuit. In the frequency response mode it selects between two levels of attenuation that scale the output voltage at J2. This can help match the levels going to your soundcard when using an external amplifier to drive the speaker under test during frequency response measurements.
- SW2 and SW3 select the main modes of the jig. When SW2 is to the left and SW3 is down, the jig is in *impedance* mode. Here, either or one of two or none of the calibration resistors is selected by the position of SW1. When SW2 is to the right the internal 8-ohm bridge resistor is shorted out, and the amplifier output is applied directly to the test terminals BP3-4. Note that the calibration resistors are also removed from circuit. So with SW2 to the right, and when SW3 is down, then the jig is in *direct* mode, where the signals at J2 and J3 are the same and a channel difference calibration can be performed. When SW2 is to the right and SW3 is up, the jig is in *frequency response* mode, where the microphone line level input from the preamplifier at J4 is connected to J3, and the attenuation of the speaker signal at J2 is selected by the position of SW1. So the only "odd" or meaningless mode is with SW2 to the left and SW3 up.

The jig has three basic modes, *Impedance*, *Frequency Response*, and *Direct*, which are selected by the positions of SW2 and SW3. SW1 selects the sub modes. Here is a listing of all modes and sub modes in a handy table form:

Jig Modes

SW1	SW2	SW3	Mode	Description
U	L	D	IMPCAL16	16-ohm calibration resistor is across BP3-BP4 for jig impedance calibration.
D	L	D	IMPCAL4	4-ohm calibration resistor is across BP3-BP4 for jig impedance calibration.
C	L	D	IMPMEAS	Measure the impedance of a device across BP3-BP4.
C	R	D	DIRECT	J2 and J3 are connected to J1 for channel balance calibration.
C	R	U	FRMEAS-0DB	Mic input J4 connected to J3; no attenuation @ J2.
U	R	U	FRMEAS-10DB	Mic input J4 connected to J3; -10dB attenuation @ J2.
D	R	U	FRMEAS-20DB	Mic input J4 connected to J3; -20dB attenuation @ J2.

U=up, D=down, C=center, L=left, R=right



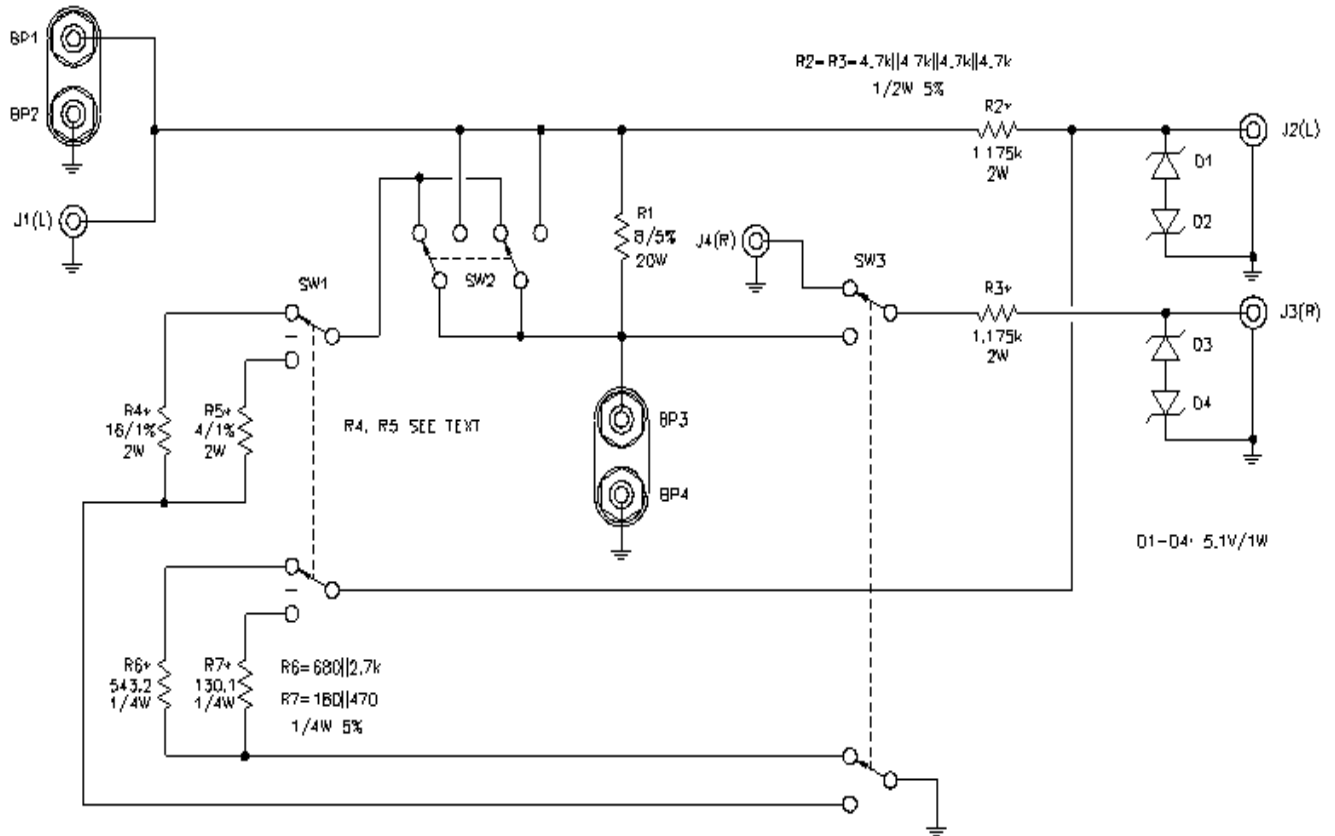
Switch positions and the modes of the jig.

Note: As long as you are careful to have SW2 to the RIGHT when applying lots of power to the jig for frequency response tests, then there is no way to accidentally "fry" the 2W

precision calibration resistors, nor the 8 ohm bridge resistor, regardless of the settings of SW1 or SW3. The 8 ohm bridge resistor is only in circuit when SW2 is to the LEFT, and the calibration resistors additionally require that S3 be in the DOWN position and S1 be in the NON-CENTERED position to be in circuit.

Note: The -10 and -20 dB attenuation is just applied to the signal at J2 (and not to J3 also, as in Wallin Jig I) and that this attenuation is only in effect when SW3 is in the UP position.

Wallin Jig II Schematic

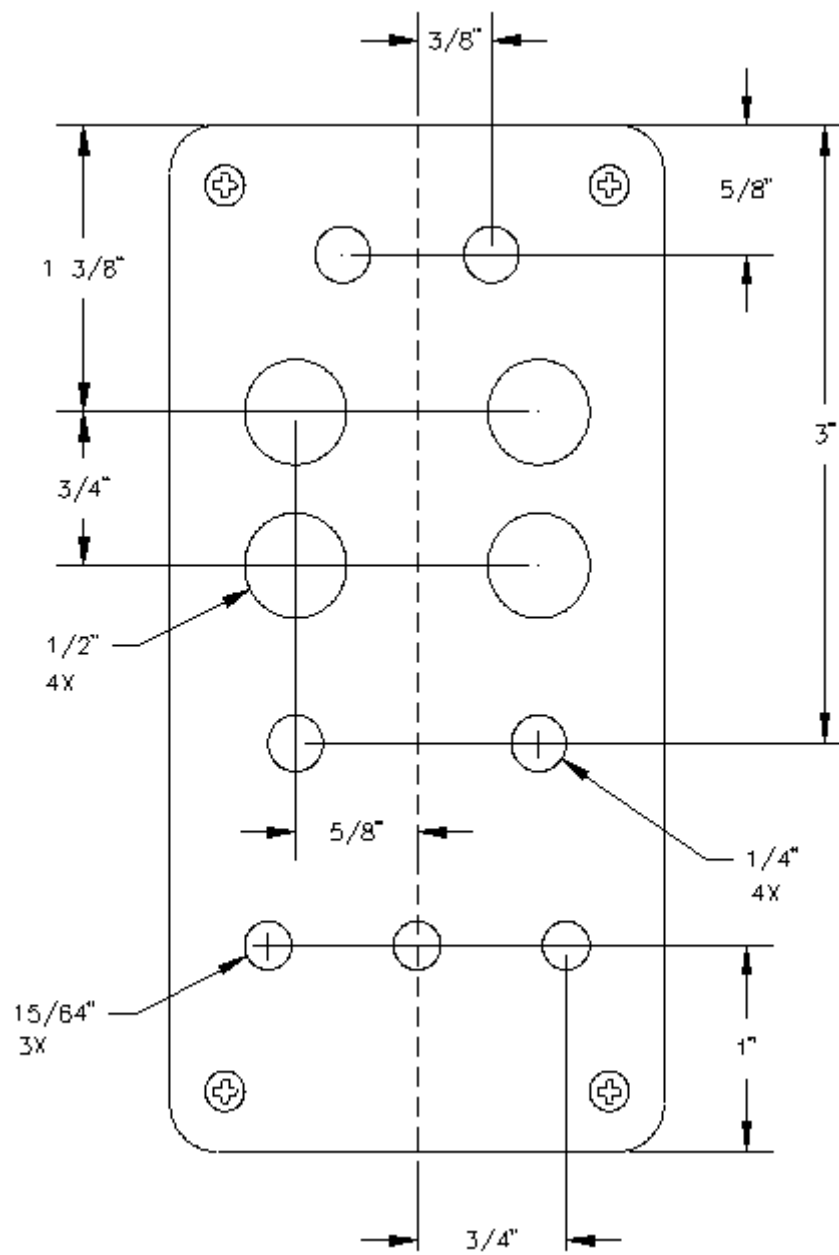


<u>ITEM</u>	<u>DESCRIPTION</u>
B1	Black plastic box, dimensions 5" x 2.5" x 2"
SW1	DPDT center-off 6A mini toggle switch
SW2, SW3	DPDT 6A mini toggle switch
BP1-2, BP3-4	Dual binding post
J1 - J4	RCA jacks, gold plated
R1	8 ohm non-inductive 20W resistor
R2, R3	1.175k 2W resistor (four 4.7k 1/2W in parallel, see text)
R4	16.2-ohm 2W resistor (four 16.2 ohm 1/2W 1% in series/parallel)
R5	4.05-ohm 2W resistor (four 16.2 ohm 1/2W 1% in parallel)
R6	543.2-ohm 1/4W (680 ohm in parallel with 2.7k ohm, both 1/4W 5%)
R7	130.1-ohm 1/4W (180 ohm in parallel with 470 ohm, both 1/4W 5%)
D1 - D4	5.1V 1W Zener diode

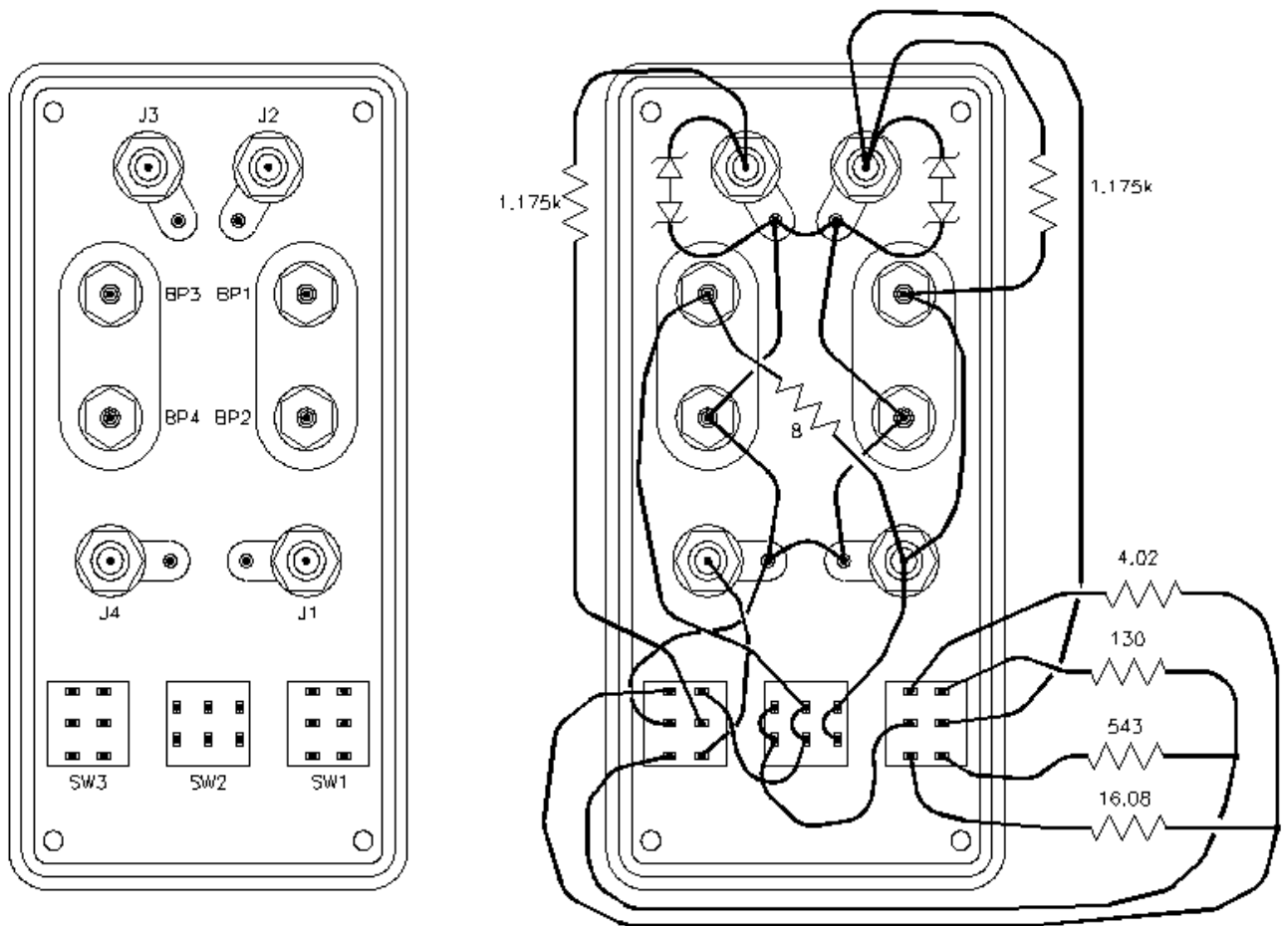
Notes:

- R2 and R3 are made by soldering four 4.7k 0.5W 5% resistors in parallel. I used heat shrink tubing to hold the resistors together before soldering. You might want to sort the individual resistors by value using a good DMM, and then combine them to get as close to 1.175k as possible for R2. This will make the dB reduction in the frequency response measurement mode more accurate. The precision of R3 is relatively unimportant, so use the remaining resistors to form this element.
- R4 is made by soldering two sets of two 16.2 0.5W 1% resistors in parallel, and then soldering these assemblies in series. R5 is made by soldering four 16.2 0.5W 1% resistors in parallel. I used a piece of heat shrink tubing to hold the resistors together both before and after soldering. Anything around 16 ohms will do, it's the precision that's important for calibrating the jig.
- R6 is made by soldering a 680 ohm 0.25W 5% resistor in parallel with 2.7k ohm 0.25W 5% resistor. R7 is made by soldering a 180 ohm 0.25W 5% resistor in parallel with 470 ohm 0.25W 5% resistor. I got all of these from an assortment sold by RS, though they may sell them in packs of five. If you have more than one resistor of each value, try to combine them so that the values are as accurate to the nominally specified value as possible. This will keep the dB reduction in the frequency response measurement mode more accurate.
- Avoid Wire Wound Resistors in this jig.
- For Diodes: The bar is the cathode or minus end and the anode is the positive end.

Physical Design



Suggested drilling guide.



View of the rear of the switch panel, and suggested wiring guide.

Construction Tips

- Mount all switches and jacks on the black cover supplied with the box - it is easier to drill and looks better to me than the aluminum cover.
- Use a sharp object like a drywall screw to make small indents at the locations of the holes before drilling. You only need hand pressure to do this in the soft plastic, and this will act as a guide for the drilling, keeping the bit from walking. To keep the plastic from cracking, start all holes with a small bit and work your way up with larger and larger bits until the holes are the appropriate size. I finished the largest holes for the binding posts using a countersink, working my way up slowly from each side until the plastic standoffs on the binding posts fit.
- The jacks used for J1 thru J4 come two to a package, with one white and one red. Use the white jacks for J1 and J2, and the red jacks for J3 and J4 (see figure 1 above). RED = RIGHT (I can safely say that now that the cold war is over).
- Bend the outer conductor solder tabs of J1 thru J4 up at ~45 degree angle after they are mounted to the back of the cover. This will help you run ground wires and attach components.
- For switches SW1 thru SW3, the hardware stack up is like this: switch body, nut, panel, star washer, and nut. So the star washer goes on the outer side of the front panel. Don't use those funky-keyed washers that may come supplied with the switch.
- "Tin" all solder lugs and wire ends before soldering them together.
- Clean the tip of your iron and add a tiny amount of fresh solder to it immediately before tinning or making a solder joint.
- Watch J1 thru J4: don't apply too much solder to the inner conductor tab or it may flow inside, making it impossible to insert RCA plugs. You pretty much have to throw out any jack that this happens to.
- Use heavier gage wire for the ground run, and also for runs going between BP1, J1, BP3, and SW2.
- Physically bind R1, R2, and R3 together using plastic cable ties. This mechanically "fixes" all three of these components, and makes them less immune to causing unintentional connectivity within the jig due to mechanical shocks and the like.

Jig Electrical Checkout

Check the wiring until you are certain that the jig is wired correctly. When you have your jig done, get an ohmmeter and perform the following measurements (note that many multimeters will not read low ohms very accurately):

Electrical Tests

SW1	SW2	SW3	Measure Between	Expected Reading
X	X	X	From barrel of J1 to barrels of J2, J3, and J4	Short (0 ohms)
X	X	X	From barrel of J1 to BP2 and BP4	Short
X	X	X	From BP1 to inner conductor of J1	Short
X	X	X	From BP1 to inner conductor of J2	1.175k +/-5%
X	X	U	From inner conductor of J4 to inner conductor of J3	1.175k +/-5%
X	X	U	From BP3 to inner conductor of J3	Open (infinite ohms)
X	X	D	From BP3 to inner conductor of J3	1.175k +/-5%
X	X	D	From inner conductor of J4 to inner conductor of J3	Open
X	L	X	From BP1 to BP3	8 ohms +/-5%
X	R	X	From BP1 to BP3	Short
C	X	U	From the inner conductor of J2 to the barrel of J2	Open
U	X	U	From the inner conductor of J2 to the barrel of J2	543.2 ohms +/-5%
D	X	U	From the inner conductor of J2 to the barrel of J2	130.1 ohms +/-5%
C	L	D	From BP3 to BP4	Open
U	L	D	From BP3 to BP4	16.2 ohms +/-1% if you used 16.2 ohm 1% resistors to make R4
D	L	D	From BP3 to BP4	4.05 ohms +/-1% if you used 16.2 ohm 1% resistors to make R5

U=up, D=down, C=center, L=left, R=right, X=don't care

Connecting The Jig to Your Computer

Connect a cable from the speaker output on your soundcard to J1. The wire corresponding to the left channel should be plugged into this jack, though either channel may do (use the right channel for the Ensoniq Soundscape).

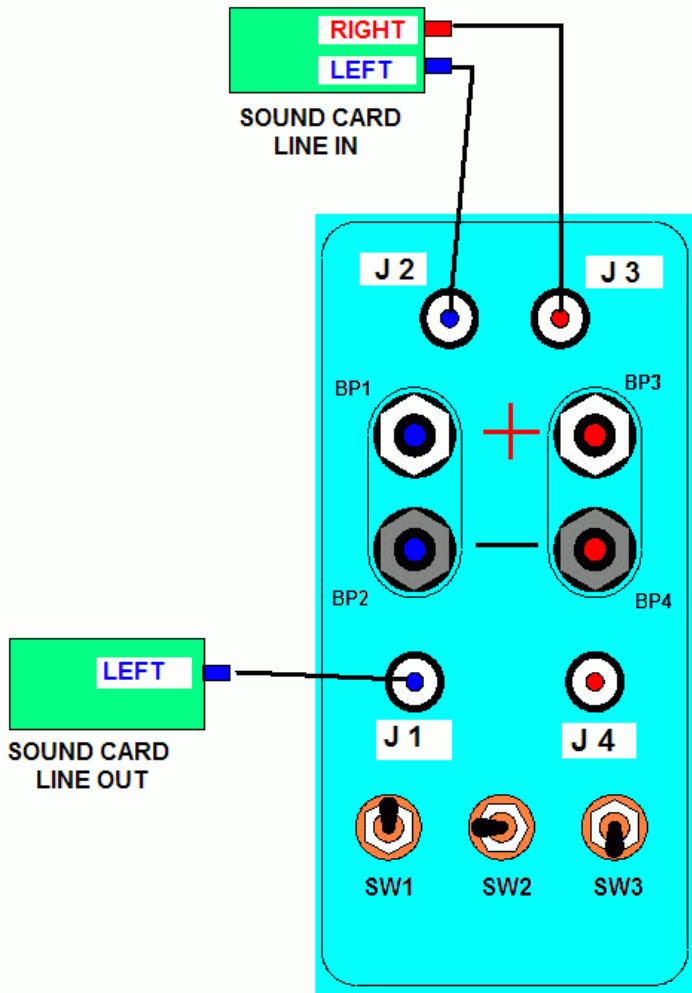
Optionally, if you want to drive the test jig with an external power amplifier, you should run the line out from your soundcard to the line in to the amplifier, then run the speaker outputs from the amplifier to binding posts BP1 (+) and BP2 (-).

Note: Be careful to remove the soundcard connection at J1 when you do this, or you could fry the soundcard and possibly more stuff inside your computer!

Connect a cable from the line-in jack on your soundcard to J2 and J3: J2 left, J3 right. If you accidentally reverse these connections, or simply aren't sure which is which, you will find this out during the channel ID check, so don't worry too much about it now. Usually Red=Right and Black=Left for the cables I've run across in my travels.

Connect some test leads with pinchy clips on one end and banana plugs on the other to BP3 (+) and BP4 (-).

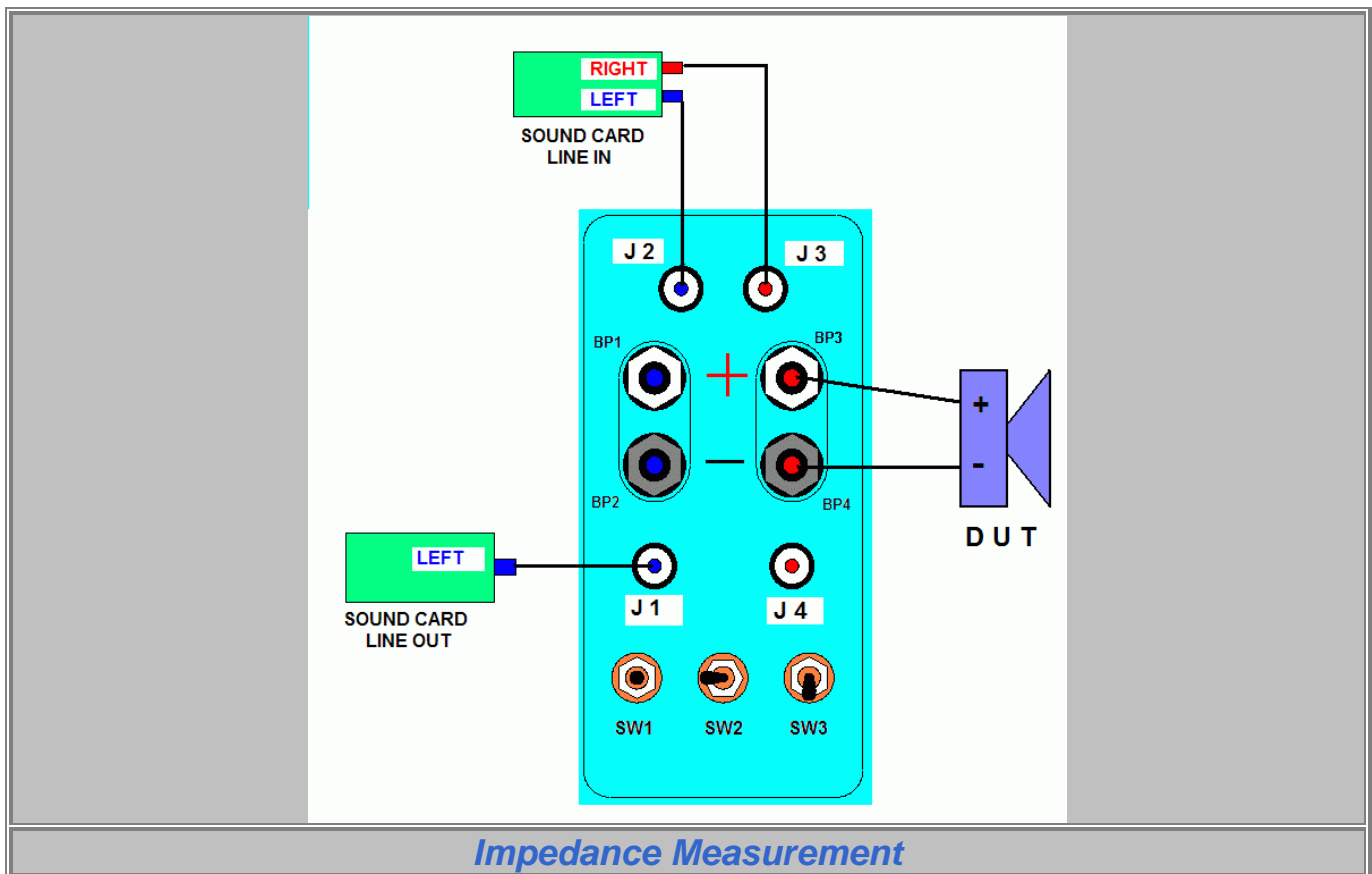
Set up

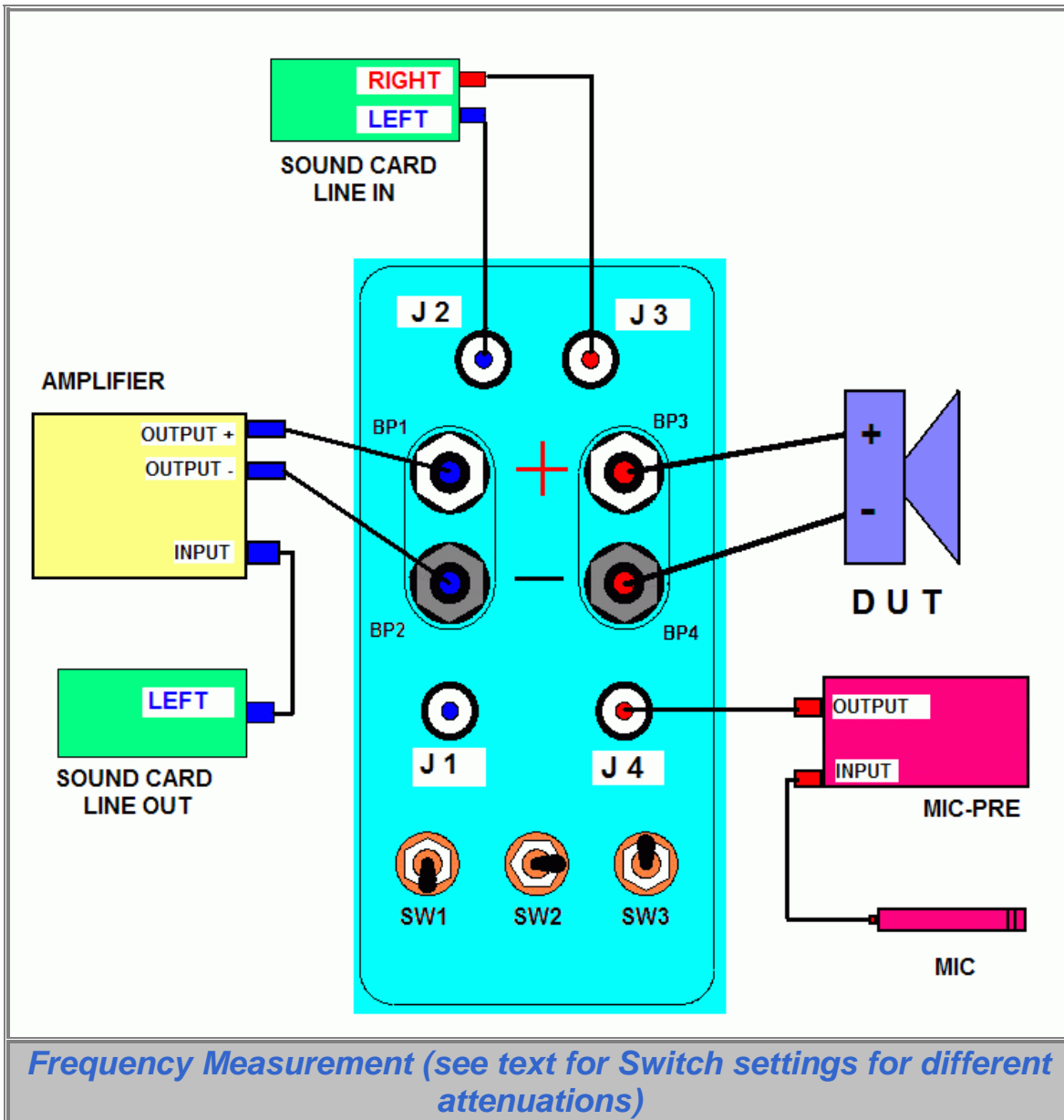


Jig Modes

SW1	SW2	SW3	Mode	Description
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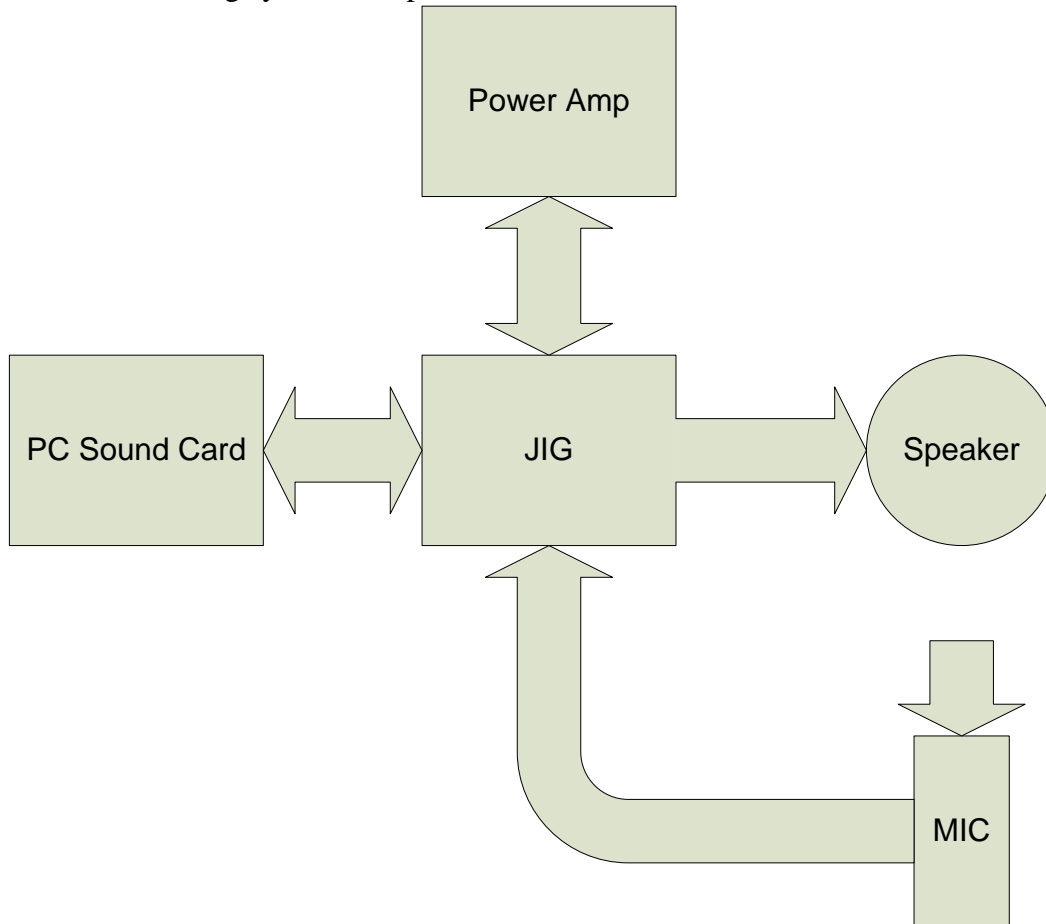




ALTERNATE WALLIN JIG DESIGNED BY GEORGE SOONG

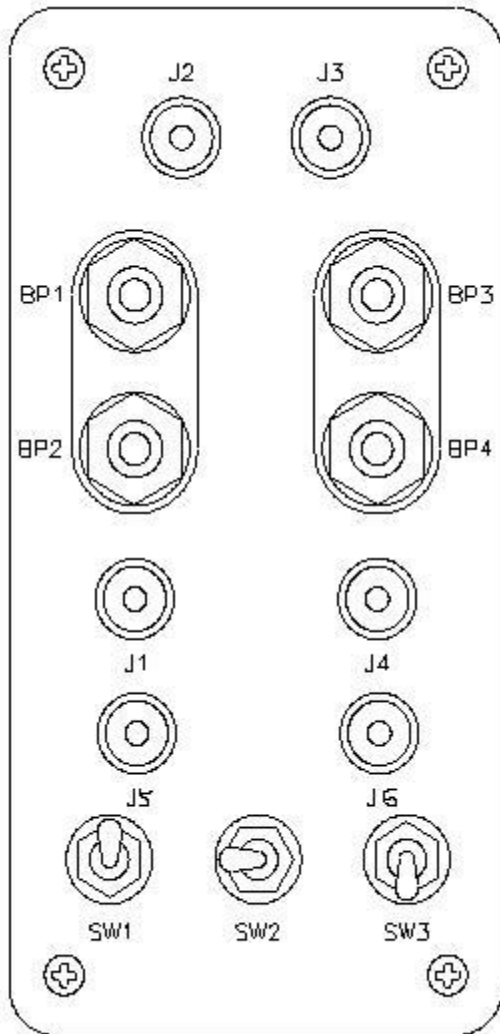
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This test JIG set up is a modified version of the Wallin Jig II (JIG II), which will be further referred to as the GS JIG. The main purpose of this GS JIG is to allow for multiple power level testing for either impedance or acoustic testing. The design is made with the following system set up in mind.



Since the GS JIG is designed for a specific power amplifier, the amplifier voltage gain must be first known in order to determine how much power is going into the speaker. The switches SW1, SW2, SW3 maintain the same functions of the original JIG II, and adds additional functionality to these switches to accommodate multiple testing power levels. One additional switch (slider switch not shown) is added to switch in the calibration mode since it is not used often, and keeps the power level at 1W during calibration. Two additional RCA connectors J5, J6 are used for sound card output signal attenuation to generate appropriate test power levels.

Below is the new layout for the GS JIG:



For the convenience of example calculation, let's assume the voltage gain of the power amplifier is 10 (20db), and we want to conduct tests using 100W drive, 10W drive, and 1W drive.

The following is the modified circuit concept. (the absolute values of the resistors depend on sound card clipping voltage and amp gain)

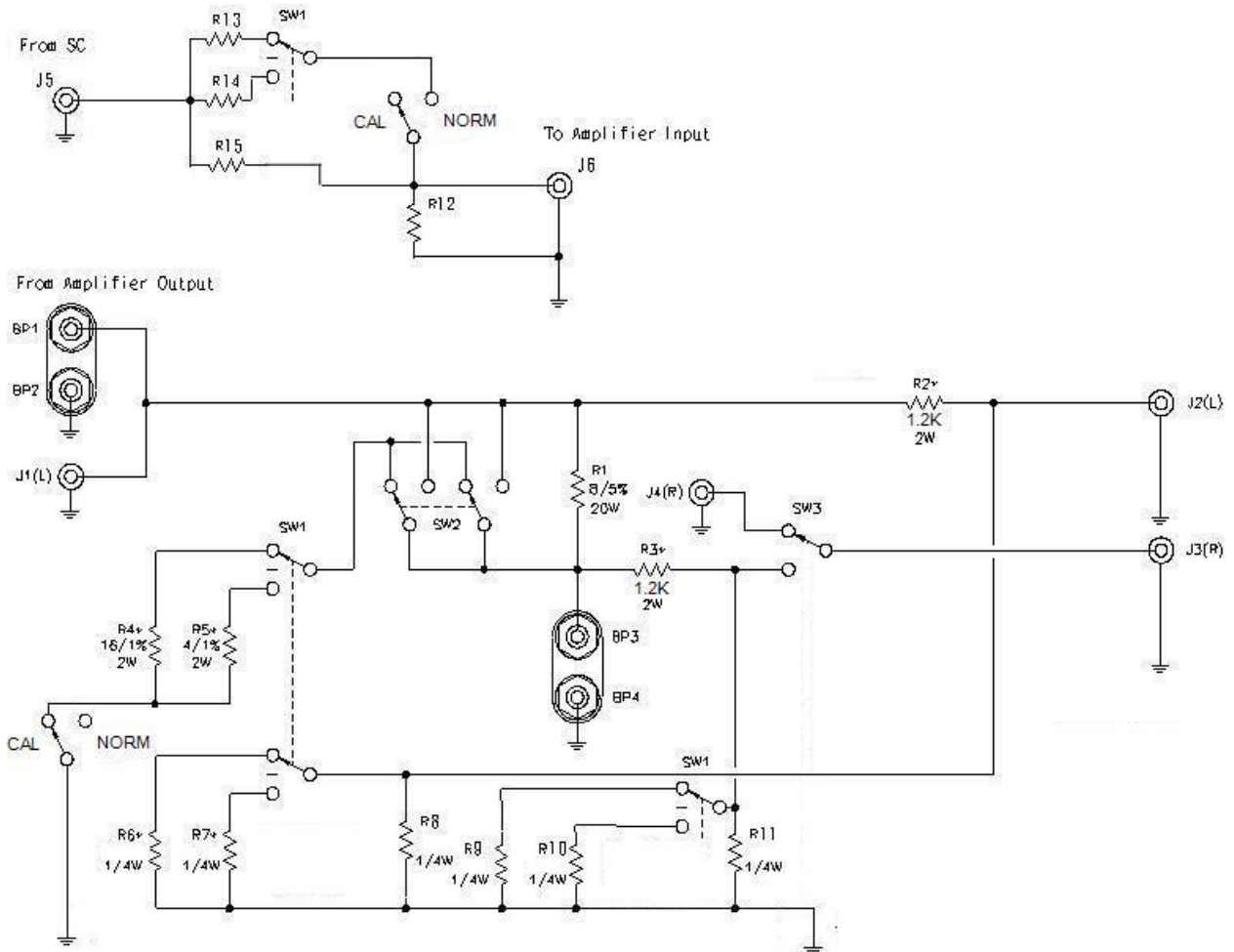
R1 – Reference resistor used for impedance measurements.

R2~R3 – Series resistor used for voltage dividing of test signals so that voltages do not damage the sound card. These values may be between 1K~1.5K, but not greater than 10% of the sound card input impedance as a rule of thumb. The amount of voltage drop is determined in combination with R6~R11.

R4~R5 – Test resistors used for calibration reference. Normally one is twice the reference resistor value and one is half the reference resistor value.

R6~R11 – Are used in combination with R2~R3 to attenuate power amp outputs going back into the sound card so that the sound card is not damaged during testing. The amount of test power, speaker load, and sound card clipping voltage determine the actual values used. If you assume an 8Ω speaker load, then you are able to drive twice the power without damaging the sound card when you test a 4Ω speaker, but you will also get half the reference and data signals when driving the planned power.

R12~R15 – Are used to determine the voltage necessary to generate the desired power level for testing. The sound card output level and the amplifier gain determine the actual values.



First we determine the R2~R3, R6~R11 resistor values for sound card protection. Selecting R2=R3=1.2KΩ is a good start since it is large enough so that R6~R11 will be at values that can be purchased. We could also adjust the final values by multiplying R2~R3, R6~R11 all by a common factor to reach stock values, or you can just select the closest value lower than the calculated values for R6~R11.

We now need to determine the voltage levels for 100W, 10W and 1W.

Since

$$P = V_{out}^2 / R_{load}$$

$$\text{Therefore, } V_{out} = (P * R_{load})^{1/2}$$

Here we assume the load is a nominal of 8Ω. Ideally you would want these to vary with true nominal impedance, but this is not possible in Speaker Workshop unless a more intelligent jig is designed that can be controlled by Speaker Workshop. We also assume that our sound card starts clipping at 3 Volts p-p.

So for protection at 1W

$$V_{out} = (P * R_{load})^{1/2} = (1 * 8)^{1/2} = 2.83 \text{ Volts rms} = 4 \text{ Volts p-p}$$

$$R_8 / (R_8 + 1.2K) = 3/4 = 0.75 \text{ Then } R_8 = R_{11} = 3.6K\Omega$$

For the protection at other wattages, the calculation is more complicated because the additional resistors are in parallel with R8 and R11 respectively when they are switched in.

For protection at 10W

$$V_{out} = (P * R_{load})^{1/2} = (10 * 8)^{1/2} = 8.9 \text{ Volts rms} = 12.6 \text{ Volts p-p}$$

$$(1 / (1/R_6 + 1/R_8)) / ((1 / (1/R_6 + 1/R_8)) + 1.2K) = 3/12.6 = 0.24 \text{ Then } R_6 = R_9 = 416.2\Omega$$

So for protection at 100W

$$V_{out} = (P * R_{load})^{1/2} = (100 * 8)^{1/2} = 28.3 \text{ Volts rms} = 40 \text{ Volts p-p}$$

$$(1 / (1/R_6 + 1/R_8)) / ((1 / (1/R_6 + 1/R_8)) + 1.2K) = 3/40 = 0.075 \text{ Then } R_7 = R_{10} = 100\Omega$$

Next we determine the R12~16 resistor values for generating the desired test power level. Since we want to make sure that the sound card has sufficient impedance and that R13~R15 does not create too much attenuation when the amps input load is connected, we first choose R12 to be 20K Ω . Let us assume the amplifier we are using has a sensitivity of 1.43 V rms (2V p-p) for 140W rms output power into 8 Ω (review the manual or test your amp before proceeding). According to the equation $P = V_{out}^2 / R_{load}$

We then have:

$$V_{out} = (P * R_{load})^{1/2} = (140 * 8)^{1/2} = 33.5 \text{ Volts rms} = 47.3 \text{ Volts p-p}$$

So the voltage gain is $47.3/2 = 23.7$

We also assume that our sound card maximum outputs are 2.2 Volts p-p. So,

For 1W test output, the input voltage we need is $4/23.7 = 0.17 \text{ Volts p-p}$

$$20K / (R_{15} + 20K) = 0.17/2.2 \text{ Then } R_{15} = 240.7K\Omega$$

For other wattages, the calculation is more complicated because the additional resistors are in parallel with R15 when they are switched in.

For 10W test output, the input voltage we need is $12.6/23.7 = 0.53 \text{ Volts p-p}$

$$2.2 * 20K / ((1 / (1/240.7K + 1/R_{13})) + 20K) = 0.53 \text{ Then } R_{13} = 84.3K\Omega$$

For 100W test output, the input voltage we need is $40/23.7 = 1.69 \text{ Volts p-p}$

$$2.2 * 20K / ((1 / (1/240.7K + 1/R_{14})) + 20K) = 1.68 \text{ Then } R_{14} = 6.2K\Omega$$

Here is a simple spreadsheet for calculation of other values:



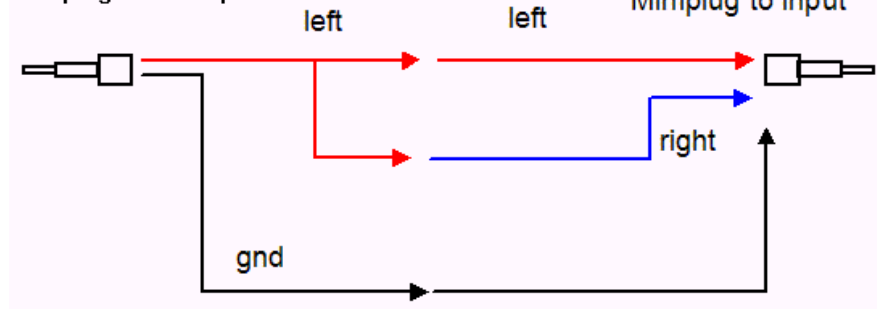
GS_Jig_Design_Description_Document.xls

CLAUDIO NEGRO'S CABLE JIG

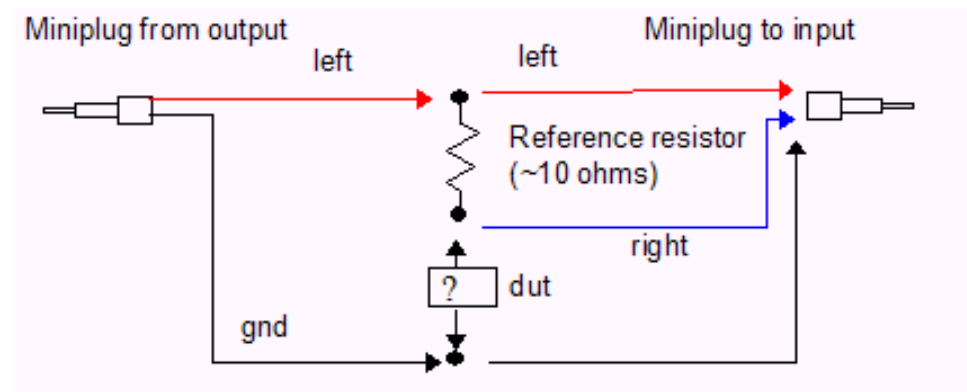
[\(TOC\)](#)

Loop Cable is used for channel calibration and for volume setup:

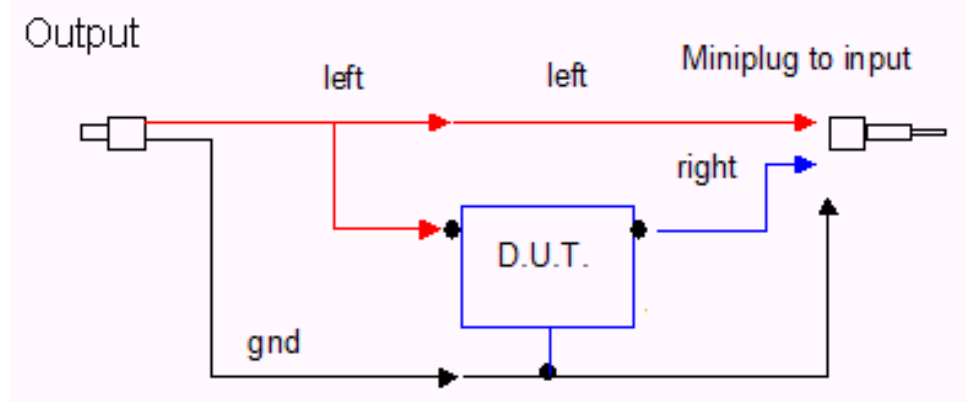
Miniplug from output



Impedance Cable is used to measure drivers impedance, to calibrate SW and to measure passive components:



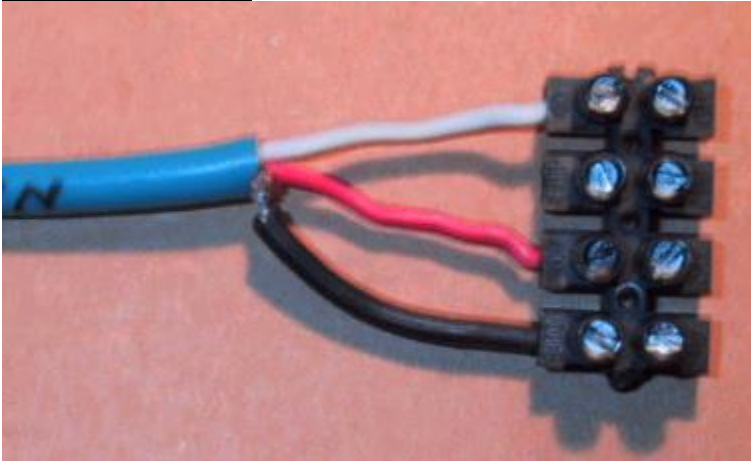
Acoustic Measurement Cable is used to measure frequency response.



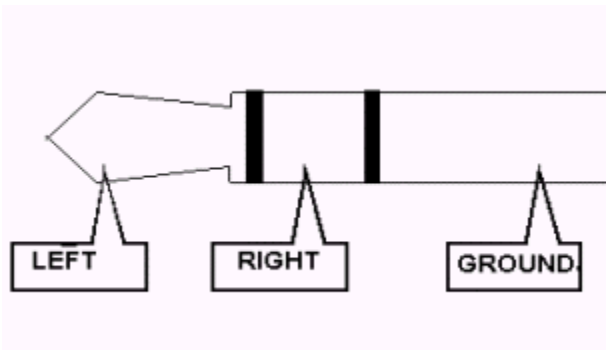
Suggestion: Use a bipolar shielded cable, 2 x 0.75 mm, well made connectors and keep the cables connected to the sound card as short as possible.

Be careful when soldering the mini-jack: remember that SW uses the left channel as REFERENCE and the right one as DATA.

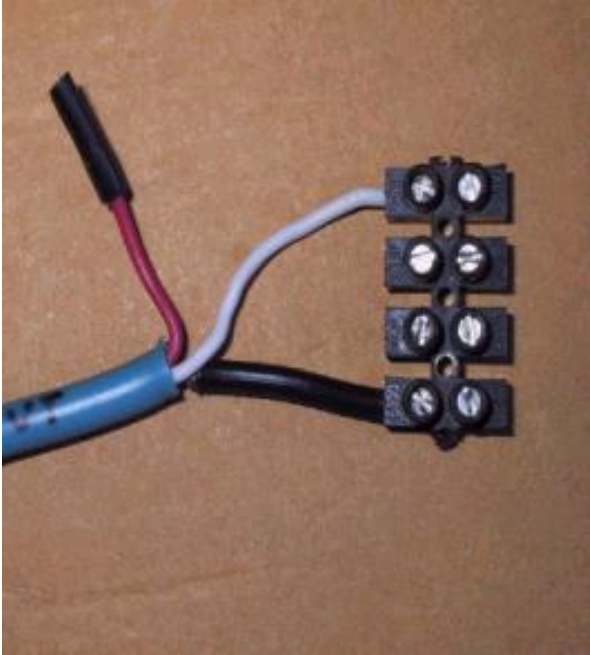
The Line In Cable looks like this:



The Mini Jack Stereo Connections look like this:

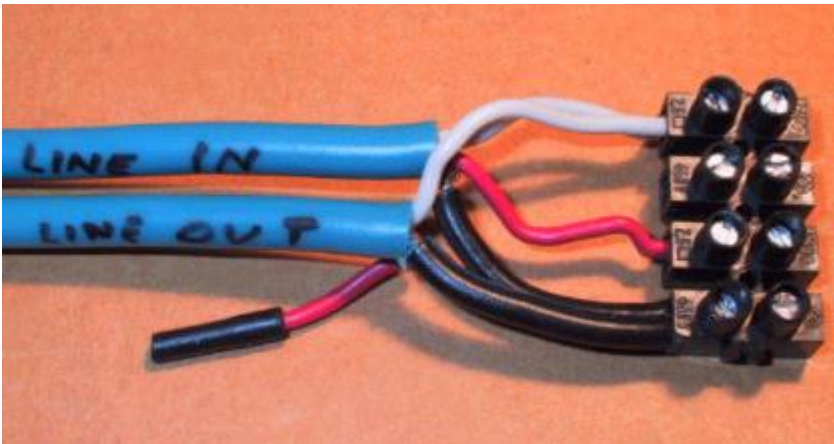


The Line Out Cable looks like this:



And **together**, the line in and line out look

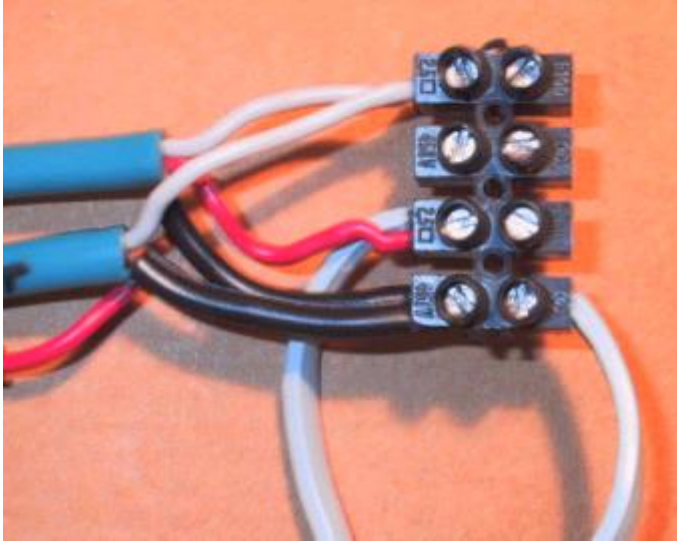
like this:



The DUT cable looks like this:

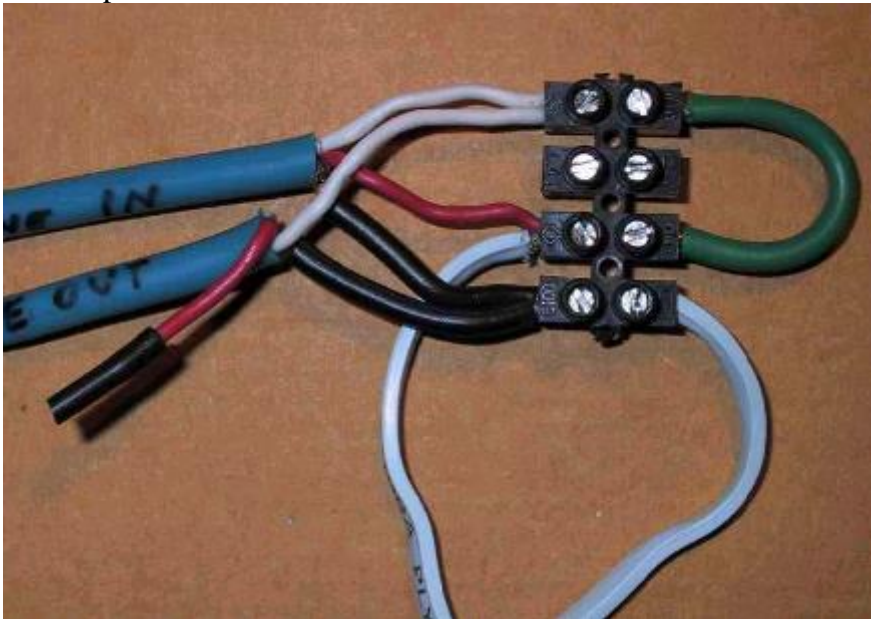


And the line in, line out, and DUT cable all connected looks like this:

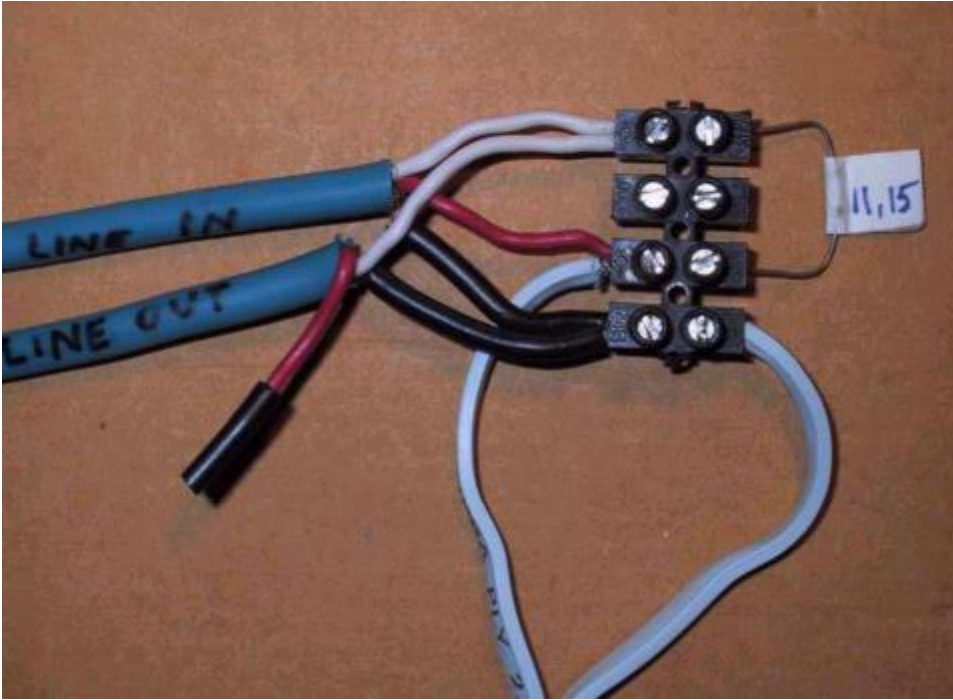


As you can see the JIG is very easy to build and we can use it as LOOP or IMPEDANCE CABLE just changing a piece of cable with a resistor.

The loop cable is connected as below:



The Impedance Cable is used for impedance measurement and is obtained by connecting the reference resistors (marked 11,15):



This is the complete JIG: the 2 blue cables with mini-jack are connected to the sound card (LINE IN and LINE OUT) while the light blue cable with two black cables fastened are where the device under test (DUT) is connected:

Note: This JIG doesn't have any divider resistor so be careful: if you have an amplified sound card you can fry it during the volumes setup: the modern sound cards are just pre-amplified and supports up to 1.5/2.0 volts input, so no risk of damaging them; while old SC (Sound Blaster AWE) are amplified so check it before connecting and be very careful during the volume adjustment.

Note: If your SC has a front module, do not connect it to the LINE IN and OUT but rather use the back connectors, as the front panel connections are noisier.

STEPS TO DESIGN AND BUILD A SPEAKER OVERVIEW **(DIRECTLY FROM THE SPEAKER WORKSHOP SITE)**

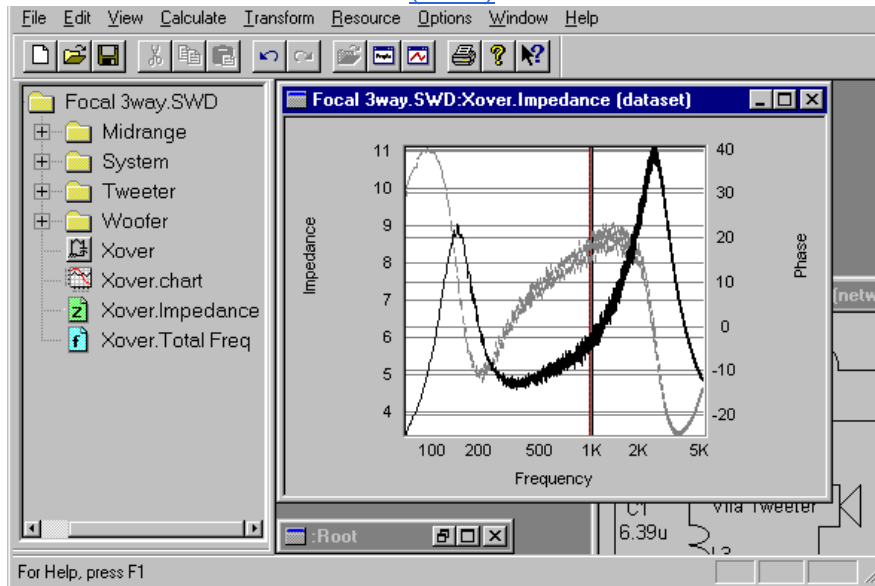
[\(TOC\)](#)

Designing a speaker can be broken down into a few steps.

- Select Speaker Type The most common types of speakers are 2-way (woofer and tweeter), 3-way (woofer, midrange and tweeter) and subwoofer (just a big woofer in a box).
- Select Drivers Based on the type of speaker, pick drivers to attain a flat amplitude vs. frequency response as well as a good impulse response. You can prototype low frequency response (woofer and midrange) using the Enclosure commands before buying these drivers.
- Buy the Drivers Purchase the drivers and wait for them to arrive. Create Driver resources to hold the measurement data.
- Measure the Drivers For the woofer and midrange measure impedance in free air and in a prototype box. Calculate T/S parameters for the woofer and midrange. Match drivers as closely as possible for stereo pairs to improve imaging.
- Design the enclosure Use the T/S parameters to design a final enclosure for the drivers. Build one.
- Measure the Drivers Mount all of the drivers in the final enclosure and measure their impedance and frequency response.
- Design the Crossover Using the measurements from the real enclosure use the Network commands to prototype a crossover.
- Test the Loudspeaker Use the Signal commands to verify operation of the crossover with a load resistor, then attach the crossover to the drivers and test the frequency response of the system, splicing near field and far field measurements.
- Test the System There are two things you can do here: either evaluate in room measurements from a listening position (custom for the room) or evaluate distortion and time and phase domain measurements in a pseudo anechoic environment to assess for general accuracy of the speaker.

PAGE SETUP (FROM SPEAKER WORKSHOP HELP FILES)

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On the left is the Project Tree. This is movable and resizable. Double click an item in the project tree to open its window on the left. Right click to get a menu of options. Drag to move resources from one folder to the next.

On the right are the resource (item) windows. Windows generally show graphics (called Charts). Any chart can show more than one data item and charts are easy to manipulate with the mouse. Charts are limited to 2 billion data points.

On top is the Menu. The inner entries of the menu will change depending on which view is on top (which resource you are looking at). In addition to the general commands (such as file save) each resource type has a set of resource specific commands (such as Record for signals).

Just below the menu is the Speaker Workshop Toolbar. These buttons are shortcuts to commands from the menu. There is a secondary toolbar that shows up depending on which resource you are currently looking at. This toolbar contains tools specific to the resource – it is called the Resource Bar.

Below the view is the Status Bar. The far left portion of the status bar shows help about the current command. Next to the help are 5 fields. When you move the cursor around in a chart the first three fields show Frequency, Amplitude, and Phase. The next two fields are only used when you move around in an impedance chart. In that case the fields show equivalent impedance. The first field is capacitance / inductance. The second field is the series resistance.

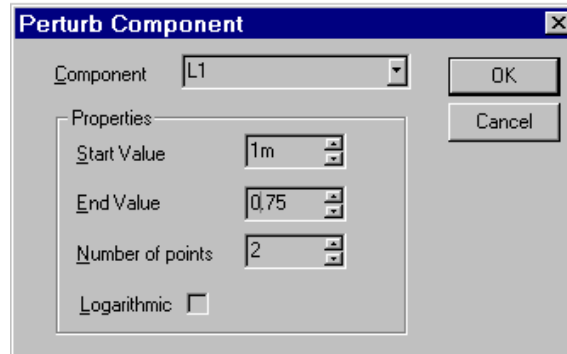
The Project Tree and the Toolbars can be detached, moved, and/or hidden easily. To detach a bar pick it up by an area outside its display area and move it into the view

window. To attach it back again just drag the bar into its original docking area. To hide a view, look at the View menu.

DATA ENTRY

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Speaker workshop has two types of entry data - integer and floating. For example, look at the perturb dialog...



In this dialog, the Start and End entries are floats and the Number of Points entry is an integer.

Integer data Enter the number manually or use the spin button to scroll the number up or down.

Float data Use the spin button to scroll the number up or down, or enter the number manually using any valid float data format.

Float data formats:

You can enter floating-point numbers using any of the following suffixes

Suffix Name Value Alternate

M	Mega	Million	
K	Kilo	Thousand	k
m	milli	1/1000	
u	micro	1/1,000,000	U
n	nano	1/1E9	N
p	pico	1/1E12	P

For example, the following numbers all mean 1500...

1500
1.5K
1.5k
.0015M
1500000m

EDITING CHARTS WITH A MOUSE

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Much of your time with Speaker Workshop is spent looking at charts and graphs. To make life simpler, there are a number of things you can do with the mouse...

- Press and drag, then release inside the data area.* This will draw a rubber-band rectangle. When you release, the chart range is set to the selection (Zoom-in).
- Right click* This brings up a menu of standard operations. To compare two graphs on the same chart, Right Click/Add/Select the other folder with the comparator graph in it.
- Click the border* This brings up a box with handles. Drag a side handle to change the X range. Drag a top/bottom handle to change the y range.
- Click a dataset* This highlights the dataset and enables Edit Delete.
- Double-click a dataset* This brings up the Chart properties dialog, with the clicked dataset ready for editing.
- Double-click a scale* This brings up the chart properties dialog, with the selected scale ready for editing.
- Press and drag a marker* A vertical rubber band line is drawn during this operation. This moves the marker to the dragged location.
- Double click the title* This brings up the chart properties dialog with the title highlighted.

Note the border click - drag is often used to change the max/min frequency range while allowing the Amplitude range to be automatic.

Of course, the location of any dataset is available just by turning on the (View) Locations bar.

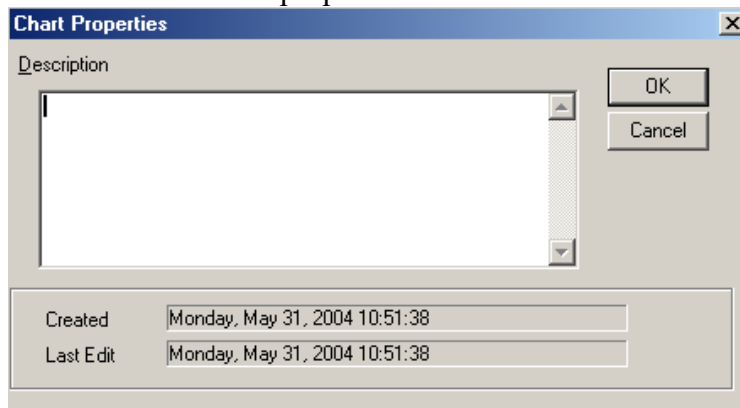
ADDING NOTES TO CHARTS (ANNOTATION)

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Much of your time with Speaker Workshop is spent looking at charts and graphs. It is easy to add notes to charts so that you can keep track of specific issues with datasets.

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

To add a note to a chart. Select Chart / Note (in the menu) or use the right mouse button/properties and add the note...



To edit an existing note Double-click the note in the chart

To delete a note Select the note and use Edit / Delete (or right mouse button Delete from the menu)

To move a note To move a note on-screen highlight the note (click on it) then press inside the box or on the box and drag the box where you want the note to be placed

To change a note frequency (time) spot Double-click the note and change the frequency/time in the Note Properties dialog.

To get multiple line notes Press Ctrl+Enter at the end of the line in the Text edit box to get a new line

Notes are text box areas that are attached to a specific frequency (or time) in a dataset. When you create a note you select what frequency to attach it to. The text box then has a line that goes from the box to the data point you have selected. You can move the box and the line will stay connected to the data point. You can move the frequency (via Edit Properties) as well.

If you just want a note in a chart you can turn off the connecting line, but the box is always defined at a specific frequency (time).

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

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

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

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

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

Note Properties Dialog

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

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