

SECTION 2 – CALIBRATION

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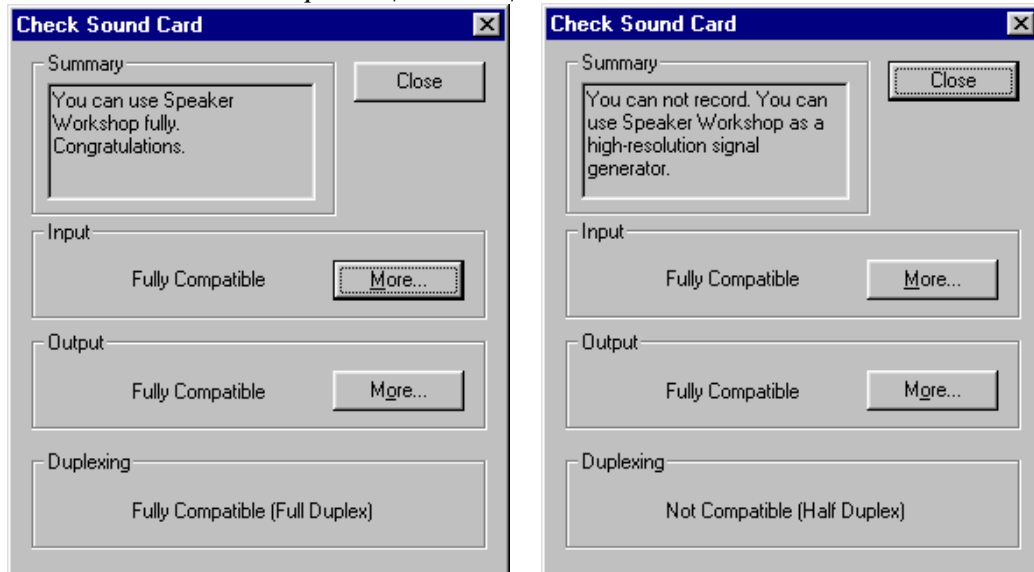
SECTION II - CALIBRATION

[\(Section Index\)](#)

SOUND CARD FUNCTIONALITY CHECK

SW needs to simultaneously play and record through the soundcard, an operating mode known as "full-duplex". To check to see if your sound card will support this mode, in SW:

1. Select the menu item *Options / Wizard / Check Sound Card*



Note: If the wizard reports back that your card isn't capable of full duplex operation, it may be just a matter of downloading and installing the latest set of software drivers from the web in order to enable this feature. If that doesn't do it - it may be time for a new soundcard if you want to do any meaningful testing with Speaker Workshop. See the Appendix for an Excel Spreadsheet of Soundcard functionality with Speaker Workshop (compiled by Claudio Negro)

SETTING SOUND CARD AND COMPUTER CONTROLS

(Section Index)

Sound Cards should be calibrated every time you switch the sampling rate or the MLS length as anti-aliasing filters in soundcards may otherwise give rise to ripples in the frequency response. For Impedance, small errors in balance between channels may give rise to large errors in measured values.

Always allow your computer and sound card to warm up for at least 15 minutes prior to performing the set up procedure.

This is saved in System folder.

Use Claudio's hook-up or use the Wallin jig as demonstrated above but without a jumper from BP3 to BP4 unless you have an old sound card that has it's own amplifier. If you are using an external amplifier and an output voltage divider with either set up, attenuation should be used when using an external amplifier to help to protect the sound card and all calibration should be done with this in place.

It is recommended to use non inductive resistors for calibration as inductance may falsely make resistors appear as if their values are larger yielding less accurate results.

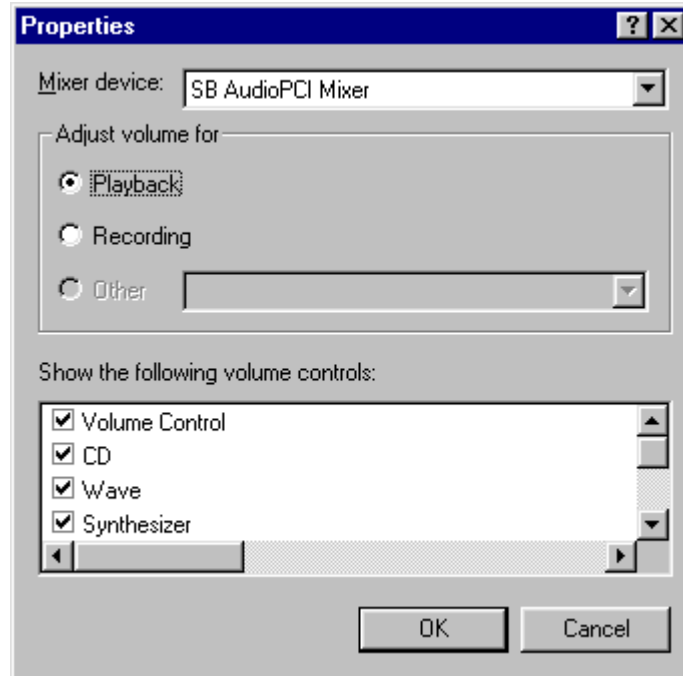
1. File/Save As "Test"
2. Close SpkrWk1
3. File/Open Test.swd

Windows adjustments

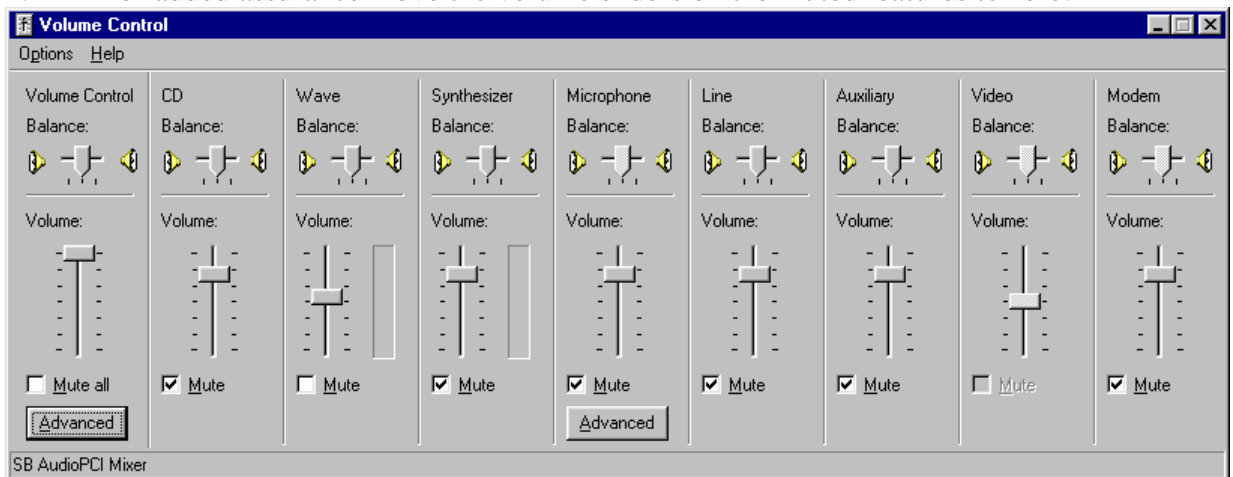
1. Go to Windows Control Panel/Multimedia/Audio/Advanced Properties (if present)
2. Set for full Acceleration
3. Set for the best available sample rate conversion for both record and for playback.

Volume Control

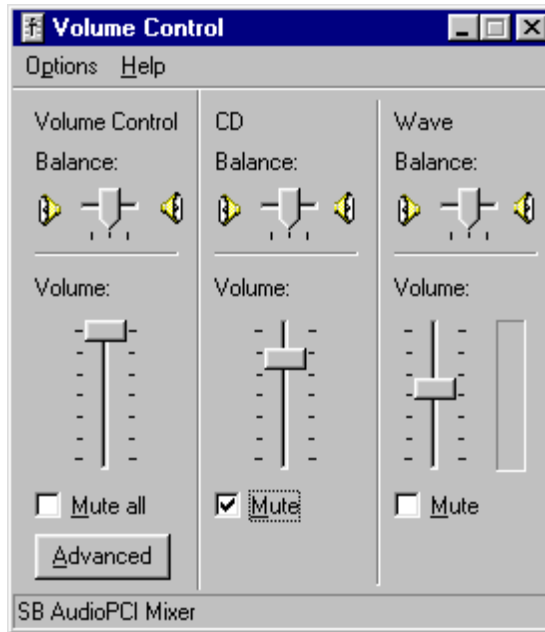
1. Open windows mixer (volume control) by either clicking on the little speaker in the lower right of the screen or Start/Programs/Accessories/Entertainment/Volume Control
2. Options/properties/open all boxes by clicking on all of the boxes.



3. OK/mute all boxes except volume control and wave.
4. For added assurance move the volume sliders on the muted features to zero.



5. Make sure that the Mute all box is unchecked on volume control and on wave. For some cards, you may need to mute the volume control, as well.
6. Hide all boxes except volume control and wave
7. Make certain that nothing is plugged into the speaker or headphone jack of the card as this may mute the line out.



8. Click on Options/Advanced and make sure that any bass, treble or tone controls are centered and that enhanced features are turned off, if present.



Note that for Soundblaster Live, it is reported at <http://www.libinst.com/SoundCardCal%20SBLive.htm> that tone controls need to set

as follows to provide a flat response:



9. Set the volume control to the maximum; SW will set the Wave control.
10. Do not use any volume attenuation on the jig during volume set up.

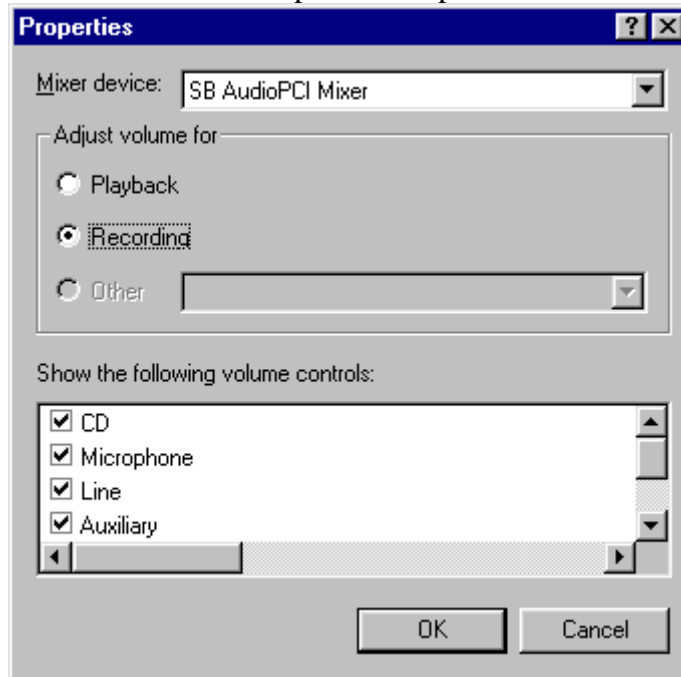
Note: if you are using a Sound Blaster Card, you must go to the Sound Blaster Control panel and mute the “What-U-hear” option. As a double precaution, bring the volume slider down to zero. For some sound cards, you may have to disable “What-U-hear” for sine waves and enable it for MLS signal recording.

For Audigy II ZS, you must choose Control Panel/Recording Control panel/Advanced /Select Analog Mix as your recording source and “record without monitoring”.

For further assistance with Audigy Cards, go to <http://audio.rightmark.org/downloads/audigy-setup.zip>

Record Control

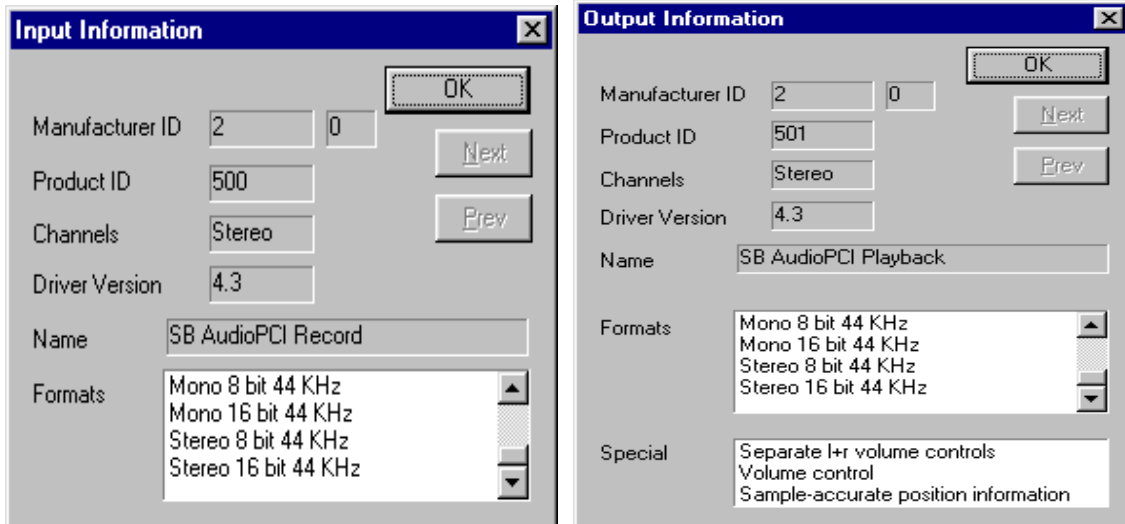
1. Open windows mixer (volume control) 2nd copy (as above)
2. Options/properties/recording [this will change the name to recording control]
3. Display all boxes by placing a check in all of the boxes.
4. Set the volume control for all options except Line In to zero.



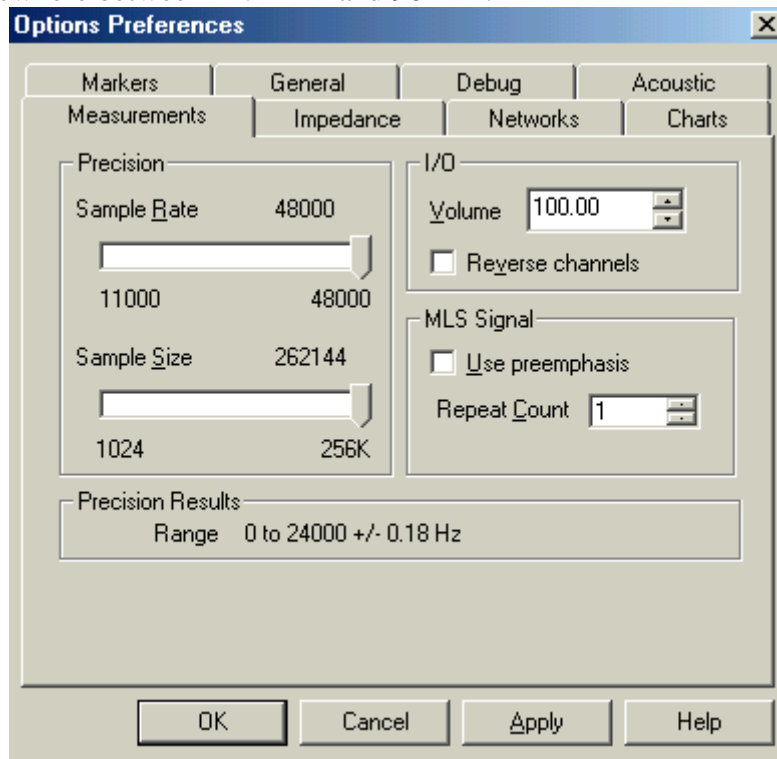
5. Select only Line In and deselect all of the other options. (Note for Audigy 2 ZS cards: substitute the Analog Mix for Line In)
6. Turn the volume all the way up.
7. Go to Options/Advanced
8. Center Base/Treble controls and line balance controls.
9. For Audigy 2 ZS cards, Select Record without Monitoring
10. Hide all controls except Line In.

In Speaker Workshop

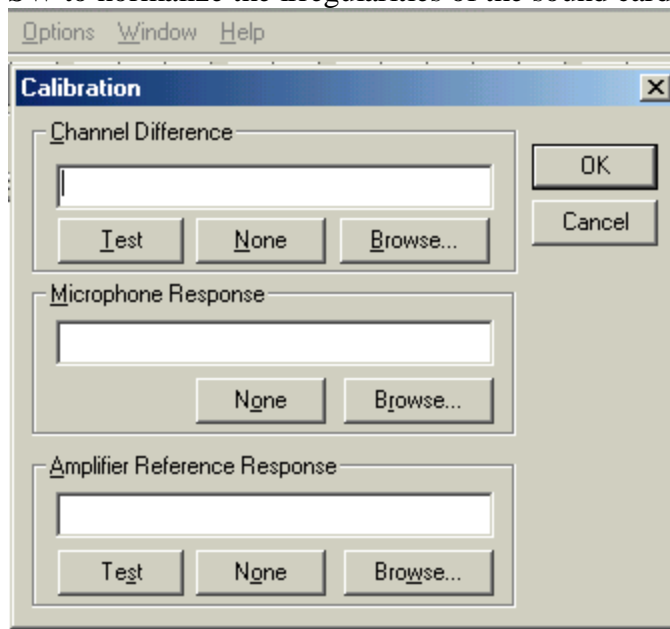
1. For audigy 2 ZS cards, go to Options/Preferences/General make certain that the 96kHz sampling rate is checked. If it is not, select it, close the program and then re-open the program.
2. Go to *Options/Wizard/Check Sound Card* to open the soundcard wizard
3. Click "More" for the input and output information
4. Look at the bottom of the Formats box for maximum Input and Output.



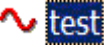

5. You will learn the maximum number of bits and frequency at which your card will sample. Copy these numbers down.
6. Close the Wizard.
7. Go to Options/Preference/Measurement
8. Move the cursor of the sample rate and sample size all the way to the right
9. Make certain that I/O volume is at 100 (this regulates the wave control in the windows mixer)
 - a. Leave Reverse channels unchecked unless you have the cables reversed for input and output
10. Check the maximum frequency sample rate as noted above. It is likely to be somewhere between 44.1 kHz and 96 kHz.

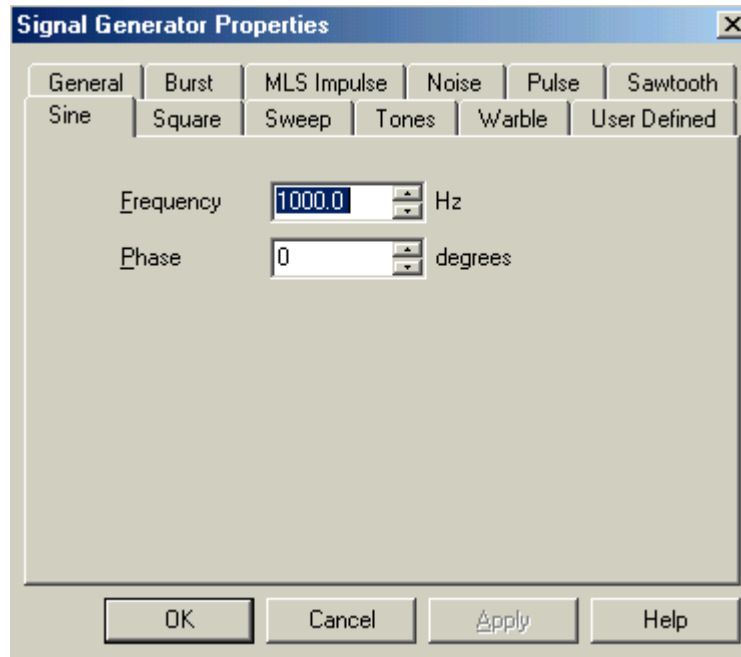


11. Set the sample size to 256K.
 - a. If measurement signal amplitudes are too low, you can try to lower the sample size in increments and re-check the pulse level.
12. Under the MLS signal box, leave use Pre-emphasis unchecked for now. Using this will lower the energy in the highest frequency ranges and will give lower frequencies more relative power.
13. Leave the repeat count at 1 for now. The sample time will increase by a factor of this times the sample time. Lengthening the measurement duration in this manner will not increase frequency resolution but does increase the signal to noise ratio. This option is useful for acoustic measurements.
14. Close this window
15. Options/Calibrate/Amplifier Reference Response/Test in loop mode to allow SW to normalize the irregularities of the sound card.

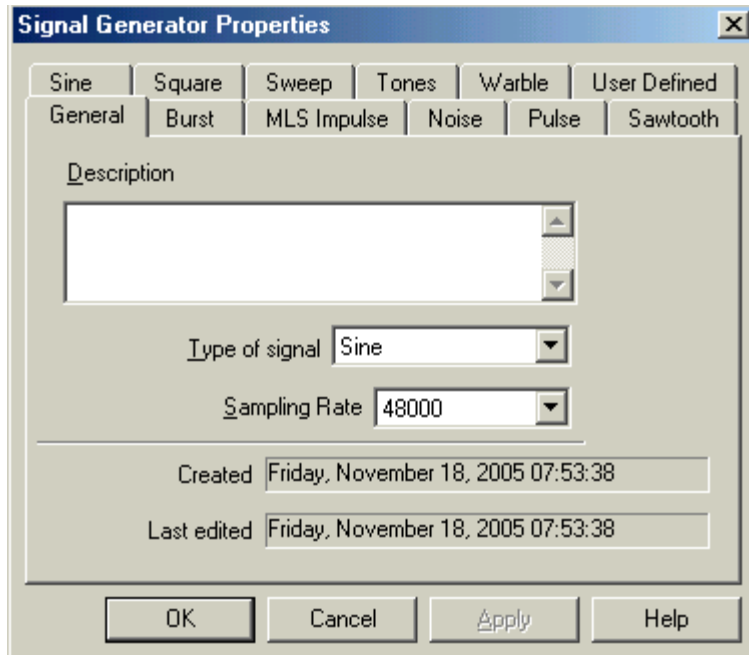


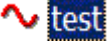
16. After the test, it will by default be named Measurement.Pulse.
17. Rename this to Soundcard.Measurement.Pulse.
18. Though some sound cards may require an amplifier to get accurate Impedance measurements however I suggest trying this without an external amplifier first. If you do require one, run the Amplifier Reference Response with the Amplifier in the loop but be very cautious about volume so you don't destroy your sound card.
19. Create a new folder and name it "Projects".
20. Right Click on this directory
21. New/Folder
22. Name this "First"
23. Right Click on this new directory
24. New/Folder
25. Name this folder "Calibration"
26. Click on Save.
27. Close the window named "root".

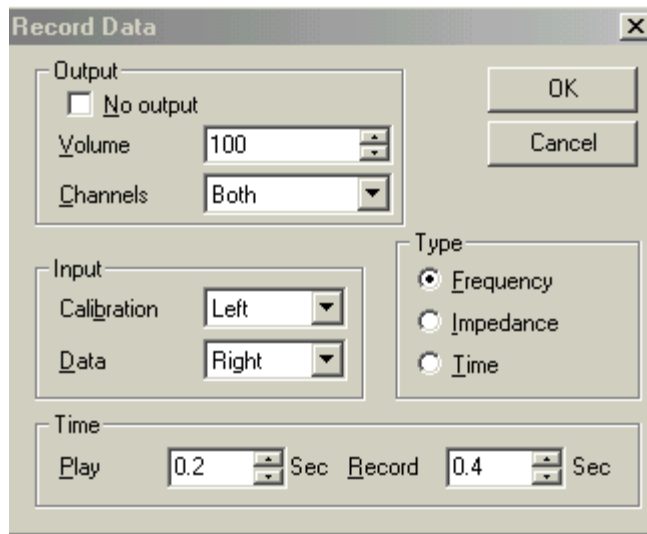
28. Click on the Menu/File/Calibration, which will open up a window named “Calibration”.
29. Highlight Calibration on the design tree.
30. Click Resource/New/Signal (creates a new signal)
31. Name this signal “test”
32. 2 new files will appear:
 - a. test 
 - b. test.output  Test.Output
33. Cut and paste into the calibration folder
 - a. Right click on the chart
 - b. Go to properties
 - c. Go to the individual signal tab of the type of signal that you want to generate. For sound card calibration, a sine wave of 1kHz is recommended.



- d. Set the options you want to use
- e. Go to the general tab and select that signal type.

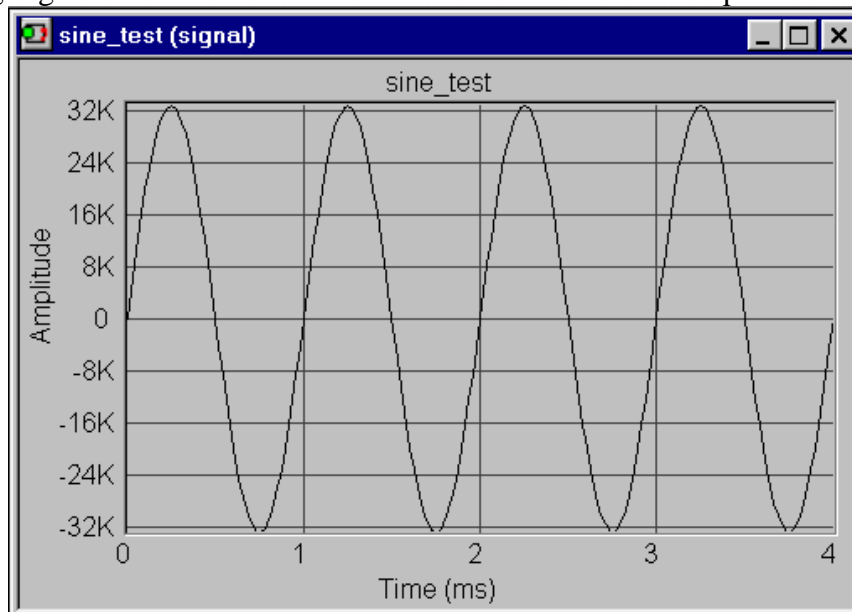


- f. Select the desired sample rate
 - g. Exit out of the dialog box and it will remain on the last signal selected in the general tab.
34. Connect Line Out and Line In of the Sound Card using a loop cable or loop connect with the jig.
 35. Make certain that the jig is set for Direct mode (SW1 Center/SW2 Rt/SW3 Down).
 36. Set Mixer Volumes (windows) to a mid low level. *Failure to start without the volume settings turned down in both play and record can result in a damaged card. Adjustment can be made after the initial test recording once you know what levels you are getting during the test.*
 37. Double click ~test  /sound/record
 38. Values should be set as follows:
 - a. Volume 100
 - b. Calibration Left
 - c. Data Right
 - d. Type Frequency
 - e. Time: Play 0.2 sec
 - f. Time: Record 0.4 sec (*Note: always want to assure that the Record Time is longer than your play time in order to catch the entire signal. This will be helpful in setting latency and in finding the start and stop of the signal response.*)



37. Click OK and 4 files will be created in the calibration file in addition to test and test.output:
- a. Test.in.l
 - b. Test.in.l.fft
 - c. Test.in.r
 - d. Test.in.r.fft

38. Highlight and click on “test.in.l” and double click into the open chart

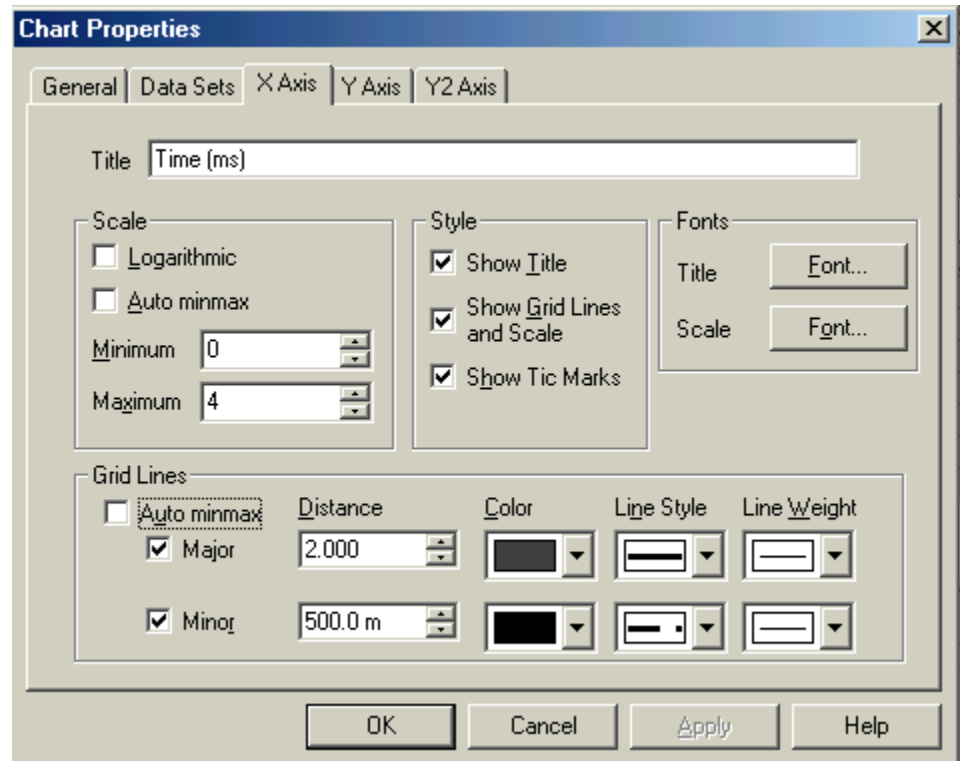


39. Right click on the chart and Chart Properties will appear.

40. Select the X tab

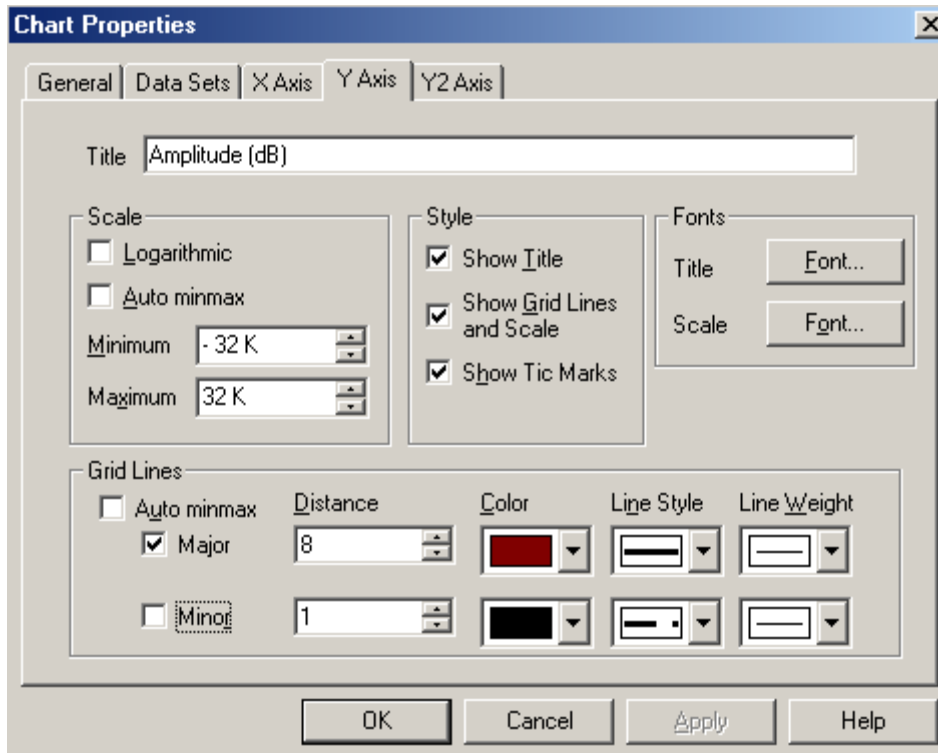
- a. Scale
 - i. Minimum 0
 - ii. Maximum 4
 - iii. Major Gridlines 2.0

iv. Minor Gridlines 500 m



b. Y axis

- i. Minimum -32k
- ii. Maximum 32k
- iii. Major Gridlines distance 8k
- iv. Don't check Minor
- v. Make sure that all Auto Minmax boxes are unchecked.
- vi. Click OK and the chart will appear as above with 4 Sine Wave cycles.



41. Right click on the chart area and make this the chart default
42. Open test.in.r
43. Adjust chart dimensions to fit in the desktop
44. Menu/Options/view/check the Vu meter to enable volume meters in the bottom left of the screen.

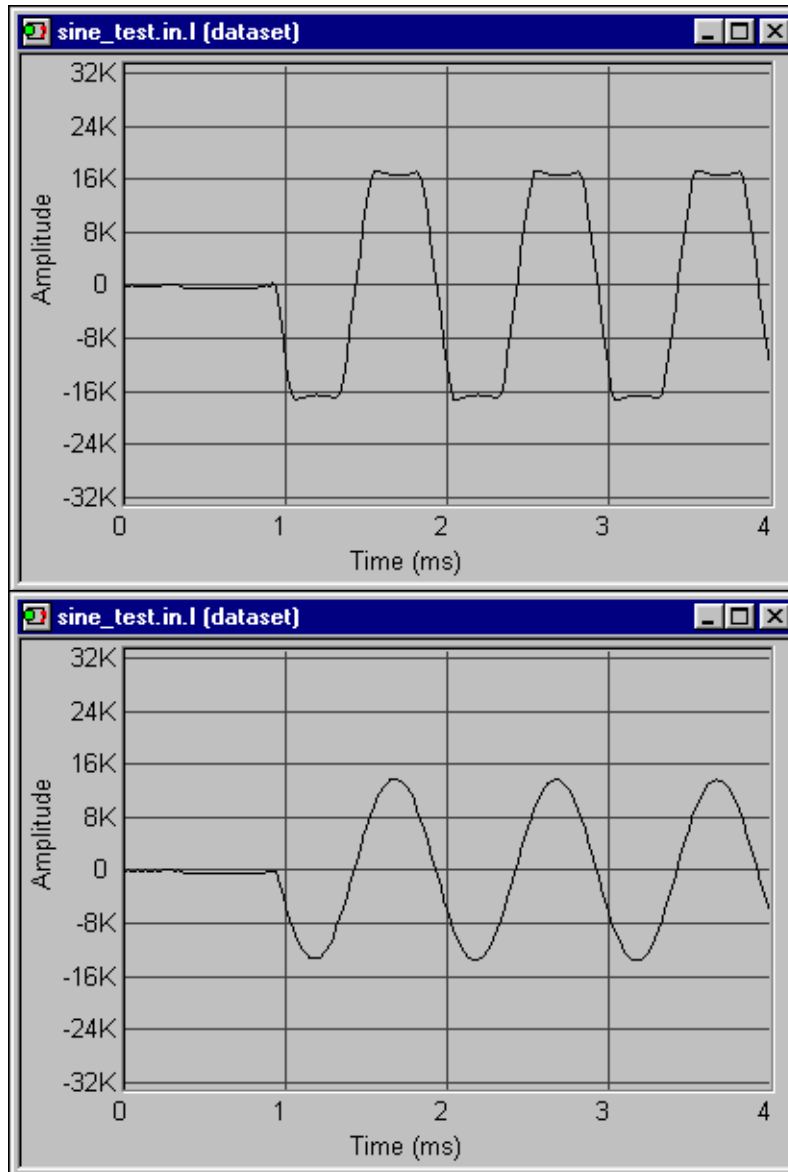
	Maximum	Average Abs	Minimum
Left	553	104	-313
Right	553	104	-313

45. Highlight test file and set it for loop
46. Set the volume to see a wave without clipping by continually increasing the volume until the wave just clips then lower it a little. *Note: some cards can be difficult to adjust using only master play volume control. You may have to adjust the line in volume (recording) and the I/O volume in SW (options/preferences/measurements) to get a sine wave that does not show signs of clipping.*

Note: An Option here is to check harmonic distortion as well.

- i. Set up the card in a loop connection from output to input
- ii. Highlight the signal window
- iii. Measure/Harmonic Distortion
- iv. Choose Spectrum at 1 power
- v. Click "OK"
- vi. This produces a window that shows various harmonics along with noise and measurement error.

- vii. Look at the 2nd and 3rd harmonics, in particular. Good sound cards can attain less than 0.1% distortion and often lower than 0.05% for the 2nd harmonic.



- 47. Note the volume readings; this is the reference level; a good target would be between 17K and 27K.
- 48. SW cannot reliably recognize the start of acoustic data if VU is below 2K.
- 49. Download QuickMix from <http://www.msaxon.com/quickmix/index.htm>



QuickMixIn.EXE

- (Terratec Sound Cards have this type of feature built into it).
- 50. After installing QuickMix: Once you have the mixer in the state you want, run QuickMix from the Start menu.

51. The checkboxes on the QuickMix panel allow you to select which mixer channels you want to save. There is usually at least a 'play' channel and a 'record' channel - the exact names vary from driver to driver - but there may be others.
52. Once you have chosen a setting, click Save//Choose a file name (the default extension is .qmx) then click 'Save' to save the settings in that file.
53. To restore a saved mixer state, you can just double-click on a .qmx file and QuickMix will restore that state. (It may be convenient to set a short cut to QuickMix).
54. You might want to name and make another file to return the mixer settings to their previous state when you are done.

Note: Adjust volumes lower than the clipping reference level (probably in the range of 23-27K) and save with QuickMix. It might be a good idea to save other volumes as well at perhaps 20K and 15K to compare which level works best for you later.

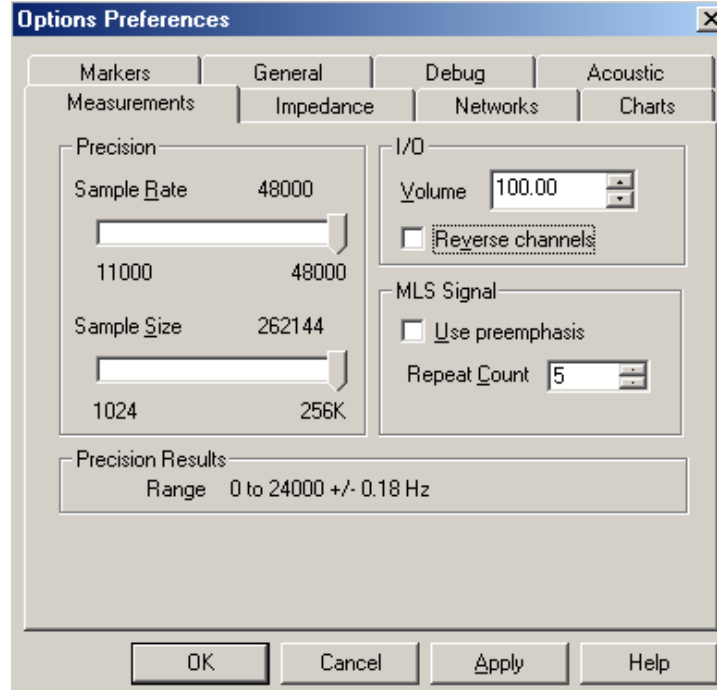
CALIBRATION OF THE SOUND CARD

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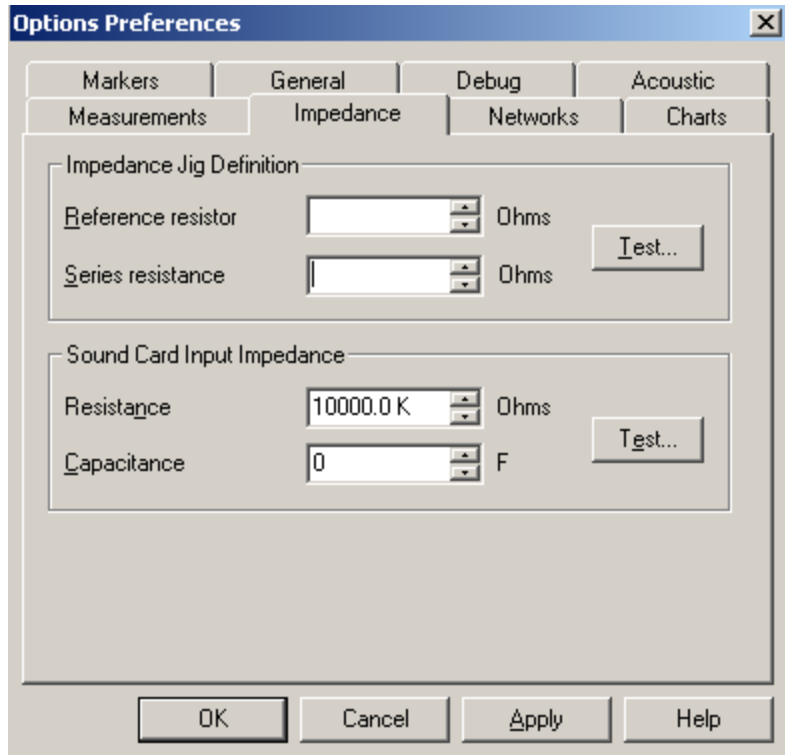
Manufacturer's specifications can be entered in the appropriate places if you would prefer. Some believe that these will be more accurate than measured specs though I disagree. I believe that using calibrated equipment better tailors testing to your own equipment and not to production line specifications. Sound Card Impedance will generally be in the 45-50K range.

Reference Resistors are best if they are the non-inductive type as the inductance in typical resistors may make the resistor seem larger than it actually is and throw off some calibrations.

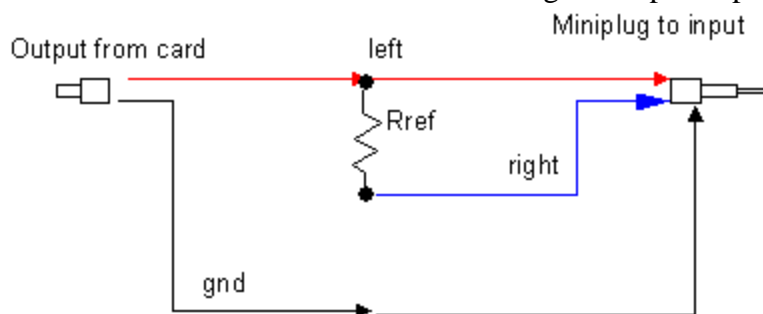
1. Select the chosen volume set through Quick Mix
2. Open SW/Options/Preference/Measurements/Select the chosen sample rate.
3. Set the sample size to the Maximum.
4. Set Repeat Count to 5.



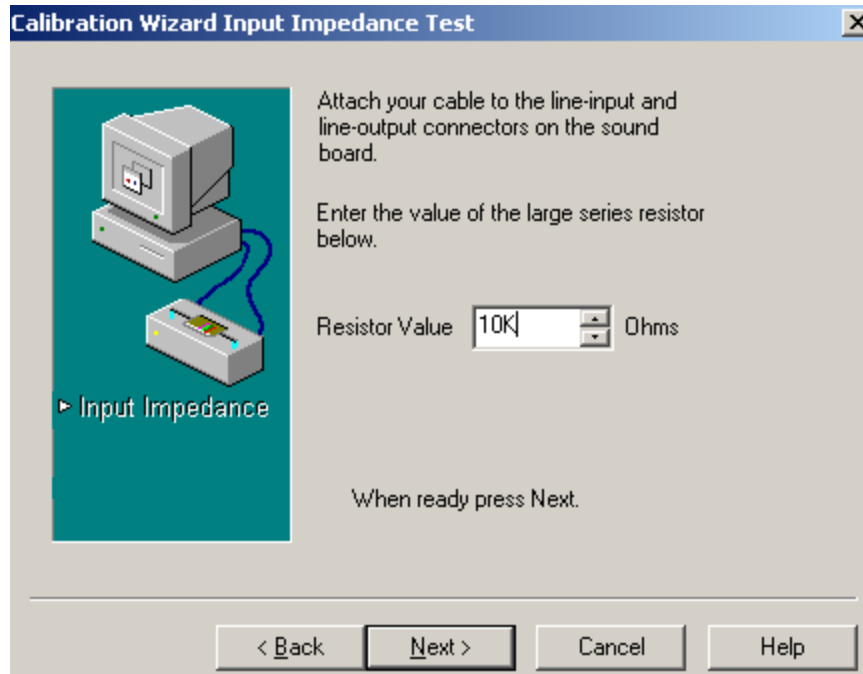
5. Click on Impedance Tab.



6. Connect the Impedance Cable to the Sound Card
7. Measure the DMM cable resistance and use this number to subtract all DMM resistance measurements (common DMM impedance is in the range of 0.3 to 0.6 ohms).
8. Measure a 10,000-Ohm Resistor with an accurate DMM.
9. For the Wallin Jig II, connect the 10,000-Ohm resistor from BP3 to BP4.
10. For the Cable (Claudio Negro) Jig, replace the loop connection with the 10 K Ohm resistor.
11. Under Sound Card Input Impedance, Press Test and a Calibration Wizard Impedance Jig Test will come up.
12. A window will come up indicating that this wizard will measure the input impedance of your sound board (card). It suggests using a cable with a short from the left output to the left input and a large (approximately 10K resistor from the left output to the right input. The results of this test for resistance and capacitance will be stored in the Preferences settings for input impedance.



13. Press Next
14. Enter the value of your measured 10 K Ohm resistor



15. Press next and SW will provide measured values for the Resistance and Capacitance Values of the Soundcard. The resistance reported is the Soundcard Impedance. Impedance values should be in the range of 40-50K for typical soundcards and near 10K for professional cards.
16. Select Finish and you will see that these numbers are automatically filled into the Sound Card Input Impedance Resistance and Capacitance Boxes. See the note below if you are using a voltage divider!!
17. If these values do not appear to be reasonable, you could use “0” for the capacitance and enter the value of the resistance listed in your users manual for the sound card. The capacitance of the sound card is unlikely to cause significant problems at lower frequencies but may affect the measurements at higher frequencies if the program does not have this value.

If you are using an amplifier with a voltage divider, you must calculate this in. The formula is

$$Z_{total} = R_1 + (Z_{SC \text{ impedance}} * R_2) / (Z_{SC \text{ impedance}} + R_2).$$

For a voltage divider with resistances of 900 ohms and 100 ohm and a sound card impedance of 47015 ohms, you would get the following:

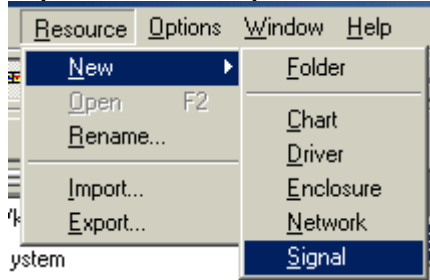
$$Z_{total} = 900 + (47015 * 100) / (47015 + 100) = \mathbf{999.79}.$$

In this case, you would manually enter (and override the values automatically entered by the Wizard) this value in the Sound Card Input Impedance instead of the Resistance value that appears on the screen Calibration Wizard Input Impedance Test.

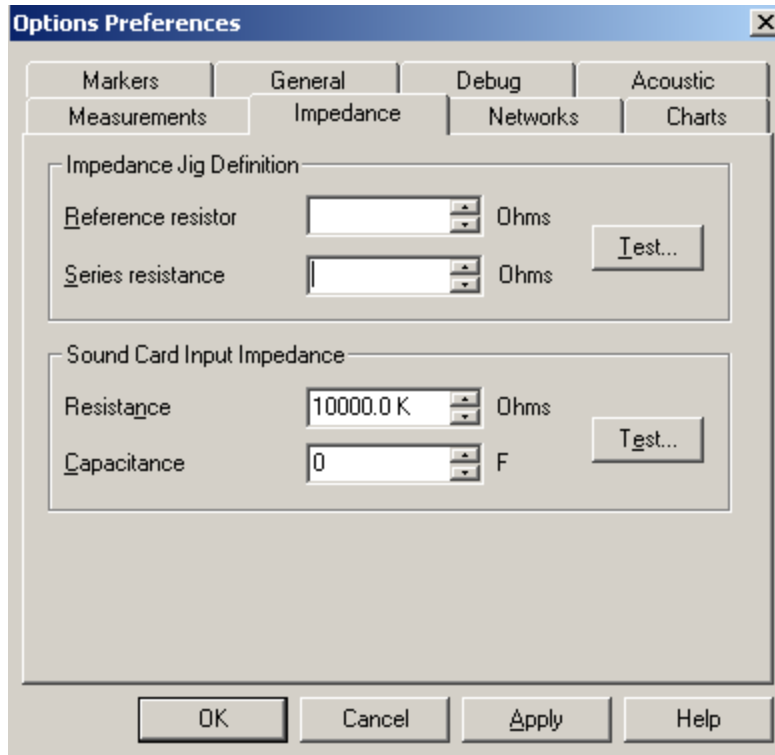
SOUND CARD CALIBRATION ALTERNATE METHOD WITH WALLIN JIG 2

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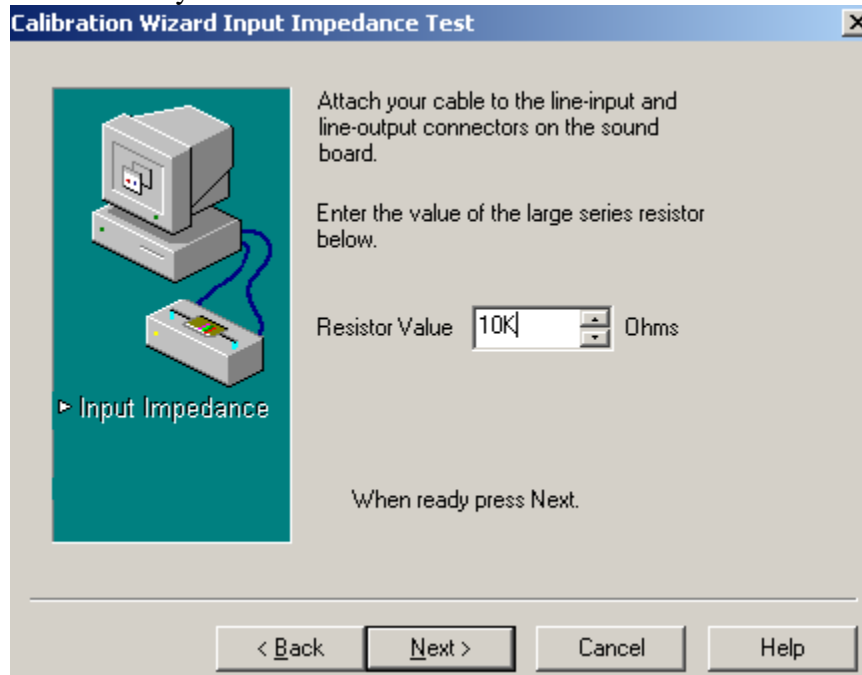
1. Connect line out (left) of the soundcard to CD in on an amplifier using a stereo 3.5 mm jack to rca stereo cable with the red (right) jack not connected.
2. Left speaker out of amplifier using a regular speaker cable goes to BP1 and BP2 on the jig
3. J2 and J3 on the jig go to line in on the sound card (left and right, respectively) using an rca stereo to stereo 3.5 mm jack adapter.
4. Set the soundcard in and out to max volume for most soundcards
5. Turn off the volume of the amplifier prior to turning the amplifier on and use the Direct mode on the amplifier (bypassing tone controls) if it has one or set tone controls to neutral.
6. Set the jig to Direct.
7. Record a Sine Wave in Speaker Workshop.



8. Gradually raise the volume of the amp until you reach an unclipped 25-27K sine wave.
9. Do not touch the volume on your amp after calibration.
10. Select Preferences/Impedance
11. Put the jig into Impmeas mode
12. Measure a 10K resistor accurately as your reference resistor
13. Place the resistor between BP3 and BP4
14. Under Sound Card Input Impedance Definition, Press Test and a Calibration Wizard Impedance Jig Test will come up.



15. A window will come up indicating that this wizard will measure the input impedance of your sound board (card). Ignore the suggested hookup on the page that comes up. The results of this test for resistance and capacitance will be stored in the Preferences settings for input impedance.
16. Press Next
17. Enter the value of your measured 10 K Ohm resistor



18. Press next and SW will provide measured values for the Resistance and Capacitance Values of the Soundcard. This will be stored and used by SW. The

- resistance reported is the Soundcard Impedance. Impedance values should be in the range of 40-50K for typical soundcards and near 10K for professional cards.
19. Select Finish and you will see that these numbers are automatically filled into the Sound Card Input Impedance Resistance and Capacitance Boxes. See the note below if you are using a voltage divider!!
 20. If these values do not appear to be reasonable, you could use "0" for the capacitance and enter the value of the resistance listed in your users manual for the sound card. The capacitance of the sound card is unlikely to cause significant problems at lower frequencies but may affect the measurements at higher frequencies if the program does not have this value.

If you are using an amplifier with a voltage divider, you must calculate this in. The formula is

$$Z_{total}=R_1+(Z_{SC\ impedance}*R_2)/(Z_{SC\ impedance}+R_2).$$

For a voltage divider with resistances of 900 ohms and 100 ohm and a sound card impedance of 47015 ohms, you would get the following:

$$Z_{total}=900+(47015*100)/(47015+100)= 999.79.$$

In this case, you would manually enter (and override the values automatically entered by the Wizard) this value in the Sound Card Input Impedance instead of the Resistance value that appears on the screen Calibration Wizard Input Impedance Test.

SETTING UP AN MLS SIGNAL

[\(Section Index\)](#)The MLS signal will be used for your impedance measurements and for your acoustic measurements. Note: Speaker Workshop uses MLS (Maximum Length Sequence) rather than sine waves for most of its measurements (unless you dictate otherwise). It is generated from a relatively simple algorithm that looks like random (white noise) but it is, in reality, periodic with a very long repeat cycle. MLS has a flat energy vs. frequency curve over a wide frequency range and it possesses a low peak to average power ratio (a low crest factor) and produces a good Signal to Noise ratio.

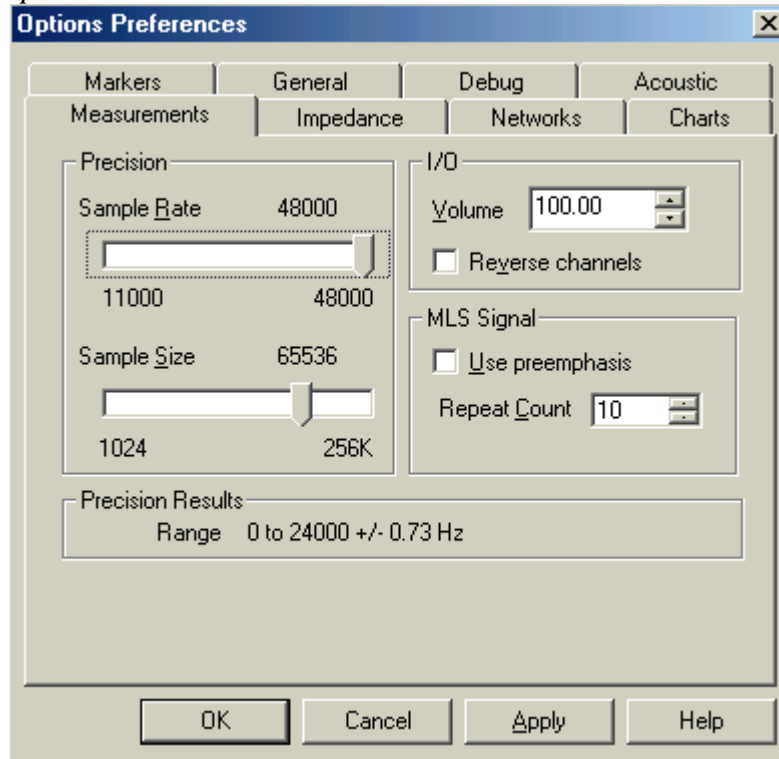
You will have to set up your sample signal type and size prior to doing a sound play or sound record.

Note: The Fast Fourier Transform (FFT) is a mathematical technique in which a voltage vs. time (time domain) signal is converted into a dB vs. frequency (frequency domain) signal. This frequency domain helps to explain the effect of frequency on the shape of the waveform. The speed and resolution of the analysis is based upon the speed of the computer, the FFT size, and the sampling rate. A sampling size of 48kHz and 4096 data points will lead to a time constant of $4096/48,000$ or .085 seconds. The frequency resolution equals the inverse of this number or $1/.085$, which equals 11.76 Hz. The frequency resolution will be better with more data points however the calculations will take longer. Accurate measurements are dependent upon choosing the largest number of data points that your time allows. If the decay time of the signal is one second and if you are using a sample rate of 48kHz, there will be 48,000 data points. In sampling for audio signals, it is generally best to use twice as many audio points as the highest frequency you wish to measure; in other words, if you wish to measure up to 20,000 Hz, you must use at least 40,000 data points. A sampling rate of 48kHz will accurately measure to 24,000 Hz.

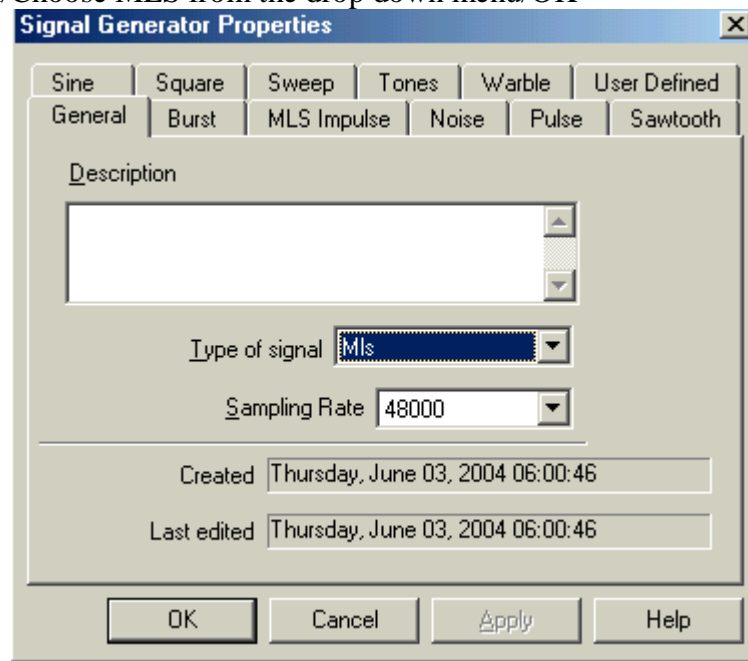
1. Resource/New/Signal
2. Name the signal, "MLS"
3. Right click /properties/MLS/OK/ and set for a loop.
4. Go to Options/Preferences/Measurements/
5. The Sample Size will vary depending upon your needs. Larger sample sizes provide better frequency domain resolution but take longer to process. Larger sizes will be particularly important in lower frequency measurements where resolution is more problematic and the signal is somewhat less susceptible to noise corruption. For higher frequency measurements, a target of around 32K might be reasonable as this provides a good balance between noise and resolution. For a system measurement, using between 32K and 64K is reasonable.
6. The Sample rate should be at the maximum sample rate that your sound card can support.
7. Repeat Count should be set as high as you wish however there are some caveats:
 - a. The more repeats, the longer the test time will be
 - b. The more repeats, the better the Signal to Noise ratio will be

- c. The more repeats, the more likely you are to capture a delayed room reflection or noise though this will generally be diluted through measurement averaging as long as the noise is not cyclical.

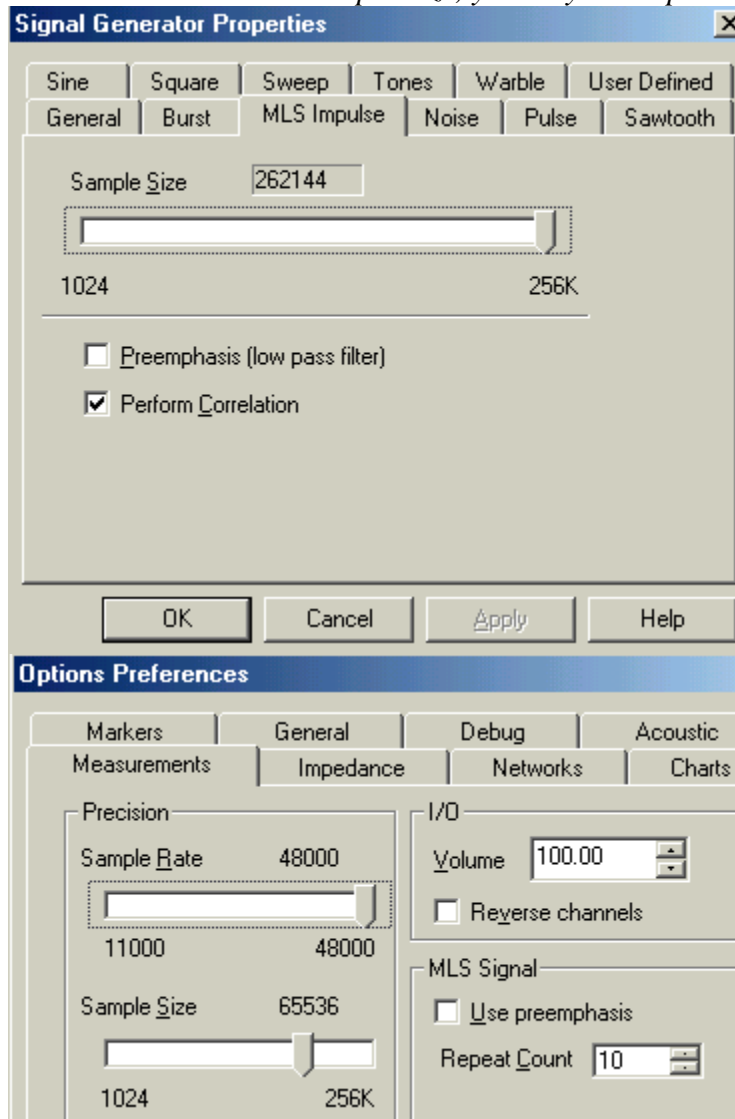
Note: The Precision Results, which will tell you how precise your measurements can be with different options.



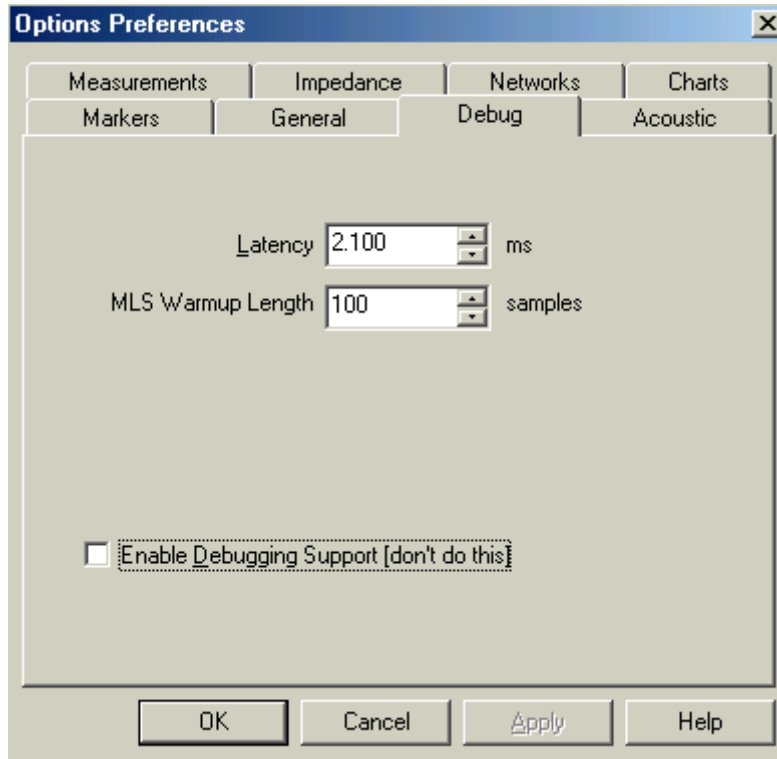
8. With the Chart Active, Right Click/Properties/General tab/Type of Signal/Choose MLS from the drop down menu/OK



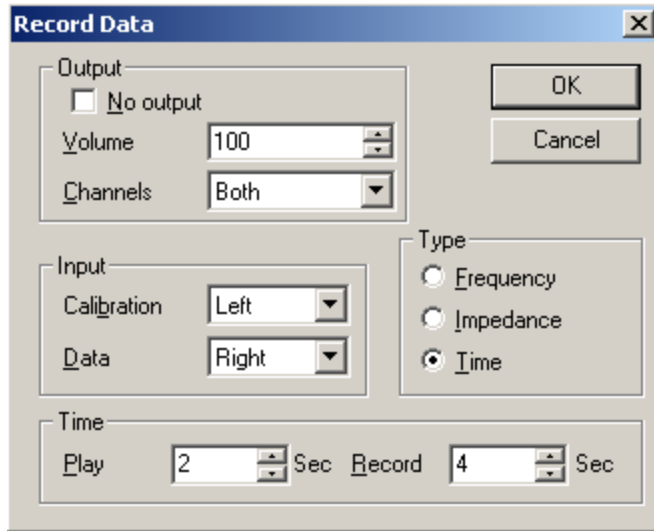
Note: You have to set the size of the MLS Impulse burst size. Right Click on the Chart/Properties/MLS Impulse then choose your burst size. If you choose a size that is smaller than that of the sample size defined under Options/Preferences/Measurements/Sample Size, you may not capture all of the data..



6. Click Ok
7. Go to Options/Preferences/Debug and set the warm up:
 - a. MLS warm up addresses the delay in getting a sound card to full operation ability (caps charging and clocks getting up to speed). This improves signal quality. A default of 100 seems to work well and is most useful with fewer samples and not quite as beneficial for more samples.



8. Go to Sound/Play (or Record)/Time, you must set this for a minimum period of time that is at least the sample size/sample rate so if your sample size is 2048 and your sample rate is 48K/sec, you need at least $2048/48000$ sec or at least 43 msec duration whereas if your sample size is 256K, you would have to play and record for a minimum of $256000/48000$ or 542 msec. You can set the time of play and record for much longer periods of time without any problem so setting 2 seconds for play and 4 for record would be reasonable as long as you don't need more than 2 seconds to capture all of the data.
9. Make sure that you set the type to the appropriate domain under Sound/Record. It would be time for an Impulse Response. Select Frequency to automatically get a frequency chart based on the recorded time data. Impedance uses the impedance in the Options/Preferences/Measurements tab to calculate impedances based on the recorded data.

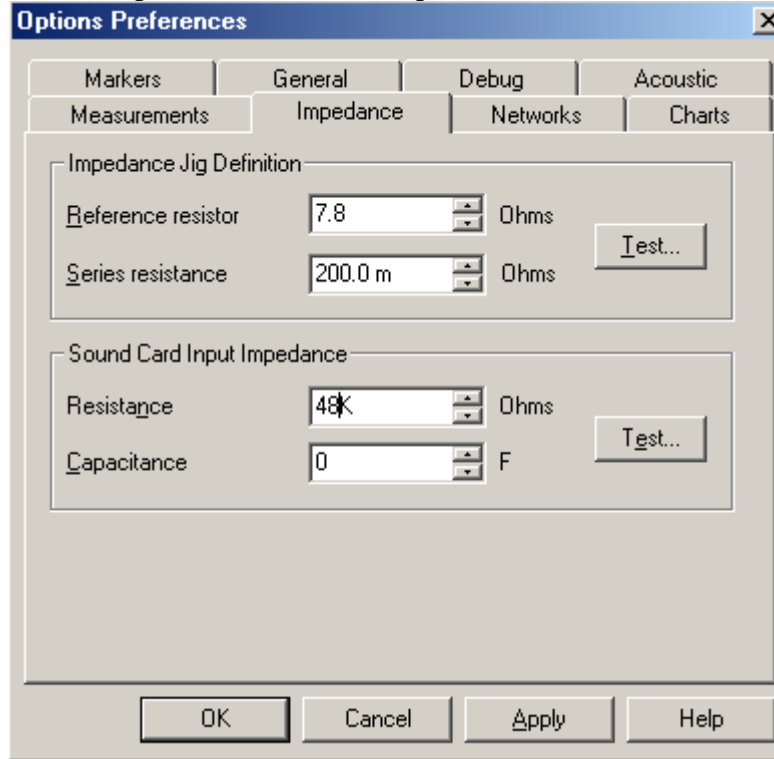


10. The recorded MLS signal will not look like the Measurement.output chart. It isn't supposed to. It should be a single pulse either positive going or negative going depending on your card. The small sine wave in the output signal is the MLS warm up feature that gets the card up to speed before performing the MLS signal.

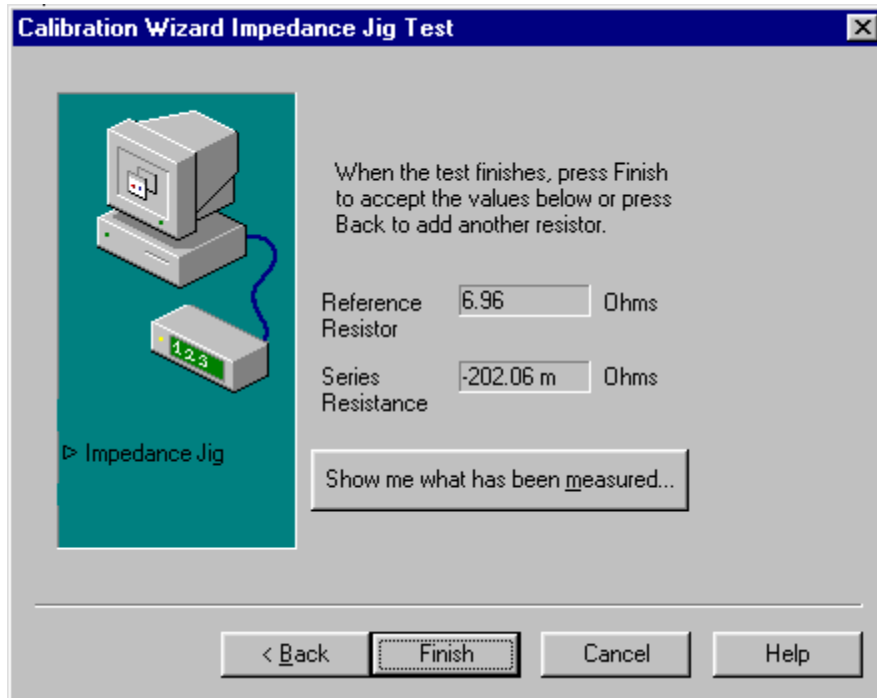
MEASURING JIG IMPEDANCE WITH THE WALLIN JIG

(Section Index)

1. Unlike the Cable Jig, the Wallin Jig has the reference resistors built into it.
2. Click on Menu/Options/Preferences/Impedance Tab



3. Measure the resistance of the jig from the barrel of J1 to BP4. This is your Impedance Jig Series Resistance. It should be in the range of .15 to 1 ohm
4. If you know the input impedance of your soundcard, enter this here. If you don't know, use the defaults recommended by SW (usually 1000K and zero Farads according to a post by Mark Zachmann).
 - a. See Sound Card Calibration for how to measure it yourself.
5. Place the Jig switches at SW1 up, SW2 left, SW3 down (Imp Cal 16)
6. Measure with a DMM the Resistance from BP3 to BP 4 and use this number as your reference resistor number for the 16 ohm calibration.
7. Press the Upper Test button/Next
8. Place the Jig switches at SW1 down, SW2 left, SW3 down (Imp Cal 4)
9. Measure with a DMM the Resistance from BP3 to BP 4 and use this number as your reference resistor number for the 4 ohm calibration.
10. Press the Upper Test button/Next
11. Now look at the table:

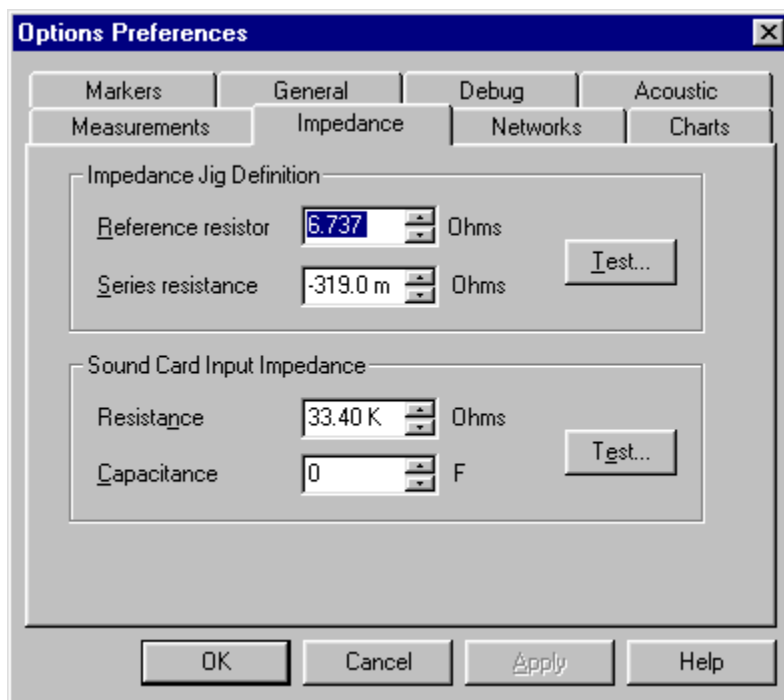


12. If you don't get something reasonable for the reference resistor (7 to 9 ohms) and something reasonable for the series resistance (less than 1/2 ohm, positive or negative) then you should check your jig connections and repeat the calibration
13. You can hit the < Back button a couple of times to get to the beginning of the calibration procedure, and then you won't have to re-enter the resistor values again since SW remembers them for you
14. Click Finish when your numbers look reasonable
15. This brings you back to the Options/Preference/Impedance Menu
16. Click Apply then OK

MEASURING JIG IMPEDANCE WITH THE CABLE JIG **(CLAUDIO NEGRO)**

(Section Index)

1. You need 3 resistors as reference to measure the cable jig
2. Reference resistor: Use a resistor around 8-10 Ohms and connect to the impedance cable in place of the 10-kOhm resistor.
3. Calibration resistor 1: Use a resistor roughly 2x the value of the reference resistor and connect to the DUT cable connector.
4. Calibration resistor 2: Use a resistor roughly ½ that of the reference resistor and connect to the DUT cable connector.
5. Measure Series Resistance: Using a DMM with low resistance measurement ability (if you have one available), connect one end of the DUT cable to the Right Line In Mini jack.
6. Enter the reference and series resistance in SW (if you did not measure series resistance, enter 0.2 ohms)



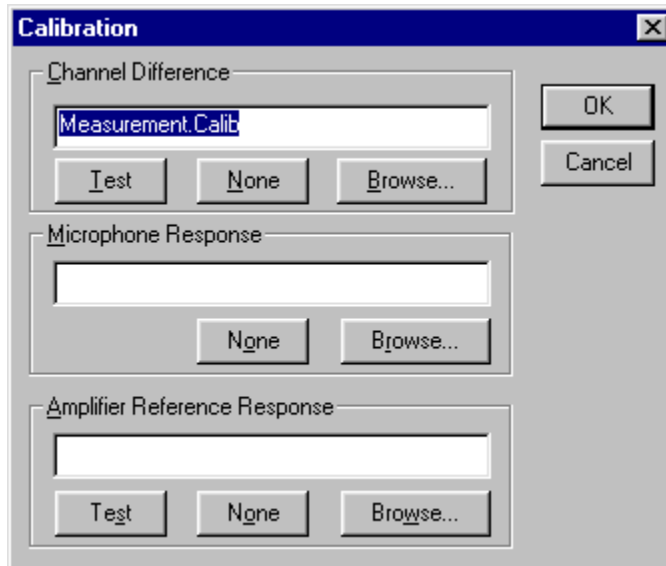
7. Click Apply/test/Next
8. Connect the reference resistor to the impedance cable and the larger calibration resistor to the DUT cable.
9. Enter the value of the reference resistor and click Next.
10. Select Next when available and substitute the smaller for the larger calibration resistor
11. Click Next when the option appears.
12. Go to Options/Preferences/Apply to make the new values effective.
13. Click OK.

CHANNEL DIFFERENCE CALIBRATION

[\(Section Index\)](#)

This adjusts for inequity between the channels in the sound card input. This can sometimes be fairly substantial. You will need to do this prior to starting any measurements, impedance or amplitude. It is best to re-do this each time you start a new measurement session as the channel differences may vary to some degree over time with certain cards.

2. Continue to use the loop cable or the jig connection set up as Direct mode (SW1 Center/SW2 Rt/SW3 Down).
3. Menu/Options/Calibrate
4. Click on the "Test" button in the "Channel Difference" section



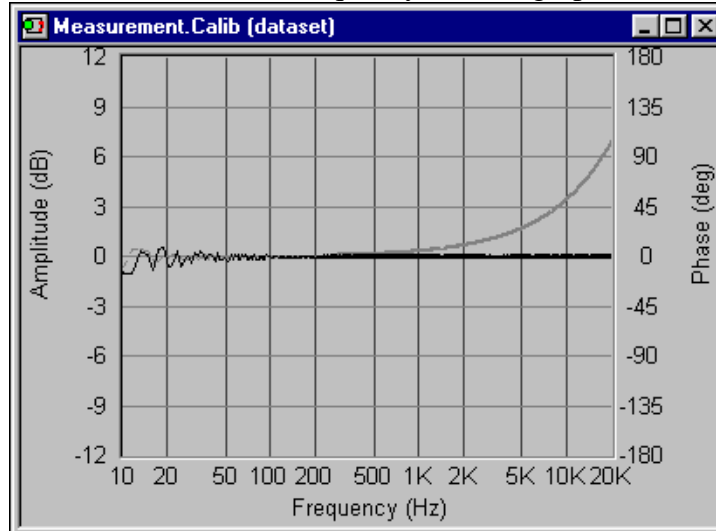
5. Click Next/Next/ then Finish when the calculations are complete and this option is no longer grayed out
6. Click OK when you are returned to the Calibration dialogue (this closes the calibration dialogue).
7. Inspect the VU meter in Speaker Workshop

	Maximum	Average Abs	Minimum
Left	23761	9169	-23506
Right	23945	8065	-23521

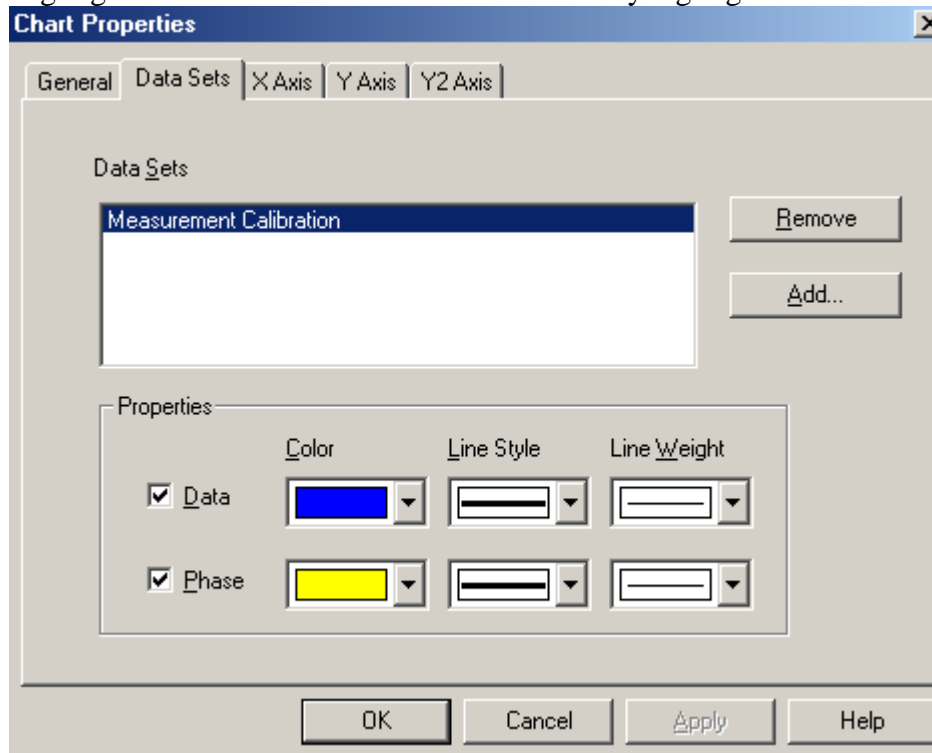
8. Re-check the volume meters (in Speaker Workshop) to assure that the clipping levels were not reached. If they clipped, you must choose a different volume set from Quick Mix.

	Maximum	Average Abs	Minimum
Left	23761	9169	-23506
Right	23945	8065	-23521

9. Go to the Design Tree/System/ double click Measurement.Calib. (where the results are stored) to obtain the frequency domain graph.

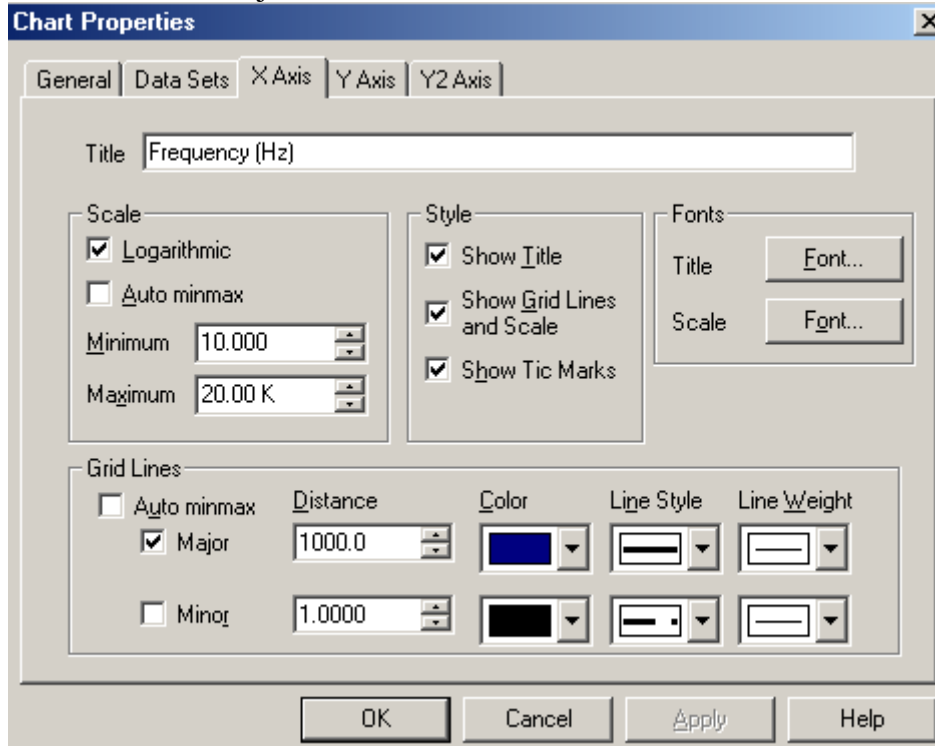


10. Right click the chart area
11. Chart Properties/Data
12. Highlight Measurement Calibration if not already highlighted



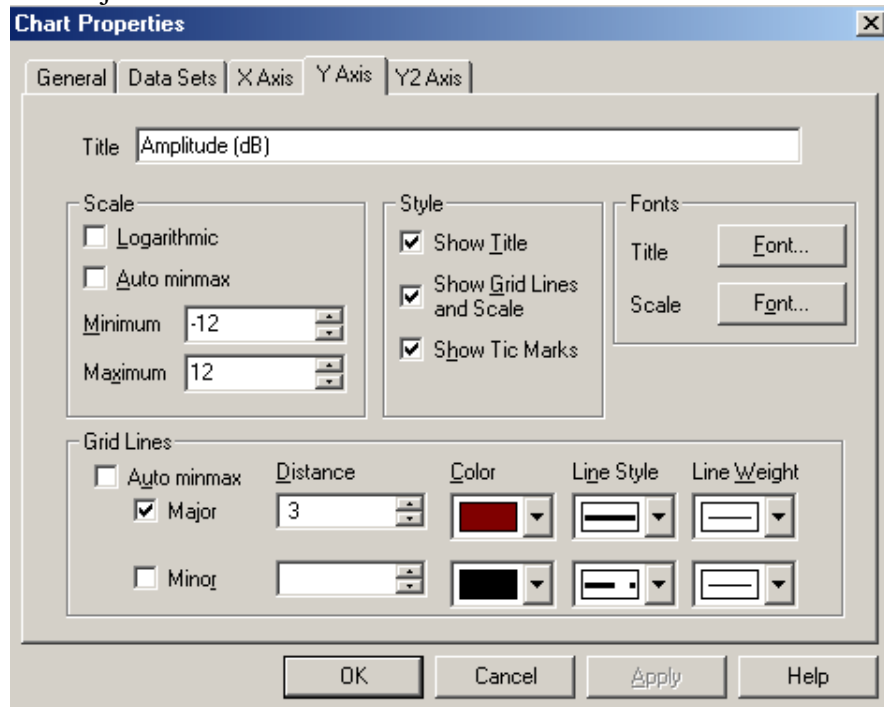
13. Check Data, Check Phase, and select your visual preferences (color, line thickness, etc.)
14. Under X Axis:
 - a. Set Title to Frequency (Hz)
 - b. Scale
 - i. Logarithmic
 - ii. Do not check Minmax

- iii. Set Min to 10
- iv. Set max to 20 K
- c. Style
 - i. Check all 3 boxes
- d. Grid Lines
 - i. Check auto minmax
 - ii. Check major
 - iii. Select the color, line style, and the other line box
 - iv. Do not check minor
- e. Set distance to Major 100.0 and Minor 10.0



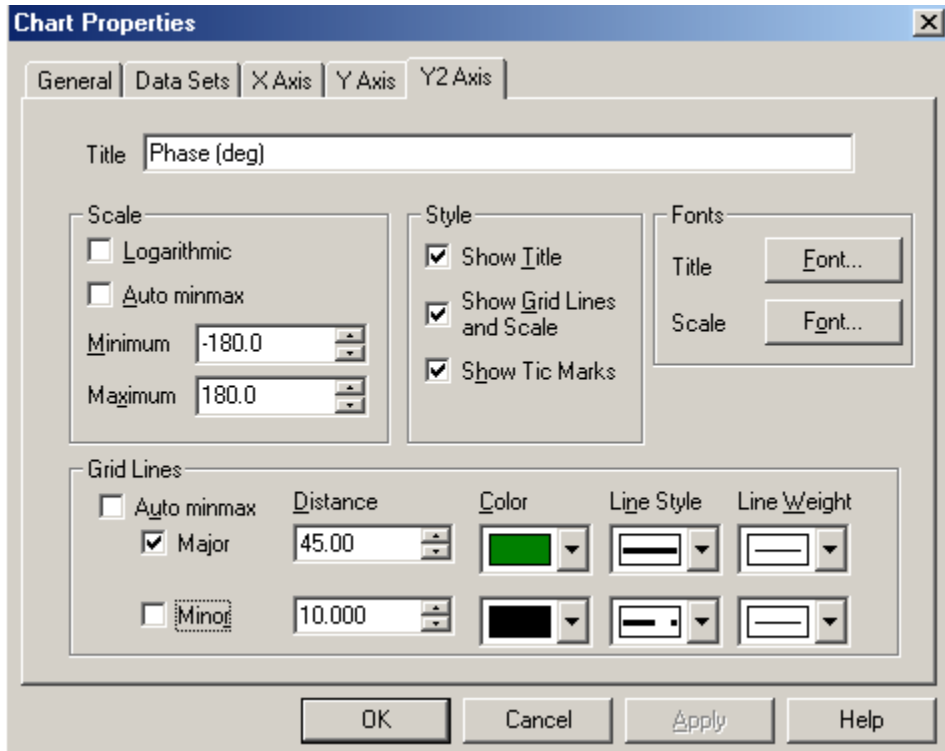
15. Under Y Axis
- a. Set Title to Amplitude (dB)
 - b. Scale
 - i. Don't check logarithmic
 - ii. Check Minmax
 - iii. Min -12
 - iv. Max 12
 - c. Style
 - i. Check all 3 boxes
 - d. Grid lines
 - i. Select Major but not Minor
 - ii. Select the color, line style, and the other line box

iii. Set Major 3

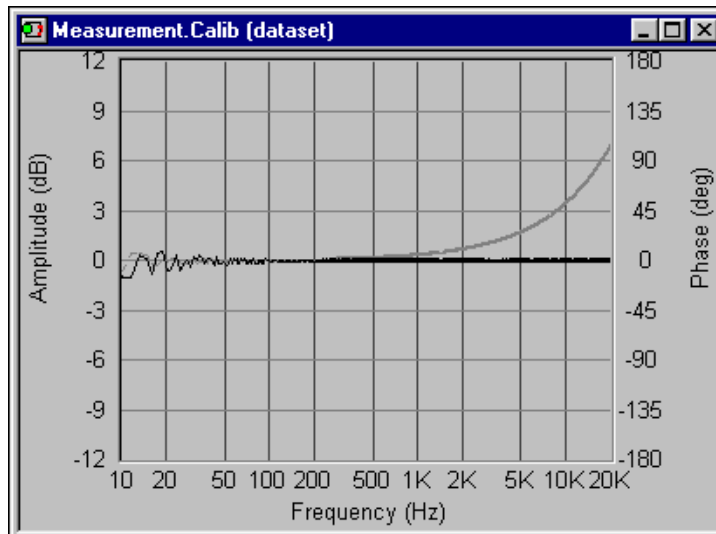


16. Under Y2 Axis

- a. Set Title to Phase (deg)
- b. Scale
 - i. Check nothing here
 - ii. Min – 180.0
 - iii. Max + 180.0
 - iv. All Minmax boxes should be unchecked
- c. Style
 - i. Check all 3 boxes
- d. Grid Lines
 - i. Check only Major
 - ii. Select the color, line style, and the other line box
 - iii. Distance
 - 1. Major 45.0

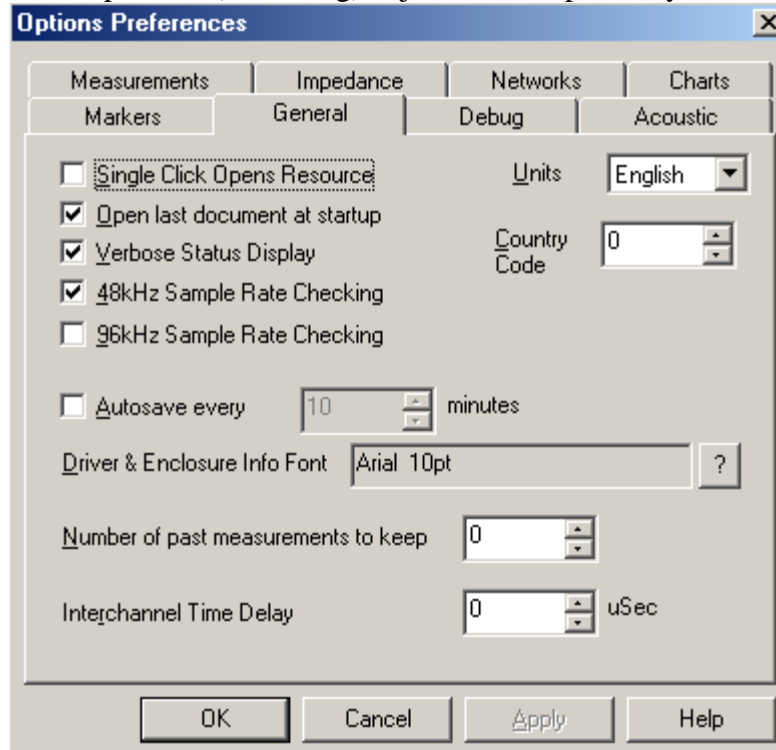


17. Close this dialogue and Right click in the chart area/Make Chart Default
18. Review Phase data to see if it is flat out to 20-24K
19. If the left and right inputs are perfectly matched at this point, you should ideally get two straight lines at 0 dB and at 0 degrees.
20. IF you have a synthesized duplex card such as Ensoniq, the phase will increase to around 180 degrees at half of the sample frequency (Nyquist rate) as a result of it being a single multiplexed analog to digital converter that switches back and forth between inputs.



- a. IF your card does this, adjust the Interchannel Delay as follows:
 - i. If it is rising or dropping:

1. Options/Preferences/General/Interchannel Delay (on the bottom of the page) and add a negative (for rising) or positive (for falling) adjustment, respectively



- ii. Re-run the Calibration and adjust the values until the phase is as flat as possible. *Note: you may want to re-scale the 2-Y axis while making fine adjustments in Interchannel delay so that the changes are more visible.*
- iii. You may have to re-set this each time you re-start Speaker Workshop.

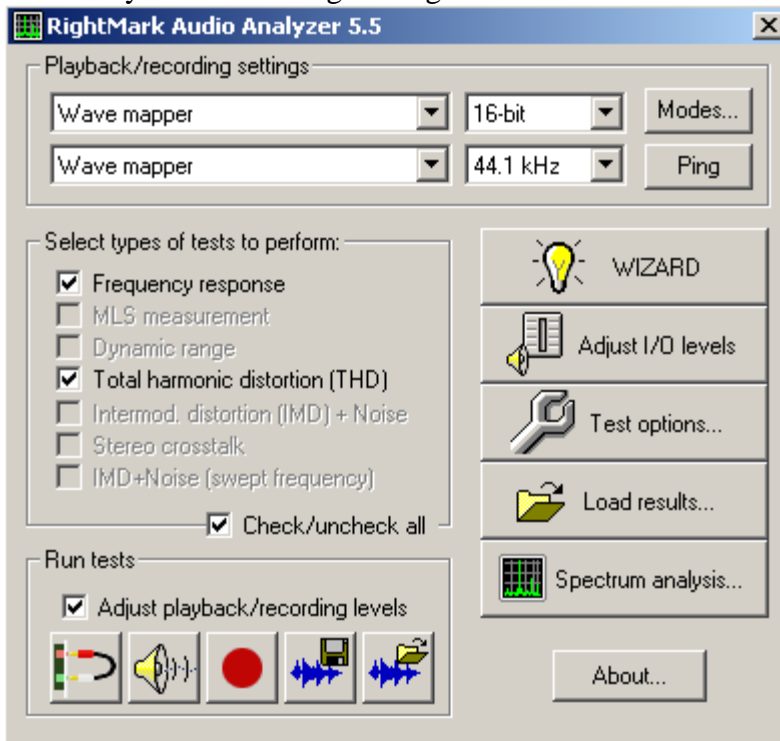
SETTING THE VOLUME USING RIGHT MARK AUDIO ANALYZER (RMAA)

(Section Index)

1. Download and Install RMAA from <http://audio.rightmark.org/download.shtml>




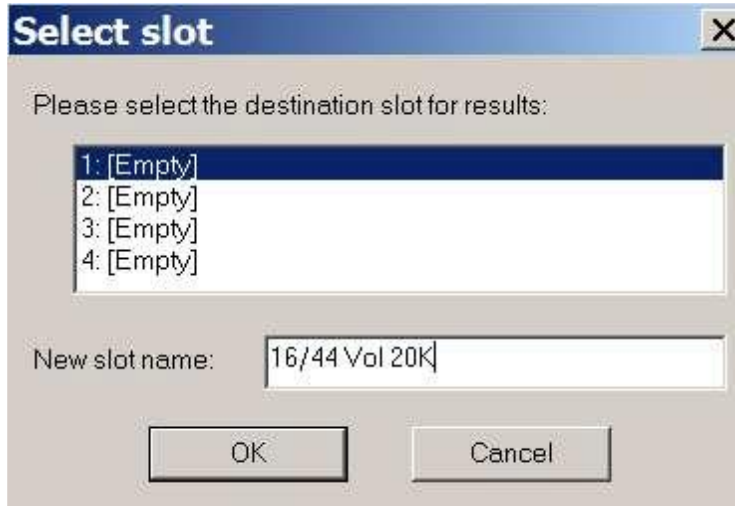
2. Open RMAA (note that a tutorial can be downloaded from the same site).
3. RMAA requires that you set up a loop of both right and left channels from the output of the card to the input of the card. A mini jack to mini jack cable is your best option here. Connecting the jig here would result in one output channel being fed into two input channels and would yield inaccurate results.
4. Go to Playback/Recording Settings and select the Soundcard.



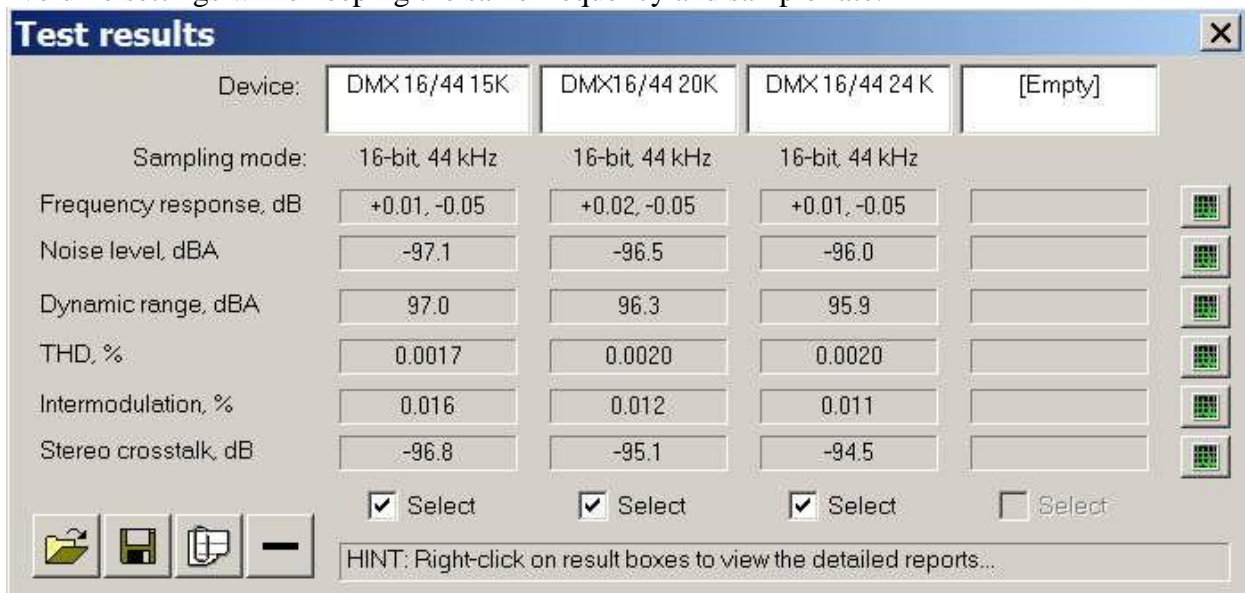
5. Select the number of bits (16 for DMX 6 Firewave card)
6. Select frequency (the card's maximum sample rate) (96kHz for DMX 6; 48kHz for SB Live)
7. You will want to repeat this procedure several times using different sample rates to compare which sample rate works best for your card
8. Deselect "Adjust Playback/Recording Levels" (we are using our volumes saved in QuickMix)
9. All other boxes should be checked (Freq Response, Noise Level, Dynamic Range, Total Harmonic Distortion, Intermodulation Distortion, Stereo Cross-talk, and Check/uncheck all).
10. Go to Test Options/Sound card and make certain that "Use WDM drivers" is not checked.





11. Click on  and after a few seconds, a window will appear in which you can save results.
12. Name this by identifying in the name: Sample Rate/Resolution_Volume Setting.



13. RMAA allows for up to 4 different settings to be saved so that you can compare other volume settings while keeping the same frequency and sample rate.



- a. Recall each volume setting saved in QuickMix and rerun the test saving each result with a descriptive name.
- b. Save the best setting by clicking 
- c. Close the test results window
- d. Click  Load results...

- e. Select the saved files.

Look for the following:

Frequency Response: we want this as close to zero as possible. (This estimates the noise level in silence in the test chain. It uses dB Fs A which means sound relative to a full-scale signal and is weighted with a special aural perception A curve. Professionals target a signal to noise ratio of -96 dB with a lower number being better.)

Dynamic Range: The greater the magnitude, the better. (This estimates a noise level with a weak signal applied and the linearity of the sound device operating at low signal levels.)

THD (Total Harmonic Distortion): target 0.002% for the first harmonic. (This defines the level of unwanted harmonics generated in a sound device. Of note, even harmonics generated in tubes make for a warm sound and high THD; odd harmonics generated by solid state transistors make for an unpleasant sound.).

1. Test for this as follows:
 - a. Highlight the signal window
 - b. Measure/Harmonic Distortion (THD)
 - c. Click the spectrum at one power option.
 - i. Click OK to show various harmonics
 - ii. Less than 0.1% is acceptable for the 2nd or 3rd harmonic.
 - iii. If the THD is too high, decrease the volume until acceptable and document the level.

Intermodulation: the lower the better. (IMD occurs when a complex signal passes through a test chain. The test signal consists of 2 harmonics of different frequencies. After passing through the test chain, the resulting signal contains different and possibly a larger number of harmonics resulting from D/A and A/D conversion, over sampling filters, and sample rate conversion.)

Stereo Crosstalk: the higher the value, the better. (Estimates channel leakage for various frequencies; stereo crosstalk interferes with stereo imaging.)

- f. Check Frequency response again to make sure that it is still a flat line; if not, all results will be invalid. Make sure that tone controls are all neutral.

LATENCY

[\(Section Index\)](#)

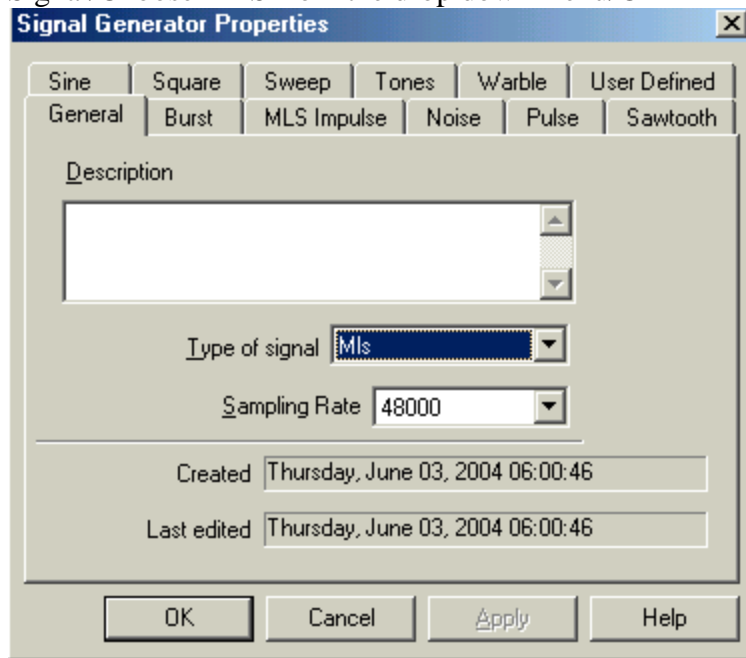
Speaker Workshop synchronizes the input and output using Windows controls and this could be slightly off depending upon the speed of the computer and sound hardware. Set this value to adjust the synchronization to compensate for those effects. A value of 0 uses the standard synchronization. Non-zero values adjust the synch point forward and backward in milliseconds.

Latency is stored in the system registry and will be used for all record operations, including measurements, and with all databases.

When working with SW, make certain that as little as possible is running in the background or is open to minimize problems with Latency variability.

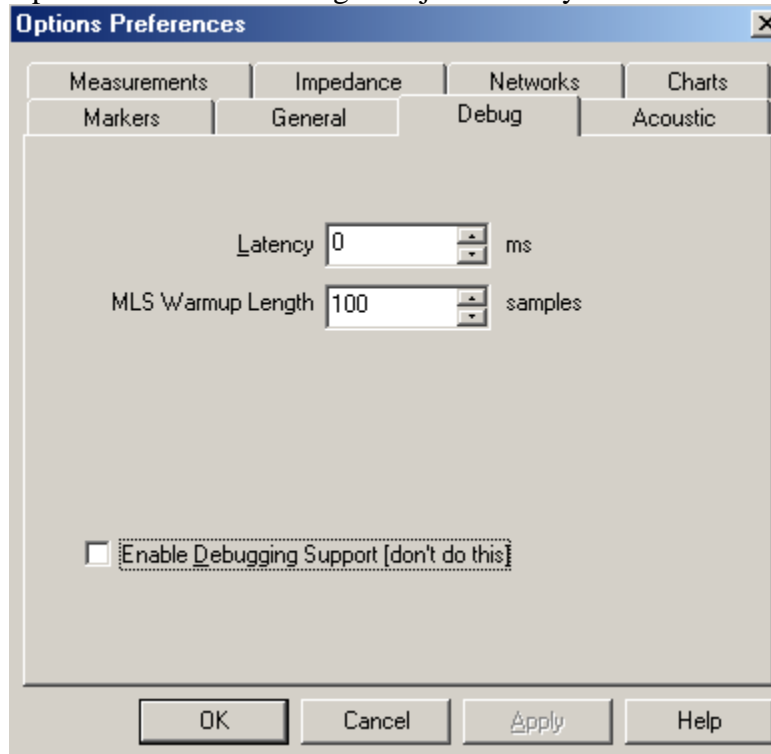
If there are problems with inconsistent latency, the phase measurements will be inaccurate and the quality of your crossover design will be poor.

1. Do a signal record using an MLS wave signal in loop mode (though this could be done with a Sine Wave, MLS will be easier and more accurate to work with) in the following way:
 - a. Make certain that the jig is set for Direct mode (SW1 Center/SW2 Rt/SW3 Down) or that you have looped the output to the input with Claudio's arrangement.
 - b. With the Chart Active, Right Click/Properties/General tab/Type of Signal/Choose MLS from the drop down menu/OK



- c. Menu/Sound/Record
2. Examine the signal in test.in.1

3. Expand the left edge of the chart to examine the signal closely. A smaller latency difference is better (5 msec is acceptable though 2msec is excellent) and introduces a smaller amount of noise.
4. Don't be concerned if the phase of the pulse starts out negative, it may do this sometimes without harm to the measurements.
5. If you have too much latency, right click on the chart, go to chart properties and set the x-axis minimum to -5ms and the maximum to +5msec so you can see how far the pulse goes negative.
6. Options/Preference/Debug to adjust Latency



- a. If the wave is already deflected upward at the zero point of the x axis, shift the latency using negative values (start with -5 and adjust)
- b. If the wave has a blank space prior to starting, shift the latency to the right using positive values.
7. Set the repeat to 1
8. Leave a little blank space between zero and the start of the pulse to assure accuracy.
9. Repeat it several times to make sure that it is staying on the positive side to assure consistency.
10. Note that the x axis may be adjusted for better resolution (0.6,1.2,1.8 ms)

PASSIVE COMPONENT TESTING WITH THE WALLIN JIG

[\(Section Index\)](#)

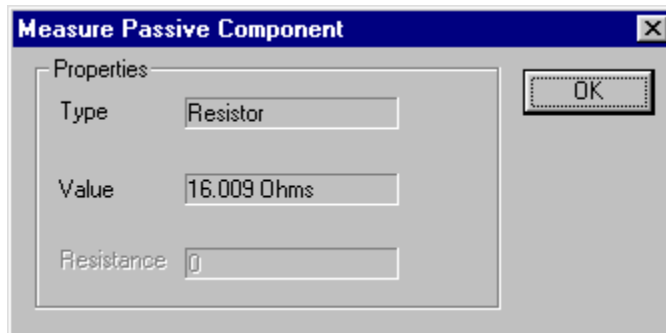
It is best to use reference values as follows when testing passive components:

Resistors less than 20 ohms

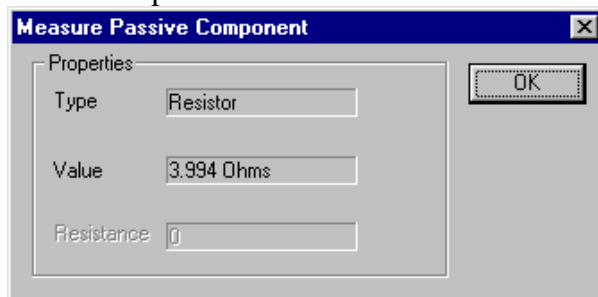
Caps 1-10 microfaradays

Inductors around 1mH.

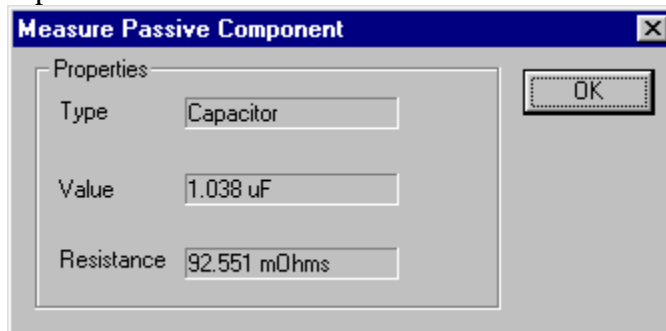
1. Place the jig in *IMPCAL16* mode (SW1 up, SW2 left, SW3 down)
2. Click on the *sine_test* (*signal* window then Menu/Measure/Passive component and the measurement should start.



3. Next place the jig in the *IMPCAL4* mode (SW1 down, SW2 left, SW3 down) and do another passive measurement.



4. Do these measurements several times to confirm that they are consistent.
5. Put the jig in *IMPMEAS* mode (SW1 center, SW2 left, SW3 down) and install a capacitor across BP3 and BP4.



6. Try this with an inductor across BP3 and BP4 as well. *Speaker Workshop can automatically identify the type of passive component it is measuring.*

PASSIVE COMPONENT TESTING **WITH THE CABLE (CLAUDIO NEGRO) JIG**

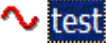
(Section Index)

When testing passives, best to use

Resistors <20ohms

Caps 1-10 microfaradays

Inductors around 1mH.

1. Leave SW settings as they were during the calibration process.
2. Continue to use the Impedance Cable with the reference resistor.
3. The DUT cable will be used to test components (resistors, caps, inductors, drivers and speakers).
4. Connect the ~16 ohm resistor to the DUT cable.
5. Expand the Calibration Folder under Test.swd and double click  and then on the Resistor Symbol with a “?” under it.
6. A new window will appear with a measurement.
7. Substitute known resistors to establish a range of values that we can trust.
8. Substitute Caps and the test will tell us Cap values with internal resistance values.
9. Substitute Inductors using the Resistor Symbol to establish a trustworthy Inductance Range.