Morset Sound Development

WinMLS 2000

for Microsoft Windows 95/98/ME/NT/2000/XP

User's Guide

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1 Introduction

WinMLS 2000 is software for performing and evaluating audio, acoustical and vibrational measurements. It was designed for MLS (maximum length sequence) measurements of acoustical systems, but other types of measurements are also supported. Measurements based on MLS allow measuring system impulse responses with high accuracy. Until recently, such measurements were impossible without custom hardware solutions. With WinMLS, however, impulse responses can be measured with great dynamic range and fine resolution on a PC with a standard full duplex[1] sound card. Powerful and flexible post-processing capabilities are integrated in the software. In order to export the measurement data, several formats are supported. If further post-processing of the measurement data than the current version of WinMLS can offer is needed, MLSSA® (DRA Laboratories) and Matlab® (MathWorks Inc.) may be used. Software that can read text-files, such as Excel (Microsoft) may also be used.
In the documentation, note that some parts may not be relevant for your type of sound card or the type of measurements you want to do. This text is written in a way that it should be understandable for persons having little experience with computers. Bold letters are used for items in the user interface, such as menus and buttons. If you have any questions or comments about WinMLS, please contact support@winmls.com, and please do visit us at www.winmls.com.

Several other documents should be used as reference.
For the installation procedure and a list of issues, please read the **Readme** file.
For an answer to a wide range of questions, please read the **FAQ** file. If you have a problem, please see if you can find the solution in the **FAQ** before contacting us.
For questions relating to sound cards, please read the **Tested Sound Cards** file.
For questions relating to the content of the menus and dialog boxes, please read the **Menus and Dialog boxes Documentation** file.

2 System Requirements

The requirements for using the WinMLS software is a PC with a standard sound card/device running any of the Windows operative systems 95, 98, ME, NT or 2000.

2.1 PC

The minimum configuration recommended is a Pentium® processor, 16 Mb Ram, 30 MB of available disk space.
If WinMLS is to be used for post-processing measurements, at least a 400 MHz processor with 64 Mb RAM is recommended. This is because a slower PC will give bad performance on plotting long sequences (measurements of large rooms) and for computing room acoustics parameters. If your PC does not perform well enough to run WinMLS 2000, we recommend that you use the previous version of WinMLS, ver. 2, that can be downloaded from our homepage.

[1] A full duplex sound card is capable of simultaneous record and playback.

2.2 Sound card

A standard Windows compatible sound card is needed to perform measurements. There is a wide range of sound card available (we prefer to call it “sound device” instead of “sound card”, since USB devices can also be used, but the term “sound card” is so widely used that we continue to use it.).
For performing measurements that require an output signal, such as loudspeaker measurements, the sound device must be full duplex. A full duplex sound card is able to record and play simultaneously. Most sound cards are full duplex, but a few sound cards are still not capable of performing MLS measurements with WinMLS. See the document “Tested Sound Cards” for sound card recommendations, more information, and a list of tested sound cards.

2.3 Hardware

If you are using WinMLS for measuring electronic equipment only, such as amplifiers, cables for connecting to the sound card is in principle all the extra equipment needed.
In order to use WinMLS for other types of measurements than electrical measurements, transducers are needed. A transducer is device for transforming a quantity to another quantity. Two well-known transducers are the microphone and loudspeaker. A microphone transforms sound pressure to voltage and a loudspeaker transforms voltage to sound pressure.
In addition to microphone and loudspeaker, which will be discussed in more detail below, WinMLS supports transducers for measuring the quantities displacement, velocity, acceleration and force.

2.3.1 Microphone

Connect the microphone to the *left* channel sound card line input (if you are to perform a 2-channel measurement, connect microphone channel 2 to the *right* input).

If you have a cheap multi-media microphone that comes with the sound card, it can be used for testing purposes and for comparison measurements not acquiring a flat frequency response. The frequency range is probably also limited. Such a microphone must usually be connected to the microphone input of the sound card, since it will provide the *phantom power supply* voltage needed.

For more information about microphones and for purchasing our low-cost microphone for performing loudspeaker frequency response measurements and amateur room acoustics measurements please visit our homepage or ask us.

For high-quality measurements we recommend dedicated microphones. However, the price may be high, and an external *phantom power supply* or a microphone amplifier is often needed.

2.3.1.1 Microphone amplifier/Phantom power supply

Most soundcards are equipped with a single channel microphone input containing an amplifier. This can be used, eliminating the need of an external amplifier. For most sound cards it is better to use the **Line** input than the **Microphone** input since it gives a better signal-to-noise ratio. But

for most purposes the **Microphone** input is usually sufficient and gives a low voltage *phantom power supply* (~1 Volts) for the condenser microphone.

If your microphone needs a greater voltage, a *phantom power supply* may be purchased or made using batteries.

If none of the above is sufficient, we recommend using a microphone amplifier. Note that a microphone amplifier may be optimized for certain types of microphones. For instance, a *phantom power supply* voltage may be provided for condenser microphones. Also, the frequency characteristic may be optimized for a maximally flat frequency response and the input impedance may have been optimized for maximum signal-to-noise ratio.

2.3.2 Loudspeaker

Several applications involve a loudspeaker, e.g. measuring the loudspeaker itself or the reverberation time of a room. For room acoustics measurements it is important that the loudspeaker gives enough power and has a flat frequency response at the range of interest. If you intend to perform room acoustics measurements according to the ISO 3382 standard, the loudspeaker should also be omni-directional. However, for reverberation time measurements the directivity of the loudspeaker is usually not that important (depends on the shape of the room).

Note that a long loudspeaker cable may cause a significant loss, therefore the cable should be a part of the configuration when calibrated measurements are to be performed.

2.3.2.1 Power amplifier

An amplifier is needed to supply enough power to the loudspeaker. The amplifier can either be an integrated part of the loudspeaker or external. Special functions, such as stereo enhancement, loudness or spatial sound must be switched off for measurement purposes. Filtering, such as bass, treble or equalization should be bypassed or set as neutral as possible.

The loudspeaker amplifier input is to be connected to the *left* channel sound card output.

3 Purchasing and Registering

WinMLS can be purchased from our homepage www.winmls.com or from our distributors. Since this is a beta version the pricing and registration is not ready. Please contact us for questions.

4 The first measurements

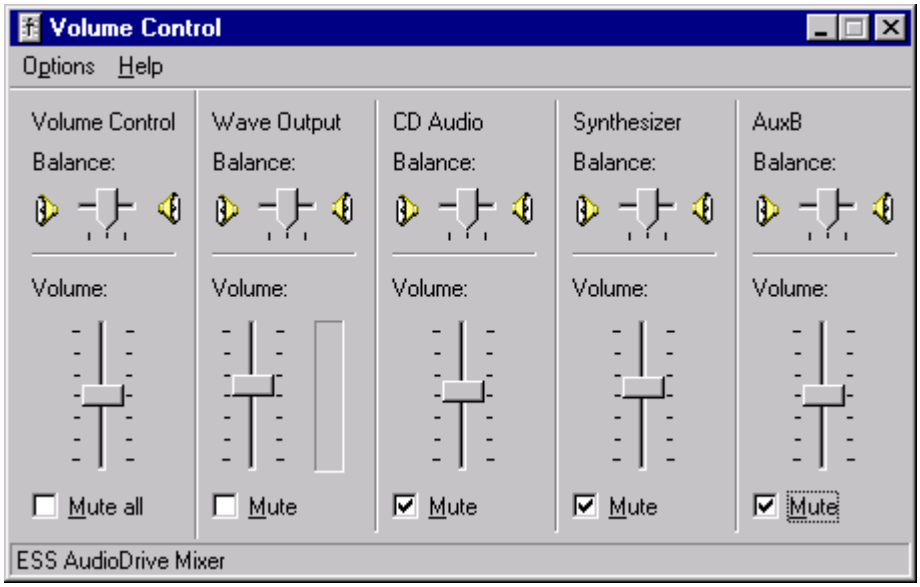
The purpose of this chapter is to give a step-by-step description of how a measurement can be performed. The loop-back response of the sound card is first to be measured, then a simple measurement using a microphone and a loudspeaker. We recommend that you take the time to do this measurement because it is a good way of learning to use the software. It will also let you know if the sound card is suited and if the settings are correct. Finally, a loop-back sound card measurement may be used to detect the sound card input/output delay (because of the converters) and to correct the sound card's influence on the measurements.

4.1 Turning off Windows Notification Sounds

Before you begin your measurements, it is recommended that the Windows notification sounds are turned off or send to another sound card if you have several cards installed. By doing this you avoid disturbing the measurements. You can turn off the system sounds in the following way:
Find the **Control Panel** (click the **Start** button and choose settings). Then double-click on the folder **Sounds**. If you use Windows95:
You see a dialog box **Sound Settings**. At the bottom of the dialog box, find **Settings** and choose **No sounds**. If you use Windows NT:
You see a dialog box **Sound Properties**. Click **Windows** under **Events**.
Then click on the list field under **Schemes**. When the list appears, choose **No Sounds**.

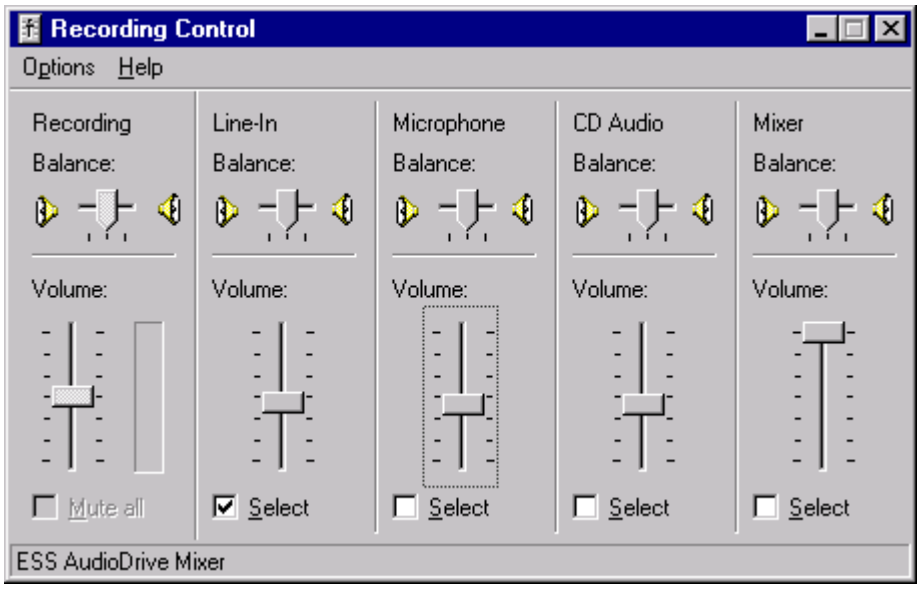
4.2 Finding and Preparing the Volume Control

If your sound card supports the WinMLS mixer, you do not need to read the following chapter since you will not use the external mixer to be discussed here. If your sound card has a mixer, but does not support the WinMLS mixer, this chapter is important. If your sound card does not have a software mixer at all, an external mixer can be used to set the levels. If you are unsure if your sound card supports the WinMLS mixer, go to **Measurement->Sound Card Settings** and try to check **Use the WinMLS mixer.** If it is impossible to turn it on, then your sound card does not support the WinMLS mixer.
The first time you perform a measurement you will probably have to adjust the input or output volume levels. Then you have to find the **Volume Control** software that is supported by your sound card. Usually this is found in the **Multimedia** folder in **Accessories,** it may also be found in the **Control Panel**. Some sound cards only support their own volume control that is found in the specific sound card folder. A typical volume control is shown in the figure below.



The volume control above shows the output volume. When using WinMLS you only use the **Wave Output** shown in the figure above. It is a good idea to mute the other possibilities as shown in the figure above. But do not mute the master **Volume Control**.

We also need to adjust the input volume. For this example it is found from the menu in the figure above. Choose **options**, then **properties** and **Adjust volume for recording**. Then the following window is displayed:



Select the input you would like to use, either line-in or microphone. While you use WinMLS, you probably need to set both the input and output level. You should therefore have one of each open at the same time. To do this, just open a new volume control in addition to the one already open.

4.3 Measurement 1: Loop-back measurement of the sound card

4.3.1 Preparation

For this measurement, we assume that no settings have been changed, WinMLS has its default configuration. If you are not sure, you can load the default configuration from the setups toolbar (**View->Setups Toolbar**). Chose the measurement setup named **MasterSetup** as shown in the figure below.

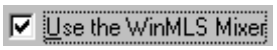
4.3.1.1 Connections

Find the output and input of the sound card. If your sound card has a **microphone** input, this may be used, we will use the **line** input in this example. Connect the output to the input (only left

channel will be used for the measurement, so you do not have to connect the right channel input and output).

4.3.1.2 Sound card settings

In **Measurement->Sound Card Settings**, make sure that the correct input and output device is chosen. In the figure below, the card VX222 is chosen as input and output device. If you have more than one sound card installed, it is strongly recommended to use the same sound card for input and output. This is because there is usually a small difference in sampling frequency and this may give a large measurement error for MLS measurements.

While you still are in the **Sound Card Settings** dialog box, make sure 

is checked. If it cannot be checked, this indicates that your sound card does not support the WinMLS Mixer. If this is the case, you will not be able to set the input and output line as described below. You should read the chapter **Finding and Preparing the Volume Control** for using the sound card mixer, if any.

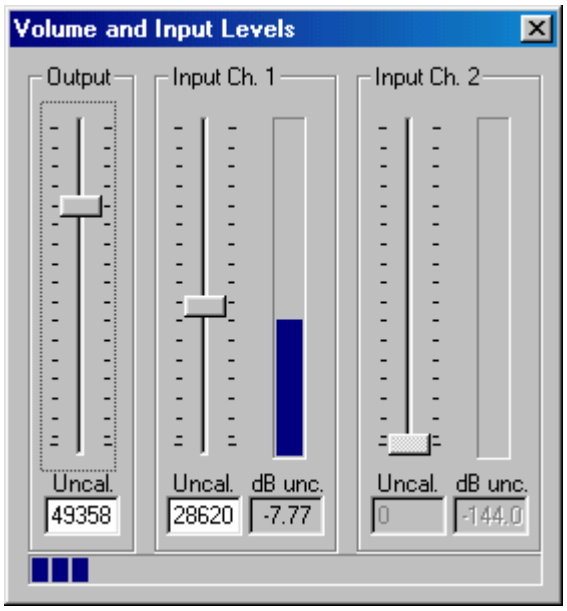
If possible, select input line as shown in the figure below. Depending on the sound card, you may choose between only one or several lines. Usually the line input is called **Line-in** or **Line**, and the microphone input is called **Mic** or **Microphone**. But the VX pocket/VX222 cards have only one analog input that can be used as both line and microphone input, thus the input line is just called **Analog Input**. The output line has usually only one choice, and it is usually called **Wave** or **Wave Output**.

Click OK to close the dialog box.

4.3.2 Measurement procedure

4.3.2.1 Set the input and output levels

Select **Measurement->Test Current Levels** to display the dialog box showing the input and output levels. If the WinMLS mixer is supported, sliders for setting the mixer levels will also be displayed. Use the sliders for adjusting the levels, if you see no sliders, please see the chapter **Finding and Preparing the Volume Control** on how to adjust the levels.



The input level must be less than zero to avoid clipping, but as near zero as possible. In the figure, the input level is –6.51 dB, which tells how far it is from clipping. If the level is lower than –40 dB or increasing the volume does not increase the level, the cables are not properly connected.
The level indicator shown to the right of the figure is colored red if there has been clipping during a measurement.

4.3.2.2 Performing the measurement

When the levels are set, select **Measurement->Start Measurement** to start the measurement.
When the measurement is finished, it should look similar to the figure below.

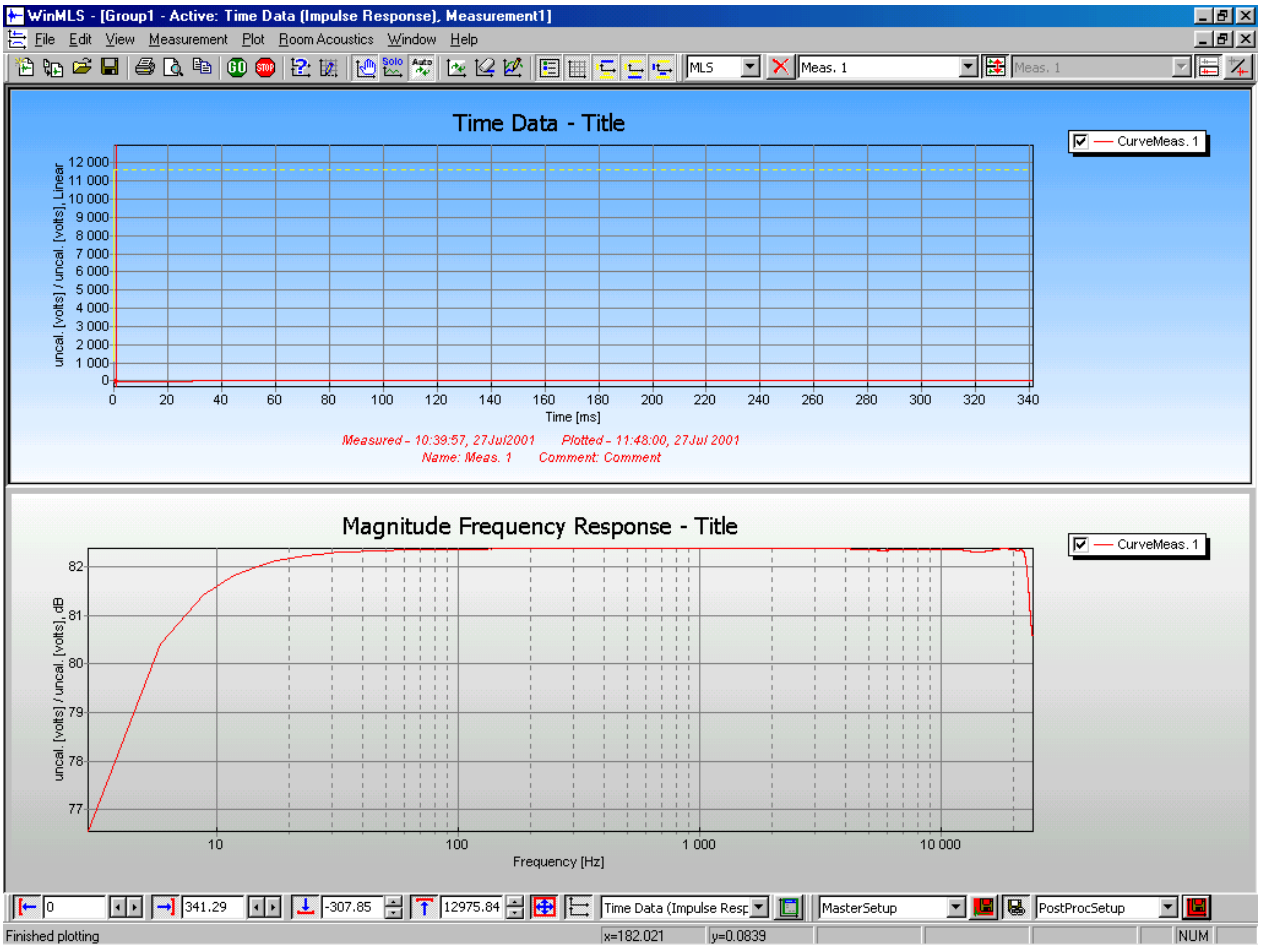
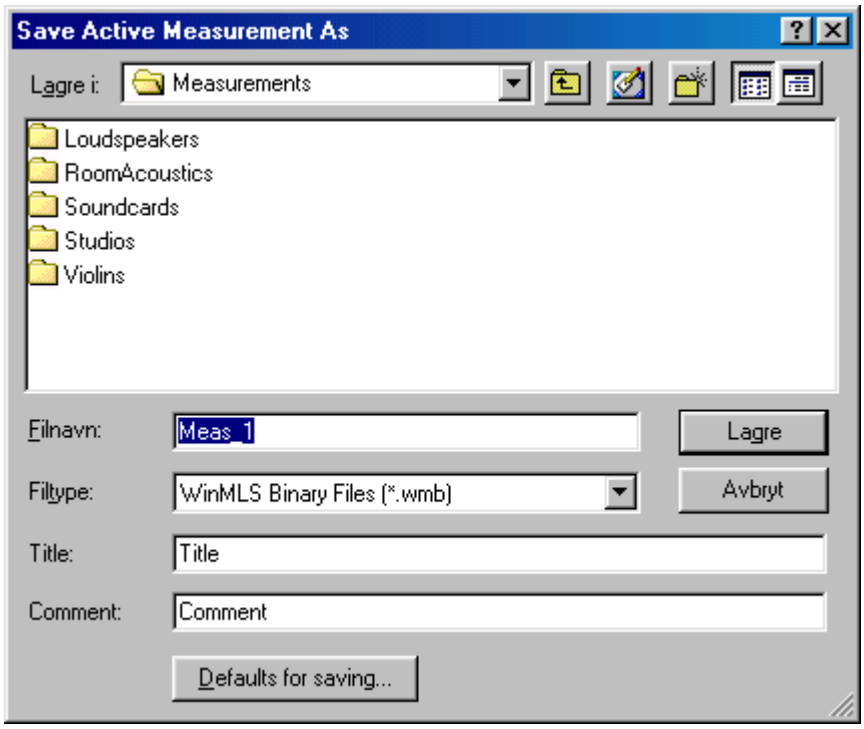


Figure 1 Screenshot after loop-back measuring the Digigram VX222 sound card.

The red curve in the upper plot displays the impulse response of the sound card and the lower plot displays the magnitude frequency response obtained from the impulse response. We see that the impulse response is very short and the frequency response is very flat in the range 20-20000 Hz.

4.3.2.3 Saving the measurement

Save the measurement by clicking the button or **File->Save Active Measurement As…**. The dialog box shown below will be displayed.

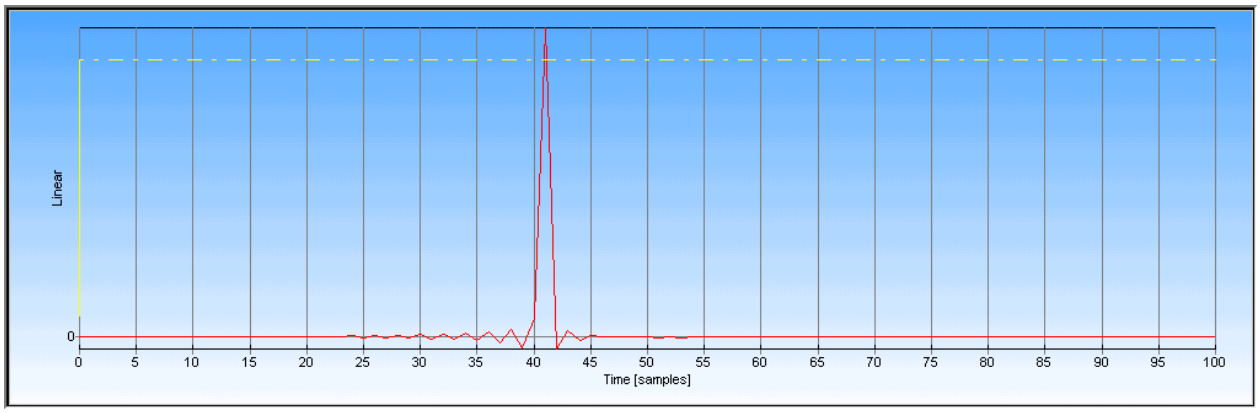


Choose a name, title and comment, then click the **Save** button.

4.3.3 Changing plot settings

We will now find the delay in unit samples. First double-click in the upper plot window. This will open the **Time Data Settings** dialog box. To the upper left select **Time [samples]** as axis unit for the x-axis:

Zoom in on the x-axis of the upper plot. Do this by typing the number 100 in the edit box to the right of the  button found at the lower left. Then the upper window will display the first 100 samples of the x-axis as seen in the figure below



From this we see that the sound card has a delay of 41 samples. It is possible to correct for this delay, how to do this is explained in the measurement part of the FAQ.

Now restore the old x-axis unit **Time [ms]** from the **Time Data Settings** dialog box and restore also the maximum axis value by clicking the  button. We will now plot the impulse response using dB as y-axis. To do this, first double-click in the plot window. This will open the **Time Data Settings** dialog box. To the upper left select **dB** as axis unit for the y-axis:

Click OK to exit the dialog box, this will update the plot with the new settings. The result should look similar to the figure below.

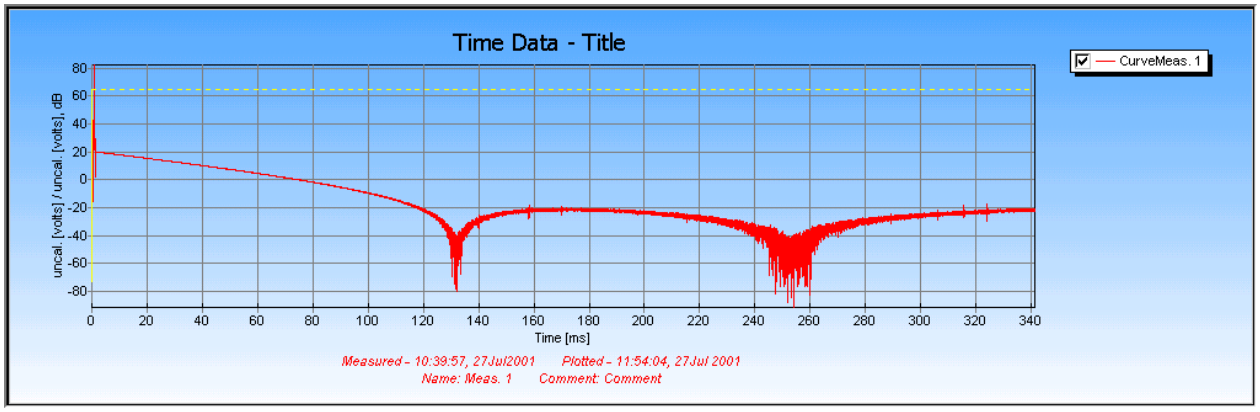
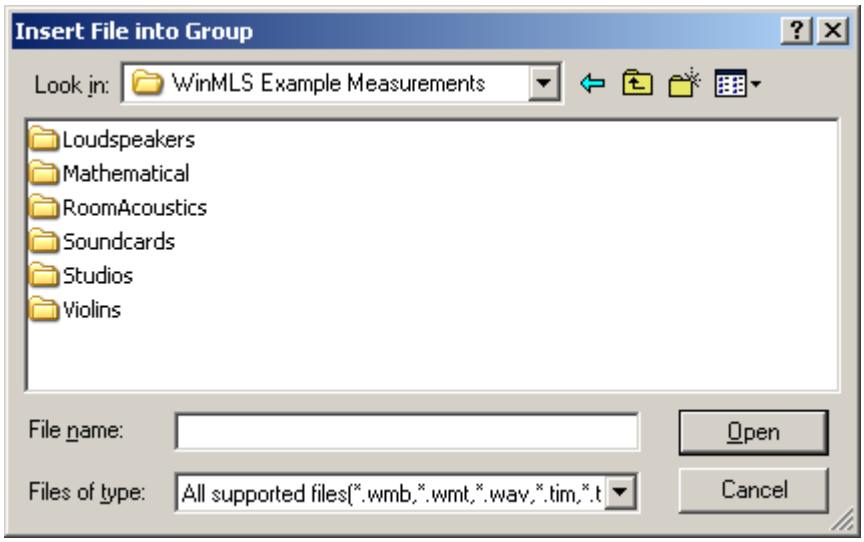


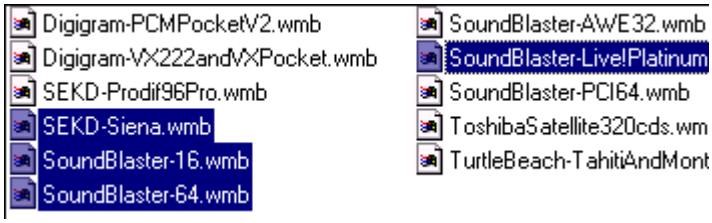
Figure 2 VX222 sound card measured loop-back, plotted using dB as y-axis.

4.3.4 Verifying the sound card measurement

If your loop-back sound card measurement performed in the chapter above looks suspicious, we recommend that you first compare it with the other measurements of sound cards. Click  or **File->Insert** to insert a new file into the current window.

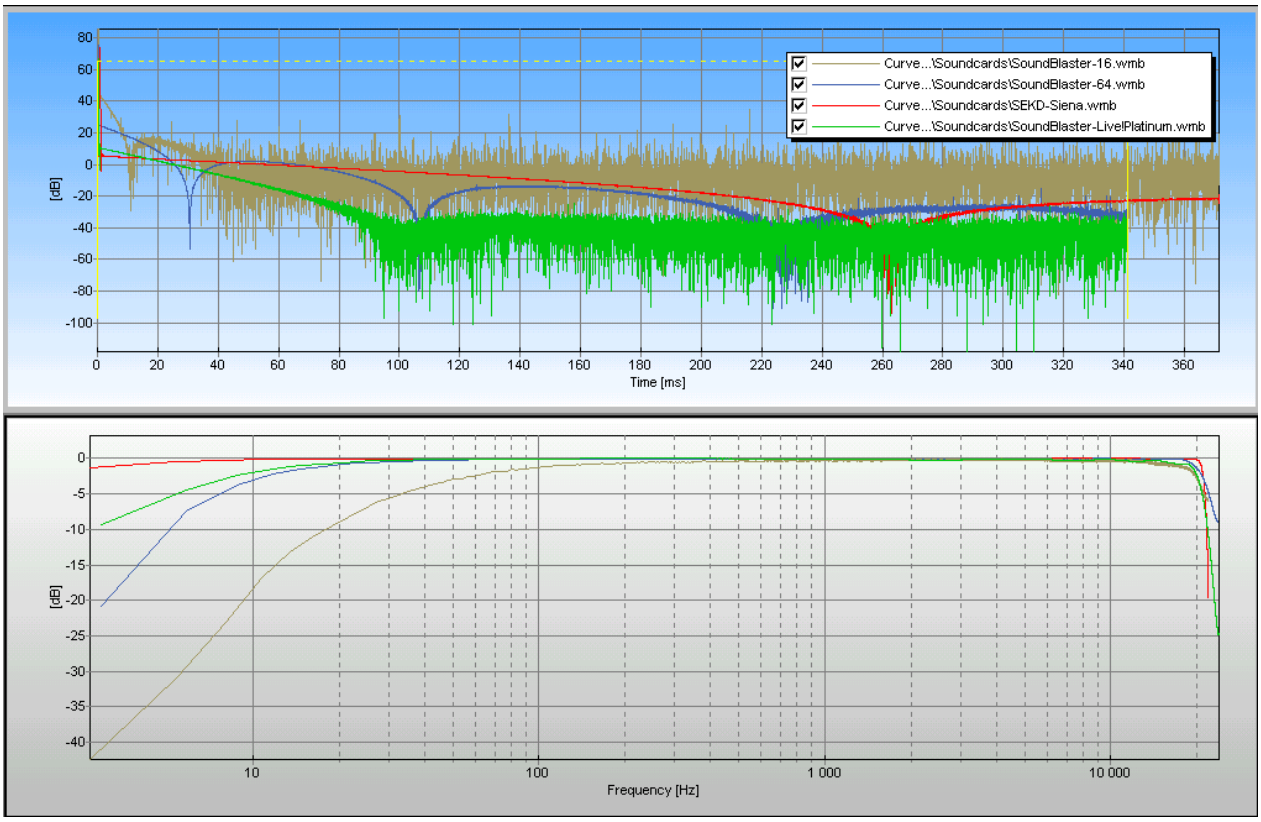


Double-click on the **Soundcards** folder shown in the figure above to display the available measurements of sound cards.
Note that you can select several files by holding the **Ctrl** key when clicking. As shown in the figure below (some of these files may not be found in your installation, but they can be found at our homepage or by contacting us)



Click 

to insert the files. The result will be similar to the figure below (some changes to the layout has been done, which will be explained later). From the figure we see the results from the different sound cards. The measurement in brown could probably be improved by optimizing the volume settings, but we see that the frequency response is smooth for all the measurements.



4.3.5 Sound Card Troubleshooting

If the response of your sound card does not look at all like the other sound cards, you have checked that your settings are correct, the cables are properly connected and the sound card is not on the list of sound cards that does not work with WinMLS, please contact support from our homepage and attach your sound card measurement file.

If the response has spurious peaks, this is an indication of distortion. Try to reduce it by decreasing the input or output level of the sound card.

4.3.6 Saving the settings

WinMLS will always save the last used settings when the software is closed to temporary setup files. You may still want to save your settings. Do this by clicking  on the **Setups Toolbar**.

4.4 Measurement 2: A simple loudspeaker measurement

This chapter assumes that the chapter above has been read. Note that for all sound cards, except for VXpocket and VX222, this measurement procedure will not measure the distance between the loudspeaker and microphone (initial time delay). In the examples we have used a typical low

quality loudspeaker and microphone that comes with the sound card, thus showing how low quality equipment may influence the measurement.

4.4.1 Preparation

Connect a speaker to output left channel. Connect a microphone to input left channel, the **microphone** input can be used if any, then make sure that **microphone** is chosen as **Input Line** in **Measurement->Sound Card Settings**.
Place the microphone in front of and near the loudspeaker.

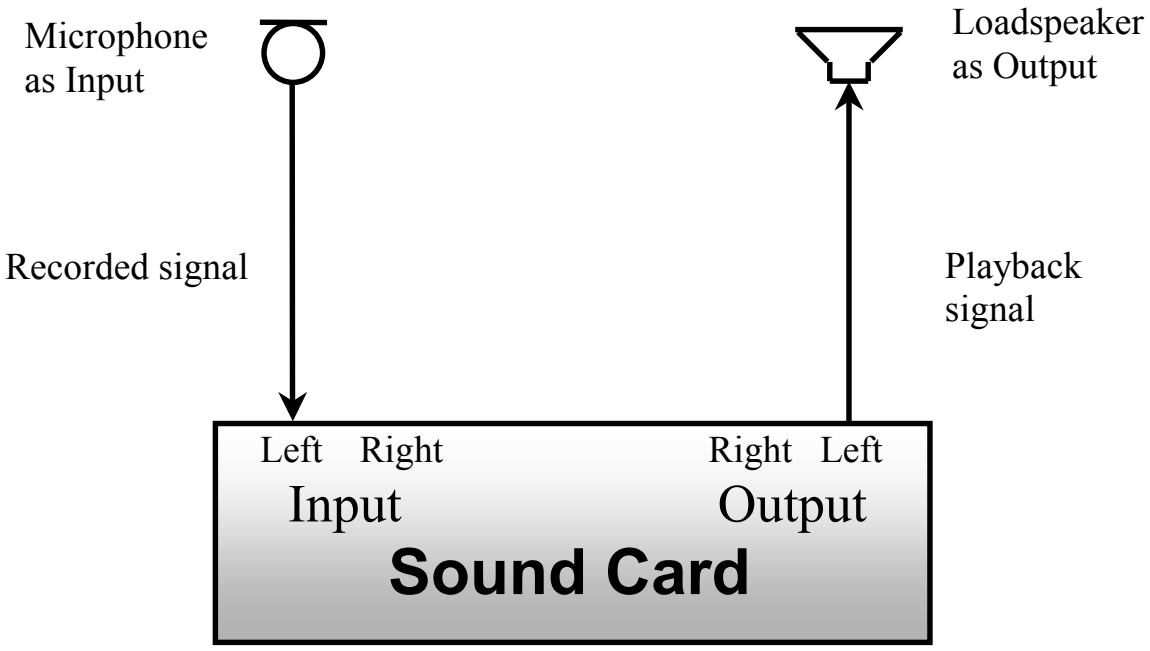


Figure 2: Schematics for one-channel measurement

4.4.2 Measurement procedure

Select **Measurement->Test Current Levels** to display the dialog box showing the input and output levels. Adjust the levels so no clipping occurs, -6 dB should be fine.

When the levels are set, select **Measurement->Start Measurement** to start the measurement. When the measurement is finished, it should look similar to the figure below.

4.4.3 Post-processing the measurement

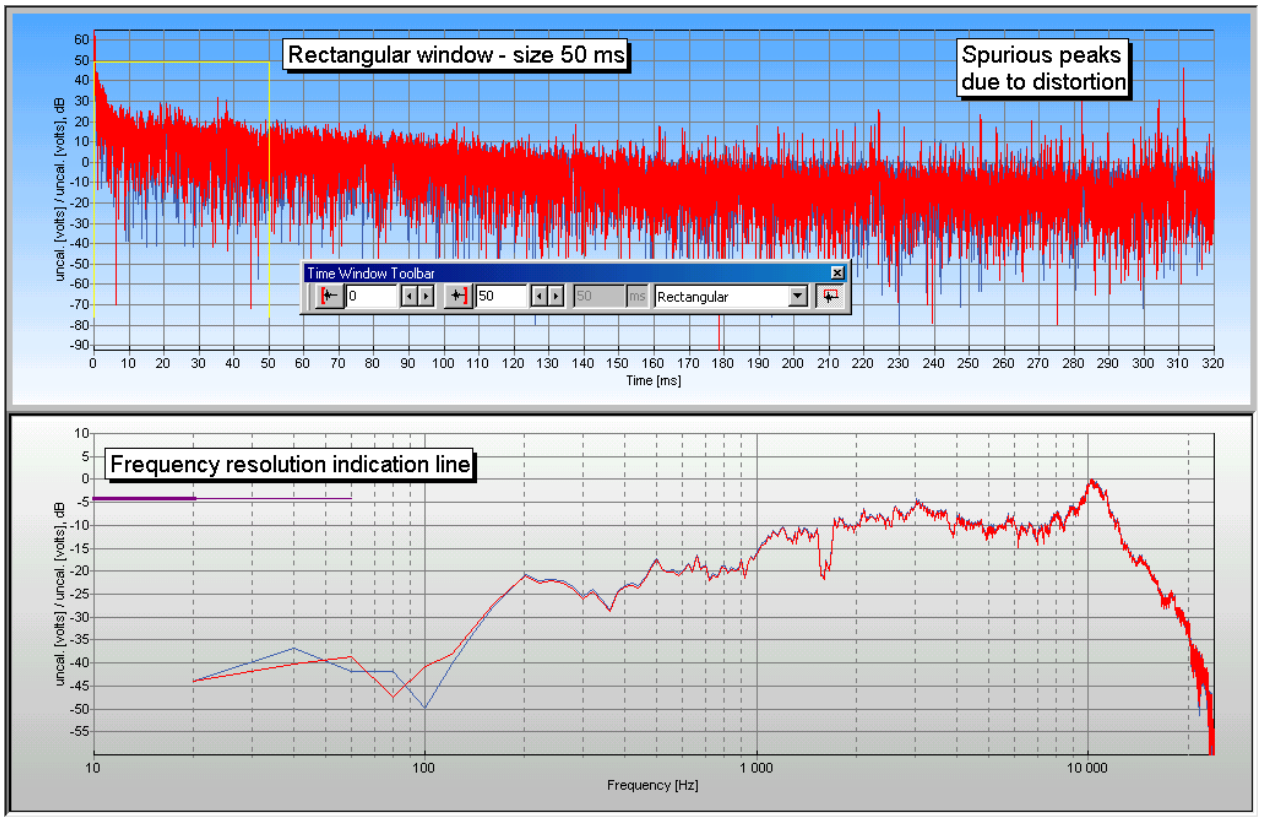


Figure 3 Multimedia microphone and loudspeaker measured on axis at a distance of ~20 cm. Time window 50 ms.

In the upper right part of the figure above, we see that there are several spurious peaks. These are due to distortion in the loudspeaker or microphone, reducing the volume did not reduce the magnitude of these peaks. These peaks do not, however, influence the frequency response shown in the lower part of the plot significantly. Note that a rectangular window is used on the impulse response of length 50 ms displayed in yellow in the figure. The limits can be changed by clicking and dragging, or from the **Time Window Toolbar** shown in the middle of the figure above.
Decreasing the window size will reduce the noise in the frequency response, but note that it will give a decreased frequency resolution and thus false results for very low frequencies. The "Frequency resolution indication line" shown in the figure above depends on the window size and is meant to warn the user where the results may not be trusted. At the thick part of the line, the results should not at all be trusted. At the thin part of the line (which is twice the size of the thick line), the result is not certain.
Note that if the frequency resolution is not sufficient, two resonance peaks may not be resolved, but shown as one peak.

4.5 Measurement 3: Measuring using loop-back

This chapter assumes that you have performed the measurements described above. The main purpose of the loop-back is to *determine the correct initial time delay* of the system being measured (e.g. the distance between the loudspeaker and the microphone). Most sound cards do not have a constant time difference in start of playback and record, it is different from one measurement to another. This makes it impossible to find the initial time delay with the usual measurement method, but this is easily detected if using the loop-back.

When using a loop-back it is also possible to detect errors in the measurement caused by non-continuous play or record. This is usually a problem only for slow PC's, but can occur if you run other tasks on your PC while performing a measurement. If such an error is detected the measurement will be repeated and a warning is given in the status bar.

4.5.1 Preparing loop-back

A loop-back is a physical connection from the output to the input of the sound card as shown in the figure below. The MLS output signal is always played in the left channel and the loop-back output signal is played in the right channel. Connect the cables as shown in the figure below,

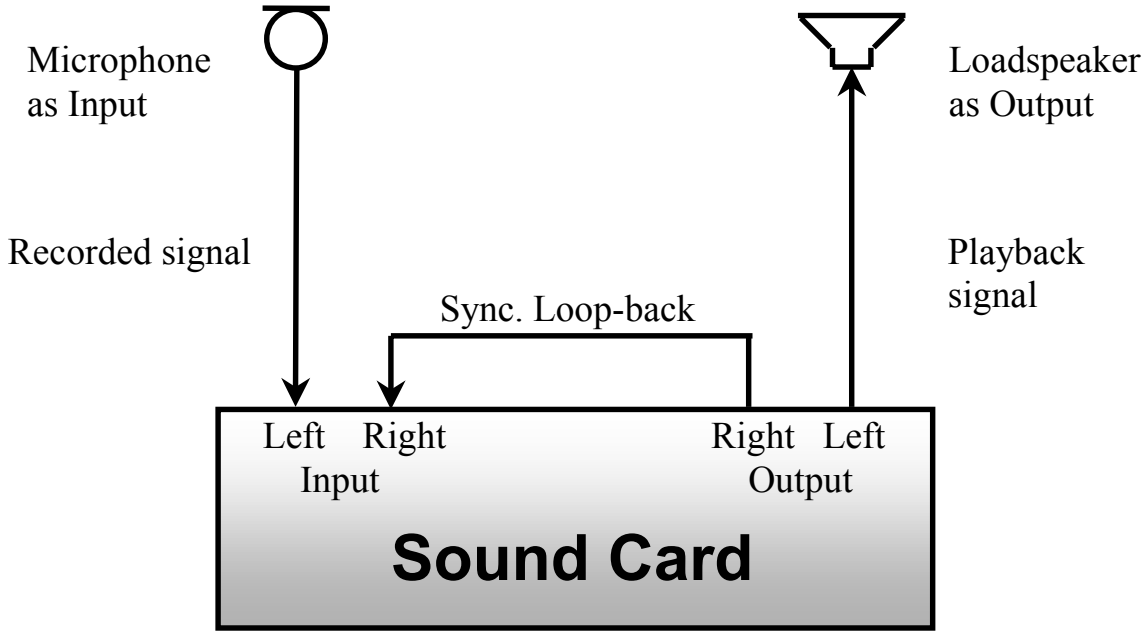


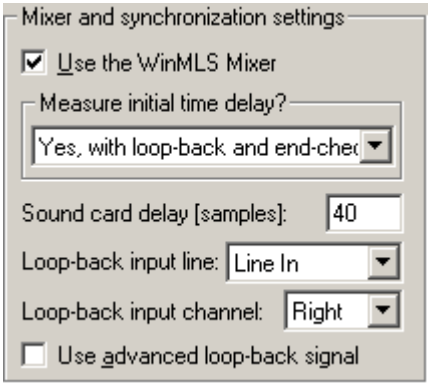
Figure 3: Schematics 1-channel measurement using feedback loop.

Line input is used for

You might need to purchase extra cables to be able to connect the loop-back.

If the input you are using for the microphone signal is mono (the microphone input is usually mono), no *right* input channel exist. Then connect the loop-back input cable to the right channel input of the *line* input instead.

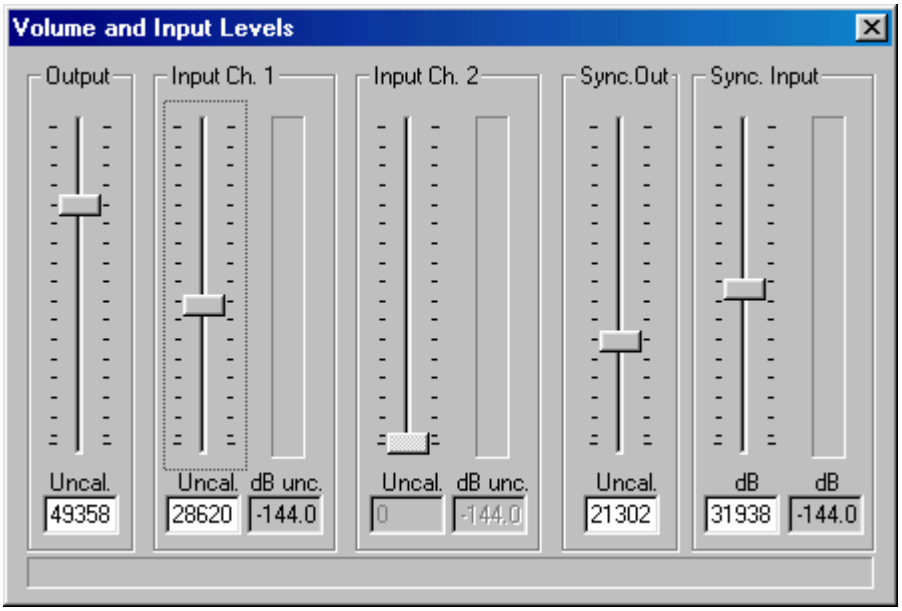
Open **Measurement->Sound Card Settings** dialog box. Make sure '*Yes, with loop-back and end-check*' is selected as shown in the figure below.



Now select the **Loop-back Input Line** you used for connecting the loop-back to, in the figure above **Line-In** is chosen. Make sure **Loop-back input channel** is set to **Right**.
The sound card delay must now be set in samples, this we already have found in the first measurement procedure we described. This number can vary from 10-80. After typing this number, click OK to exit the dialog box.

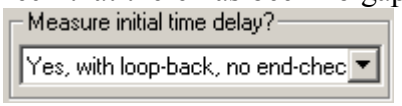
4.5.2 Loop-back measurement procedure

Select **Measurement->Test Current Levels** to display the dialog box showing the input and output levels. If your sound card supports the WinMLS mixer, the volume controls **Sync. Out** and **Sync. Input** will be displayed as shown to the right in the figure below. Use these controls to set the volume of the loop-back synchronization signal.



Select **Measurement->Start Measurement** to start the measurement. An error message will be given if the loop-back volume is too low or too high.
The result of a successful measurement will now have almost the correct initial time delay. But as we have seen, the sound card has a delay of some samples, which is usually less than 1 millisecond. To subtract this delay a *measurement system correction* of the sound card must be performed as described in the FAQ.

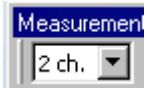
4.5.3 This happens during a loop-back measurement

1. When the measurement is started a synchronization signal is first sent through the right channel output to the loop-back. The synchronization signal is recorded and is later used to find the correct initial time delay.
2. The MLS signal is now played on left channel output, through the speaker. The signal is recorded on the microphone.
3. The synchronization signal is played once more on the right channel output. This synchronization signal is compared against the first synchronization signal and the time difference is computed. The purpose of this is to check that there has been no gaps in the playback or record. This check is not performed if 

is set in **Measurement->Sound Card Settings...**.
4. If no error has occurred, the system impulse response is found using cross correlation. The sound card delay (pipeline delay in sigma-delta converters) is added in order to obtain the whole system response (the default is 40 samples, but should be measured for the sound card as shown above).

4.6 Performing a 2-channel measurement

The number of channels to be measured is set in the **Measurement Settings toolbar**. The figure shows the setting for performing a two-channel measurement.



A 2-channel measurement is similar to a one-channel measurement. If you are measuring using the loop-back, special consideration must be taken as described below. The rest of the procedure is very similar to a one-channel measurement as described in the previous chapters.

4.6.1 Synchronization Loop Connection for 2-channel Measurement

To perform a 2-channel measurement in loop-back mode, the loop-back must be mixed with the measurement signal in one of the channels. How this is best done depends on if your sound card has one or two input lines.

4.6.1.1 2-channel loop-back for sound cards with one input only

If your sound card only has one input line, e.g. Line input, a device has to be used for summing the signals, see the figure below. An audio mixer or a passive device (consisting of 3 resistors) can be used, you may also contact us for purchasing such a device.

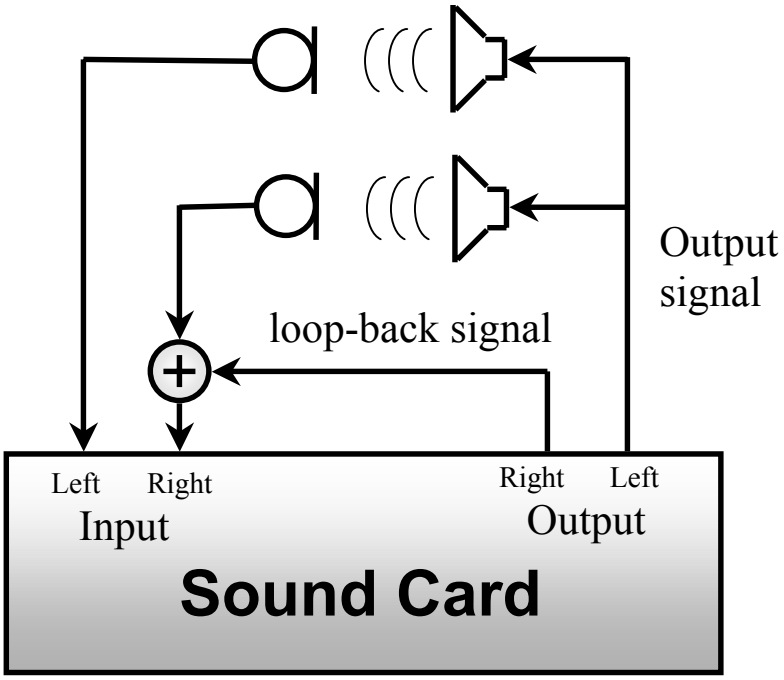


Figure 3: Schematics for two-channel measurement using synchronization loop-back for one single input.

4.6.1.2 2-channel loop-back for sound cards with two physical inputs (e.g. line and microphone input)

If your sound card has two input lines, e.g. **Line** and **Microphone** input, no external device is needed to connect the loop-back. Choose different lines for **Input Line** and **Loop-back Input Line** found in the **Measurement->Sound Card Settings** dialog box. If your **Loop-back Input Channel** is mono, make sure the right channel is chosen as shown below.

Connect, using a cable, the output of the synchronization loop (which is on the right channel) to the line and channel you have chosen as input for the synchronization loop. Note that you may have to split the cable you are connecting to your output to access the output-signal that is on left channel.

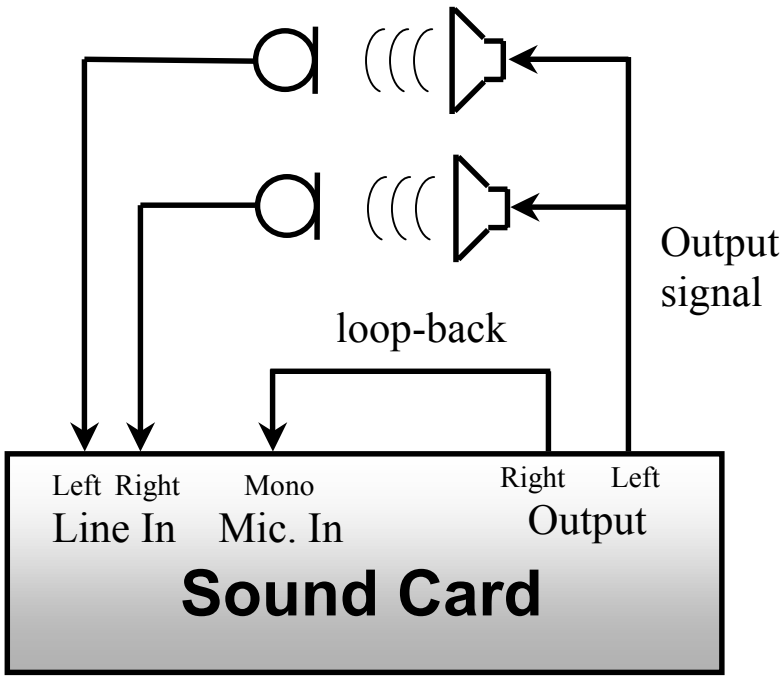


Figure 4: Schematics for two-channel measurement using synchronization loop-back for two physical inputs.

4.6.2 Synchronization Loop-back Connection for Mono Full Duplex Sound Cards

Read this chapter only if the sound card is only capable of *one*-channel simultaneous play and record. If you have such a card you cannot do 2-channel measurements. The card will support 1-channel measurements without using a synchronization loop. But if the card has both a **microphone** input and a **line** input and if it supports the WinMLS mixer, WinMLS also support

1-channel measurements using synchronization loop for such cards. To do this, you must choose different inputs for the **Input Line** and the **Loop-back Input Line** as described in the chapters above. Then make the synchronization loop by connecting a cable from the sound card output to the input you chose as **Loop-back Input Line.** Note that you need to split your output cable from the sound card. One part is for the synchronization loop that we just showed how to connect. The other part is for the output signal.