

# RTL Utility



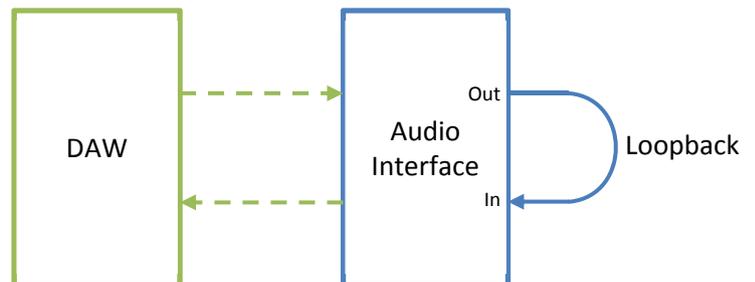
## What is RTL?

RTL is the Round Trip Latency of your DAW and audio interface.

When your DAW sends data to your audio interface for playback, it doesn't send a continuous stream of data one bit at a time. What it does is fill up a section of RAM called a buffer and sends that in one message when it is ready. Before sending the next message it has to fill the buffer again. This wait time introduces a latency, or delay, between something happening in your DAW and when you actually hear it.

When you are recording, the audio interface buffers and sends data to your DAW in a similar fashion. This introduces latency into your recordings.

If you send a signal from your DAW, out through the audio interface and back in via a loopback patch, then there will be a round trip latency which is the sum of the output and input delays. This is the RTL.



## What are the components of RTL?

RTL is composed of the delay due to output buffering in the DAW, delays in your audio interface and the delay due to input buffering. Note that for analogue audio interfaces there are delays associated with A/D and D/A conversion, while for digital audio interfaces there may be delays associated with line transceivers and format conversion.

## Why should I be interested in low latency?

Most DAW software automatically compensates for audio interface latency so that recorded tracks are put in the correct place on your project timeline. However you may notice the latency if you are monitoring inputs live. Artists and musicians can find this quite disconcerting and it can impact their ability to maintain good timing in their performance. It is because of this that many people are interested in how well audio interfaces perform at low buffer settings.

## How does RTL Utility make its measurements?

RTL Utility measures the RTL by outputting a carefully designed chirp signal and recording the looped back audio. Initially, an estimate of the RTL is made by looking for the start of the chirp. This estimate is usually accurate within 1 or 2 samples.

The next step involves 16x oversampling the recorded signal and carefully matching it to an oversampled version of the test signal. A couple of DSP tricks are used to make sure that subsample accuracy is maintained even in the presence of moderate levels of noise or interference.

## How to make measurements

### *Mute your speakers*

Firstly, mute or disconnect your speakers to avoid nasty surprises. The test signal is output at a peak level of -10dBFS to avoid discomfort, but depending on the configuration of your audio interface it is possible (but unusual) to inadvertently create a nasty feedback loop.

### *Make the loopback patch*

Next, make a physical patch (with a cable) from your chosen output to your chosen input. Ideally, the patch would have unity gain. If it's hard to achieve this with your setup, then err on the side of caution and attenuate the signal path (i.e. don't add gain).

### *Start RTL Utility and select & configure your audio interface*

Make sure you have the right device type and device selected in the RTL Utility and select which input and output channels you would like activated. Next set the sample rate using the dropdown box. Lastly, set the buffer size using the drop down.

Note: for some audio interfaces (such as RME Fireface), there may be only one buffer size available in the dropdown box. To change to a different buffer size, launch the device's control panel and change the buffer setting in there. When you return to RTL Utility, just press the "Restart device" button and it should pick up the change and update the drop down box to the newly selected buffer size.

### *Make your measurement*

To make your measurement, just press the "Measure RTL" button. When finished, the results will be displayed in the "Most Recent Results" panel on the "Main" tab. The results will also be logged to a table in the "Log" tab. If there are problems with making the measurement, a dialog box will let notify you (see troubleshooting section).

You'll also see a couple of shortcut buttons for selecting the next highest sample rate or the next highest buffer size. See the section on troubleshooting for some information on issues that can happen with these buttons.

### *Review your results*

Switch to the "Log" tab review your results. The table defaults to showing basic details for each test. However there are plenty more columns available and these can be accessed by right clicking on the header row.

If you can't see the contents of a cell, just double click on it to auto size the column.

Results from the current test session can be copied to the clipboard in CSV format for pasting into a spread-sheet programme (note that all columns will be copied). A log of all measurements ever made by the current user on the current computer can also be accessed.

## Troubleshooting

### *Audio device configuration issues*

Please note that there are a few circumstances where changing a configuration won't work and your audio device will be deselected:

- If the current buffer size isn't supported at the next sample rate setting (e.g. changing from 48kHz to 88.2kHz at a buffer size of 64 with an RME Fireface)
- Some interface drivers (e.g. RME Fireface) remember the last buffer setting per sample rate setting – so sometimes swapping between sample rates will fail because the driver will also try and change the buffer size
- RTL Utility just doesn't interact nicely with some drivers (e.g. ASIO4ALL driver sometimes won't switch neatly from 48kHz to 88.2kHz)

If this any of these happen to you, just reconfigure the audio device and carry on.

### *Only one buffer size listed*

For some audio interfaces (such as RME Fireface), there may be only one buffer size available in the dropdown box. To change to a different buffer size, launch the device's control panel and change the buffer setting in there. When you return to RTL Utility, just press the "Restart device" button and it should pick up the change and update the drop down box to the newly selected buffer size.

### *Couldn't reliably detect test signal*

If you get this message then you need to check a couple of things.

- If the dialog box displays only the noise floor, then the test signal wasn't present:
  - Check that your loopback cable is properly connected
  - Check that you have activated the correct input and output channels (if in doubt try selecting all of them)
  - Check the channel gain/mute settings on your interface (and its control software)
- If the dialog box tells displays both the noise floor and return loss, then the test signal wasn't detected amongst the noise and you need to try and decrease either the noise floor, the return loss, or both:
  - Check that your loopback cable is properly connected
  - Check the channel gain settings on your interface (and its control software)
  - Check that other audio signals aren't being mixed into the input channel (whether this be within your interface control software or an external mixer)

## Credits

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