

# RTL Utility

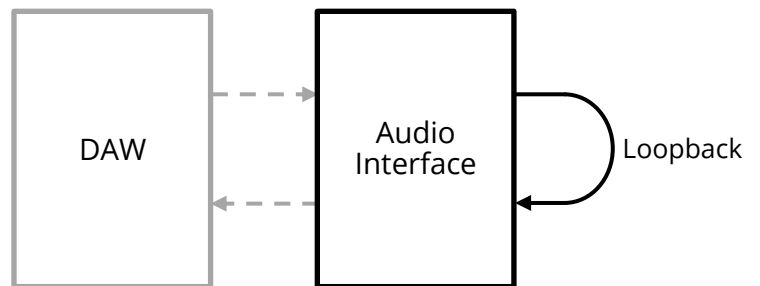
## What is RTL?

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. If you have an acoustically coupled path (i.e. the signal path includes a speaker and a microphone), then there will be a latency component due to the acoustic delay.

## 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 sequence of impulses and recording the looped back audio. The algorithm checks and accounts for any noise while detecting the impulses.

## How to make measurements

### *Mute your speakers\**

Firstly, mute or disconnect your speakers to avoid any chance of a nasty feedback loop which may occur if you have inadvertently patched in a microphone somewhere. The impulses in the test signal are output at a peak level of 0dBFS but are only a few samples wide so you would only hear a very short burst of clicks if you don't mute.

### *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 too much 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: audio interfaces which adhere to the ASIO specification, there will 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 it should already have updated the buffer size in the selection box – but if it hasn't, just press the "Reset Audio Device" button to pick up the change.

### *Make your measurement*

To make your measurement, press the "Measure RTL" button. When finished, the results will be displayed in the "Last Measurement" 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 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

\*RTL Utility can also make measurements using acoustic coupling between your speakers and a microphone. Obviously, you need to unmute your speakers and disconnect any physical patches in this

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.

## GUI Features

Just in case it wasn't obvious, here are some handy things you can do in RTL Utility:

- Application
  - You can zoom the size of the interface elements by using the buttons in the bottom right, or by using Ctrl-Mouse Wheel on the background. Double-click the background to reset to 100% zoom.
- Main tab
  - Use the Store/Compare button to store the current measurement. After any subsequent measurements, the tooltips for the different values will show what the stored value was, and the delta. This can be handy for checking speaker delays for PA systems.
  - Hover over the reported RTL result to show extra details on the components of the reported RTL (e.g. shows the safety offset for CoreAudio)
- Log tab
  - Re-order columns by dragging the column header left or right
  - Sort on any column by single clicking the column header, click again for reverse sort
  - Right-Click a column header to auto size columns, or to show/hide other columns
  - Click on a row and press delete to delete it from the table (no further confirmations will be asked!)
- About tab
  - Click "Check version" to see if there is a newer version of RTL Utility available on the web site
  - Click the Oblique Audio URL at the bottom to go to our website
- Results file
  - Unless you manually edit or delete this file, it will hold records of all your past measurements until it reaches 10 Mbytes in size, at which time old records will be deleted as new ones come in.
- Application log file
  - This contains detailed data to help diagnose problems with measurements or audio device drivers.
  - Unless you manually edit or delete this file, it will hold records of all your past measurements until it reaches 10 Mbytes in size, at which time old records will be deleted as new ones come in.

## Command Line Interface

Since v1.0.9, RTL Utility supports a command line interface (CLI). This is useful if you want to script some tests using python or similar.

Command	Alt	Description	
--version	-v	Display current application version.	(1)
--listdevices	-ld	List available audio devices by device type. Any other commands will be ignored if this is specified.	(1)
--type	-t	Set audio device type (ASIO, CoreAudio, DirectSound, Windows Audio, etc).	
--device	-d	Set input audio device by name. Will also be used for output device if output device is not supplied.	
--deviceout	-do	Set output audio device by name. Input device will be used if this is not supplied.	
--listrates	-lr	List available sample rates for specified device.	(2)
--listbuffers	-lb	List available buffer sizes for specified device.	(2)
--controlpanel	-cp	Attempt to open control panel for specified device.	(2)
--rate	-r	Set sample rate in Hz (e.g. 48000).	
--ratekhz	-rk	Set sample rate in kHz (e.g. 48).	
--buffer	-b	Set buffer size in samples (e.g. 256).	
--inputs	-i	Select input channels using integer bitmask.	(3)
--outputs	-o	Select output channels using integer bitmask.	(3)
--format	-f	Specify output format: simple, csv, csvwithheader, xml, json.	

(1) These commands can only be used separately, and no test will be performed.

(2) Any of these commands can run together, but no test will be performed.

(3) Integer bitmasks have first channel at bit 0, second channel at bit 1 etc. A value of 1 means only the first channel is selected. A value of 4 means only the third channel is selected. A value of 3 means the first and second channel are both selected.

Return codes:

- 0 - OK
- 1 - Command unrecognised
- 2 - Unable to initialise device
- 3 - Unable to detect signal
- 4 - Invalid measurement

Some tips for using the CLI:

- Show the help by calling RTL Utility with `-h`, `--help` or `help`
- Results are printed to `stdout`, errors to `stderr`
- Windows doesn't like GUI apps which take arguments, so the command prompt won't reappear properly.
  - Wait until you see `--DONE--`, then press return to see the prompt
  - Here's an example of calling from PowerShell in the same directory as the application: `& '.\RTL Utility.exe' -t "ASIO" -d "Yamaha Steinberg USB ASIO" -rk 48 -b 64`

- macOS also doesn't like mixing GUI and console apps, so you need to call the executable inside the application package.
  - You can create a helper script like this:
    - ```
echo "\"/Applications/RTL Utility.app/Contents/MacOS/RTL Utility\" \"\$@\" 2> /dev/null\" > rtl.sh
```
    - ```
chmod +x rtl.sh
```
  - You can then call it like this: `./rtl.sh --help`
  - Note that the application is not able to change some values for some devices (e.g., ASIO devices must have their buffer size changed from the device control panel). In this case, you can use `--controlpanel` to launch the control panel for the specified device, then call the CLI again with matching parameters. You can also use `--listrates` and/or `--listbuffers` to determine allowable values.
- Results of CLI commands are still saved to the application log and results file which are accessible from the GUI
- Note that you often get invalid measurements from the CLI for Windows Audio devices operating in WASAPI shared mode.
  - Try and reset the sample rates of the device, and ensure sample rate matches for both input and output.
  - WASAPI exclusive mode can be set with `--type "Windows Audio (Exclusive Mode)"`.

## Troubleshooting

### *Audio device configuration issues*

Please note that there are a few circumstances where changing a configuration may not 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 96kHz at a buffer size of 32)
- Some interface drivers remember the last buffer setting per sample rate setting – so sometimes swapping between sample rates will fail if the driver changes the buffer size but does not properly notify the application of that change
- RTL Utility just doesn't interact nicely with some drivers (e.g. ASIO4ALL driver sometimes won't switch neatly from 48kHz to 88.2kHz)
- You can't use a device in Windows Audio (Exclusive Mode) if it's being used for anything else. Try making another device the default device.

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

### *Only one buffer size listed*

For audio interfaces which adhere to the ASIO specification, there will 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 displays both the noise floor and return level, then the test signal wasn't detected amongst the noise and you need to try and decrease either the noise floor, or increase the return level, 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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