Reducing latency when using

Solocontutti

A guide for the perplexed

What is latency and why is it important

- Latency is the delay between a sound being produced and a sound being heard

  - For example – if somebody is playing an instrument 10m away from you, then because the delay between a sound being produced and it being heard by you is 1/30th of a second (or about 33 milliseconds) because sound travels at about 300 metres per second
  - In Solocontutti the delay is the sum of all the delays in the various components

- Latency, or delay, is important because your ear is very sensitive to timing differences, and if the delay is too long you will unconsciously interpret this as slowing down the tempo.

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What causes latency in Solocontutti?
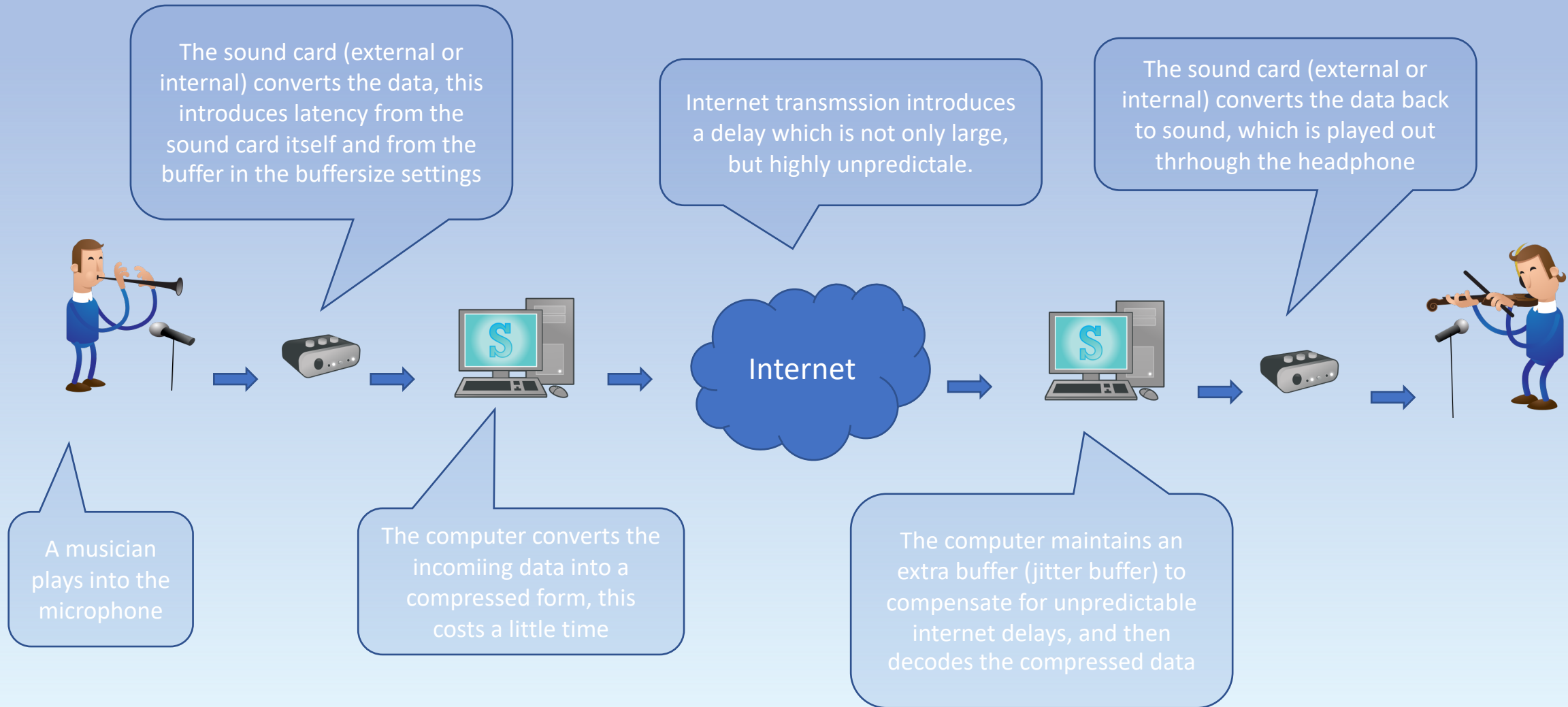
Sound Card Delay

- Run the latency wizard to see what your soundcard latency is. Anything above about 5ms could be a problem.

- Many internal sound cards (e.g. the standard Realtek chips) can introduce serious delays.

- In Windows the DirectSound and Windows drivers can introduce serious delays (sometimes severals 10ths of a second). If you use an internal sound card try to use the ASIO4ALL driver

- Generally an external audio device will give the best result and also allow you to use a high quality microphone

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Delay in the PC/device (outbound)

- Coding/Decoding delay is very low and can't be modified.

- The buffering delay can be adjusted in the sound settings. This is a minimum of 64 samples which is equivalent to 1.3ms delay for internal buffering. A buffer of 480 samples is already 10ms delay, which is significant. Keep your buffer as small as possible.

- There is a catch – the smaller the buffer the more powerful a device you need.

- In the settings you should try reducing the buffer size and activating the test sound. If the test sound starts distorting, set the buffer size one higher and leave it there.

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Internet delay

- This is the most significant source of delay, not because the average transmission time is long (it can be as low as 5ms) but because the variation can be very large (100's of ms).

- Don't use 3G or 4G, the delays are too long. 5G is supposed to be good, but not really available yet.

- WiFi will work, but generally produces extra delay and tendes to reduce quality because it's fairly unreliable for this sort of application. Prefereably connect directly to the cable.

- Bandwidth is almost completely irrelevant and has little to do with transmission times.

Incoming delay on the PC/device

- When the data comes in the receiver needs to compensate for unpredicability of the internet. This is done with a buffer called a jitter buffer and is usually the largest single source of delay. The larger the unpredictability of internet the larger a buffer you need and the more delay you get. Without the jitter buffer you would lose a lot of data and the quality would degrade.

- Solocontutti uses an "adaptive jitter buffer" which means it adapts to changing internet conditions to keep a constant quality. This is why the delay can vary slightly during a session

- The parameters of the jitter buffer can be modified in the tuning dialogue to choose the best tradeoff between quality and delay. There is a separate video tutorial on this

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Incoming sound card delay

- The same rules apply to the incoming delay on the sound card: keep the buffer size as low as possible, use the right drivers and preferably use a external sound device.

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Summary

- Use low-latency sound devices and drivers
- Keep the internal buffer size as low as possible
- Use the best quality internet you can – preferably with a wired connection
- If you want, adjust the parameters of the jitter buffer

- And of course – if someone is watching Netflix or downloading a large file whilst you are playing, this won't help at all.

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