



## Reducing audio delay (latency) in RAVIN

The information in this article applies to:

RAVIN 1.1 (R8) or higher, RAVIN Media Server

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### Summary

This article discusses some common causes of audio delay and offers techniques for reducing it.

### Symptoms

Audio latency is 1 or more seconds.

### Cause

Audio latency can be introduced by the Media Server capturing or playing out audio, or by other activity on the network. Recording does not introduce any delay as recording is done in the same time-envelope as the audio being captured or played out.

### Resolution

To reduce the delay that the Media Server introduces on either end, you will want to change the **FrameQueueLength** and **FrameMultiplier** values in the XML for the advanced sessions used for recording and playout on the respective Media Servers.

**FrameQueueLength** represents the number of audio frames in the queue passed to the audio device (microphone or speaker), while **FrameMultiplier** represents a factor of frame sizes within that queue. A factor of **1** is the lowest value and indicates a frame size of 10ms, a factor of **2** indicates a frame size of 20ms. A queue length of **10** indicates that 10 frames (each of factor **FrameMultiplier**) are in the queue. For example, the default values for **FrameQueueLength** and **FrameMultiplier** are **10** and **2** respectively. This means that the queue consists of 10 frames, each of 20ms. In other words, the total length of the queue is 200ms (10 x 20).

To reduce delay, you can reduce these values. For example, reduce **FrameQueueLength** to **8** and **FrameMultiplier** to **1**. This will make the queue 8 frames of 10ms each - i.e. 80ms vs. the default of 200ms. HINT: I would not drop

the queue length below 5 ! - and definitely NEVER below 3 as you may run into a situation where the queues run dry out of the soundcard (audio capture) or into the soundcard (audio playout).

The reason for the default of 10 length times multiplier 2 is for compatibility with most soundcards on the market-- including USB devices which don't always like a frame size of 10ms (some of them tend to fail and either deliver really bad audio or simply lock up the entire machine).

So, assuming that your values for these parameters are at their defaults, you're going to have a worst case scenario of 200ms on the capture end and 200ms on the play-out end. To that you must add network delay to come up with the overall delay. So, if the delay is 2 seconds currently (and the RAVIN defaults are as above), you're looking at a network delay of around 1.6 seconds (1,600ms). That's REALLY bad !!!

To begin tuning, try changing the **FrameMultiplier** on each Media Server first to a value of 1 and set the **FrameQueueLength** to a value of 8. If the delay is still bad, change the **FrameQueueLength** to 5 but watch out for choppy audio. Also, try performing some roundtrip tests on the network between the two machines to see what kind of delay there is. Note that using PING to measure the roundtrip is not necessarily a good test as ICMP packets that PING sends out tend to have higher priority on the network than regular UDP traffic. Also, please check whether QoS or some other traffic prioritization is setup on the network and whether the channels in RAVIN are configured for QoS (we use DSCP packet marking for prioritization).

Check the advanced XML config for your sessions and look for attributes named **FrameQueueLength** and **FrameMultiplier**. If you see them; modify them, otherwise add them in as new attributes - e.g.

```
.  
. FrameQueueLength="8"  
FrameMultiplier="1"  
.   
.
```

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