Notes on a path to lower JamKazam latency

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I've been able to keep total latency at 19-21 mS which becomes hardly perceptible. That's with somebody 200 miles away. It's like playing with somebody across the room or that drags the beat a bit. It's not the same as being right next to somebody but it's a decent option.

I have a Mac. There may be some PC things I'm missing. Let me know and I'll add them.

The latency train has several key components

Settings and equipment anywhere can make a difference. Key components: Your AI—>Interface from AI to Computer—>JamKazam settings—>Computer Processing Load/Speed—>Interface between computer and router—>Router Settings—>Internet Speed—>JamKazam's Audio Server. And lastly…your co-jammer(s) is the other half of the problem/solution.

Thinking about these as separate targets for improvement might help you break it down into a more digestible problems.

What's your latency?

When in a session (Solo or with somebody) you can see your Audio Interface (AI's) latency. Click on the green/yellow/red dots. The AI Latency number should be low. Between 2.5 and 3 mS is about as good as you'll get. Between 8 and 10 is ok but there's probably room for improvement. When you are connected to somebody else you can mouse over their green/yellow/red dots and see a more complete set of numbers. You can see their AI latency but also the total that includes the components from your computer to theirs.

Here's what I've tried so far…

Hardwire to router!

Absolutely no WiFi. Shortest run possible. If your laptop doesn't have an ethernet port you can get an adaptor. Choose the port that is fastest. USBC > USB2. There are firewires that are faster and slower than USBC.

Choose your AI (Audio Interface, also often Digital Interface)
Everybody seems to love the Focusrite. I bought it based on that. But, Motu is faster than Focusrite. I went from 8.4 to 2.8 on an older Mac with just that change. That improvement was confirmed with a friend who changed out to a Motu as well. Another friend with a Motu gets 2.5 mS on a PC. I don't know about other AIs.

AI Drivers
Get the custom driver for your AI. It will be optimized for lowest latency which isn't the case for your default. For Motu on a Mac the driver drops 5 mS. Remember that when you upgrade operating systems you might (probably will) need to get an updated driver. If your AI latency goes up or if you have other new symptoms it's possible that the stock driver got re-installed and you need to replace it again. Check for firmware updates for your AI as well.

Buffer Size and Sampling Rate
The jury is still out on this. There may be other bottlenecks within your AI and the USB interface that limit your ability to take advantage of this. 48kHz may be as high as you can go on a USB2.0 box.

Theoretically, setting buffer size as low as possible (32 samples on mine) and sampling rate as high as possible (96 KHz highest supported by JK) will give you the lowest latency. Do this in your AI control program (Focusrite) or your DAW (Digital Audio Workstation ie Ableton Lite, Reaper, ProTools, GarageBand etc). This seems a bit counter-intuitive but the equation works out.** You may lose some quality and get some occasional crackle or data loss…but lower latency in king.

If you have more than one DAW check them all and go back occasionally and make sure they all say the same thing. Usually in the preferences. The setting on one may have a universal effect on the others. I suspect they control/modify the driver.

Frame Rate
Set the frame rate on JamKazam to 1 mS. JK seems to occasionally re-set this so check it every now and then. You may also have to re-install your Audio Source in JK every now and then for the same reason.

Router and Port Forwarding

Set up Port Forwarding (Peer-to-Peer). Confirm that your ISP allows it. That dropped about 10 mS. Log into your router, go to advanced settings. You may have to shut down or allow exceptions to your firewall. JK's lack of explanation of settings isn't helpful. But…under Manage—>Network Settings—Route select Prefer Peer-to-Peer if you have this set up. If you select Prefer Lowest Latency I suspect that it checks every now and then to make sure you are using the lowest latency path.

Have adequate bandwidth and speed. Use a service like https://www.speedtest.net to see how you are doing. Fast up and download is nice but speed and jitter have bigger impacts. A ping in the single digits is best. Your ISP may be able to help you diagnose problems. Sometimes restarting your router or having them cycle their equipment helps.

Optimize Your Computer

Anything that takes up processing power can slow things down. Aside from buying a more powerful rig there are things you can do.

Get rid of unnecessary programs in your start up menu. A lot of things get put there with out you knowing and they probably don't need to be running.

There are some other tips for optimizing your computer for recording here. https://support.focusrite.com/hc/en-gb/articles/207355205-Optimising-your-PC-for-Audio-on-Windows-10 and https://support.focusrite.com/hc/en-gb/articles/207546515-Optimising-your-Mac-for-Audio. I've already included the most important ones that I think apply to JamKazam.

JamKazam

They appear to be working on their service and that's good. There's probably room to improve the latency of their program but also the data transmission/connection. There are two ways to connect supported by JK. Peer-to-Peer and Client-Server.

The Peer-to-Peer model has the lowest latency but requires more band width, processing power and computer/router settings. You each send each other data and process it on your machine.

The Client-Server model involves an Audio Server at the hub that mixes and sends out a combined mix. This is simpler for the user but adds latency at the hub. I think about 10 mS.

When you go to Manage—>Network Settings—>Route you are choosing one of these or the third option to have JK choose the faster of the two for you. I suspect that when you select the third option it diagnoses the two and picks one for you. I don't know if it does that once or if it keeps checking throughout the session. If it were checking frequently that might add to jitter or cause little hiccups.

What happens when some users in a session are Peer-to-Peer and some are Client-Server? I'm not sure how JK manages it but it doesn't appear to be a big problem.

> Hybrid models are a combination of peer-to-peer and client-server models. A common hybrid model is to have a central server that helps peers find each other....There are a variety of hybrid models, all of which make trade-offs between the centralized functionality provided by a structured server/client network and the node equality afforded by the pure peer-to-peer unstructured networks. Currently, hybrid models have better performance than either pure unstructured networks or pure structured networks because certain functions, such as searching, do require a centralized functionality but benefit from the decentralized aggregation of nodes provided by unstructured networks

Things that may have helped

- Short high quality, shielded cables for mics
- Short high quality, shielded (Cat8) ethernet cable
- Replace provided USBC—>USB2 cable provided with AI with high quality, shielded USBC to USBC cable for DI to computer
- Use a condenser mic which is more responsive than a dynamic mic
- USB mics should not be used. They have high latency.

While In a Session

Here are some things that may make a difference. They mostly reduce competition for internet bandwidth or processing power. Some are more important for recording because they cause interruptions. They also increase jitter (variance of latency). Every little bit helps.

- Ask your spouse to quit streaming Netflix or stop working from home
- Be close to your mic. Each foot adds a mS of un-measured latency
- Turn WiFi off to computer
- Disconnect unneeded peripherals.
- Set phone, etc to airplane mode to reduce competition for bandwidth
- Turn of auto-sleep
- Stop back-ups
- Turn of Auto-Update (PC settings)
- Get the junk off your computer desktop (Mac monitors it constantly)
- Turn off tool bar animations (Mac)
- Disable firewalls
- Turn off Bluetooth.
- Stop the spotlight indexing if it is running (Mac)
- Hit the Resync button in JK every now and then
- Distance adds latency
- The number of people adds latency

Living With Your Latency
With any luck you can bring your latency down to 20 mS or less. It's still going to be different from sitting next to somebody.

Use a metronome or back-up program like StrumMachine. With a total latency at 19-21 mS it can become hardly perceptible. It's like playing with somebody across the room or that drags the beat a bit. But, if someone in the session is used to compensating for people like that (catching up at the end of phrases, etc) or doesn't have rock solid timing than you can spiral.

The metronome doesn't spiral. You just can't "lock in" like you do in the "epic" session. But, I find that after a tune or two your brain accepts that it's a bit out of sync from where you'd like it to be but it's steady and after that it becomes a non-issue.

If you want to try StrumMachine I recommend running it from your phone or iPad through a second channel (if you have one) on your AI. You'll need a mini—>quarter patch cable. The patch cable should be good quality and ideally shielded. The first one I had a bit of noise. I leave my phone unplugged. The power cable is unshielded and was picking up an AM sports station. I also use a pre-amp for mine. The output of the phone is underpowered and you have to push both the phone and AI gain to get enough volume. I've put the chords in for 600 or so tunes so you can probably find most. The link is on www.TaterJoes.com.

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Explaining effect of buffer size and sample frequency on latency:

> Buffer sizes are usually configured as a number of samples, although a few interfaces instead offer time-based settings in milliseconds. The choices on offer are normally powers of two: a typical audio interface might offer settings of 32, 64, 128, 256, 512, 1024 and 2048 samples. You can calculate the theoretical latency that a particular buffer size setting will give you by doubling this number—to reflect the fact that audio is buffered both on the way in and the way out—and dividing the result by the sampling rate. So, for example, at a standard 44.1kHz sample rate, a buffer size of 32 samples should in theory result in a round-trip latency in seconds of (32 x 2) / 44100, which works out at 1.45 milliseconds.

The bottom line of this quoted paragraph is that latency is proportional to sample size (it goes up, latency goes up) and inversely proportion to sample frequency (it goes up, latency goes down). Therefore: to get the lowest latency you want the smallest sample size and the highest sample frequency that still allows acceptable sound quality and data transmission.

However, it has been pointed out that this may be moot or at least of limited efficacy due to other bottlenecks in the system. In particular, the transfer rate of USB audio systems or your AI. MOTU says this "MOTU's expertly engineered USB drivers, which deliver class-leading, ultra-low 2.5 ms Round Trip Latency (at 96 kHz with a 32 sample buffer)." Changing from 48kHz to 96kHz currently only gives a slight decrease in latency. I will update once MOTU provides a new driver for Big Sur operating system.