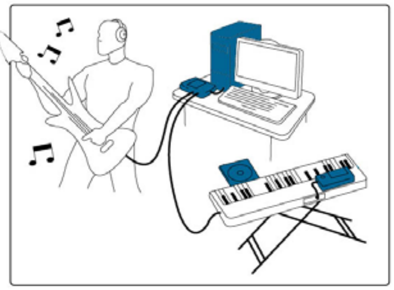
Get Your Groove Back

How recording delay affects playing



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Digital technology has caused many changes in the way music and audio are created and recorded. Today's digital audio workstations are smaller and more feature-rich than the analog recording set-ups of the past. Digital technology has also drastically reduced the cost of producing professional audio. These benefits have opened up the world of professional audio to the consumer market, and have made it much easier for musicians to record themselves. However, recording that "killer" guitar part is more complex than simply downloading some software off the internet, plugging a guitar into the computer's soundcard, and pressing the record button. Beware of the pitfalls of digital recording.

Delay – The Drawback of Digital

It all used to be simple when an acoustic signal, picked up by a microphone and converted to an electrical signal, moved along the microphone cable to the mixer, then to the recorder, and then back to the musician at the speed of light. Everything happened instantly. These days the electrical signal is converted into a digital signal, or binary number, and passed to the computer for processing. Unlike the analog realm, the processing in the digital domain does not occur right away. Computers introduce recording delays (latency) due to their sequential nature of operation. At the end of the paper we will explain precisely why this happens.

The Gift of The Groove

Hearing yourself back delayed can cause timing problems and throw off one's sense of the groove. "Groove" is colloquially known as the "ability to play in time", a quality of music that is hard to quantify. An experienced musician can easily tell when another player is "in the groove," but that defining point of "grooving" is not absolutely clear. Groove actually refers to creative timing and not to perfect timing, because if perfect timing equaled a perfect groove, every piece of music created with computers would sound right. Yet we know that quanitized music feels robotic -- if all notes are spaced apart with mathematical precision the music lacks that human "feel". Staying in the groove then is a matter of playing ahead of or behind the beat by the slightest amount of time. Recording engineers know that shifting certain tracks (such as rhythm guitar) against the rest of the band by as little as several milliseconds can sometimes liven up an otherwise dull performance. Naturally, when trying to get that perfect take, even a tiny delay becomes detrimental. So how much is too much? At first, a simple estimate.

Perception of Delay
Few musicians can produce a run of 64th notes at a sustained rate, so we'll say that a 32nd note is the shortest note a person can physically play. At a tempo of 120 beats per minute, a 32nd note has the duration of 62.5ms, which is the time it takes for the nerve impulses to reach the hand and for the hand to move on the musical instrument. The speed of our perception is much higher than that.

A German scientist named Helmut Haas in 1949 published one of the earliest papers on the perception of delay. His findings are known as the Haas Effect, which states that if the echo follows the original sound very closely, the two will be perceived as a single sound. The echo could be delayed by as much as 30 ms before our brain will recognize and treat it as a separate sound. The Haas effect comes close to understanding the delayed monitoring phenomenon, but we need to take it one step further to apply to our situation. What if there is no direct sound at all, as is the case with electric instruments plugged directly into the computer? How does the delay affect the timing of a musician when there is no direct sound present?

Experimental Setup
A simple experiment helped us understand exactly how much delay is too much. We tested two musicians: a bass player, and a DJ-turntablist. Each was tested separately and was instructed to play a fast rhythmic figure over an over again. Bass players and DJs can play percussive parts and are therefore very sensitive to small timing inconsistencies. Testing a drummer or a percussionist would produce even better results, but drums are a rather loud instrument. In practice, it would be hard to mask the direct sound reaching the drummer's ear.

Our test subjects heard a drum beat over the headphones and were instructed to play their rhythmic figures precisely on the beat. The signal from their instrument was fed through a variable digital delay, the output of which was mixed in with the drum pattern and sent back to the headphones. This way, the musicians' hands were effectively "disconnected" from their ears. The goal of the study was to find the point at which the delay was noticeable to the musician. We conducted a number of trials with varied delay lengths, from 0mS to 50mS in small increments.

Results and Conclusions
Right away it became apparent that neither musician could play while hearing himself through a 50 ms delay. We lowered the numbers until the point when each musician could comfortably execute their fastest run. Then we raised the delay to find the "sweet spot". All experiments were conducted multiple times for statistical validity. We concluded that the DJ could reliably notice the delay down to as little as 12ms, and the bass player noticed it all the way down to 9ms.

Can people successfully perform while hearing themselves back delayed? Of course they can. Humans can adjust to almost anything. But this delay does not go unnoticed. The brain has to work extra hard to readjust and filter out the unwanted sounds, which results in earlier fatigue. A musician would get tired faster and his "chops" won't last for long. In order to keep the musician relaxed and the music grooving, delays in monitoring should be brought down to a minimum.

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Zero Latency Monitoring

One solution that has been used to avoid the problem of latency is “Zero Latency Monitoring.” This method allows the I/O device to send the input signal directly to the output before it is converted into a digital signal. The delayed feedback problem is solved if the musician can play while monitoring the input signal only. However, in some cases a performer may need to hear the processed signal while recording, which renders this method ineffective. As an illustration, consider that the direct signal has to be significantly altered (compressed or equalized) for either aesthetic or corrective purposes and the musician needs to hear that altered sound in order to play right.

Sources of Delay

Some delays occur even before the audio gets to the computer, such as the analog to digital conversion, which is performed by the audio interface and which takes approximately 1ms. Inside the computer, the device driver takes the audio from the USB or 1394 jack to the operating system, which in turn hands it to the recording or processing application. Each step in the resulting “conveyor” takes up additional time, causing the delays to stack up. When the audio is recorded/processed and is ready for playback it moves through the same conveyor backwards. The D/A conversion inside the audio interface takes another 1ms. Further, the audio in the computer flows in chunks, known as buffers. The buffer size refers to exactly how many samples will fit into one chunk of data that is being sent at a time. Typical buffer sizes are 64…512 samples. It takes time to fill up or empty the buffer, but configuring one’s system to the smallest buffer size is not necessarily the answer. More CPU power is being used when the buffer size is smaller, and this can cause digital errors, so you always have to find a balance between the acceptable delay and ending up with “glitches” in your recording.

ASIO to the Rescue

Steinberg™, the German manufacturer of Cubase™, Nuendo™ and other recording software developed the ASIO specification, a way to reduce latency by not letting the operating system handle the audio, and instead passing it straight from the application to the output jack. ASIO drivers, available from several manufacturers including CEntrance, remove the extra “handling” step, thereby reducing latency. ASIO drivers are the answer, but they need to be configured properly for your particular system to minimize delay. Most ASIO drivers will let the user configure sampling rate and bit depth. One needs to keep in mind that audio with a higher sampling rate and bit depth takes longer to process and can cause additional latency. It takes some time to experiment with the settings to “tune” the digital recording system just right. Once tuned however, it should be stable for a long while.

One thing is good about computers. Unlike pianos or cellos, they shouldn’t need to be tuned every time the weather changes outside.

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