ENCO Recommendation

Digigram and Audio Drops

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Several different anomalies can result in audio drops or muting. The audio mutes during the middle of playback then returns later on its own, the level indicators go blank during this period, the DAD counters continue to run and there are underruns reported in the debug file.

The level indications are derived from calls to the audio card and are therefore only an indication of what the card is outputting. If there is a significant data underrun that causes the buffered data to fall behind what the card needs to stay in time, the card will mute the audio (and level info) until it is able to recover sync. The audio card knows the timing based on it's reference clock, so the card continues to report timing data to DAD thus the counters continue to run. Underruns are reported until the data can catch up at which point the card returns to outputting audio.

These are classic signs of a throughput issue. Obviously these can be caused by server problems, either physical like drive problems or logical like a file system requiring maintenance. But if the server checks out OK, don't rule out network problems. A network environment can be extremely complicated to diagnose and any problem at the physical, link and/or protocol layers can cause trouble.

However, we know from experience that most common network throughput problems can be traced to a few simple things: Improper preferred protocol selection and/or superfluous or improperly ordered IP/IPX bindings in the network client interface, improper FDX/HDX negotiation between the workstation NIC and network switch, or believe it or not, an extremely common cause, faulty CAT5E cabling due to incorrect RJ-45 connector selection and installation (connectors for stranded wire installed on solid wire). As basic as some of this sounds, you must make certain these things are OK before doing any more involved trouble shooting.

Assigning arbitrarily larger numbers of buffers to the Digigram driver will not necessarily help throughput problems. The Digigram driver is only designed to use a maximum of 4 buffers per audio stream. The maximum number of streams per physical I/O channel is limited by the DIGIGRAM_MAX_MIXES parameters in the DAD.CFG file. This is usually set to 3 although an early default setting was 5. So a simple formula determines how many buffers you need:

Total buffers = physical I/O channels X 4 buffers per stream X max streams

For a PCX-822 with 2 physical inputs (1 stereo pair) and 8 physical outputs (4 stereo pairs) and with max mixes set to 5 we get: Total Buffers = (2 + 8) X 4 X 5 = 200 buffers. Assigning any more buffers for this card will server no purpose other than to waste memory. However, since these buffers are allocated at startup, assigning too few WILL cause erratic operation.

As for the buffer size, this is a trade off between network latency and audio startup delay. Larger buffers can compensate for moderate network latency but the price you pay is more delay between pressing a button and audio playback actually starting. With modern 100 Mbps and 1 Gbps networks, the default 32K buffer size is totally adequate for even linear PCM audio. Older installations with slower servers may benefit from raising the buffer size to 64K, but if you need to go beyond that, you should really be taking a thorough look at your network.