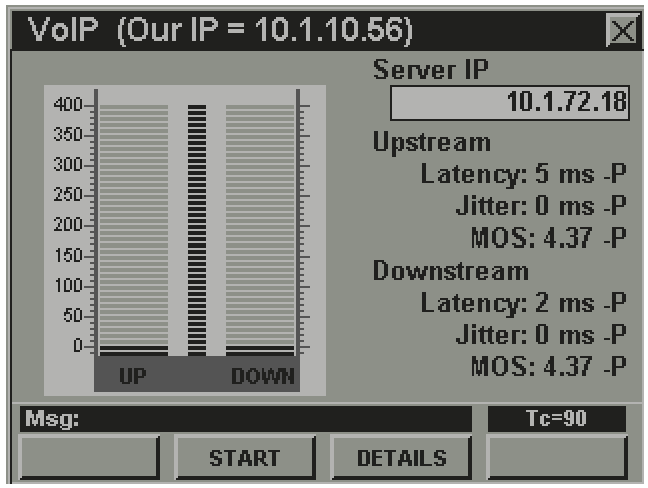
860 DSPi VoIP Testing

Cable operators test their networks to ensure signal quality and signal leakage complies with FCC guidelines. With VoIP, new parameters have to be monitored, analyzed, and tested to ensure that customers are getting the best service.

VoIP Services

The main impairments for VoIP are latency, packet loss, and jitter.



Latency

Performing all the functions that are required to process and packetize voice signals and then transport those from the origination point to the receiving point in any IP architecture – including PacketCable – takes time. Each function requires fractions of a second, but the total amount of time varies based on the architecture of the device as well as the amount of traffic to be processed. This time delay is known as latency. Most network latency occurs after the packets leave the endpoint, or gateway. Every time a packet encounters a network router, a few milliseconds or more of additional latency is introduced. Therefore, unless the signal is kept within a carefully managed intranet or similar network, there is no control over the number of router-to-router hops that a packet may make. It is necessary to monitor the total latency that a packet experiences is necessary to maintain a high-quality signal transmission.

Delays below 150 milliseconds are considered acceptable for most VoIP communications. Delays ranging between 150 and 300 ms are acceptable, depending on the voice quality desired, but delays over 300 ms are unacceptable. Delays on VoIP sessions are measured in two categories – fixed and variable.

Fixed Delays Include the Following:

- **Propagation delay:** The time necessary for the packet to be transmitted over the physical link. This delay is usually bound by physical characteristics of the transmission media (using a fiber optic circuit, it would be bound by the speed of light).
- **Serialization delay:** The time necessary to place the bits from the transmission buffer into the transmission media. The higher the speed, the less serialization delay.
- **Processing delay:** Includes the time necessary to code, compress, decompress and decode the voice signal, and the time necessary to collect enough voice samples to be placed on the payload for a data packet. This varies, depending on the algorithm used.

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1-800-344-2412 or 317-895-3600
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Variable Delay Includes the Following:

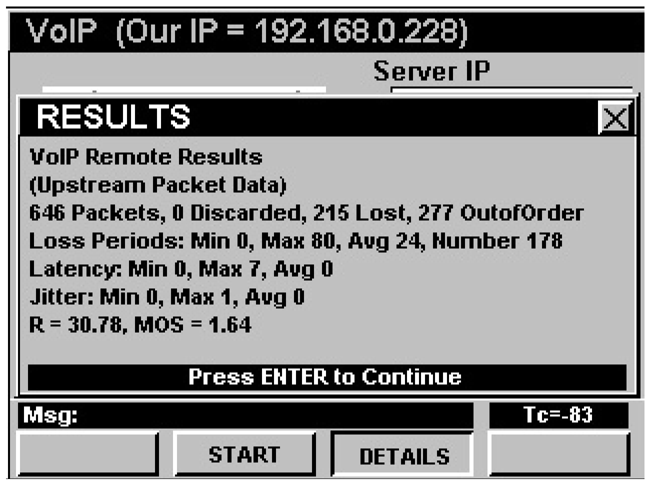
- **Queuing Delay:** The time a packet has to wait in a router before it can be serviced. This delay will occur at every router in the path of a VoIP session.

Jitter

In addition to being sent over an unpredictable number of router hops, packets are routed from one router to another, using different assigned routes – each of which has a different amount of traffic to handle. Therefore, packets from the same voice conversation will experience differing amounts of latency as they head toward their destination. These variable delays produce jitter – a phenomenon that comes from different packets arriving at the destination at different points in time. Gateways use buffers to collect and hold the packets and put them back in the proper order. This process has to be optimized, so as not to introduce its own unacceptable latency. Again, jitter must be effectively monitored to be sure it it being properly handled.

Dropped Packets

When traffic levels rise to a level that overloads a router, the device may intentionally drop packets to relieve congestion. Error-checking has been built into the protocols and is used to maintain data integrity for HSD and digital video. This procedure requires additional overhead, and is not optimized for voice signals (VoIP uses UDP). A certain number of dropped packets (less than one percent, typically) can be tolerated by the human ear before signal degradation is perceived, but beyond that amount, call quality can degrade to unacceptable levels.



(UDP) User Datagram Protocol: UDP does not provide the reliability and ordering guarantees that TCP does. Datagrams may arrive out of order or go missing without notice. Without the overhead of checking to verify that every packet actually arrived, UDP is faster and more efficient for time-sensitive purposes, such as VoIP.

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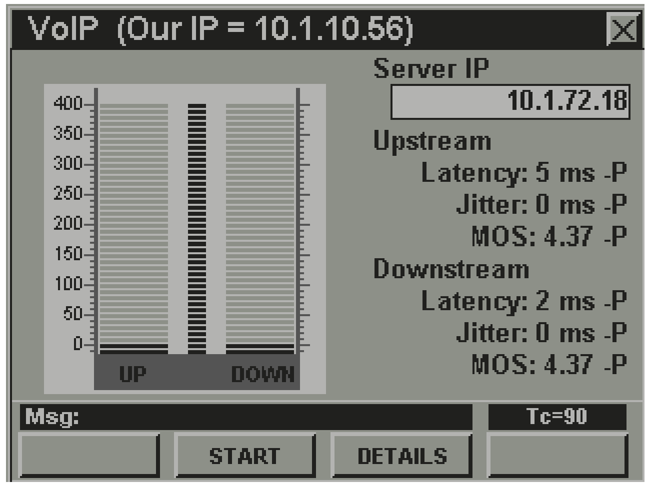
Mean Opinion Scores



Speech quality is usually evaluated on a five-point scale, known as the mean-opinion score (MOS) scale. In speech quality testing, an average is taken over a large number of speech data, speakers, and listeners. The five points of quality (from one to five) are: bad, poor, fair, good, and excellent. Quality scores of 3.8 or higher generally imply high levels of intelligibility, speaker recognition, and naturalness. MOS is a global method used to evaluate the user's acceptance of a transmission channel or speech output system. It reflects the auditory impression of speech by a listener. The listener is asked to rate his impression of subjective scales such as: intelligibility, quality, acceptability, naturalness, etc. The MOS gives a wide variation among listener scores and does not give an absolute measure since the scales used by the listeners are not calibrated. Using this method, a score from 4 to 5 is considered toll quality; 3 to 4 communication quality; and less than 3, synthetic quality. This method is subjective.

The 860 DSPi VoIP RTP test uses an objective model that predicts human speech quality. This test transmits data through the network, comparing the timing of the received and transmitted data to assess distortions. The 860 DSPi VoIP RTP test calculates the R-factor, which is a measure of voice quality ranging from a best case of 100 to a worst case of 0. The R-factor uniquely determines the familiar mean-opinion score (MOS), which is the arithmetic average of opinion when *"excellent"* quality is given a score of 5; *"good"* a 4; *"fair"* a 3; *"poor"* a 2; and *"'bad"* a 1. The parameters for the computation of the R-factor are codec impairments, delay (latency), delay variation (jitter), and packet loss. This provides a means to estimate the subjective mean-opinion score (MOS) rating of voice quality over these planned network environments.

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