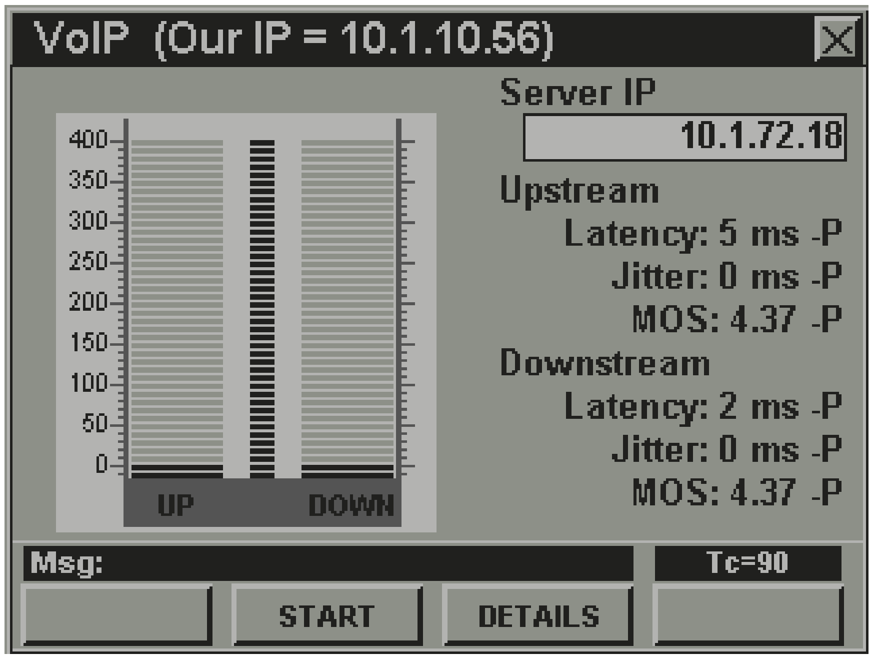
TRILITHIC **Application Note**

# 860DSPi Testing VoIP

Cable operators test their networks to ensure signal quality and signal leakage to comply with FCC guidelines. But with VoIP, HSD, and Digital Video new parameters have to be monitored and analyzed or tested to ensure that customers are getting the best service.

## VoIP SERVICES

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




**Latency:** Performing all the functions that are required to process and packetize voice signals and then transport them from the original point to the receive point in any IP architecture, including PacketCable, takes time. Each particular function requires tiny fractions of seconds, but the total amount of time varies based on the architecture of the device as well as the amount of traffic that has 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 a packet takes. Monitoring the total latency a packet is experiencing is necessary to maintain a high-quality signal transmission.

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

**Fixed delays include the following:**

- **Propagation delay:** The time it takes 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).

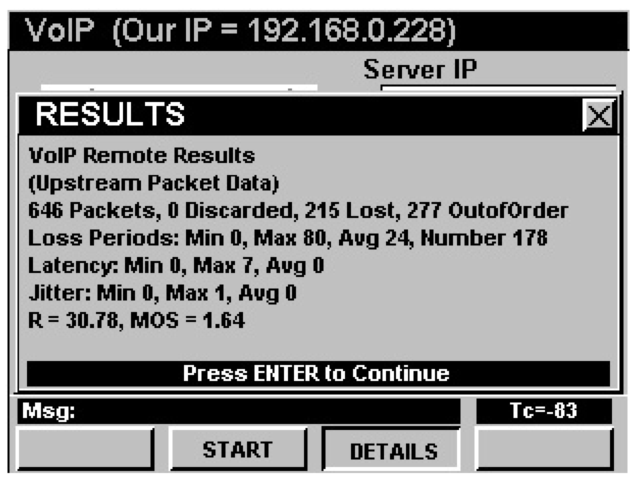
- **Serialization delay:** The time it takes to place the bits from the transmission buffer into the transmission media. The higher the speed, the less serialization delay.
- **Processing delay:** Includes the time it takes to code, compress, decompress and decode the voice signal, and the time it takes to collect enough voice samples to be placed on the payload for a data packet. This varies, depending on the algorithm used.

**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 it has 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. But 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's being properly handled.




**Dropped packets:**

When traffic levels rise to a level that overloads a router the device may intentionally drop packets to relieve the congestion. Error-checking has been built into the protocols and is used to maintain data integrity for HSD and Digital Video. But this procedure requires additional overhead, and isn't optimized for voice signals (VoIP uses UDP). A certain number of dropped packets (less than 1 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.

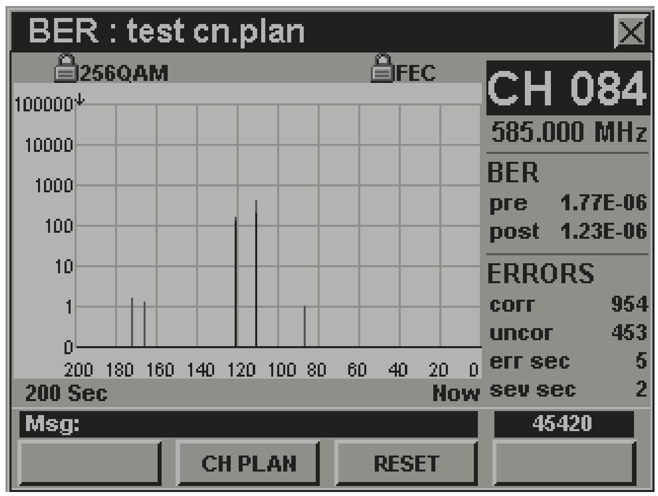
***(UDP) User Datagram Protocol** - UDP does not provide the reliability and ordering guarantees that TCP does. Datagram's may arrive out of order or go missing without notice. Without the overhead of checking if every packet actually arrived, UDP is faster and more efficient for time-sensitive purposes,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

## 860DSPi MER AND BER:

When testing QAM signals use the 860DSPi to test both Modulation Error Ratio (MER) and Bit Error Rate (BER). That's because MER and BER measurements detect different types of impairments.

MER is the measurement in dB of the RMS error magnitude over the average symbol magnitude. The higher the error magnitude, the poorer the MER. MER essentially assigns a value to the "fuzziness" of the symbol cluster). The larger or fuzzier the cluster becomes, the poorer the MER. Consequently, the farther the dots move from their ideal locations, the poorer the MER. Each symbol or "dot" on the constellation is framed by decision boundaries. When the carrier falls inside the boundaries, the information is transmitted without errors. MER determines how much margin the system has before failure. Low MER is not noticeable on the subscriber's picture until system failure. MER is a good figure of merit for the QAM signal.

Why measure BER, because MER is a poor indicator of fast, intermittent transients.




Impairments include:

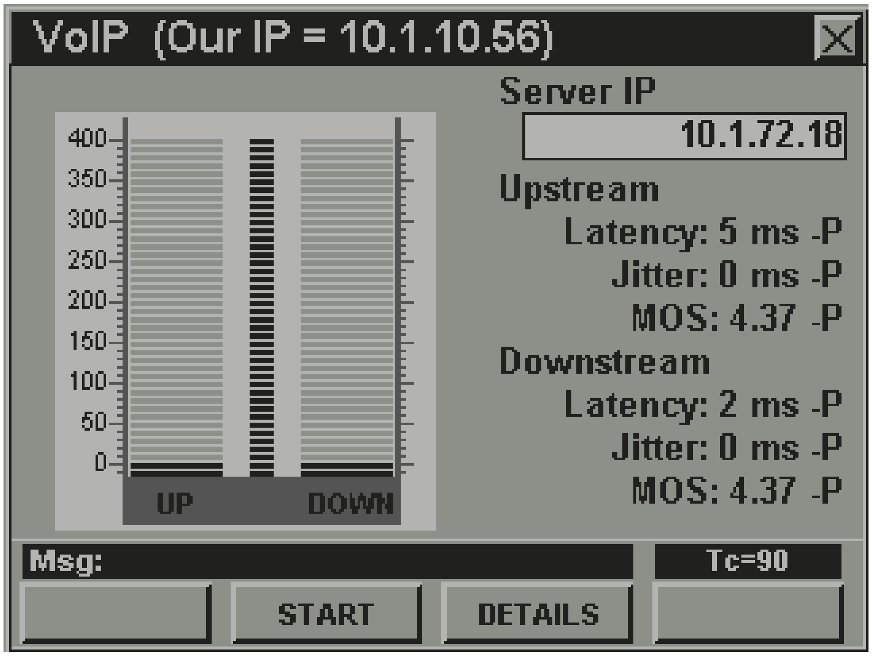
- Laser clipping
  - the most common cause
- Sweep system interference
- Corroded or loose connections
- Microphonics

Therefore, if you have high MER. but errors are present, they are probably caused by intermittent interference. This shows up on a constellation diagram as a lone dot that is away from the main cluster. The best way to test this with the 860DSP is the BER mode. Gray lines are Pre-BER errors and Black lines are Post-BER errors. It is much easier to see BER errors in this mode than one lone dot on the constellation and the time duration can be set from 200 seconds to 600 seconds.

For Additional Help Contact
Trilithic Applications Engineering
1-800-344-2412 or 317-895-3600
support@trilithic.com or www.trilithic.com

## 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 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. But this method is subjective.

The 860DSP VoIP RTP test is an objective model that predicts human speech quality. This test transmits a file through the network, comparing the received and transmitted files to assess distortions. A basic result of the 860DSP VoIP RTP test is the calculation of the R-factor, which is a simple 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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