

860DSPi VoIP RTP Find & Fix

Typical Measurements

Network Timers

Post-Dial Delay and Dial Tone Delay typical DTD on a regular PSTN line is usually around 200 ms and for most PacketCable environments. Speech Power, Echo & Noise: Significant number of calls with Echo detected (about 40%). Typically any EPL (Echo Path Loss) value in the 25dB range is good. This means that the perceived echo is 25dB (or more) quieter than the initial voice utterance, and not very perceivable. EPD (Echo Path Delay) is usually close to the RTD (Round Trip Delay) measurements, therefore perceived EPD will be about the same delay as RTD.

Voice Quality

Based on MSO data and other VoIP deployments, the MOS thresholds are now set to 3.8 as a cut-off (for G.711 codec implementations). This revised threshold will give a lot more passing than failing. These Voice Quality scores are mostly affected by occasional In-Between Clipping events, as well as Frame Muting and Speech Distortion. The Clipping events are generally caused by poor performing VAD (Voice Activity Detection), where the speech is being cut or clipped abruptly, instead of transitioning smoothly between periods of silence and periods of speech activity. When Clipping Ratio exceeds 1.0%, there will be some degradation of the voice quality. Clipping, Frame Muting, and Speech Distortion are the analog approximation of packet loss. Generally this number should be less than 1.0%, and when it is not, the MOS scores are significantly affected. Packet Loss is a common problem in VoIP. Speech Distortion also seems to be an issue, usually caused by IP impairments (e.g. Packet loss or Jitter). There also appears to be consistent DTMF dropping, typically an MTA provisioning issue.

MOS (Mean Opinion Score) – 860 DSPi VoIP RTP Algorithm

- Statistics-based algorithm used to derive opinion scores from QoS measurements on the perceived voice quality of a call.
- Enables the measurement of end-user satisfaction based on *impairments that affect perceived voice quality*: codec impairments, delay (latency), delay variation (jitter), and packet loss
- Transforms this information into an approximation of the MOS (1-5)

VoIP Problems and How to Fix With the 860DSPi

Connection Status Failure

- Test could not connect
 - Check home wiring

Network Timer Failure

- Indications of excessive latency in the Ethernet path
 - Check router configuration
- Indications of excessive latency in the RF path
 - This indicates a plant issue and rely on the techs field knowledge of the 860 DSPi to find and fix the impairment
 - Perform the VoIP RTP test
 - MOS, Latency, & Jitter
 - Perform the Ping test
 - Latency
 - Packet Loss

Intermittent DTMF Failure

- A different DTMF digit fails on separate consecutive calls
- This is attributed to packet loss in the network
- Verify on the 860 DSPi
 - Perform the VoIP RTP test
 - MOS, Latency, & Jitter
 - MER, BER, & Signal Levels
 - SSR Mode
 - Ingress and Noise
 - Perform the Ping test
 - Latency
 - Packet Loss

Consistent DTMF Failure

- Consistent failure on the customers CPE (Phone) but tests pass on 860 DSPi
 - CPE is faulty
- Consistent failure with CPE but CPE is known to pass a test from another location
 - This indicates a faulty eMTA not interpreting DTMF correctly

Low MOS Scores - Indications of Jitter, Excessive Latency in the RF Path

- This typically requires further testing to isolate where in the network if any that calls pass or fail
- Typically at this point if the call passes in the hub directly from the CMTS
 - Then this indicates a plant issue and rely on the techs field knowledge of the 860 DSPi to find and fix the impairment
 - Verify MER, BER, Signal Levels, Ingress, and Noise
 - SSR mode
 - Test for ingress and or noise
 - Perform the VoIP RTP test
 - MOS, Latency, & Jitter
 - Perform the Ping test
 - Latency & Jitter

Systems That Worked Fine at QPSK and are Now 16QAM and Having Issues

- At slower transmit speeds and lower modulation methods, packet loss is lower and the system can handle worse signal quality
- Use the 860DSPi to find and fix
 - Perform the VoIP RTP test
 - MOS, Latency, & Jitter
 - SSR mode
 - Ingress and Noise
 - Perform the Ping test
 - Latency, Packet Loss, & Jitter

Speech Quality Failure

- Indications of Jitter, excessive latency in the RF path
- This typically requires further testing to isolate where in the network if any that calls pass or fail
- Typically at this point if the call passes in the hub directly from the CMTS
 - Then this indicates a plant issue and rely on the techs field knowledge of the 860 DSPi to find and fix the impairment
 - SSR mode
 - Test for ingress and or noise
 - Perform the VoIP RTP test
 - MOS, Latency, & Jitter
 - Perform the Ping test
 - Latency & Jitter

VoIP Transmission Failure

- Clipping or lost packets
- Typically at this point if the call passes in the hub directly from the CMTS
 - Then this indicates a plant issue and rely on the techs field knowledge of the 860 DSPi to find and fix the impairment
 - SSR mode
 - Test for ingress and or noise
 - Perform the VoIP RTP test
 - MOS, Latency, & Jitter
 - Perform the Ping test
 - Latency, Packet Loss & Jitter

Delay Failure

- Indications of Jitter, excessive latency in the RF path
- This typically requires further testing to isolate where in the network if any that calls pass or fail
- Typically at this point if the call passes in the hub directly from the CMTS
 - Then this indicates a plant issue and rely on the techs field knowledge of the 860 DSPi to find and fix the impairment
 - SSR mode
 - Test for ingress and or noise
 - Perform the VoIP RTP test
 - MOS, Latency, & Jitter
 - Perform the Ping test
 - Latency & Jitter

Echo Failure

- This could be a Echo delay or a Echo Path Loss problem
 - First determine if the customer is having an issue with “Listener Echo” or “Talker Echo”. If the echo is occurring on the opposite end of the phone line then the issue is “Listener Echo” and is an issue with the customer. If the issue is occurring on the customer’s end of the phone call then it is a “Talker Echo” issue and is with the person on the other end.
 - A quick test for echo would be to press a DTMF tone on the customer’s telephone and check to see if this eliminates the echo. A DTMF tone will disrupt the RTP stream and cause it to realign.
 - Be sure the ECAN (Echo Canceller) is activated in your customers MTA.
 - A customer’s wireless phone/headset can introduce a certain level of echo. Try asking your customer to turn on the speakerphone, if there isn’t echo then the wireless phone is at fault.

- If the problem is Echo delay and the call passes in the hub at the CMTS
 - Then this indicates a plant issue and rely on the techs field knowledge of the 860 DSPi to find and fix the impairment
 - SSR mode
 - Test for ingress and or noise
 - Perform the VoIP RTP test
 - MOS, Latency, & Jitter
 - Perform the Ping test
 - Latency & Jitter

Most of the above problems will drill down to latency and jitter or packet loss. The VoIP test in the 860 DSPi displays these tests and will give a MOS for both the forward and reverse. This will be your most valuable tool. This will show where these problems occur forward or reverse. The way to fix these problems is the same as in the past divide and conquer. This means from the point at which the problem occurs work back to the headend until there is no problem. Then the problem is between the last point and this point. This will help to isolate the affected area. Also, Leakage, Return Monitoring, and Sweeping the system both forward and return are vital to provide quality VoIP service.

Notes

The High QoS for VoIP is granted to EMTA's by the CMS (Call Management Server). An EMTA logging onto the system sets up access to the CMS with a dialog with the CMTS using the normal low-level QoS. Once into the CMTS, the EMTA contacts the CMS. The CMS recognizes the caller as an EMTA, consults its instructions for call QoS and sets up the equipment in the call path, including the terminal devices' QoS.

The Packet Cable standard provides an alternate way of establishing High QoS for a selected terminal device. The CMTS itself can be configured to grant a specified QoS to the device when it logs onto the CMTS without reference to a CMS.

- The same Cable Modem MAC cannot be configured on both the HSD CMTS and the VoIP CMTS
 - 860 DSPi Dual MAC option

If the customer complains of any of these problems:

- They cannot dial 800 numbers or that they cannot call certain phone numbers or they cannot be called via certain phone numbers the problem is with their old phone company.
 - The previous phone company must release that phone number.