

## 860 DSPi VoIP RTP Find and Fix

---

### Typical Measurements

#### Network Timers

Post-dial delay (PDD) and dial tone delay (DTD) on a regular PSTN line is usually around 200 ms for most packet cable environments. Within the speech power, echo, and noise timers, a significant number of calls with echo are detected (about 40%). Typically, an echo path loss (EPL) value in the 25 dB range is considered to be good. This means that the perceived echo is 25 dB (or more) quieter than the initial voice utterance, and not very perceivable. Echo path delay (EPD) is usually close to the round trip delay (RTD) measurements, therefore perceived EPD will be about the same 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 permit more passing than failing measurements. 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 voice activity detection (VAD), where the speech is being cut or clipped abruptly, instead of transitioning smoothly between periods of silence and periods of speech activity. When the 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. This number should generally 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 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).

For Additional Help Contact  
Trilithic Applications Engineering  
1-800-344-2412 or 317-895-3600  
[support@trilithic.com](mailto:support@trilithic.com) or  
[www.trilithic.com](http://www.trilithic.com)

860 DSPi VoIP RTP Find and Fix  
P/N 0010275059 – Rev 01/10  
1 of 6

## **VoIP Problems and How to Fix With the 860 DSPi**

### **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, and 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, and jitter
  - MER, BER, and 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

For Additional Help Contact  
Trilithic Applications Engineering  
1-800-344-2412 or 317-895-3600  
[support@trilithic.com](mailto:support@trilithic.com) or  
[www.trilithic.com](http://www.trilithic.com)

860 DSPi VoIP RTP Find and Fix  
P/N 0010275059 – Rev 01/10  
2 of 6

## **Low MOS Scores – Indication 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, and jitter
    - Perform the ping test
      - Latency and jitter

## **Systems Which Worked Correctly at QPSK and 16 QAM are Now Having Issues**

- At slower transmit speeds and lower modulation methods, packet loss is lower and the system can handle worse signal quality
- Use the 860 DSPi to find and fix
  - Perform the VoIP RTP test
    - MOS, latency, and jitter
  - SSR mode
    - Ingress and noise
  - Perform the ping test
    - Latency, packet loss, and 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
  - 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, and jitter
    - Perform the ping test
    - Latency and jitter

For Additional Help Contact  
Trilithic Applications Engineering  
1-800-344-2412 or 317-895-3600  
[support@trilithic.com](mailto:support@trilithic.com) or  
[www.trilithic.com](http://www.trilithic.com)

860 DSPi VoIP RTP Find and Fix  
P/N 0010275059 – Rev 01/10  
3 of 6

**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, and jitter
    - Perform the ping test
      - Latency, packet loss and 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, and jitter
    - Perform the ping test
      - Latency and jitter

For Additional Help Contact  
Trilithic Applications Engineering  
1-800-344-2412 or 317-895-3600  
[support@trilithic.com](mailto:support@trilithic.com) or  
[www.trilithic.com](http://www.trilithic.com)

860 DSPi VoIP RTP Find and Fix  
P/N 0010275059 – Rev 01/10  
4 of 6

## Echo Failure

- This could be an echo delay or an 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” is the issue. If the issue is occurring on the customer’s end of the phone call, “talker echo” is the 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 is not echo then the wireless phone is at fault.
- If the problem is echo delay and the call passes in the hub at the CMTS:
  - A plant issue exists and the technicians’ field knowledge – with the help of an 860 DSPi – is necessary to find and fix the impairment.
    - SSR mode
      - Test for ingress and or noise
    - Perform the VoIP RTP test
      - MOS, latency, and jitter
    - Perform the ping test
      - Latency and 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 paths. This will be your most valuable tool. This will show where these problems occur in the forward or reverse paths. 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

For Additional Help Contact  
Trilithic Applications Engineering  
1-800-344-2412 or 317-895-3600  
[support@trilithic.com](mailto:support@trilithic.com) or  
[www.trilithic.com](http://www.trilithic.com)

860 DSPi VoIP RTP Find and Fix  
P/N 0010275059 – Rev 01/10  
5 of 6

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 previous phone company.
- The previous phone company must release that phone number.

For Additional Help Contact  
Trilithic Applications Engineering  
1-800-344-2412 or 317-895-3600  
[support@trilithic.com](mailto:support@trilithic.com) or  
[www.trilithic.com](http://www.trilithic.com)

860 DSPi VoIP RTP Find and Fix  
P/N 0010275059 – Rev 01/10  
6 of 6