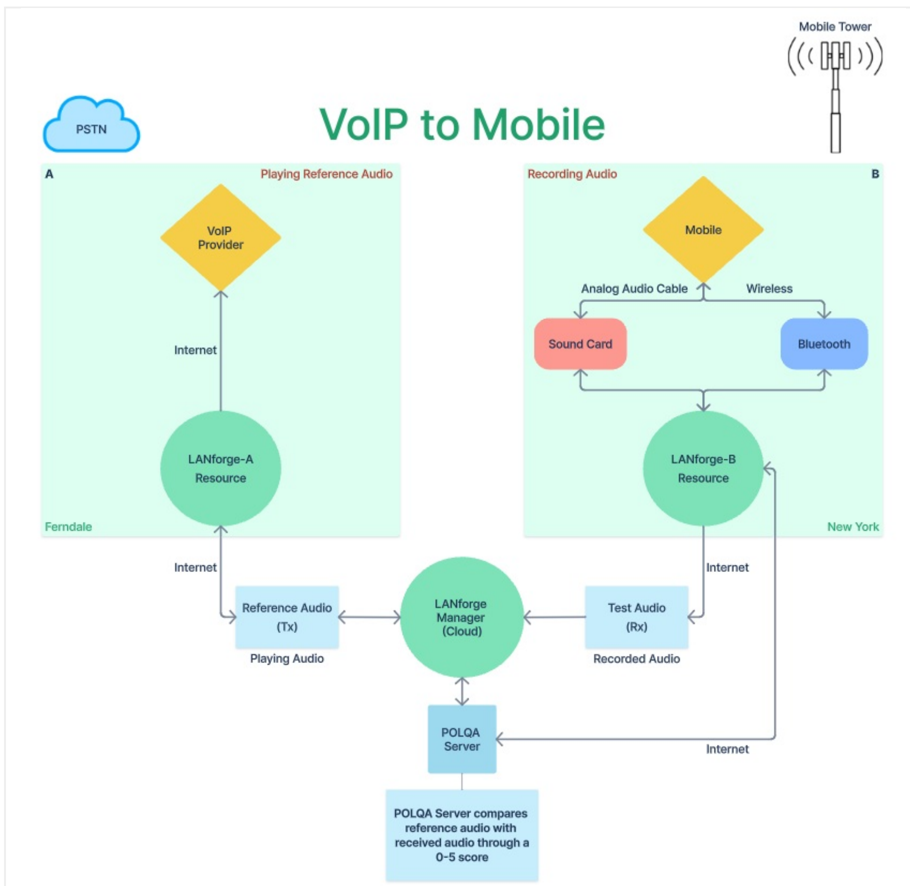
Network Testing and Emulation Solutions

sales@candelatech.com
support@candelatech.com
+1 (360) 380-1618 [PST, GMT -8]

Audio Quality Testing: VoIP/SIP and mobile calls using POLQA (Advanced Setup)

Goal: Evaluate the voice/speech audio quality made between VoIP-SIP and mobile calls through POLQA scoring server where both the endpoints are located at different locations.

Consider an example:
LANforge-A (LF resource system, Ferndale location) makes 20 multiple single calls using VoIP-SIP towards mobile phone device which is connected to LANforge-B (LF resource system, New York location). Both of the LANforge resources are connected together to the LANforge manager cloud instance. LANforge-A plays a reference audio file over the VoIP-SIP call. The call is being recorded by LANforge-B from the mobile device using Bluetooth or audio cable. After the call completes, both the reference audio file and recorded audio file are evaluated by LANforge manager (Cloud) using the POLQA server which is installed on LANforge-B. The POLQA server scores the recording based on audio quality loss during the call. Multiple location resources can be further clustered from LANforge manager for mesh testing (optional).



1. Requirements:
   A. LANforge systems (version 5.4.8). One cloud manager and minimum two resources.
   B. LANforge licenses.
   C. POLQA server with required licenses
   D. POLQA standard reference audio files.

E. Bluetooth USB dongle.

F. Analog sound card and audio cables. (If testing over analog audio cable)

G. VoIP service provider. (Customer provided)

H. Mobile device (Android or IOS) having Bluetooth and active SIM/eSIM card. (Customer provided)

I. Mobile network like VoLTE, VoNR, etc. (Customer provided)

J. Internet access. (Customer provided)

2. Configurations:

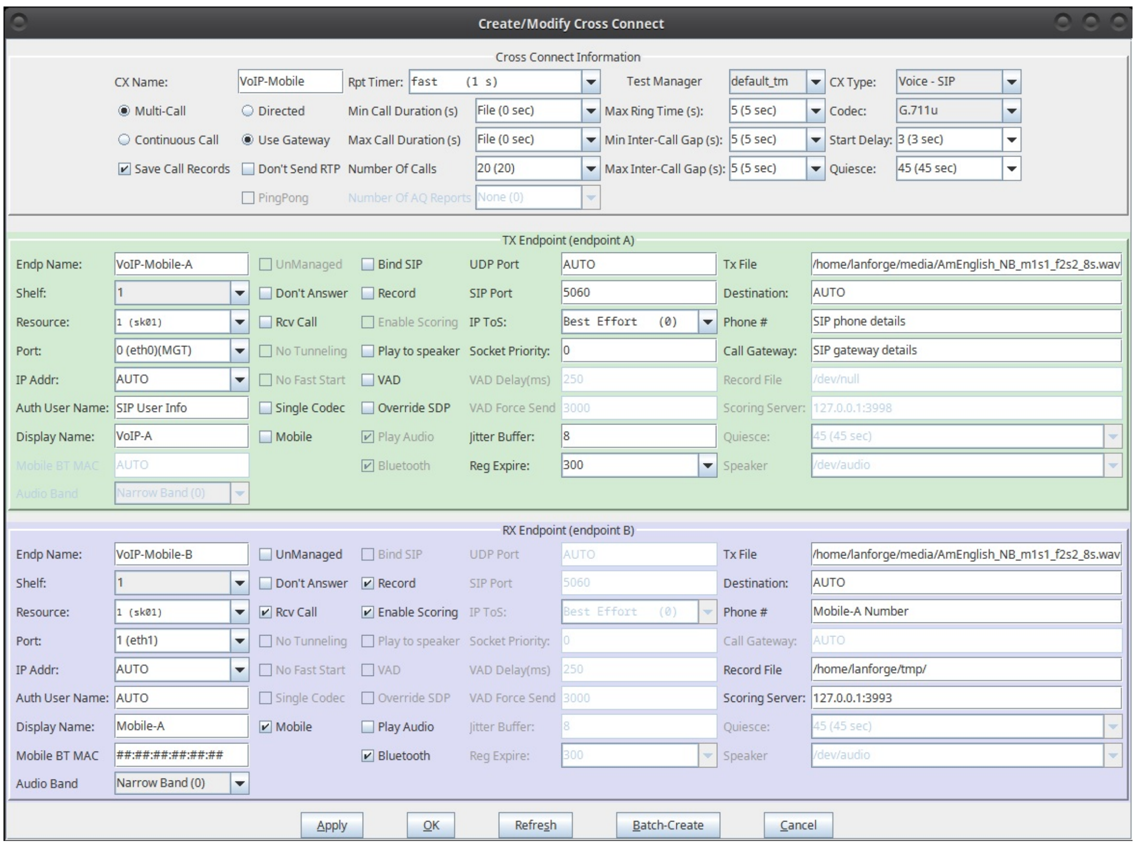
   A. Clustering between one LANforge manager (Cloud) towards two or multiple LANforge resources should be done till here.

   B. LANforge and POLQA licenses are installed.

   C. AQ configuration: Follow `/home/lanforge/audio-bluetooth/README.txt` on all LANforge resources.

   D. Then reboot all the systems.

   E. On the LANforge manager (cloud), open the **GUI**.
   Under **VoIP/RTP** tab, select **Create**.



A. Cross Connect details to be filled are:

   I. **Cross Connect Information:**

      i. **CX name:** VoIP-Mobile

      ii. Select **Multi-Call** checkbox.

      iii. Select **Save Call Records** checkbox to save recordings for further analysis.

      iv. Select **Use Gateway** checkbox.

      v. **Min/Max Call Duration:** File

      vi. **Number Of Calls:** 20

      vii. **Min/Max Inter Call Gap:** 5 sec

      viii. Rest can remain defaults

   II. **TX Endpoint A:** Fill the TX Endpoint A with VoIP-SIP details.

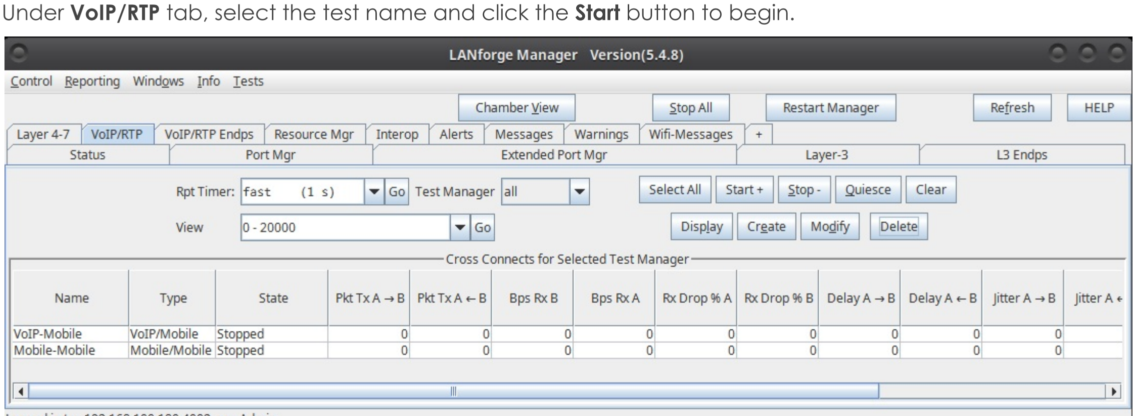
      i. **Resource:** LANforge-A resource Hostname
      (Ferndale location system in this example)

        ii. **Port:** Management Port with Internet access.
        iii. **Auth User Name:** VoIP-SIP User info
        iv. **Display Name:** VoIP-SIP Name
        v. Deselect **Rcv Call** checkbox.
        (VoIP-SIP is going to make calls and not receive in this case.)
        vi. Deselect **Mobile** checkbox.
        (VoIP-SIP does not need Mobile checkbox.)
        vii. **Tx file:** /home/lanforge/media/AmEnglish_NB_m1s1_f2s2_8s.wav
        viii. **Destination:** AUTO
        ix. **Phone:** VoIP-SIP phone number
        x. **Call Gateway:** VoIP-SIP Call Gateway info

    III. **RX Endpoint B:** Fill the RX Endpoint B with mobile details.
        i. **Resource:** LANforge-B resource Hostname
        (New York location system in this example)
        ii. **Port:** Management Port with Internet access.
        iii. **Auth User Name:** AUTO
        iv. **Display Name:** Mobile Name
        v. **Mobile BT MAC:** Mobile bluetooth mac address
        vi. Select **Rcv Call** checkbox.
        vii. Select **Mobile** checkbox.
        viii. Select **Record** checkbox.
        ix. Select **Enable Scoring** checkbox for POLQA.
        x. **Audio Band:** Narrow Band
        xi. Select **Bluetooth** checkbox.
        (Deselect this option for analog sound card option.)
        xii. **Tx file:** /home/lanforge/media/AmEnglish_NB_m1s1_f2s2_8s.wav
        xiii. **Destination:** AUTO
        xiv. **Phone:** Mobile number
        xv. **Record File:** Recording folder path
        xvi. **Scoring Server:** POLQA Server Address

  B. Select **Apply**, **OK**

3. Options to start the test:
  A. Under **VoIP/RTP** tab, select the test name and click the **Start** button to begin.



  B. Using **Command Terminal** and get the test results in **.csv** format.
    A. Open a command terminal as a user
    B. `cd /home/lanforge/Documents`
    C. `git clone https://github.com/greearb/lanforge-scripts`

D. `cd lanforge-scripts/py-scripts/`

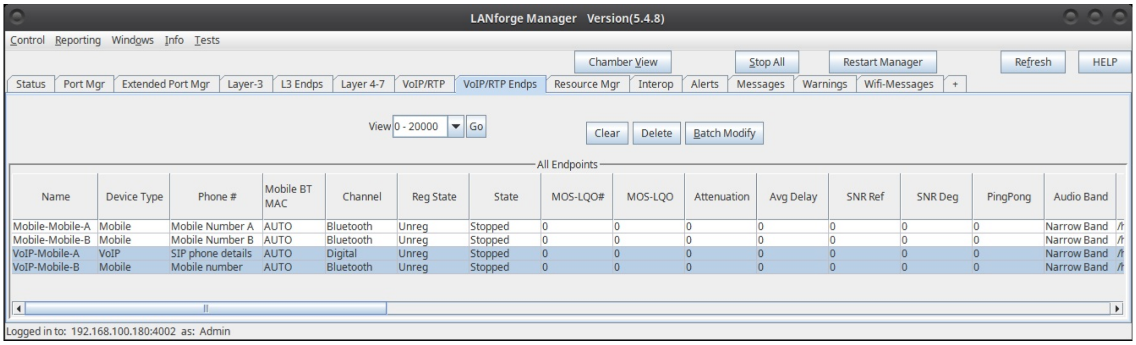
E. `git pull`

F. `./run_voip_cx.py --host localhost --cx_list VoIP-Mobile --csv_file /home/lanforge/report-data/my_test_reports.csv`

G. This command can be integrated for further automation.

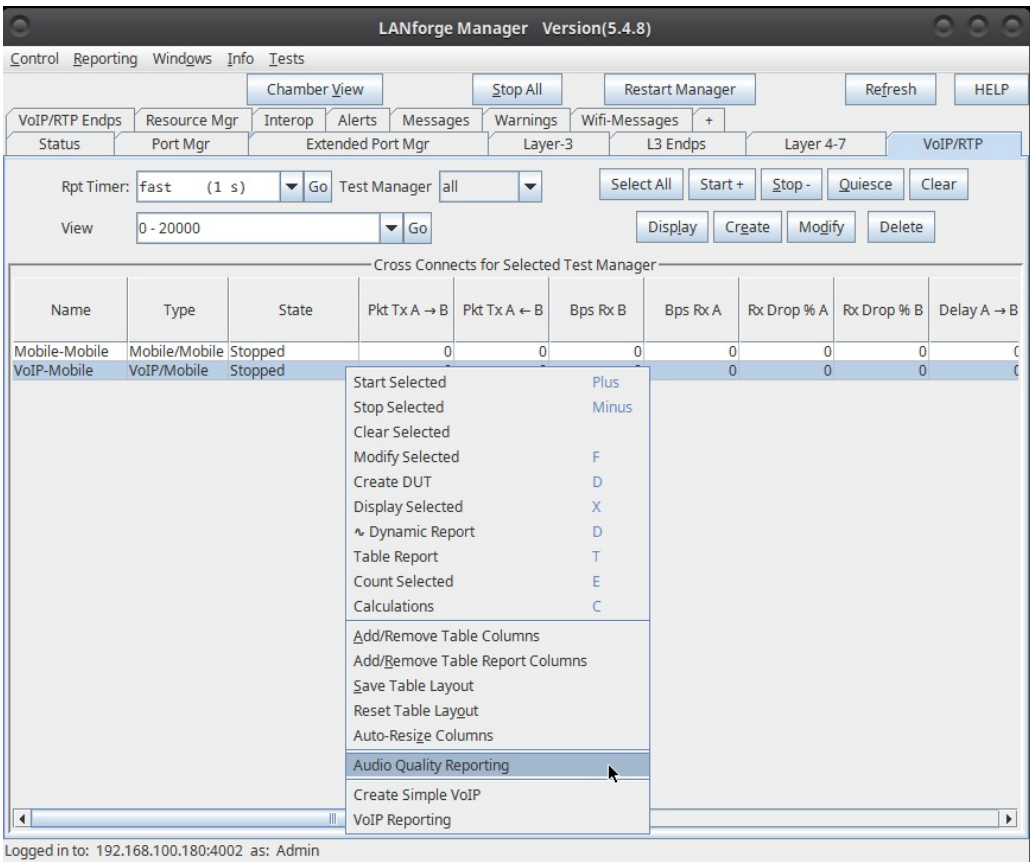
4. AQ Test Results:

   A. Option 01: Under **VoIP/RTP Endp** tab, current results will be shown in column/row structure once started.

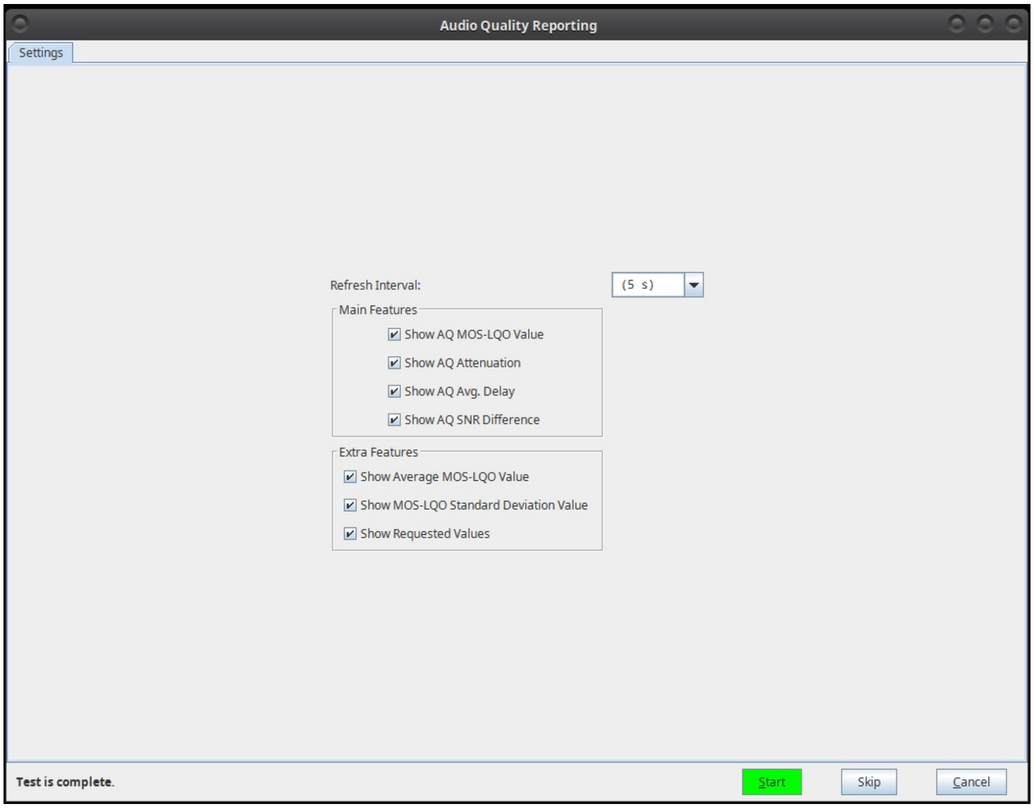


   B. Option 02: Using live graphical reporting.

      A. Under **VoIP/RTP** tab, right click on the selected AQ test name, and select **Audio Quality Reporting**.



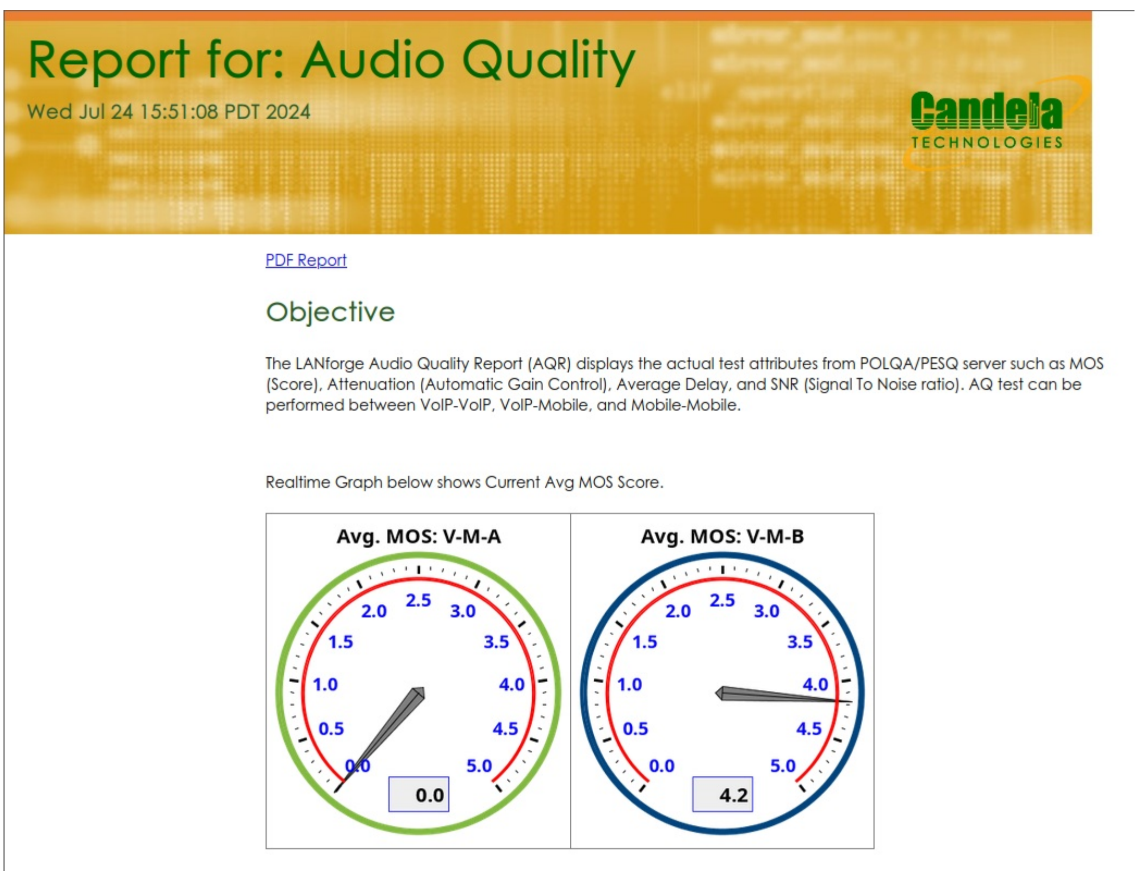
      B. Select the required configuration and **Start** the monitoring.



C. Once started, we see Live view of graphical test monitoring which shows detailed reporting.

D. Use **Save HTML** or **Save PDF** to get detailed report including **.csv** data when test is finished.

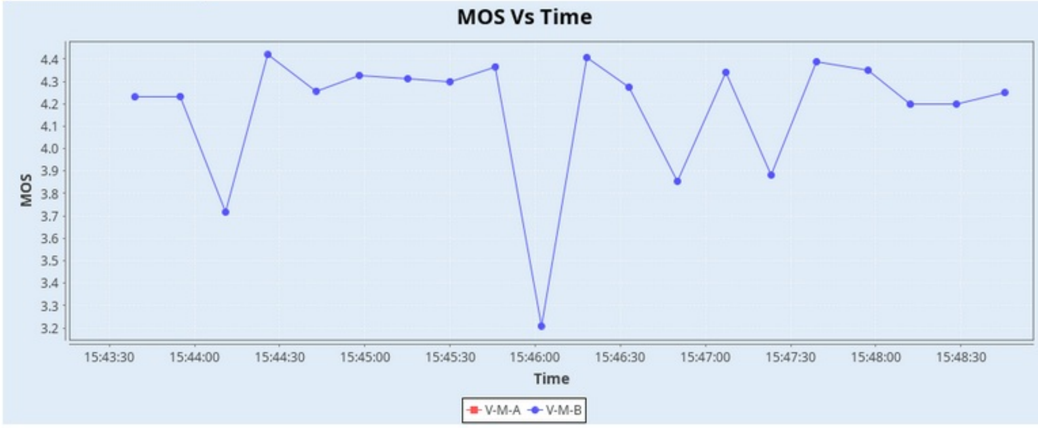
5. Sample screenshots of Live AQ Reporting.
   A. Screenshot 01



B. Screenshot 02

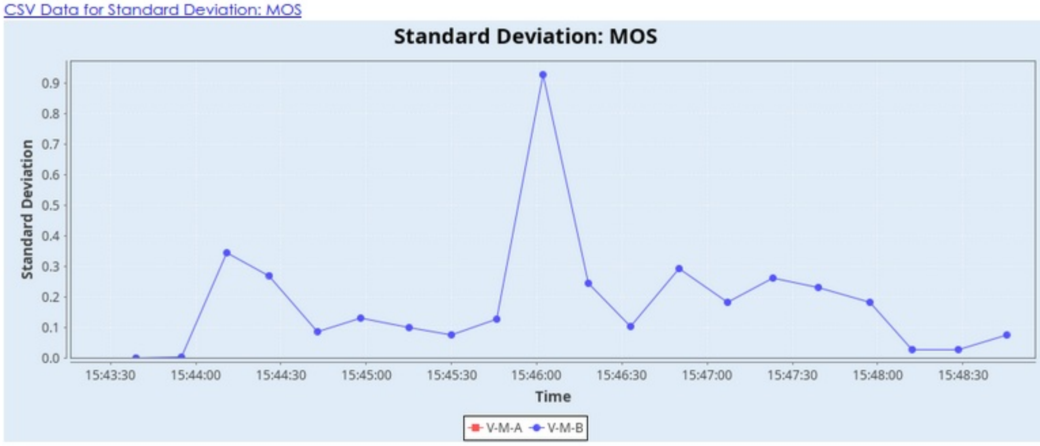
Realtime Graph below shows MOS-LQO score from recording endpoints.

CSV Data for MOS Vs Time



Realtime Graph below shows MOS Standard Deviation.

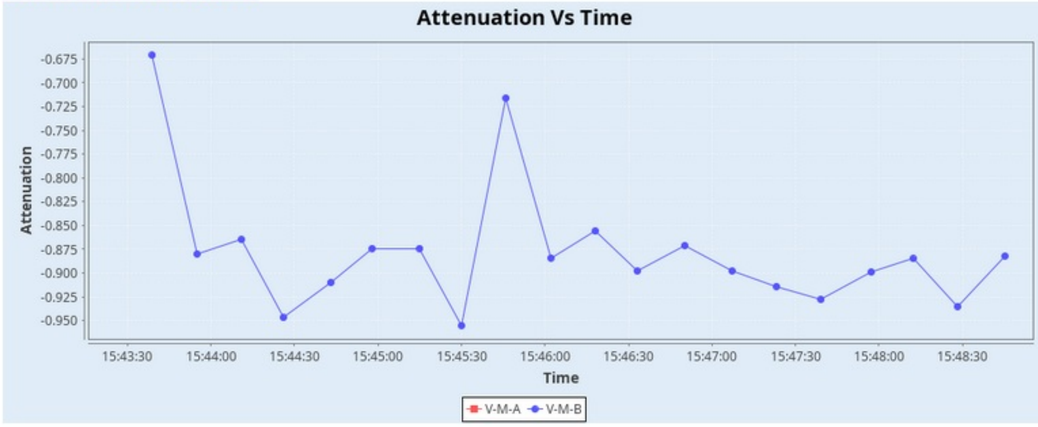
CSV Data for Standard Deviation: MOS



C. Screenshot 03

Realtime Graph below shows AQ Attenuation (AGC) from recording endpoints. Unit: dB

CSV Data for Attenuation Vs Time



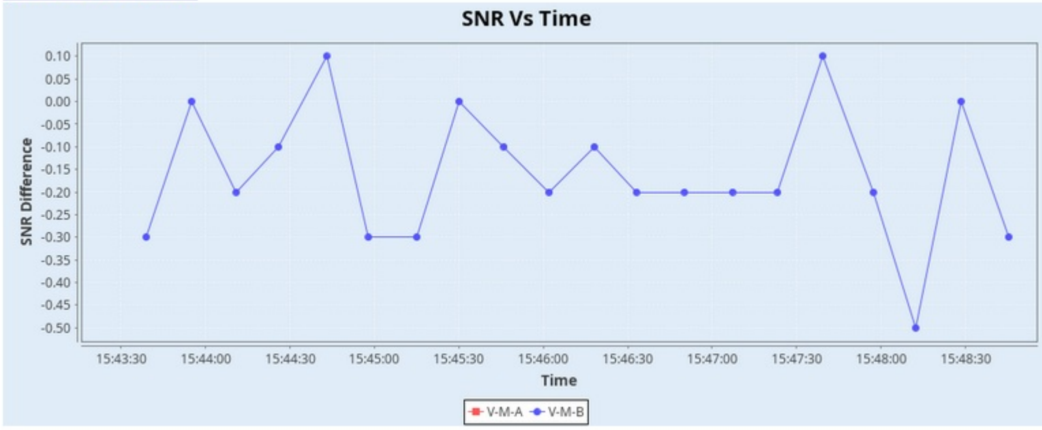
Realtime Graph below shows AQ Avg Delay from recording endpoints. Unit: ms

CSV Data for Avg. Delay Vs Time

D. Screenshot 04

Realtime Graph below shows difference between SNR Reference and SNR Degraded from recording endpoints. Unit: dB

CSV Data for SNR Vs Time



Requested Values:

Endpoint Name	V-M-A	V-M-B
Resource	1 (sk01)	1 (sk01)
Port	eth0	eth1
Device Type	VoIP	Mobile

6. Further analysis: If **Save Call Records** option is true, received audio file along with the reference audio file can be evaluated manually on POLQA server to get more advanced report. Sample Advanced Report

7. If you need assistance, you can contact us at support@candelatech.com