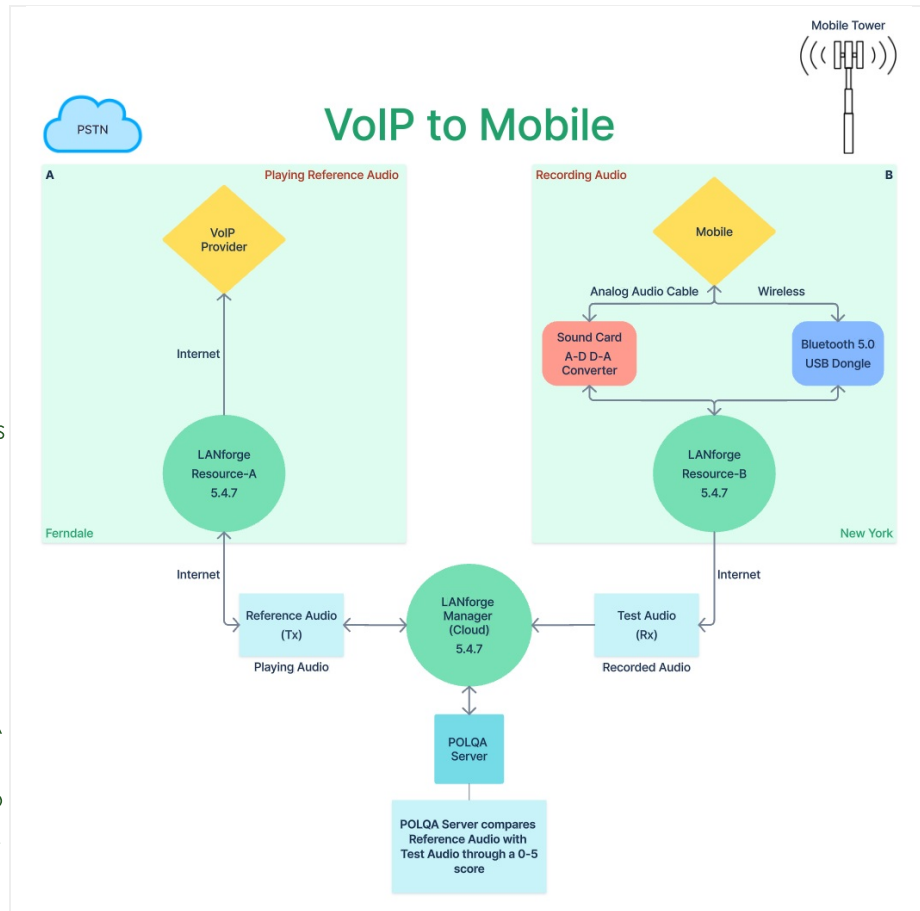


VoIP-Mobile Audio Quality Testing using POLQA/PESQ

Goal: Evaluate the voice call quality made between VoIP-SIP and Mobile call through POLQA scoring server. PESQ can also be used in place of POLQA here.

Consider an example:
LANforge-A (resource) makes a VoIP-SIP phone call towards a Mobile device which is connected to LANforge-B (resource) via LANforge manager (cloud). LANforge-A plays a reference audio file over the VoIP-SIP call. The call is being recorded by LANforge-B. After the call completes, both the reference audio file and recorded audio file are evaluated by LANforge manager using the POLQA server. The POLQA server scores the recording based on audio quality loss during the call.



1. Requirements:

- LANforge systems (version 5.4.7). One manager and atleast one resource.
- LANforge licenses
- VoIP service provider
- POLQA license and server
- POLQA standard reference audio files
- Mobile Phone having Bluetooth connection and active SIM card.
- Bluetooth USB dongle
- Analog audio cable
- USB sound card
- Internet access
- Mobile network

2. Configuration:

- Connection between one or more LANforge resource systems towards one LANforge manager (Cloud)

should be done till here.

- B. LANforge and POLQA licenses are installed.
- C. Installation steps: Follow /home/lanforge/audio-bluetooth/README.txt on all LANforge resources.
- D. After installation, please **reboot** all the LANforge systems.
- E. On the LANforge manager (cloud), open the **GUI**.
In the VoIP/RTP tab, select **Create**.

The screenshot shows the 'Create/Modify Cross Connect' window. It is divided into three main sections: 'Cross Connect Information', 'TX Endpoint (endpoint A)', and 'RX Endpoint (endpoint B)'.
- **Cross Connect Information:** CX Name: VoIP-Test, Rpt Timer: Fast (1 s), Test Manager: default_tm, CX Type: Voice - SIP. Call type options include Multi-Call, Directed, Continuous Call, and Use Gateway. Call duration and inter-call gap settings are set to File. Max Ring Time is 20s, Start Delay is 15s, and Quiesce is 45 (45 sec).
- **TX Endpoint (endpoint A):** Endp Name: VoIP-Test-A, Shelf: 1, Resource: 1 (sk01), Port: 0 (eth0)(MGT), IP Addr: AUTO, Auth User Name: SIP user info, Display Name: AUTO. It includes checkboxes for UnManaged, Bind SIP, Don't Answer, Record, Rcv Call, Enable Scoring, No Tunneling, Play to speaker, No Fast Start, VAD, Single Codec, and Override SDP. UDP Port is AUTO, SIP Port is 5060, IP ToS is Best Effort (0), Socket Priority is 0, VAD Delay is 250, VAD Force Send is 3000, Jitter Buffer is 8, and Reg Expire is 300. Tx File is /home/lanforge/media/AmEnglish_NB_m1s1_f2s2_8s.wav, Destination is AUTO, Phone # is SIP Phone Number, Call Gateway is SIP Call Gateway Details, Record File is AUTO, and Scoring Server is 127.0.0.1:3998.
- **RX Endpoint (endpoint B):** Endp Name: VoIP-Test-B, Shelf: 1, Resource: 3 (sk03), Port: 0 (eth0)(MGT), IP Addr: AUTO, Auth User Name: Mob user info, Display Name: AUTO. It includes checkboxes for UnManaged, Bind SIP, Don't Answer, Record, Rcv Call, Enable Scoring, No Tunneling, Play to speaker, No Fast Start, VAD, Single Codec, and Override SDP. UDP Port is AUTO, SIP Port is 5060, IP ToS is Best Effort (0), Socket Priority is 0, VAD Delay is 250, VAD Force Send is 3000, Jitter Buffer is 8, and Reg Expire is 300. Tx File is /home/lanforge/media/AmEnglish_NB_m1s1_f2s2_8s.wav, Destination is AUTO, Phone # is Mobile Number, Call Gateway is AUTO, Record File is /home/lanforge/tmp/, and Scoring Server is 127.0.0.1:3998. Buttons at the bottom include Apply, OK, Refresh, Batch-Create, and Cancel.

A. Cross Connect details to be filled are:

- I. **TX Endpoint A:** The VoIP-SIP endpoint performs an outbound call over the Internet towards the Mobile phone mentioned in RX Endpoint B. During this call, SIP plays an audio Tx File over the call.
 - i. **Phone:** SIP Phone info
 - ii. **Call Gateway:** SIP Call Gateway info
 - iii. **Auth User Name:** SIP User info
 - iv. **Resource:** LANforge resource-A (hostname sk01 from Ferndale location in this example)
 - v. **Port:** Management Port with Internet access (eth0 in this example)
 - vi. **Display Name:** SIP Name or mac_address
 - vii. **Tx file:** reference audio file to be played on call
- II. **RX Endpoint B:** The Mobile connected to LANforge resource-B receives an inbound call from VoIP-SIP over the Internet. LANforge resource-B starts recording the active incoming audio call via Bluetooth channel or analog audio cable option at the specified Record File location. The recorded file is then evaluated by POLQA server against the original Tx File.
 - i. **Phone:** Mob phone number
 - ii. **Call Gateway:** AUTO
 - iii. **Auth User Name:** AUTO
 - iv. **Resource:** LANforge resource-B (hostname sk03 from New York location in this example)
 - v. **Port:** Management Port with Internet access (eth0 in this example)
 - vi. **Display Name:** Mobile mac_address

vii. **Tx file:** same audio file as TX Endpoint-A Tx File

viii. Checkboxes:

i. **Mobile:** True

ii. **Rcv Call:** True (become a receiver)

iii. **Bluetooth:** True (Record call through Bluetooth channel optional)

Bluetooth: False (Record call through analog audio cable)

iv. **Record:** True

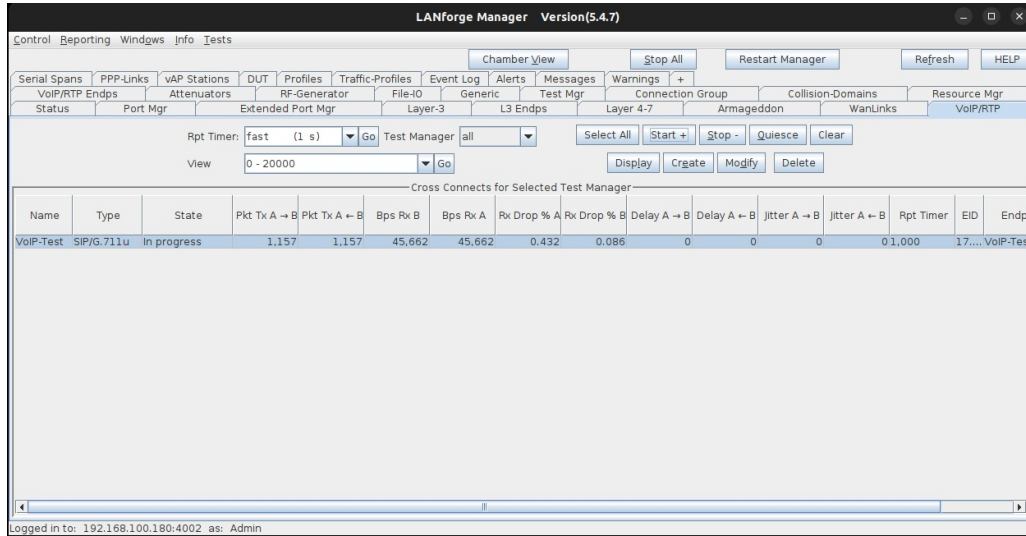
v. **Enable Scoring:** True (Enable POLQA scoring)

ix. **Record File:** Recording folder path

x. **Scoring Server:** POLQA Server Address

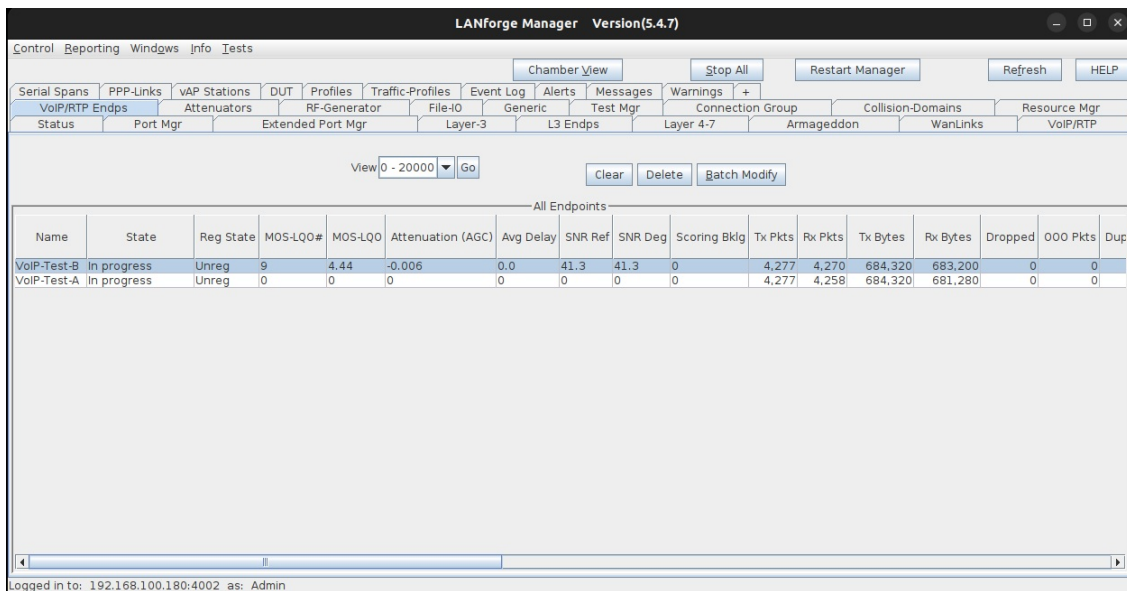
B. Select **Apply**, **OK**, and **START** the test.

C.



F. Go To **VoIP/RTP Endps** tab to get the **POLQA Scores**.

(MOS-LQO, Attenuation AGC, Avg. Delay, SNR Reference, SNR Degraded).



3. Sample **POLQA Score Report** from POLQA server.

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

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