FREQUENTLY ASKED QUESTIONS

Telecom Range of Products

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How to Configure and Troubleshoot Peer to Peer Calling?

For Peer to Peer calling between two ETERNITY systems, you need:

- VoIP Server Cards/VoIP Modules in each system
- IP connectivity between two sites through VPN or through public static IP/ DynDNS

You can also conduct Peer to Peer calling between ETERNITY and a Standard SIP phone or NAVAN CNX200.

Configuration:

- For Peer to Peer configuration between two ETERNITY Systems (PE/GE/ME/LE), refer to the following link: http://www.matrixtelesol.com/faqs/eternity-pe-ge-me-le/Peer-to-Peer-calling.pdf

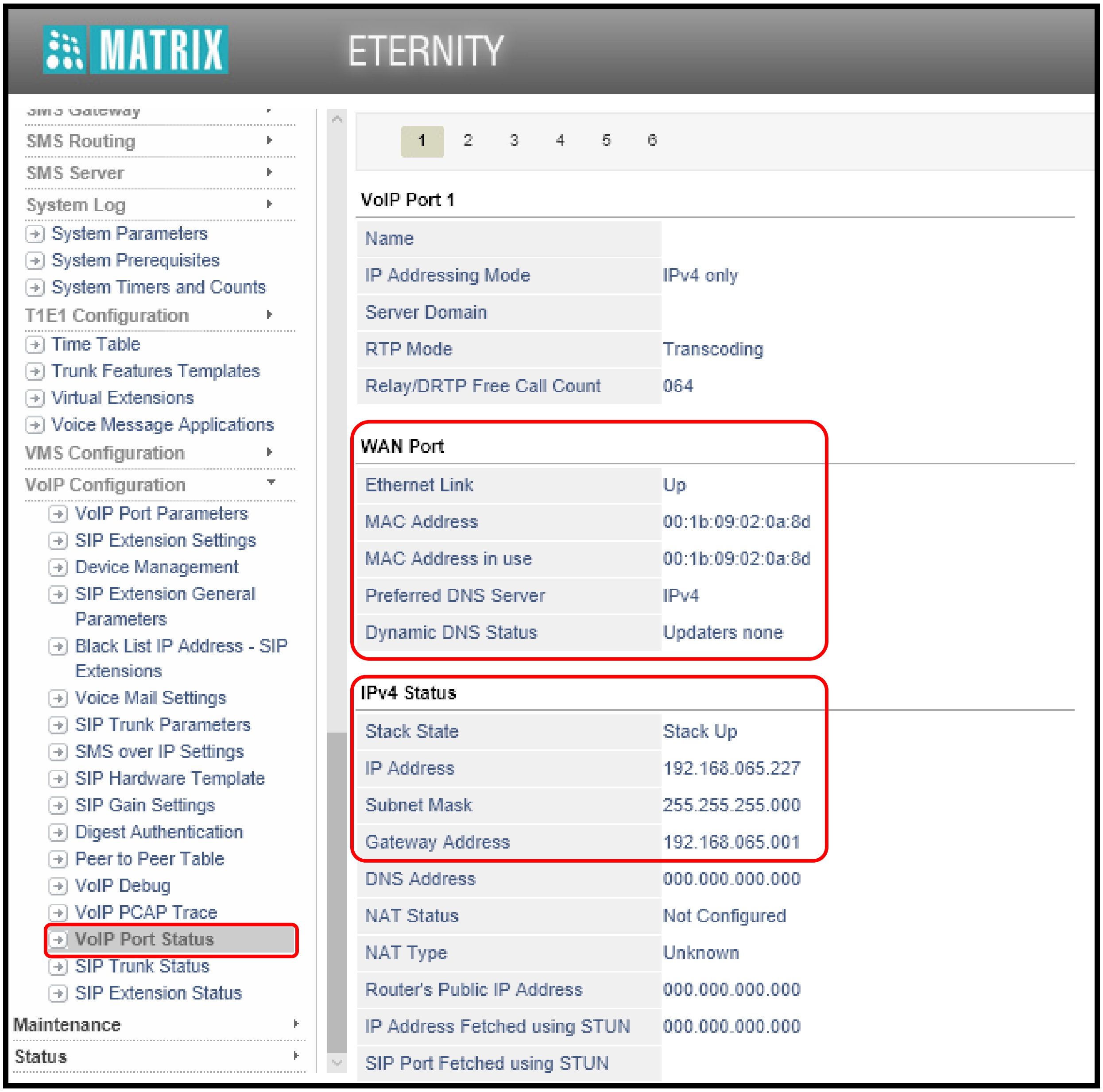
- For Peer to Peer configuration between ETERNITY NE and NAVAN refer to the following link: http://www.matrixtelesol.com/faqs/navan-cnx200/Peer-toPeer-calling-between-ETERNITY-NE-and-NAVAN.pdf

- For Peer to Peer calling between ETERNITY PE/GE/ME/LE and standard SIP phone (SPARSH VP110), refer to the following link: http://www.matrixtelesol.com/faqs/eternity-pe-ge-me-le/Peer-to-Peer-Calling-between-ETERNITY-and-SPARSH-VP110.pdf

Troubleshooting:

After completing the configuration as per the guidance provided in the above links, the calling should ideally start. If it doesn't please follow the below mentioned steps:

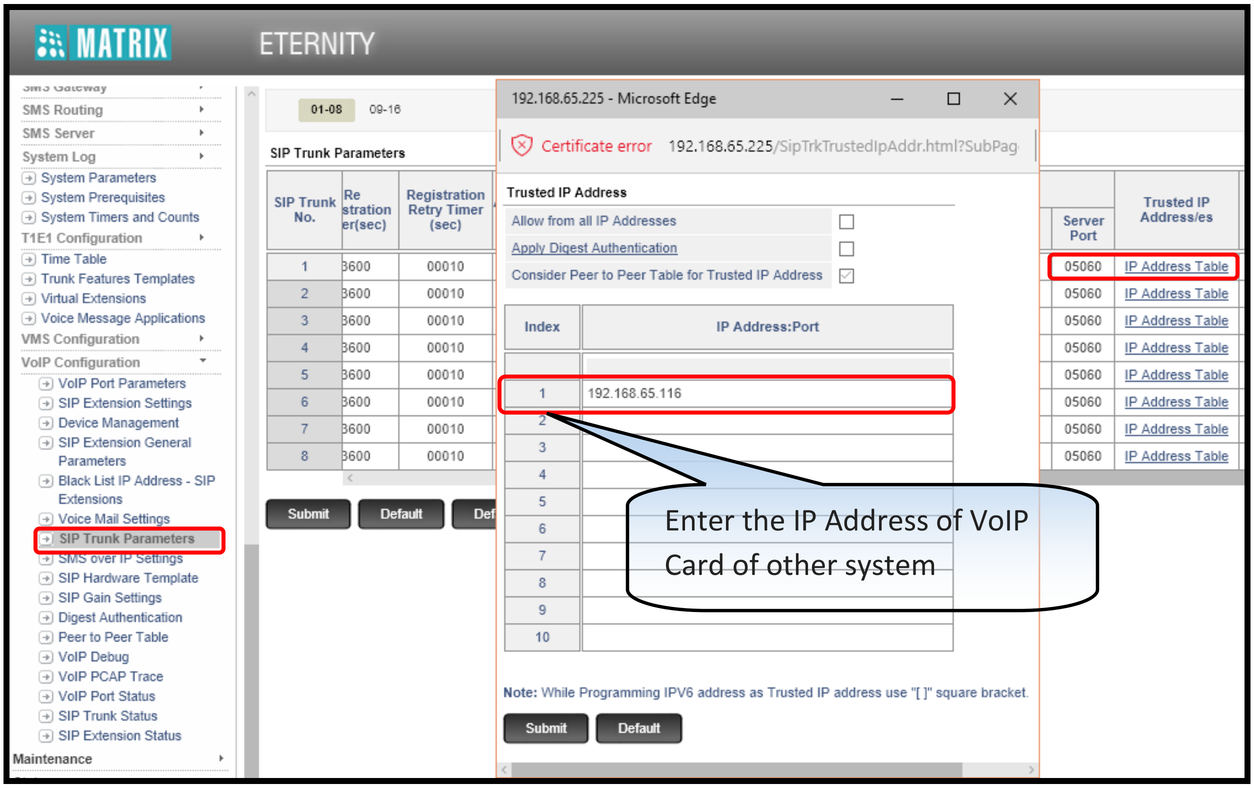
Step 1: Check the status of VoIP Port under VoIP Port Status in VoIP Configuration.



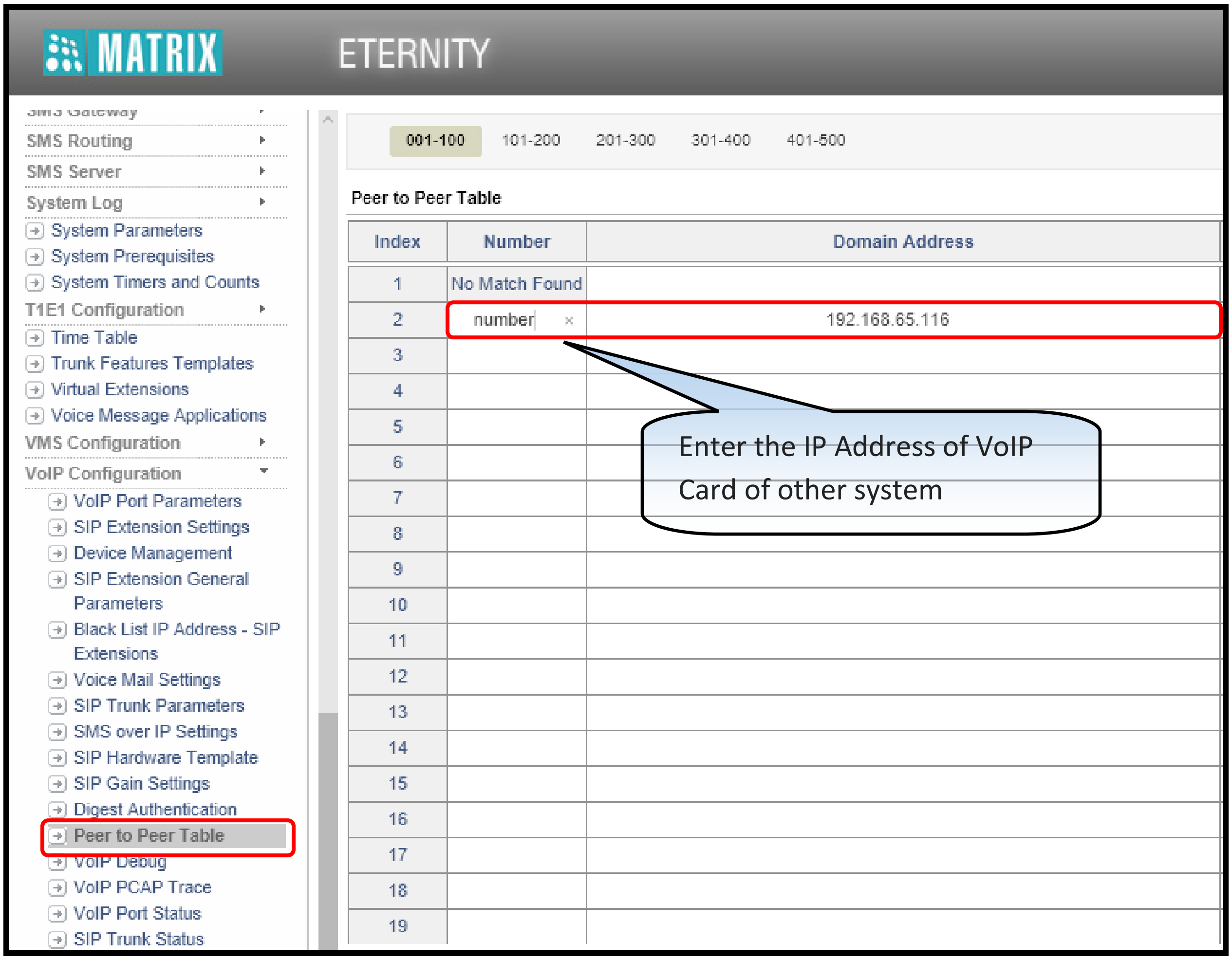
Step 2: Check the connectivity of VoIP Card within the network. Check PING response of the IP assigned to the respective WAN Port.

Step 3: Check the IP address entered in the Peer to Peer table and trusted IP table in both the sides. Make sure that the IP of VoIP Port has been entered in these tables.

Trusted IP table:

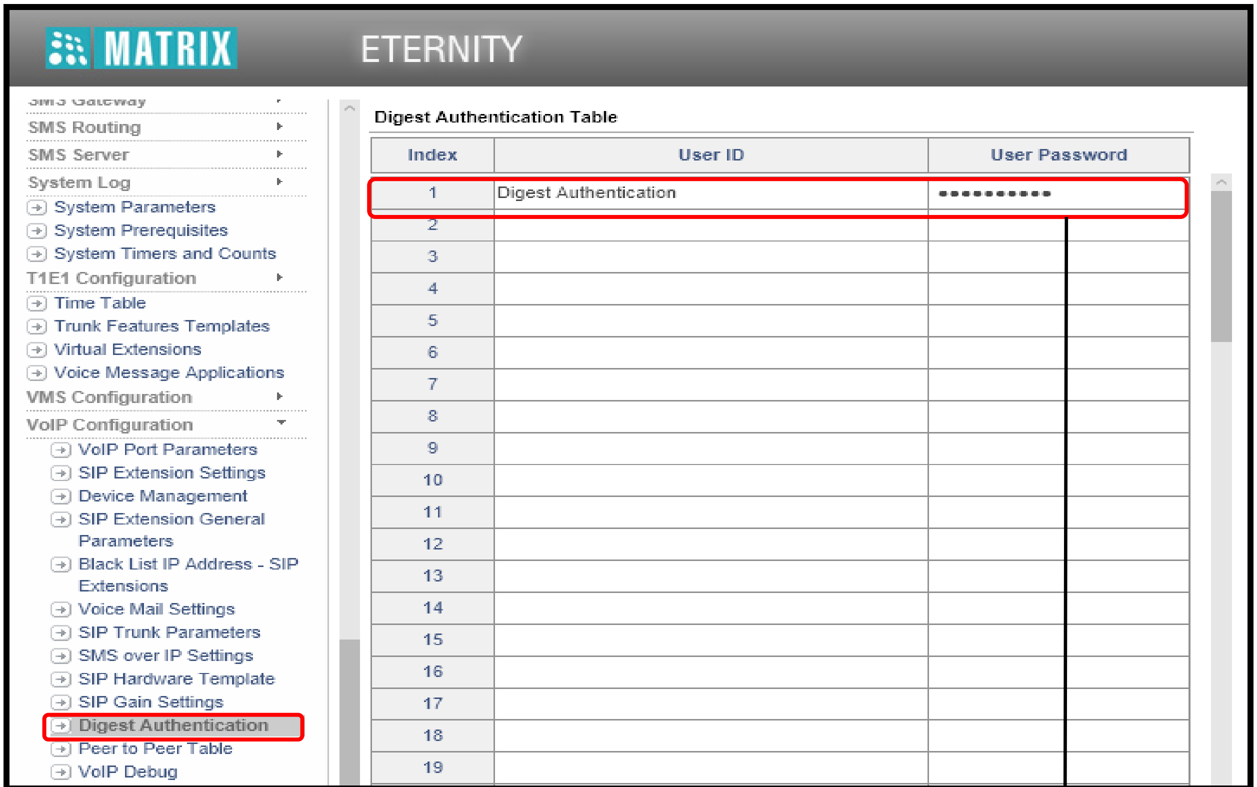


Peer to Peer table:

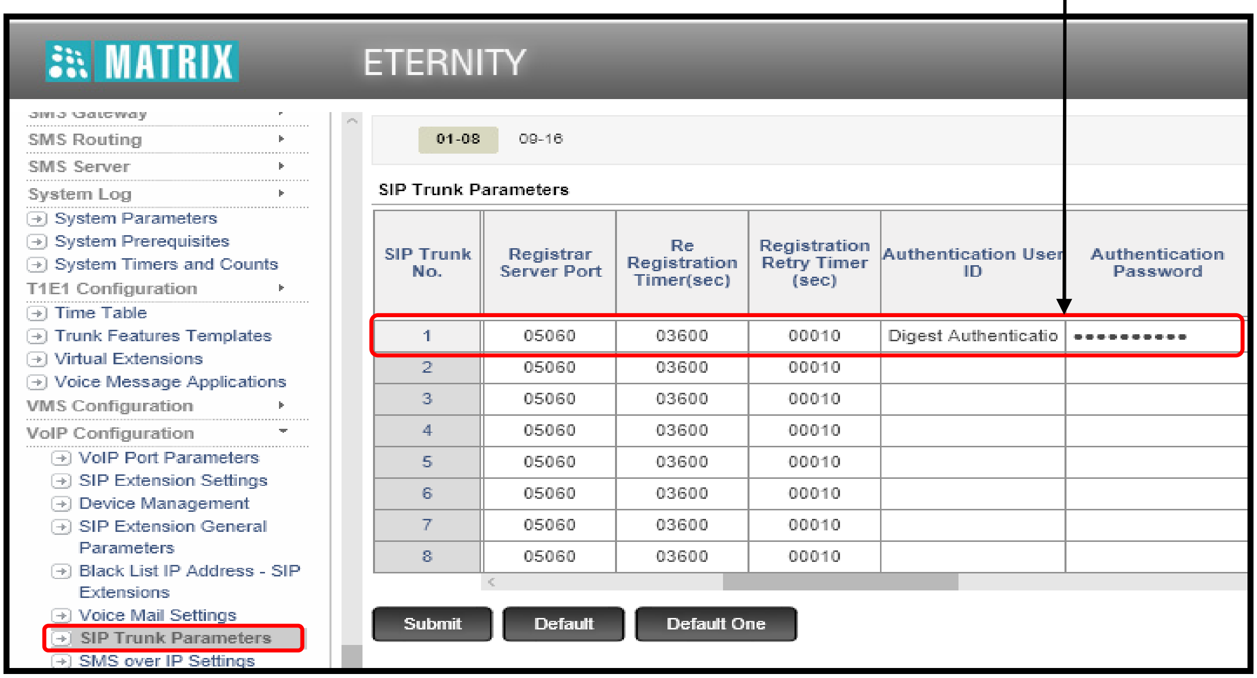


Step 4: If you have enabled Digest Authentication, make sure you enter the authentication ID and password of system A in the SIP Trunk Parameters of system B and vice versa.

System A:



System B:



Step 5: If the system version is V12R4.1 or lower and you are using white list IP addressing, make sure you enter the IP Address of VoIP Card of both systems on the opposite side.

For more information on the white list IP addressing, refer to the following link:

https://www.youtube.com/watch?v=ru7_THWtdFQ

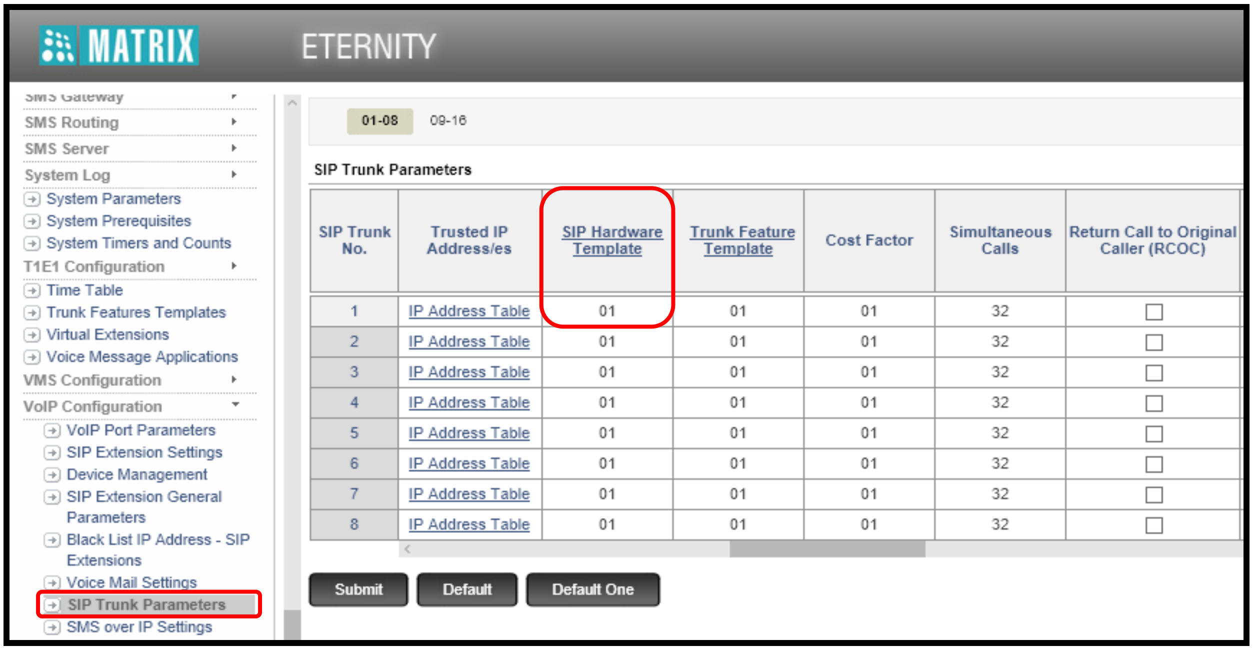
Step 6: For the NAT scenarios, make sure that proper port forwarding is done at both the sites. Without proper port forwarding, Peer to Peer calling will not work. Disable **SIP ALG** in Router/Firewall if you want to use Peer to Peer calling on Public Network.

For more information on port forwarding refer to the following link:

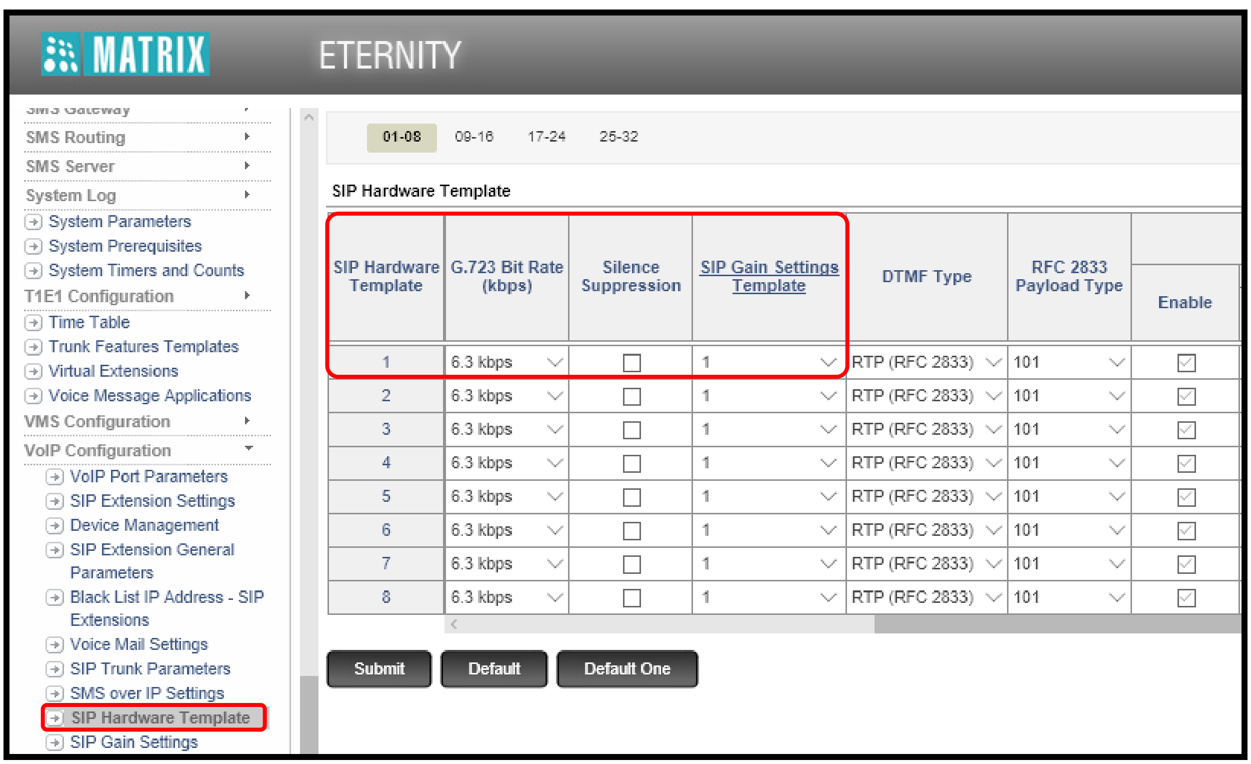
http://www.matrixtelesol.com/mtsm/MTSM-40.pdf

Step 7: If you are using DynDNS, don't forget to configure STUN and check the status of STUN in VoIP Port Status. NAT Status will show 'successful' if configured correctly.

Step 8: For low speech volume on SIP Trunk, change the gain settings under SIP Gain Settings in the SIP Hardware Template. For that, you will be required to check the SIP hardware template assigned to the respective SIP trunk.

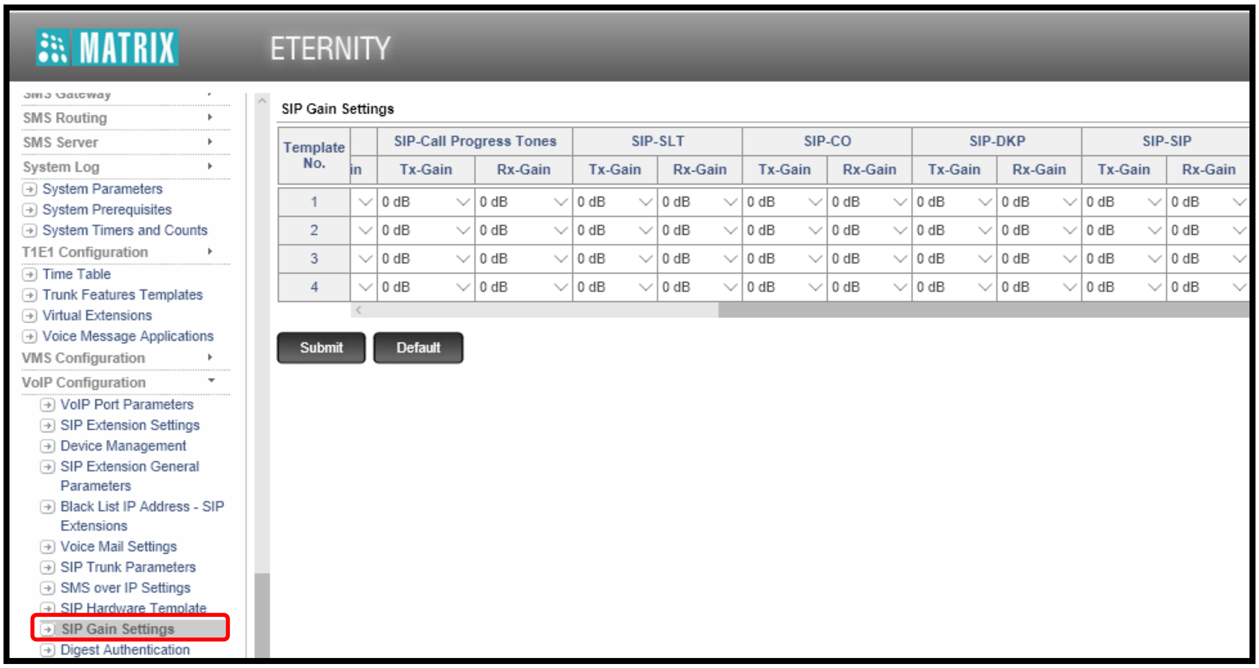


Go to SIP Hardware Template and check the SIP Gain Settings Template assigned to that particular SIP Hardware Template.

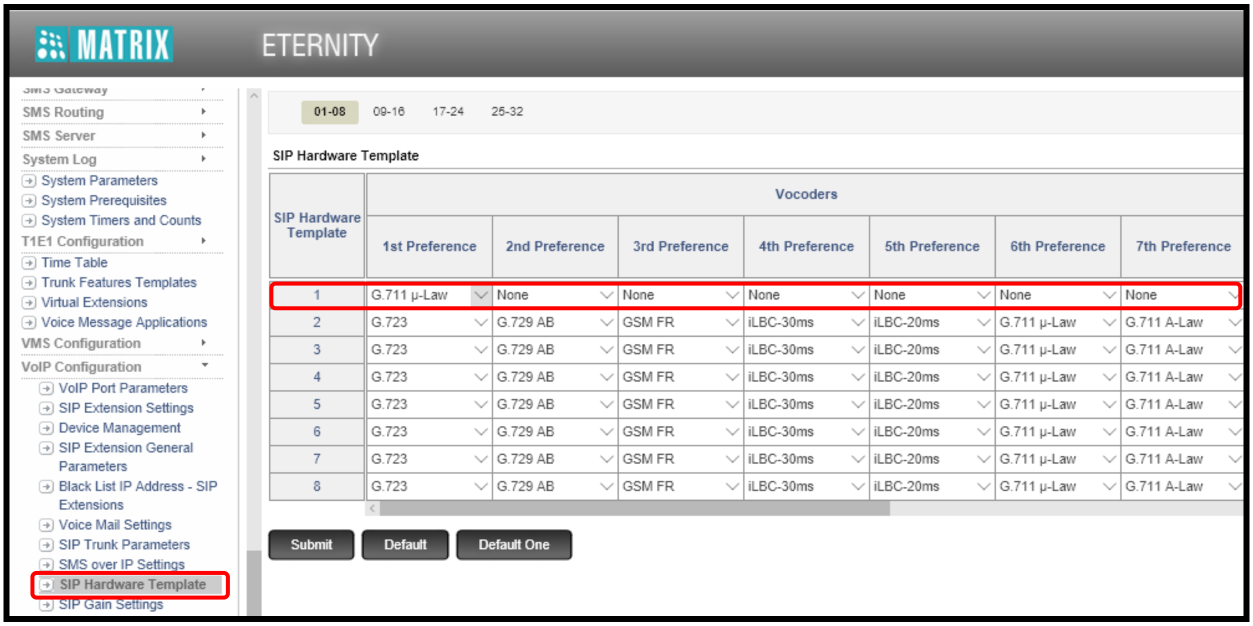


Tx-Gain refers to the transmit gain whereas Rx-Gain refers to the Receiving gain. Positive gain will result in increase in the volume whereas negative gain will reduce the volume. Tune the gain settings according to your requirement.

SIP gain settings template looks as follows:



Step 9: For disturbance/noise on SIP Trunk, make sure you select same speech vocoders at both the sites. If possible, select G.711 µlaw as first preference and all other preferences as none.



Make sure that you change the vocoders in the SIP hardware template assigned to your SIP Trunk.