

FREQUENTLY ASKED QUESTIONS

Telecom Range of Products

What
When
Which
Where
How
Who
Why



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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.

The screenshot shows the MATRIX ETERNITY web interface for VoIP Configuration. The left sidebar contains a navigation menu with the following items: Gateway, SMS Routing, SMS Server, System Log, System Parameters, System Prerequisites, System Timers and Counts, T1E1 Configuration, Time Table, Trunk Features Templates, Virtual Extensions, Voice Message Applications, VMS Configuration, VoIP Configuration, and Maintenance/Status. Under VoIP Configuration, 'VoIP Port Status' is highlighted with a red box. The main content area shows 'VoIP Port 1' configuration. The 'WAN Port' section is highlighted with a red box and contains the following data:

WAN Port	
Ethernet Link	Up
MAC Address	00:1b:09:02:0a:8d
MAC Address in use	00:1b:09:02:0a:8d
Preferred DNS Server	IPv4
Dynamic DNS Status	Updaters none

The 'IPv4 Status' section is also highlighted with a red box and contains the following data:

IPv4 Status	
Stack State	Stack Up
IP Address	192.168.065.227
Subnet Mask	255.255.255.000
Gateway Address	192.168.065.001
DNS Address	000.000.000.000
NAT Status	Not Configured
NAT Type	Unknown
Router's Public IP Address	000.000.000.000
IP Address Fetched using STUN	000.000.000.000
SIP Port Fetched using STUN	

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:

The screenshot shows the ETERNITY configuration interface. On the left is a navigation menu with 'SIP Trunk Parameters' highlighted. The main area is divided into two panes. The left pane shows a table of SIP Trunk Parameters:

SIP Trunk No.	Registration (sec)	Registration Retry Timer (sec)
1	3600	00010
2	3600	00010
3	3600	00010
4	3600	00010
5	3600	00010
6	3600	00010
7	3600	00010
8	3600	00010

The right pane shows the 'Trusted IP Address' configuration. It includes checkboxes for 'Allow from all IP Addresses', 'Apply Digest Authentication', and 'Consider Peer to Peer Table for Trusted IP Address'. Below these is a table for the Trusted IP Address:

Index	IP Address:Port
1	192.168.65.116
2	
3	
4	
5	
6	
7	
8	
9	
10	

A callout bubble points to the first row of this table with the text: 'Enter the IP Address of VoIP Card of other system'. To the right of the main configuration area is a table with two columns: 'Server Port' and 'Trusted IP Address/es'. It contains multiple rows, each with '05060' in the first column and 'IP Address Table' in the second column.

Peer to Peer table:

The screenshot displays the MATRIX ETERNITY configuration interface. On the left is a navigation menu with categories like SMS Gateway, System Log, T1E1 Configuration, VMS Configuration, and VoIP Configuration. The 'Peer to Peer Table' option under VoIP Configuration is highlighted with a red box. The main area shows a 'Peer to Peer Table' with columns for Index, Number, and Domain Address. A red box highlights the second row, where the 'Number' field contains 'number' and the 'Domain Address' field contains '192.168.65.116'. A callout bubble points to the 'Number' field with the text: 'Enter the IP Address of VoIP Card of other system'.

Index	Number	Domain Address
1	No Match Found	
2	number x	192.168.65.116
3		
4		
5		
6		
7		
8		
9		
10		
11		
12		
13		
14		
15		
16		
17		
18		
19		

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:

The screenshot shows the MATRIX ETERNITY web interface. On the left is a navigation menu with categories like SMS Gateway, System Log, T1E1 Configuration, VMS Configuration, and VoIP Configuration. The 'Digest Authentication' option under VoIP Configuration is highlighted with a red box. The main content area displays the 'Digest Authentication Table' with the following data:

Index	User ID	User Password
1	Digest Authentication	*****
2		
3		
4		
5		
6		
7		
8		
9		
10		
11		
12		
13		
14		
15		
16		
17		
18		
19		

The first row of the table is highlighted with a red border. A vertical arrow points from the 'Digest Authentication' text in the first row of this table down to the 'Digest Authentication' text in the table below.

System B:

The screenshot shows the MATRIX ETERNITY web interface. On the left is a navigation menu with categories like SMS Gateway, System Log, T1E1 Configuration, VMS Configuration, and VoIP Configuration. The 'SIP Trunk Parameters' option under VoIP Configuration is highlighted with a red box. The main content area displays the 'SIP Trunk Parameters' configuration page for the date range 01-08 to 09-16. The table below shows the parameters for SIP Trunk No. 1 through 8:

SIP Trunk No.	Registrar Server Port	Re Registration Timer(sec)	Registration Retry Timer (sec)	Authentication User ID	Authentication Password
1	05060	03600	00010	Digest Authentication	*****
2	05060	03600	00010		
3	05060	03600	00010		
4	05060	03600	00010		
5	05060	03600	00010		
6	05060	03600	00010		
7	05060	03600	00010		
8	05060	03600	00010		

The first row of the table is highlighted with a red border. A vertical arrow points from the 'Digest Authentication' text in the first row of this table up to the 'Digest Authentication' text in the table above.

At the bottom of the page, there are three buttons: 'Submit', 'Default', and 'Default One'.

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.

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01-08 09-16

SIP Trunk Parameters

SIP Trunk No.	Trusted IP Address/es	SIP Hardware Template	Trunk Feature Template	Cost Factor	Simultaneous Calls	Return Call to Original Caller (RCOC)
1	IP Address Table	01	01	01	32	<input type="checkbox"/>
2	IP Address Table	01	01	01	32	<input type="checkbox"/>
3	IP Address Table	01	01	01	32	<input type="checkbox"/>
4	IP Address Table	01	01	01	32	<input type="checkbox"/>
5	IP Address Table	01	01	01	32	<input type="checkbox"/>
6	IP Address Table	01	01	01	32	<input type="checkbox"/>
7	IP Address Table	01	01	01	32	<input type="checkbox"/>
8	IP Address Table	01	01	01	32	<input type="checkbox"/>

Submit Default Default One

Navigation Menu:

- SMS Gateway
- SMS Routing
- SMS Server
- System Log
- System Parameters
- System Prerequisites
- System Timers and Counts
- T1E1 Configuration
- Time Table
- Trunk Features Templates
- Virtual Extensions
- Voice Message Applications
- VMS Configuration
- VoIP Configuration
 - VoIP Port Parameters
 - SIP Extension Settings
 - Device Management
 - SIP Extension General Parameters
 - Black List IP Address - SIP Extensions
 - Voice Mail Settings
 - SIP Trunk Parameters**
 - SMS over IP Settings

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

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01-08 09-16 17-24 25-32

SIP Hardware Template

SIP Hardware Template	G.723 Bit Rate (kbps)	Silence Suppression	SIP Gain Settings Template	DTMF Type	RFC 2833 Payload Type	Enable
1	6.3 kbps	<input type="checkbox"/>	1	RTP (RFC 2833)	101	<input checked="" type="checkbox"/>
2	6.3 kbps	<input type="checkbox"/>	1	RTP (RFC 2833)	101	<input checked="" type="checkbox"/>
3	6.3 kbps	<input type="checkbox"/>	1	RTP (RFC 2833)	101	<input checked="" type="checkbox"/>
4	6.3 kbps	<input type="checkbox"/>	1	RTP (RFC 2833)	101	<input checked="" type="checkbox"/>
5	6.3 kbps	<input type="checkbox"/>	1	RTP (RFC 2833)	101	<input checked="" type="checkbox"/>
6	6.3 kbps	<input type="checkbox"/>	1	RTP (RFC 2833)	101	<input checked="" type="checkbox"/>
7	6.3 kbps	<input type="checkbox"/>	1	RTP (RFC 2833)	101	<input checked="" type="checkbox"/>
8	6.3 kbps	<input type="checkbox"/>	1	RTP (RFC 2833)	101	<input checked="" type="checkbox"/>

Submit Default Default One

Navigation Menu:

- SMS Gateway
- SMS Routing
- SMS Server
- System Log
- System Parameters
- System Prerequisites
- System Timers and Counts
- T1E1 Configuration
- Time Table
- Trunk Features Templates
- Virtual Extensions
- Voice Message Applications
- VMS Configuration
- VoIP Configuration
 - VoIP Port Parameters
 - SIP Extension Settings
 - Device Management
 - SIP Extension General Parameters
 - Black List IP Address - SIP Extensions
 - Voice Mail Settings
 - SIP Trunk Parameters
 - SMS over IP Settings
 - SIP Hardware Template**
 - SIP Gain Settings

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:

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SIP Gain Settings

Template No.	SIP-Call Progress Tones		SIP-SLT		SIP-CO		SIP-DKP		SIP-SIP	
	Tx-Gain	Rx-Gain	Tx-Gain	Rx-Gain	Tx-Gain	Rx-Gain	Tx-Gain	Rx-Gain	Tx-Gain	Rx-Gain
1	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB
2	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB
3	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB
4	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB

Submit Default

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

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01-08 09-16 17-24 25-32

SIP Hardware Template

SIP Hardware Template	Vocoders						
	1st Preference	2nd Preference	3rd Preference	4th Preference	5th Preference	6th Preference	7th Preference
1	G.711 μ-Law	None	None	None	None	None	None
2	G.723	G.729 AB	GSM FR	iLBC-30ms	iLBC-20ms	G.711 μ-Law	G.711 A-Law
3	G.723	G.729 AB	GSM FR	iLBC-30ms	iLBC-20ms	G.711 μ-Law	G.711 A-Law
4	G.723	G.729 AB	GSM FR	iLBC-30ms	iLBC-20ms	G.711 μ-Law	G.711 A-Law
5	G.723	G.729 AB	GSM FR	iLBC-30ms	iLBC-20ms	G.711 μ-Law	G.711 A-Law
6	G.723	G.729 AB	GSM FR	iLBC-30ms	iLBC-20ms	G.711 μ-Law	G.711 A-Law
7	G.723	G.729 AB	GSM FR	iLBC-30ms	iLBC-20ms	G.711 μ-Law	G.711 A-Law
8	G.723	G.729 AB	GSM FR	iLBC-30ms	iLBC-20ms	G.711 μ-Law	G.711 A-Law

Submit Default Default One

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

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