

FREQUENTLY ASKED QUESTIONS

ETERNITY PE/GE/ME/LE, SPARSH VP110

What
When
Which
Where
How
Who
Why

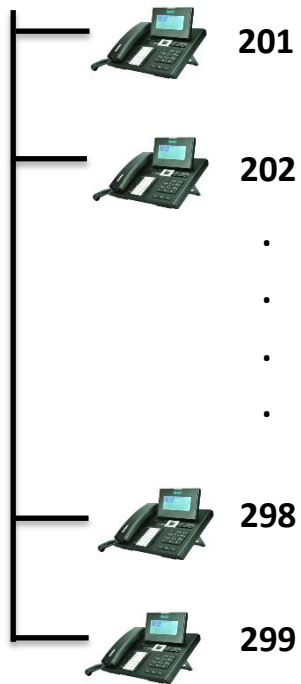
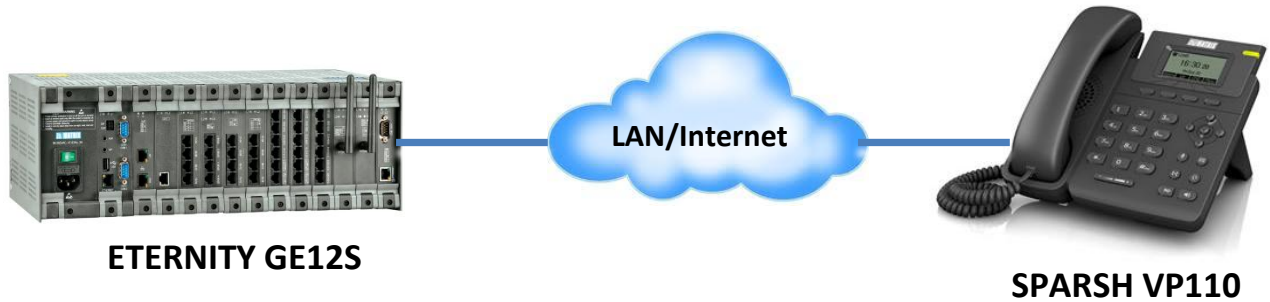


Date: 23rd December, 2014

Version: V1. R1

Author: Tapan Upmanyu

How to configure Peer to Peer calling in SPARSH VP110?



You can follow the below given steps to make a Peer to Peer calling in SPARSH VP110.

Step1: Enter IP details of VoIP card of ETERNITY

MATRIX ETERNITY

Voice Message Applications

VMS Configuration

- VMS Ethernet Port Parameters
- Ethernet Port Status
- Voice Mail Auto Attendant Profile
- Notification via Call - Profile
- VMS General Parameters
- Memory Status
- SMTP Settings
- Distribution List
- Graph
- General Mailbox Settings
- Extensions Over Q-SIG
- Debug

VoIP Configuration

- VoIP Port Parameters**
- SIP Extension Settings
- SIP Extension General Parameters
- Voice Mail Settings
- SIP Trunk Parameters

WAN Port

MAC Address: 00:1b:09:02:18:f0

Use MAC Cloning:

Clone MAC Address:

Connection Type: Static

PPPoE User ID:

PPPoE Password:

PPPoE Service Name:

IP Address: 192 . 168 . 051 . 184

Subnet Mask: 255 . 255 . 255 . 000

Default Gateway: 192 . 168 . 051 . 001

Domain Name Server (DNS)

DNS Address Assignment: Static

DNS Address: 192 . 168 . 051 . 001

DNS Domain Name:

Step2: Enable Peer to Peer trunk in ETERNITY

MATRIX ETERNITY

Voice Message Applications

VMS Configuration

- VMS Ethernet Port Parameters
- Ethernet Port Status
- Voice Mail Auto Attendant Profile
- Notification via Call - Profile
- VMS General Parameters
- Memory Status
- SMTP Settings
- Distribution List
- Graph
- General Mailbox Settings
- Extensions Over Q-SIG
- Debug

VoIP Configuration

- VoIP Port Parameters
- SIP Extension Settings
- SIP Extension General Parameters
- Voice Mail Settings
- SIP Trunk Parameters**
- SMS over IP Settings
- SIP Hardware Template

SIP Trunk Parameters

SIP Trunk No.	VoIP Port No.	Enable SIP Trunk	Name	SIP ID	SIP Trunk Mode	Check SIP ID during incoming call	Check Proxy Address for Incoming SIP Message	Check Proxy Port for Incoming SIP Message	Treat Incoming Peer-To-Peer call as
1	01	<input checked="" type="checkbox"/>			Peer-To-Peer	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Station
2	00	<input type="checkbox"/>			Proxy	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Trunk
3	00	<input type="checkbox"/>			Proxy	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Trunk
4	00	<input type="checkbox"/>			Proxy	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Trunk

Submit Default Default One Advance

Step3: Open SPARSH VP110's GUI

MATRIX

Login SPARSH VP110

Username

Password

Step4: Go to Features and then General Information

MATRIX SPARSH VP110 Log Out

Status Account Network DSSKey **Features** Settings Directory Security

Forward&DND

General Information

Audio

Intercom

Transfer

Call Pickup

Remote Control

Phone Lock

ACD

SMS

Action URL

Power LED

General Information

Call Waiting	<input type="text" value="Enabled"/>	<input "="" type="button" value="?"/>
Call Waiting On Code	<input type="text"/>	<input "="" type="button" value="?"/>
Call Waiting Off Code	<input type="text"/>	<input "="" type="button" value="?"/>
Auto Redial	<input type="text" value="Disabled"/>	<input "="" type="button" value="?"/>
Auto Redial Interval (1~300s)	<input type="text" value="10"/>	<input "="" type="button" value="?"/>
Auto Redial Times (1~300)	<input type="text" value="10"/>	<input "="" type="button" value="?"/>
Key As Send	<input type="text" value="#"/>	<input "="" type="button" value="?"/>
Reserve # In User Name	<input type="text" value="Enabled"/>	<input "="" type="button" value="?"/>
Hotline Number	<input type="text"/>	<input "="" type="button" value="?"/>
Hotline Delay(0~10s)	<input type="text" value="4"/>	<input "="" type="button" value="?"/>
Busy Tone Delay (Seconds)	<input type="text" value="0"/>	<input "="" type="button" value="?"/>
Return Code When Refuse	<input type="text" value="486 (Busy Here)"/>	<input "="" type="button" value="?"/>
Return Code When DND	<input type="text" value="480 (Temporarily Not Av)"/>	<input "="" type="button" value="?"/>
Call Completion	<input type="text" value="Disabled"/>	<input "="" type="button" value="?"/>
Time-Out for Dial-Now Rule	<input type="text" value="1"/>	<input "="" type="button" value="?"/>
RFC 2543 Hold	<input type="text" value="Disabled"/>	<input "="" type="button" value="?"/>
Use Outbound Proxy In Dialog	<input type="text" value="Enabled"/>	<input "="" type="button" value="?"/>

NOTE

Call Waiting
This call feature allows your phone to accept other incoming calls during the conversation.

Key As Send
Select * or # as the send key.

Hotline Number
When you pick up the phone, it will dial out the hotline number automatically.

Step5: Go to Features and then General Information

Send Pound Key	Disabled	?
Fwd International	Enabled	?
Diversion/History-Info	Enabled	?
Allow Trans Exist Call	Enabled	?
Auto-Logout Time(1~1000min)	5	?
Call Number Filter	-,	?
Use Logo	System logo	?
Allow IP Call	Enable	?
IP Direct Auto Answer	Enable	?
Call List Show Number	Disabled	?
Voice Mail Tone	Enable	?
DHCP Hostname	Matrix SPARSH VP110	?
Reboot In Talking	Disabled	?

Buttons: Confirm, Cancel

Callout: Enable Allow IP Call Field

Step6: Enter SIP account details

MATRIX SPARSH VP110

Log Out

Account

Register

Register Status	Disabled	?
Line Active	Enabled	?
Register	Disabled	?
Label		?
Display Name		?
Register Name		?
User Name	3301	?
Password	*****	?
Enable Outbound Proxy Server	Disabled	?
Outbound Proxy Server		Port 5060 ?
Transport	UDP	?
NAT		?
STUN Server		Port 3478 ?
SIP Server 1		?
Server Host		
Server Expires	3600	?
Server Retry Counts	3	?

NOTE

Display Name
SIP service subscriber's name which will be used for Caller ID display.

Register Name
SIP service subscriber's ID used for authentication.

User Name
User account, provided by VoIP service provider.

NAT Traversal
Defines the STUN server will be active or not.

Callout: Enable Line Active and Disable Register

Step7: Enter SIP Server details

The screenshot shows the 'Account' configuration page for a 'Register' in the MATRIX SPARSH VP110 system. The 'Register Status' is 'Disabled'. The 'SIP Server 1' section is highlighted with a red box, showing the following details:

Line Active	Enabled
Register	Disabled
Label	
Display Name	
Register Name	
User Name	3301
Password	••••••••
Enable Outbound Proxy Server	Disabled
Outbound Proxy Server	
Port	5060
Transport	UDP
NAT	Disabled
STUN Server	
Port	3478
SIP Server 1	
Server Host	192.168.51.184
Port	5060
Server Expires	3600
Server Retry Counts	3
SIP Server 2	

NOTE:

- Display Name**: SIP service subscriber's name which will be used for Caller ID display.
- Register Name**: SIP service subscriber's ID used for authentication.
- User Name**: User account, provided by VoIP service provider.
- NAT Traversal**: Defines the STUN server will be active or not.

The screenshot shows the 'WAN Port' configuration page. A callout box points to the 'IP Address' field, which is highlighted with a red box. The IP Address is 192.168.051.184. Other fields include Subnet Mask (255.255.255.000) and Default Gateway (192.168.051.001).

Enter the IP Address of WAN Port of VoIP card in SIP Server 1 (Server host)

VoIP card IP details on ETERNITY side

IP Address	192	168	051	184
Subnet Mask	255	255	255	000
Default Gateway	192	168	051	001

Domain Name Server (DNS)

DNS Address Assignment	Static			
DNS Address	192	168	051	001

Step8: Registration status after completion of configuration

The screenshot shows the Matrix SPARSH VP110 web interface. The 'Account' tab is selected, and the 'Register' section is active. The 'Register Status' is set to 'Disabled'. A callout box points to this status and contains the following text:

Register Status will show Disabled even though the Peer to Peer link is up, you have to make a call to ETERNITY's Extension to test the link

The interface also shows various configuration options for the register, including Line Active (Enabled), Register (Disabled), Label, Display Name, Register Name, Transport (UDP), NAT (Disabled), STUN Server (Port 3478), SIP Server 1 (Server Host: 192.168.51.184, Port: 5060, Server Expires: 3600, Server Retry Counts: 3), and SIP Server 2.

NOTE

- Display Name**
SIP service subscriber's name which will be used for Caller ID display.
- Register Name**
SIP service subscriber's ID used for authentication.
- User Name**
User account, provided by VoIP service provider.
- NAT Traversal**
Defines the STUN server will be active or not.

For more information, contact
Matrix Technical Training Team
Training@MatrixComSec.com

Disclaimer: The information contained in this e-mail and/or attachment may contain confidential or privileged information. Unauthorized use, disclosure or copying is strictly prohibited and may constitute unlawful act and can possibly attract legal action, civil and/or criminal. The contents of this message need not necessarily reflect or endorse the views of Matrix ComSec Pvt Ltd on any subject matter. Any action taken or omitted on this message is not entirely at your risk and the originator of this message nor does Matrix ComSec Pvt Ltd take any responsibility or liability towards the same. If you are not the intended recipient, please notify us immediately and permanently delete the message.