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Unified SIP Features

(iPECS-SMB-TRA-01-030)

13 Oct, 2017

REVISION HISTORY

ISSUE	DATE	DESCRIPTION OF CHANGES
1.0	13-Oct-17	Preliminary release

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Table of Contents

- Basic SIP configuration
- Specific settings for SIP Service
- Specific configurations
- Trace related with SIP
- Trouble Shooting

Basic Configuration

PGM 210

- Enter DNS Address
 - If Domain Name is used for SIP server, DNS Query will be sent to DNS Address.

The screenshot displays the iPECS eMG80 Administration interface. The left sidebar shows a navigation menu with 'SIP Data' expanded and 'SIP Common Attributes(210)' selected. The main content area shows a table of configuration parameters for SIP Common Attributes (210). The 'Primary DNS Address' field is highlighted with a red box and contains the value '61.41.106.223'. Below the table, there is a section for 'SIGNAL TLS OPTION' with several parameters.

Order	Attribute	Value	Range	Remark
1	Primary DNS Address	61.41.106.223	Max 32 Characters	SYSTEM will be restarted after [SAVE]
2	Secondary DNS Address		Max 32 Characters	SYSTEM will be restarted after [SAVE]
3	Local Server UDP Port	5060	Port	SYSTEM will be restarted after [SAVE]
4	Local Server TCP Port	5060	Port	SYSTEM will be restarted after [SAVE]
5	Local Server TLS Port	5061	Port	SYSTEM will be restarted after [SAVE]
6	Check Message Send Timer	120	10-3600sec	
SIGNAL TLS OPTION				
1	TLS Version	TLS1.0		SYSTEM will be restarted after [SAVE]
2	Crypt Mode	RSA		SYSTEM will be restarted after [SAVE]
3	First TLS	None		SYSTEM will be restarted after [SAVE]
4	Second TLS	None		SYSTEM will be restarted after [SAVE]
5	Persistent Level	TRANSACTION_USER		SYSTEM will be restarted after [SAVE]
6	Capacity Level	70	0-100	SYSTEM will be restarted after [SAVE]
7	Connection Reuse(TLS)	ON		

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Basic Configuration

PGM 126

- Enter Registration ID – check **[CheckBOX]** before saving data.
 - Enter index number and Click [Load] - index1 is used in capture.
 - Enter “Registration User ID” – **ID@serveraddress**.
 - Enter “Authentication User ID” and “Authentication User Password”.
 - Set “User ID Usage” – ON and set “User ID Register” – Register/Provision.

Favorite PGM SIP CO Attributes(133) SIP User ID Attributes(126)

Enter SIP User ID Index Number (1 - 140) : 1 Load Save

SIP User ID Index 1

Order	Attribute	Value
	CID Password	
1	Registration User ID	123456@150.150.150.95
2	Authentication User ID	Ericsson
3	Authentication User Password	test
4	Contact Number	
5	Contact Display Name	
6	Asc Station Number	
7	User ID Register	Register
8	Authorized Representative ID Table Index	0
9	User ID Usage	ON
10	Ring Route Type	ID ASSIGNED STATION
11	DID Conversion Type	Use 'as is' (no treatment)
12	Number of Digits Expected from DID Circuit (2-4) Expected from DID Circuit	4
13	DID Digit Mask (4 digits: *,#,0-9)	****

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Basic Configuration

PGM 133

- Soft Switch Type : It depends on the server type tested with LIK/eMG/UCP.
- Default : Normal
- Proxy Server Address : SIP Server Address - IP Address or Domain.
- Domain : SIP Server Domain - IP Address or Domain.

The screenshot displays the iPECS Administration interface for configuring SIP CO Attributes (133). The interface includes a sidebar with navigation options, a main configuration table, and a footer with copyright information.

Navigation Sidebar:

- Station Data
- Board Based Data
- CO Line Data
- System Data
- Station Group Data
- ISDN Line Data
- SIP Data**
 - SIP Common Attributes(210)
 - SIP Trunk Status Overview
 - > SIP CO Attributes(133)**
 - SIP Registration Status Overview
 - SIP UID Alloc Status Overview
 - SIP User ID Attributes(126)
 - SIP Phone Attributes(211)
 - SIP Phone Provisioning(212)
 - Provisioning File View&Delete
- Tables Data

Main Configuration Area:

Favorite PGM: SIP CO Attributes(133)

Enter CO Range (1 - 74): [] [Load] [Save] [Register] [UnRegister]

Uncheck All	Attribute	Value	Range
<input checked="" type="checkbox"/>	Soft Switch Type	Normal	
<input checked="" type="checkbox"/>	Proxy Server Address	150.150.150.95	IP Address
<input checked="" type="checkbox"/>	Use Outbound Proxy	OFF	
<input checked="" type="checkbox"/>	Connection Mode	UDP	
<input checked="" type="checkbox"/>	Caller Name Service	Use	
<input checked="" type="checkbox"/>	181 Being Forwarded	Unused	
<input checked="" type="checkbox"/>	100 rel	OFF	
<input checked="" type="checkbox"/>	Use single codec only	OFF	
<input checked="" type="checkbox"/>	Use rport method	OFF	
<input checked="" type="checkbox"/>	Domain	150.150.150.95	Max 32 Characters
<input checked="" type="checkbox"/>	Invite Acceptance	From All	
<input checked="" type="checkbox"/>	Contact Address Domain	SIP Device Addr	
<input checked="" type="checkbox"/>	From Address Domain	SIP Device Addr	
<input checked="" type="checkbox"/>	Firewall IP Apply	ON	

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Basic Configuration

PGM 133

- P-Asserted-ID, Remote-Party-ID :
 - Set Usage of P-Asserted-ID and Remote-Party-ID
- From ID, P-Asserted-ID, Contact, Remote-Party-ID
 - Select outgoing CLI rule.

The screenshot displays the iPECS eMG80 Administration interface. The left sidebar shows the navigation menu with 'SIP Data' expanded and 'SIP CO Attributes(133)' selected. The main configuration area is titled 'Favorite PGM SIP CO Attributes(133)'. It contains several sections:

- ID Presentation Option**: Includes 'Secondary Domain', 'Secondary Proxy Server UDP Port', and 'Port'. Buttons for 'Save', 'Register', and 'UnRegister' are present.
- ID Usage**: A table with the following rows:

Usage	Setting
<input checked="" type="checkbox"/>	P-Asserted-ID Use
<input checked="" type="checkbox"/>	Remote-Party-ID Use
<input checked="" type="checkbox"/>	Privacy(CLIR) Presentation Anonymous Name & Anonymous Number
- ID Individuality**: A table with the following rows:

Individuality	Setting
<input checked="" type="checkbox"/>	From ID Extension SIP-User-ID-Table
<input checked="" type="checkbox"/>	From Display SYS RULE
<input checked="" type="checkbox"/>	P-Asserted-ID Extension SIP-User-ID-Table
<input checked="" type="checkbox"/>	P-Asserted-ID Display SYS RULE
<input checked="" type="checkbox"/>	Contact ID Extension SIP-User-ID-Table
<input checked="" type="checkbox"/>	Remote-Party-ID Extension SIP-User-ID-Table
- Offnet Call Route ID Transit**: A table with the following rows:

Transit	Setting
<input checked="" type="checkbox"/>	From/Contact ID SYS ATD
<input checked="" type="checkbox"/>	From Display SYS RULE

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Basic Configuration

PGM 133

ID Settings	Description
Extension SIP-User-ID Table	<ul style="list-style-type: none">• Follow PGM111 - SIP USER TABLE INDEX number.• SIP USER TABLE INDEX number must be set PGM126-index.
Extension Outgoing CLI	<ul style="list-style-type: none">• Follow ISDN Outgoing CLI rule.• PGM113, PGM151, PGM201 ..
Authorized Representative ID	<ul style="list-style-type: none">• Follow PGM126 - Authorized Representative ID Table Index
Fixed Table	<ul style="list-style-type: none">• Follow PGM133 - SIP User ID Fixed Table Index• Normally main ID will be used.
Original CLI	<ul style="list-style-type: none">• Use CLI of incoming Trunk. – Call Forward, Call Transfer case.
Extension	<ul style="list-style-type: none">• Follow PGM133 - ID Individuality- From ID.

Display Settings	Description
SYS RULE	<ul style="list-style-type: none">• Use Station Name of related station.<ul style="list-style-type: none">- Normal outgoing call : name of outgoing station.- Call forward : name of forwarding station- Co-Co Call forward : name of attendant station.
Original	<ul style="list-style-type: none">• Use name received from incoming Trunk. If there is no name, CLI will be used as diaplay.

Basic Configuration

PGM 133

- Registration UID Range: Enter index number of PGM126 related with Registration ID

The screenshot displays the iPECS Administration interface for configuring SIP CO Attributes (133). The left sidebar shows the navigation menu with 'SIP CO Attributes(133)' selected. The main configuration area is titled 'Favorite PGM SIP CO Attributes(133)'. The configuration table includes the following fields:

Field	Value	Unit/Description
Contact Address Domain	SIP Device Addr	
From Address Domain	SIP Device Addr	
Firewall IP Apply	ON	
Diversion Recursing	Recursing	302,Blind Transfer
VSF Answer Response	200 OK	
RTP Diversion Method	Recursing	
Virtual SIP Channel Mode	No	
Proxy Registration Timer	3600	
Proxy Server UDP Port	5060	Port
Proxy Server TCP Port	5060	Port
Proxy Server TLS Port	5061	Port
Registration UID Range	1 - 1	Max 140 Entries
DTMF Type	INBAND	
Action with REG Failure	IDLE	CO State
Media Port	6000 - 7036	UDP Port
Secondary Proxy Server		
Secondary Proxy Server Address		IP Address
Secondary Domain		Max 32 Characters

Buttons for 'Save', 'Register', and 'UnRegister' are visible on the right side of the configuration table.

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Basic Configuration

SIP UID Alloc Status

- User can see the ID allocation status.

The screenshot shows the iPECS UCP100 Administration interface. The top navigation bar includes 'Administration' and 'Maintenance' tabs, along with 'Change Language' and 'Log Out' buttons. The left sidebar contains a navigation menu with the following items: System Data, Station Group Data, ISDN Line Data, SIP Data (expanded), SIP Common Attributes(210), SIP Trunk Status Overview, SIP CO Attributes(133), SIP Registration Status Overview, **SIP UID Alloc Status Overview** (highlighted), SIP User ID Attributes(126), SIP Phone Attributes(211), and SIP Phone Provisioning(212). The main content area displays a table with the following data:

Index	Station
1	1002, 1035, 1036, 1033, 1037, 1003, 1038, 1039, 1010, 1032, 1013, 1041, 1051, 1052, 1043, 1031, 1024, 1025, 1027, 1028, 1029, 1030, 1034,
2	1007, 1026,
3	1001,
4	1000,
5	
6	
7	
8	
9	
10	
11	

Basic Configuration

SIP Registration Status

- User can see the Registration status.
 - Register/Provision
 - Registration Status in case of Register

The screenshot shows the iPECS UCP100 Administration interface. The top navigation bar includes 'Administration' and 'Maintenance' tabs, along with 'Change Language' and 'Log Out' buttons. The left sidebar contains a menu with the following items: System Data, Station Group Data, ISDN Line Data, SIP Data (expanded), SIP Common Attributes(210), SIP Trunk Status Overview, SIP_CO.Attributes(133), **SIP Registration Status Overview** (highlighted in blue), SIP UID Alloc Status Overview, and SIP User ID Attributes(126). The main content area displays a table titled 'SIP Registration Status Overview' with the following data:

Index	Registration User ID	SIP Status
1	12345678@150.150.150.95	Provision
2	12345679@150.150.150.95	Provision
3	12345670@150.150.150.95	Provision
4	1018@150.150.150.95	Register - Terminated
5		
6		
7		
8		
9		
10		

Basic Configuration

SIP Trunk Status

- User can see the Admin Status for Trunk
 - 150.150.131.207 for Trunk 7-8.
 - sipconnect.qsc.de for Trunk 9-14.

The screenshot displays the iPECS UCP100 Administration interface. The left sidebar shows a navigation menu with 'SIP Trunk Status Overview' highlighted. The main content area shows a table with the following data:

Index	Proxy Address	Domain	COL Range	SIP Group	UID Range	State	UIDSEL
1	150.150.131.207	pchan.com	7 - 8	0	0 - 0	Idle	UID_Index_1
2	sipconnect.qsc.de	sipconnect.qsc.de	9 - 14	0	0 - 0	Idle	UID_Index_1
3			-		-		
4			-		-		
5			-		-		
6			-		-		
7			-		-		
8			-		-		
9			-		-		
10			-		-		
11			-		-		
12			-		-		

Basic Configuration

Incoming call

- Incoming CLI is matched with PGM126 – index .
 - Follow PGM 126 - Ring Route Type.
(ID Assigned station, Ring assignment, DID conversion, MSN-DID conversion)
- Incoming CLI is not matched with PGM126 – index .
 - Follow ISDN rule. ->PGM145, PGM151.

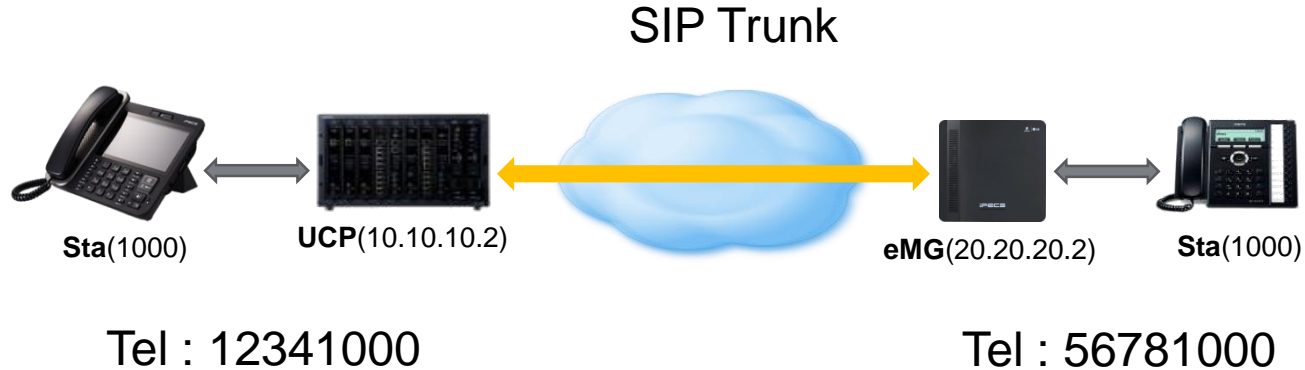
The screenshot displays the iPECS UCP100 Administration interface. The left sidebar shows a navigation menu with 'SIP User ID Attributes(126)' highlighted. The main content area shows the configuration for 'SIP User ID Index 1'. A table lists various attributes with their values and ranges. The 'Ring Route Type' attribute (row 10) is highlighted with a red box and set to 'ID ASSIGNED STATION'.

Order	Check All	Attribute	Value	Range
		CID Password	<input type="text"/> Go to Setting	
1	<input type="checkbox"/>	Registration User ID	12345678@150.150.150.95	Max 64 Characters
2	<input type="checkbox"/>	Authentication User ID	<input type="text"/>	Max 64 Characters
3	<input type="checkbox"/>	Authentication User Password	<input type="text"/>	Max 32 Characters
4	<input type="checkbox"/>	Contact Number	12345678	Max 16 Characters
5	<input type="checkbox"/>	Contact Display Name	<input type="text"/>	Max 21 Characters
6	<input type="checkbox"/>	Asc Station Number	<input type="text"/>	
7	<input type="checkbox"/>	User ID Register	Provision ▾	
8	<input type="checkbox"/>	Authorized Representative ID Table Index	0	0 - 2400
9	<input type="checkbox"/>	User ID Usage	ON ▾	
10	<input type="checkbox"/>	Ring Route Type	ID ASSIGNED STATION ▾	
11	<input type="checkbox"/>	DID Conversion Type	DID Digit Conversion ▾	
12	<input type="checkbox"/>	Number of Digits Expected from DID Circuit	4	2-4
13	<input type="checkbox"/>	DID Digit Mask	****	4 Digits: *,#,0-9
14	<input type="checkbox"/>	SMS Received Station Number	<input type="text"/>	

Basic Configuration

Trunk configuration

- Configure Basic SIP configuration between UCP and eMG80.



Basic Configuration

Trunk configuration

- Configure Basic SIP configuration with F/W device.
 - Set Port Forwarding in each F/W – SIP port and RTP ports
 - Set F/W IP in PGM102 - Firewall IP Address and PGM132 - Firewall IP Address.

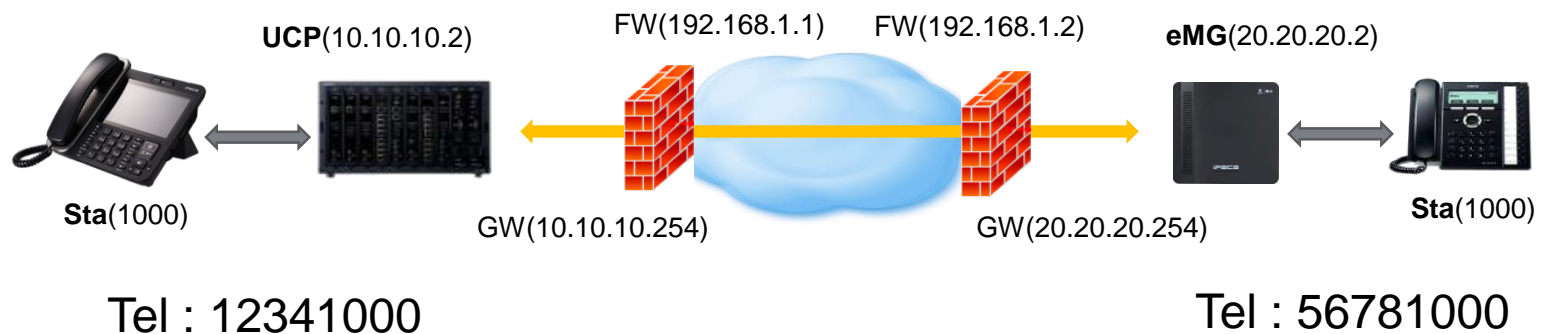


UCP:

- Slot number of VOIM (2402)

eMG80:

- Slot number of VOIB (14)

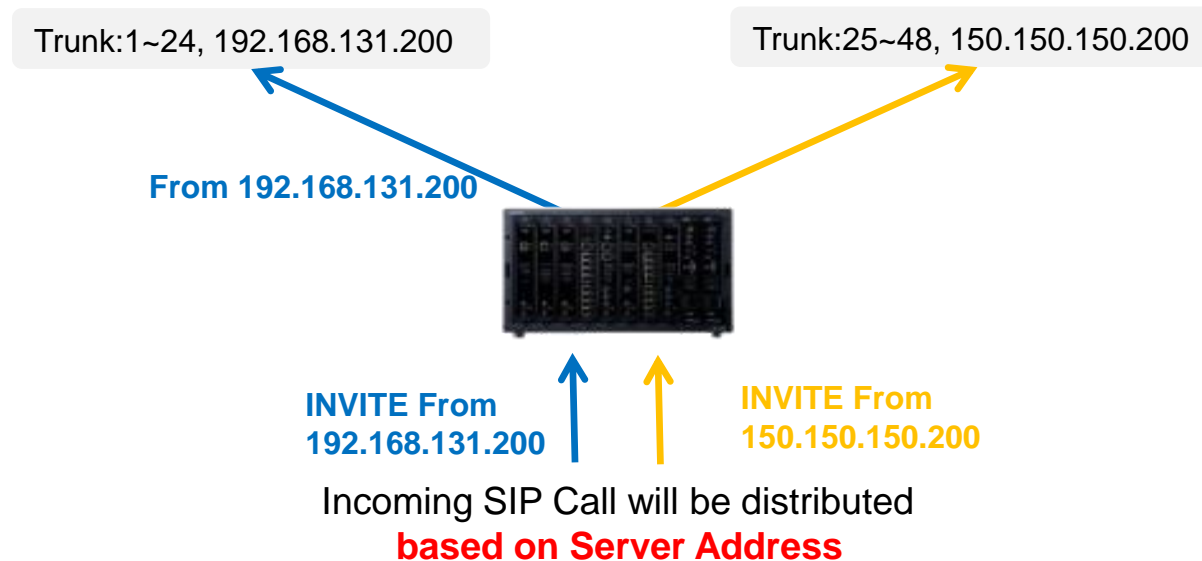


Specific settings for SIP Service

SIP Group for SIP Trunk Control

All SIP Trunk will be grouped based on PGM133 – Proxy Server Address.

- For Co 1~24, Proxy Server Address is 192.168.131.200.
- For Co25~48, Proxy Server Address is 150.150.150.200.
- Then two SIP trunk Group will be stored. → **Co range must be sequential.**



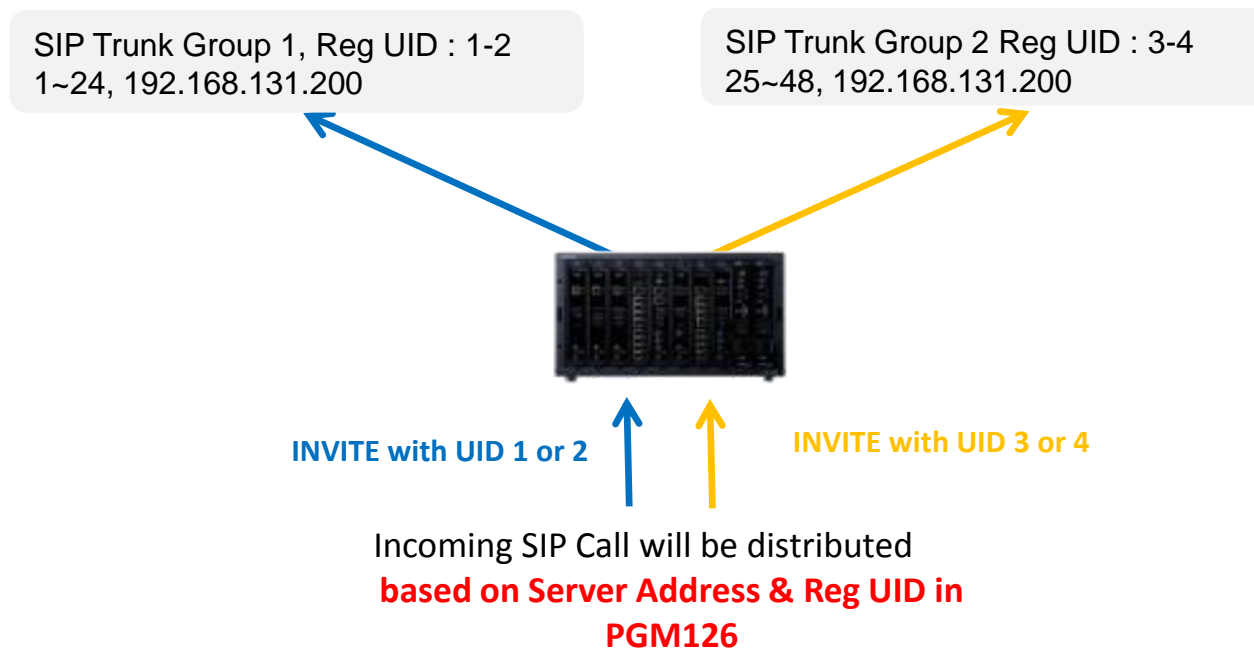
Specific settings for SIP Service

PGM133 - SIP Trunk Group

It is independent with PGM140 – CO/IP Group.

- SIP Trunk Group setting is just used in SIP Trunk.

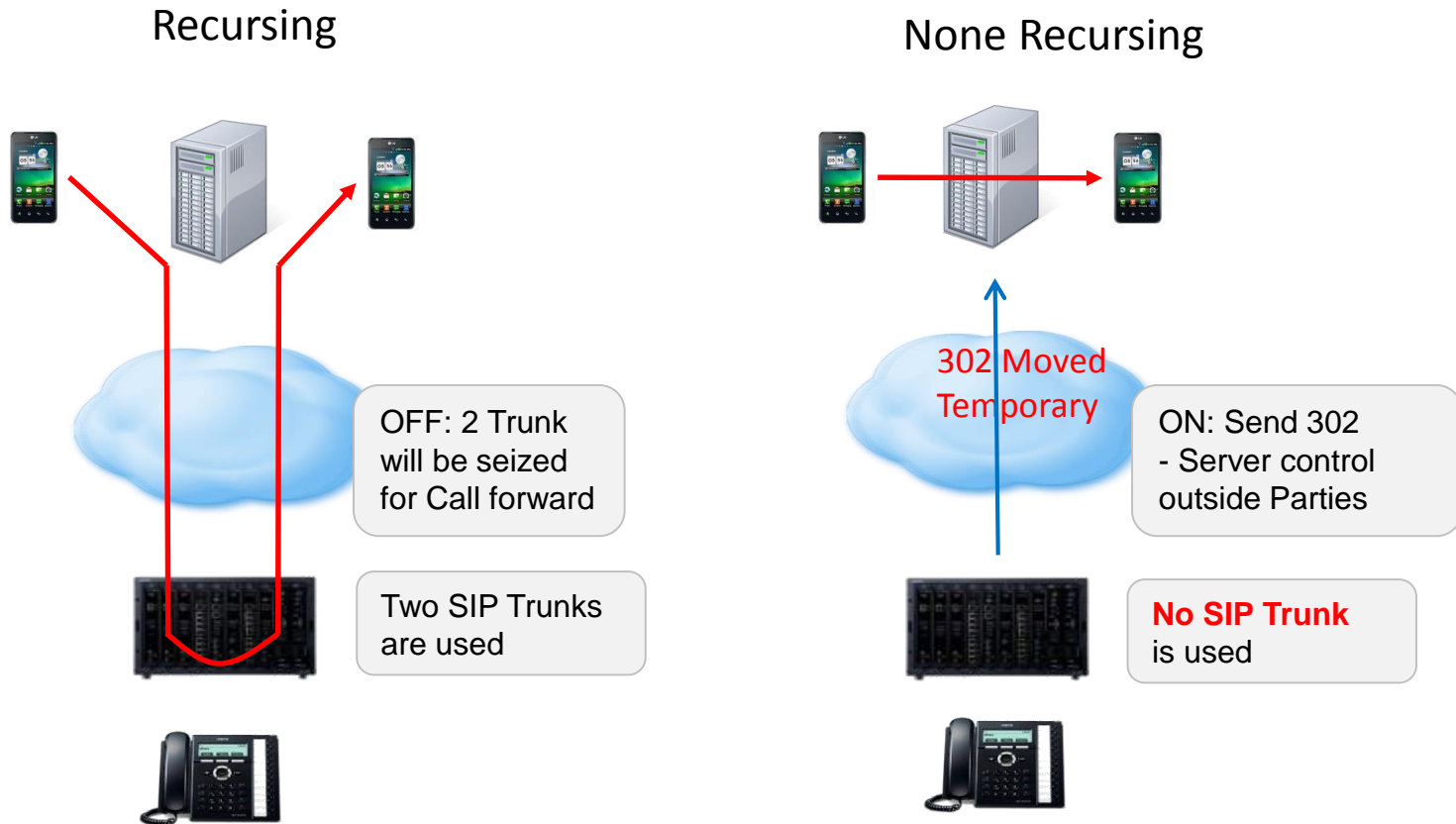
With different PGM133-SIP Trunk Group value, it will be treated as **different SIP Group** even though **PGM133-Proxy Server Address is same.**



Specific settings for SIP Service

PGM133 –Diversion Recursing

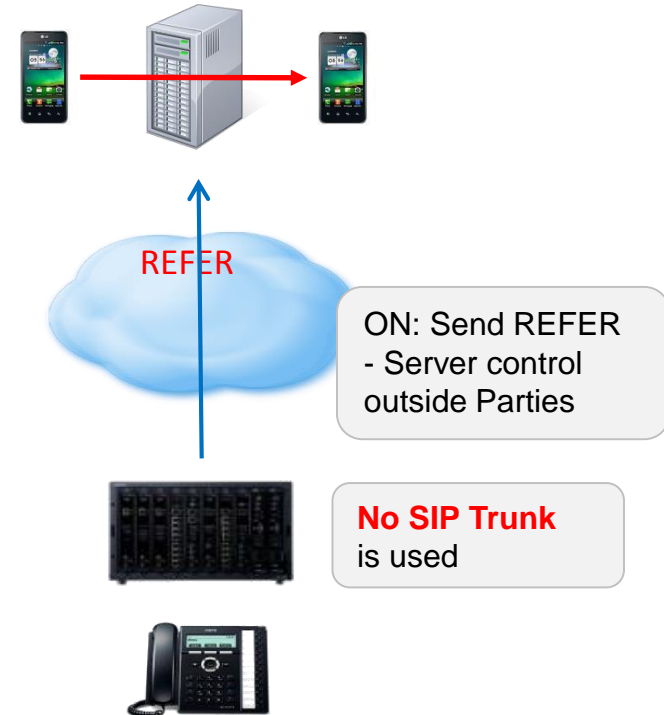
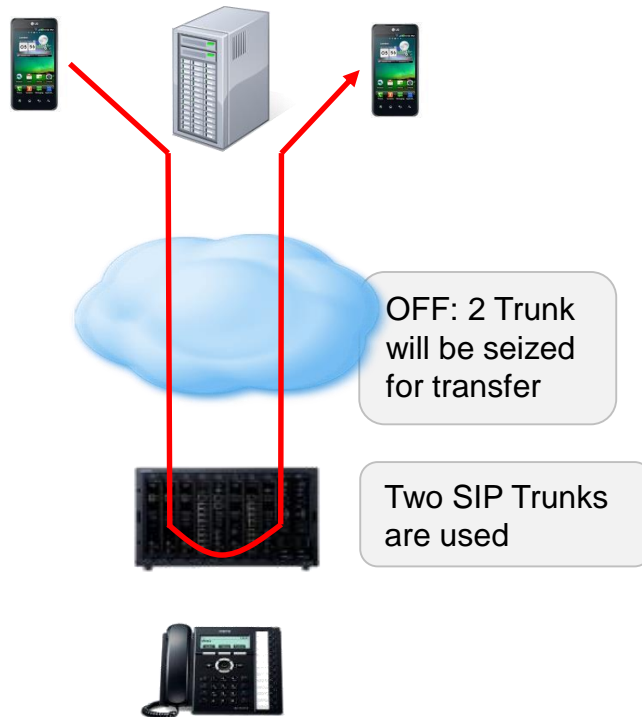
Some Proxy Server supports 302 response for Call Forward.



Specific settings for SIP Service

PGM133 – Send REFER for Transfer

Some Proxy Server supports REFER message for Transfer.
P133, “Send Refer for Transfer”



Specific settings for SIP Service

PGM133 – SIP User ID SELECTION

PGM111- SIP USER TABLE INDEX, SIP USER TABLE INDEX2, SIP USER TABLE INDEX3 are used for different service provider.

SIP USER TABLE INDEX	1
SIP USER TABLE INDEX 2	2
SIP USER TABLE INDEX 3	3

PGM111

SIP User ID Index 1

Order	<input type="checkbox"/> Check All	Attribute	Value	Range
		CID Password	<input type="text"/> <input type="button" value="Go to Setting"/>	
1	<input type="checkbox"/>	Registration User ID	11111111@1.1.1.1	Max 64 Characters

SIP User ID Index 2

Order	<input type="checkbox"/> Check All	Attribute	Value	Range
		CID Password	<input type="text"/> <input type="button" value="Go to Setting"/>	
1	<input type="checkbox"/>	Registration User ID	22222222@2.2.2.2	Max 64 Characters

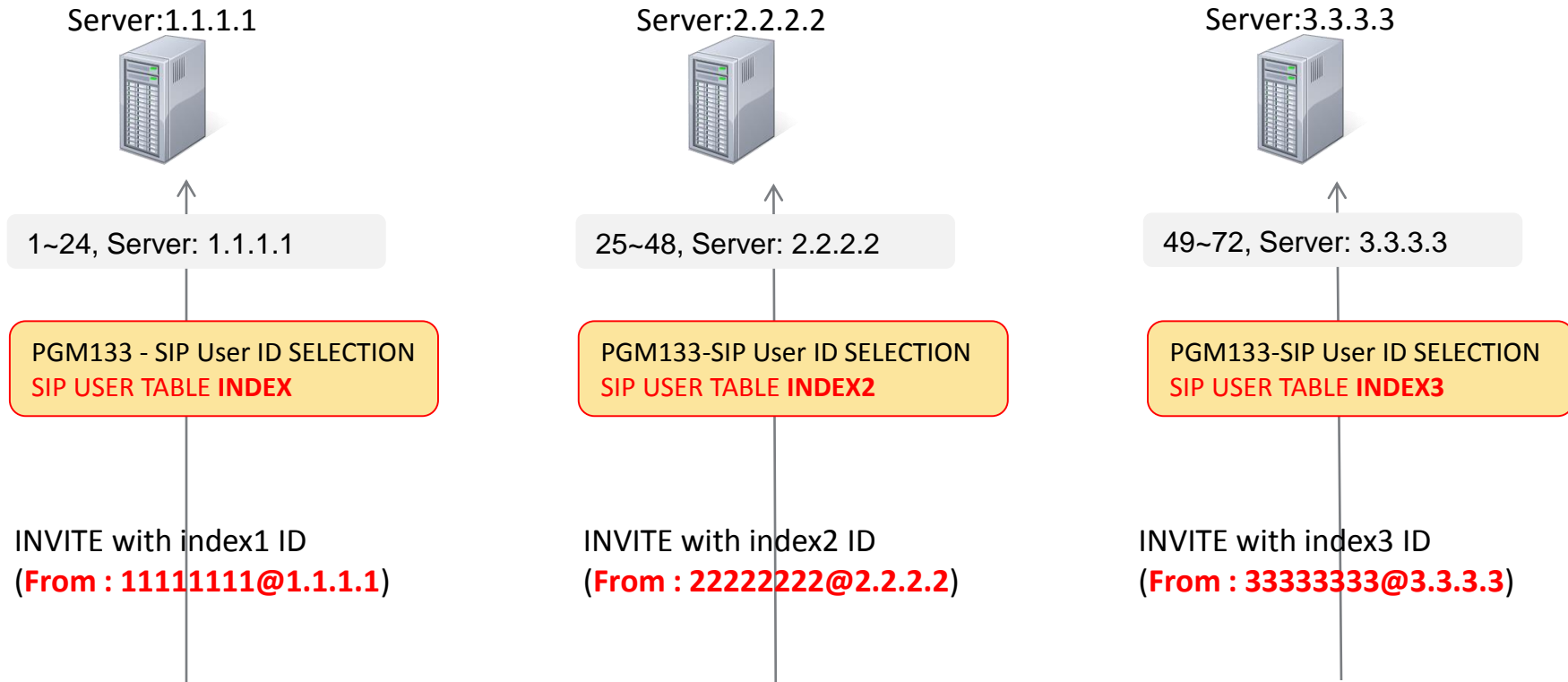
PGM126

SIP User ID Index 3

Order	<input type="checkbox"/> Check All	Attribute	Value	Range
		CID Password	<input type="text"/> <input type="button" value="Go to Setting"/>	
1	<input type="checkbox"/>	Registration User ID	33333333@3.3.3.3	Max 64 Characters

Specific settings for SIP Service

PGM133 – SIP User ID SELECTION



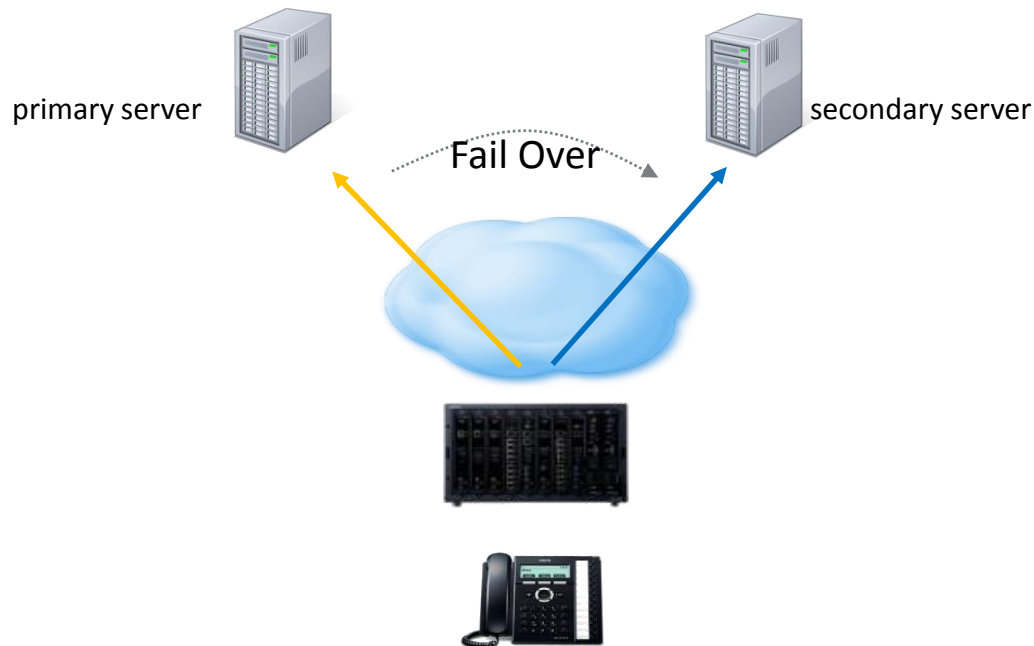
From ID will be selected by SIP User ID SELECTION



Specific settings for SIP Service

PGM133 – Secondary Proxy Server

- PGM133-Soft Switch Type : KT, KT Centrex.
 - Follow specific redundancy flow.
- PGM133-Soft Switch Type : Normal
 - PGM133-Redundancy Usage : Fail Over – If there is no response from Proxy Server.
 - PGM133-Redundancy Usage : Load Balancing – Use Proxy Server and Secondary Proxy Server evenly.



Specific settings for SIP Service

PGM133 – Soft Switch Type(DNS REDUN), PGM210 - DNS SRV Usage

- DNS Service Query is performed to get IP address.
- System sends INVITE to all IP addresses evenly.
- System accepts INVITE from all IP addresses.
- Example : Server Domain : sipconnect.qsc.de

59642320-[Sipm_SipTimerCallBack] DNS Redun, now!!!(Provider:2)

59642320-Sipm_GetDnsQueryResult Start: sipconnect.qsc.de

59642331-Sipm_SipResolverReportDataEv : mode=TRANSPORT BY NAPTR, nextmode=TRANSPORT BY 3WAY SRV, query=sipconnect.qsc.de

59642339-Sipm_SipResolverReportDataEv : mode=TRANSPORT BY 3WAY SRV, nextmode=IP BY HOST, query=sipconnect.qsc.de

59642339-Sipm_SipResolverReportDataEv Srv:1, Host:2, IP:0

59642339-Sipm_SipResolverReportDataEv : srv=_sip._udp.sipconnect.qsc.de (SIP service query)

59642339-Sipm_SipResolverReportDataEv : host=duro01.sipconnect.qsc.de, priority:10, weight:10, port:5060, proto:0

59642346-Sipm_SipResolverReportDataEv IP_BY_HOST: 213.148.136.222(providerIndex:2, NumHost:1)

59642346-Sipm_SipResolverReportDataEv IP_BY_HOST(mode:NAPTR): 1-st Server:213.148.136.222

59642346-Sipm_SipResolverReportDataEv : mode=IP BY HOST, nextmode=IP BY HOST, query=duro01.sipconnect.qsc.de

59642346-Sipm_SipResolverReportDataEv Srv:0, Host:1, IP:0

59642346-Sipm_SipResolverReportDataEv : host=duro02.sipconnect.qsc.de, priority:20, weight:10, port:5060, proto:0

59642354-Sipm_SipResolverReportDataEv IP_BY_HOST: 213.148.136.190(providerIndex:2, NumHost:0)

59642354-Sipm_SipResolverReportDataEv IP_BY_HOST(mode:NAPTR): 0-st Server:213.148.136.190

59642354-Sipm_SipResolverReportDataEv : mode=IP BY HOST, nextmode=Undefined, query=duro02.sipconnect.qsc.de

Protocol	Length	Info
DNS	77	Standard query 0x924f NAPTR sipconnect.qsc.de
DNS	116	Standard query response 0x924f NAPTR sipconnect.qsc.de NAPTR 100 50 s
DNS	87	Standard query 0x9250 SRV _sip._udp.sipconnect.qsc.de
DNS	207	Standard query response 0x9250 SRV _sip._udp.sipconnect.qsc.de SRV 20 10 5060 duro02.sipconnect.qsc.de SRV 10 10 5060 duro01.sipconnect.qsc.de A 213.148.136.190 A 213.148.136.222
DNS	84	Standard query 0x9251 A duro01.sipconnect.qsc.de
DNS	100	Standard query response 0x9251 A duro01.sipconnect.qsc.de A 213.148.136.222
DNS	84	Standard query 0x9252 A duro02.sipconnect.qsc.de
SIP	870	Request: REGISTER sip:150.150.150.95 (1 binding)
SIP	753	Status: 200 OK (1 binding)
DNS	100	Standard query response 0x9252 A duro02.sipconnect.qsc.de A 213.148.136.190

Specific settings for SIP Service

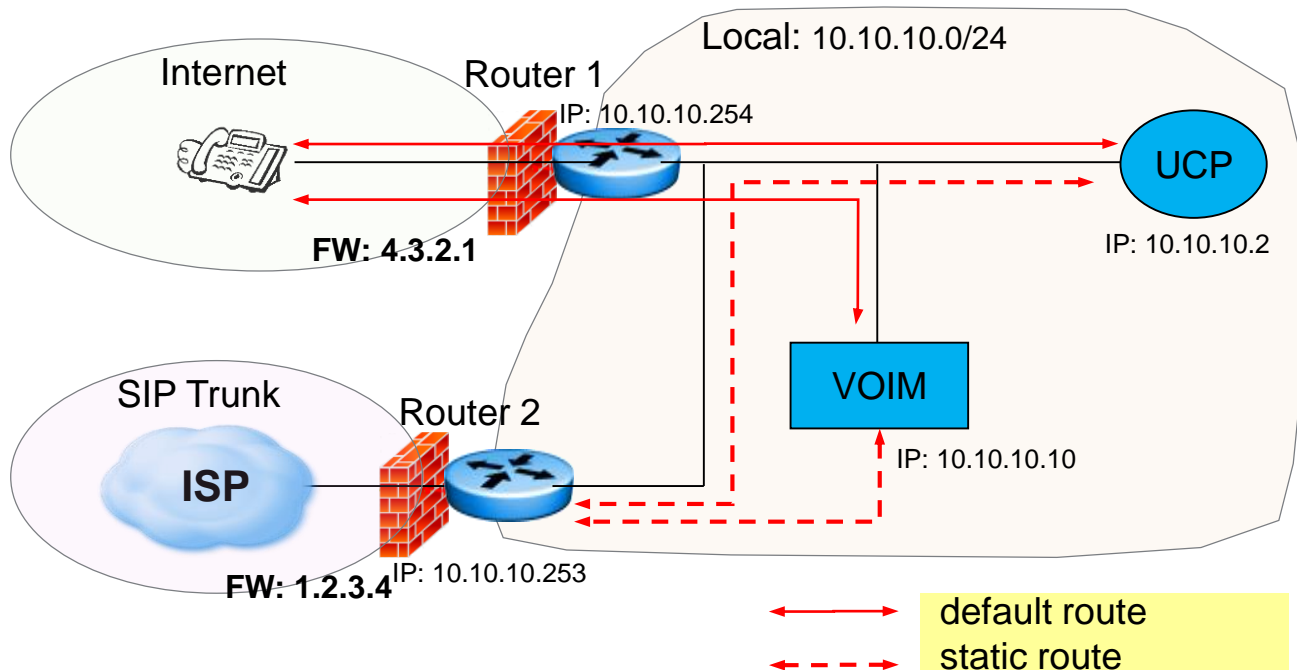
PGM133 – Soft Switch Type(DNS REDUN), PGM210 - DNS SRV Usage

- List of service provider using DNS REDUN.
 - (1)sip-corporate.tele2.se, (2)nexvortex.com, (3)sipconnect.qsc.de
- Set PGM210 – Primary DNS Address.
- Set PGM210 – DNS SRV Usage.
- Set PGM133 – Soft Switch Type : DNS REDUN.
- Set PGM133 – Proxy Server Address : nexvortex.com.
- Set PGM133 – Domain : nexvortex.com.
- Press PGM133 – [REGISTER] button.
- Check IP Addresses resolved in system trace or ethereal trace.

Specific Configurations

Dual Broadband

- SIP trunk is provided by an Internet Service provider (ISP) and its network is divided from default network (internet) by using second router device.



Specific Configurations

Dual Broadband

- The static routes feature can be used to communicate with second broadband network. So you need to know the **network address of SIP and media servers from ISP.**

Hot Desk Attributes(250)

System Call Routing(251)

CO Call Rerouting(252)

VM COS Attributes(253)

Static Route Table(254)

Access Control List(255)

Attendant Ring Mode (257)

System Speed Dial

Index	Feature	Value
1	Net Address	1.2.3.4
	Net Mask	255.255.255.255
	Gateway IP Address	10.10.10.253
2	Net Address	
	Net Mask	
	Gateway IP Address	

Specific Configurations

Dual Broadband

- Static Route, VOIM/VCIM/VVMU/VOIB8-24

The screenshot shows a web-based configuration interface for a network device. On the left is a dark blue sidebar with navigation links: Home, LAN, System (highlighted), System II, Security, Upload, Reset, WANU, LLDP, and Logout. The main content area has a top navigation bar with links: System Settings, DiffServ, DSP, Trace, H.323, SIP, Relay, Static Route (highlighted), ACL, and Fault Log. Below this is the 'Static Routing Table' section, which includes an 'Index' dropdown menu. The 'Static Routing' configuration area contains two input fields: 'Network Address' with the value '1.2.3.4' and 'Gateway' with the value '10.10.10.253'. A red box highlights these two fields. Below the input fields are buttons for 'Save', 'Add' (highlighted with a red box), 'Delete', 'Delete All', and 'Help'. At the bottom, there is a table header with columns for 'No.', 'Network Address', and 'Gateway'.

Specific Configurations

Dual Broadband

- SIP Signaling

- 1) UCP sends SIP packets to **router 2** by static route.

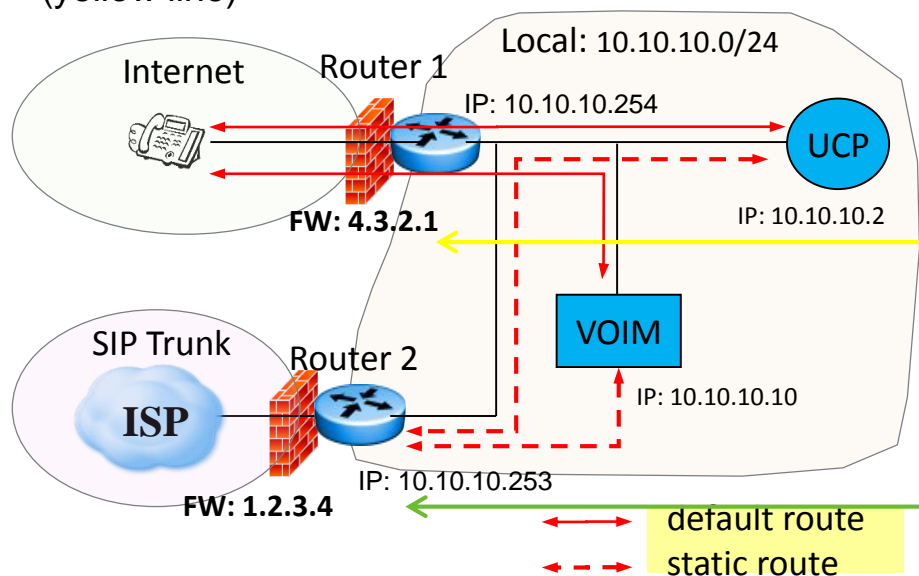
- 2) Configure the firewall IP address of **router 2** and enable 'USE Board IP for SIP' field for the dedicated SIP trunk. (green line)

- Media Exchange Between VOIM and SIP Media Server

- VOIM sends RTP packets to SIP media server by static router.

- Media Exchange Between VOIM and Remote Device

- Configure the firewall IP address of **router 1** at 'RTP Packet Relay Firewall IP Address' field. (yellow line)



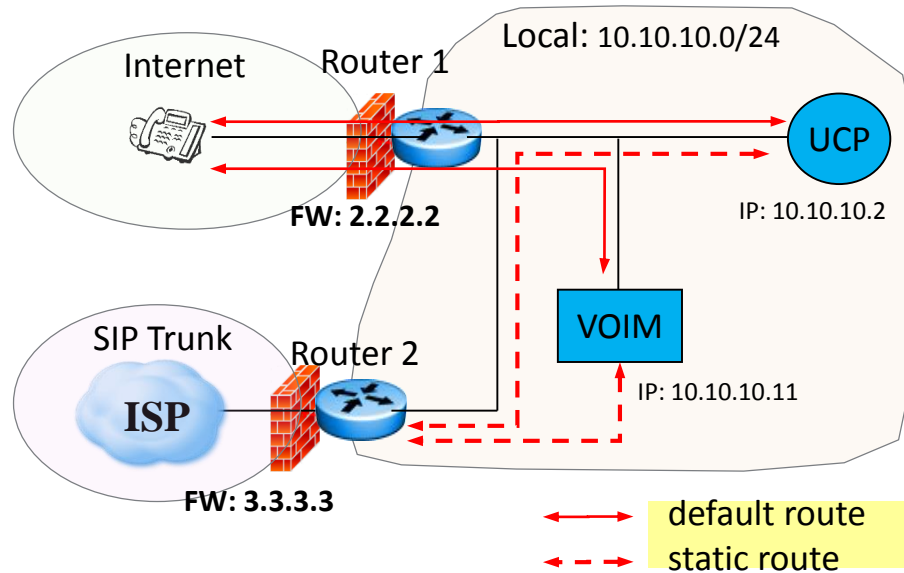
Board Base Attributes(132)	
Attribute	Value
Router IP Address	10.10.10.254
Device Codec Type	System Codec ▼
Firewall IP Address	1.2.3.4
RTP Packet Relay Firewall IP Address	4.3.2.1
RTP Security	▼
T-NET Enable	OFF ▼
T38 Enable	OFF ▼
USE Board IP for SIP	ON ▼

Specific Configurations

Example

- Configure Dual Broadband case.

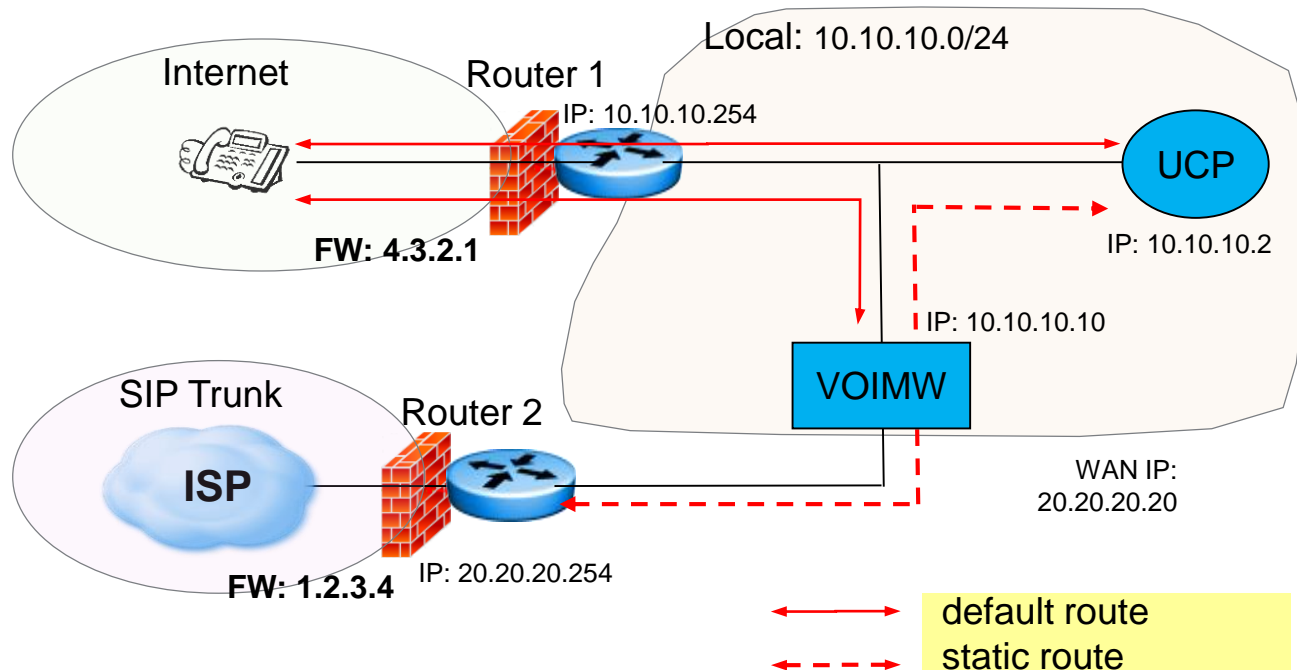
Condition:



Specific Configurations

Dual Broadband with WANU

- Router1 and Router2 are separated network.
- WANU must control SIP and RTP packets to relay Local and WAN network.
 - Refer WANU_Overview.ppt for WANU setting.



Specific Configurations

Dual Broadband with WANU

- Router IP address of UCP will be WANU local IP.

UCP IP Address	10.10.10.2
UCP MAC Address	B40EDC281BCA
UCP Subnet Mask	255.255.255.0
Router IP Address	10.10.10.10

Specific Configurations

Dual Broadband with WANU

- Router IP address of VOIM will be WANU local IP.
- Set Firewall IP and Relay Firewall IP.
- Set USE Board IP for SIP

1	<input type="checkbox"/>	Router IP Address	<input type="text" value="10.10.10.10"/>	IP Address
2	<input type="checkbox"/>	Device Codec Type	<input type="text" value="System Codec"/>	
3	<input type="checkbox"/>	Firewall IP Address	<input type="text" value="1.2.3.4"/>	IP Address
4	<input type="checkbox"/>	RTP Packet Relay Firewall IP Address	<input type="text" value="4.3.2.1"/>	IP Address
5	<input type="checkbox"/>	RTP Security	<input type="text" value="ON"/>	
6	<input type="checkbox"/>	T-NET Enable	<input type="text" value="OFF"/>	
7	<input type="checkbox"/>	T38 Enable	<input type="text" value="OFF"/>	
8	<input type="checkbox"/>	USE Board IP for SIP	<input type="text" value="ON"/>	

Specific Configurations

Dual Broadband with WANU

- Wanu Config

Config WAN MFIM VOIP VSF PC Port

WANU Configuration

Run 'Update' submenu to apply changed configurations.

WANU Control
WANU : Enabled Disabled

Accepting WAN Side Access
ICMP (ping) : On Off
Web : On Off
Maintenance : On Off

Accepting LAN Side Access
ICMP (ping) : On off

Voim Function
VOIM Gateway : Enabled Disabled

ALG Function
SIP : On Off
SIP Seize : First Round
SIP poll : seconds, (default 10)
H.323 : On Off
H.323 Seize : First Round
H.323 GK : On Off
H.323 poll : seconds, (default 10)

Save WANU Settings

WAN Network Configuration

Run 'Update' submenu to apply changed configurations.

WAN Configuration

IP Address:
Netmask:
Gateway:
Firewall: x

Save WAN Settings

MFIM Configuration

Run 'Update' submenu to apply changed configurations.

MFIM Configuration

IP Address: x
[Public Port] [Private Port]

IPKTS Protocol Port (default UDP 5588):	<input type="text" value="5588"/>	<input type="text" value="5588"/>
IPKTS Protocol Port (default TCP 5588):	<input type="text" value="5588"/>	<input type="text" value="5588"/>
TCP maintenance Port (default TCP 5003):	<input type="text" value="5003"/>	<input type="text" value="5003"/>
Web Server Port (default TCP 80):	<input type="text" value="80"/>	<input type="text" value="80"/>
Web TLS Server Port (default TCP 443):	<input type="text" value="443"/>	<input type="text" value="443"/>
TFTP Server Port (default UDP 69):	<input type="text" value="69"/>	<input type="text" value="69"/>
FTP Control Server Port (default TCP 21):	<input type="text" value="21"/>	<input type="text" value="21"/>
FTP Data Server Port (default TCP 9700..9709):	<input type="text" value="9700"/> - <input type="text" value="9709"/>	<input type="text" value="9700"/>
SMTP Server Port (default TCP 25):	<input type="text" value="25"/>	<input type="text" value="25"/>
SNMP Server Port (default UDP 161):	<input type="text" value="161"/>	<input type="text" value="161"/>
NTP Server Port (default UDP 123):	<input type="text" value="123"/>	<input type="text" value="123"/>
UC Web Port (default TCP 8899):	<input type="text" value="8899"/>	<input type="text" value="8899"/>
UC XML Port (default TCP 7878):	<input type="text" value="7878"/>	<input type="text" value="7878"/>
UCP Telnet Port (default TCP 23):	<input type="text" value="23"/>	<input type="text" value="23"/>
UCP SSH Port (default TCP 22):	<input type="text" value="22"/>	<input type="text" value="22"/>
UCP Rsync Port (default TCP/UDP 873):	<input type="text" value="873"/>	<input type="text" value="873"/>

Save MFIM Settings

Specific Configurations

Dual Broadband with WANU

- WANU Configuration

Index	0	1
VOIP IP Address	10.10.10.2	10.10.10.10
SIP Proxy Address	255.255.255.255	255.255.255.255
SIP Proxy(2nd. or Subnet)	255.255.255.255	255.255.255.255
SIP Server Port	5060	0
SIP Server TCP Port	5060	0
SIP Server TLS Port	5061	0
H.323 Proxy IP Address	255.255.255.255	255.255.255.255
H.323 Server Port	0	1720
Q.931 Signal Ports	0 - 0	2048 - 2559
H.245 Signal Ports	0 - 0	2560 - 3071
RAS Signal Port	0 - 0	2048 - 3071
Remote Maint Port	5003	5004
Web Server Port	80	8000
Data Sharing Ports	0 - 0	0 - 0
RTP P1 Ports	0 - 0	10000 - 10083
RTP P2 Ports	0 - 0	10084 - 10167
RTP Signal Ports	0 - 0	10168 - 10251
RTP T.38 Ports	0 - 0	10252 - 10335
ALG Total Channels	120	24
ALG Max SIP Channels	120	0
ALG Max H.323 Channels	0	24
ALG SIP Order	-1	-1
ALG SIP Call Direction	TxRx	TxRx
ALG H.323 order	-1	-1
ALG H.323 Call Direction	TxRx	TxRx
ALG H.323 Gatekeeper	No	No

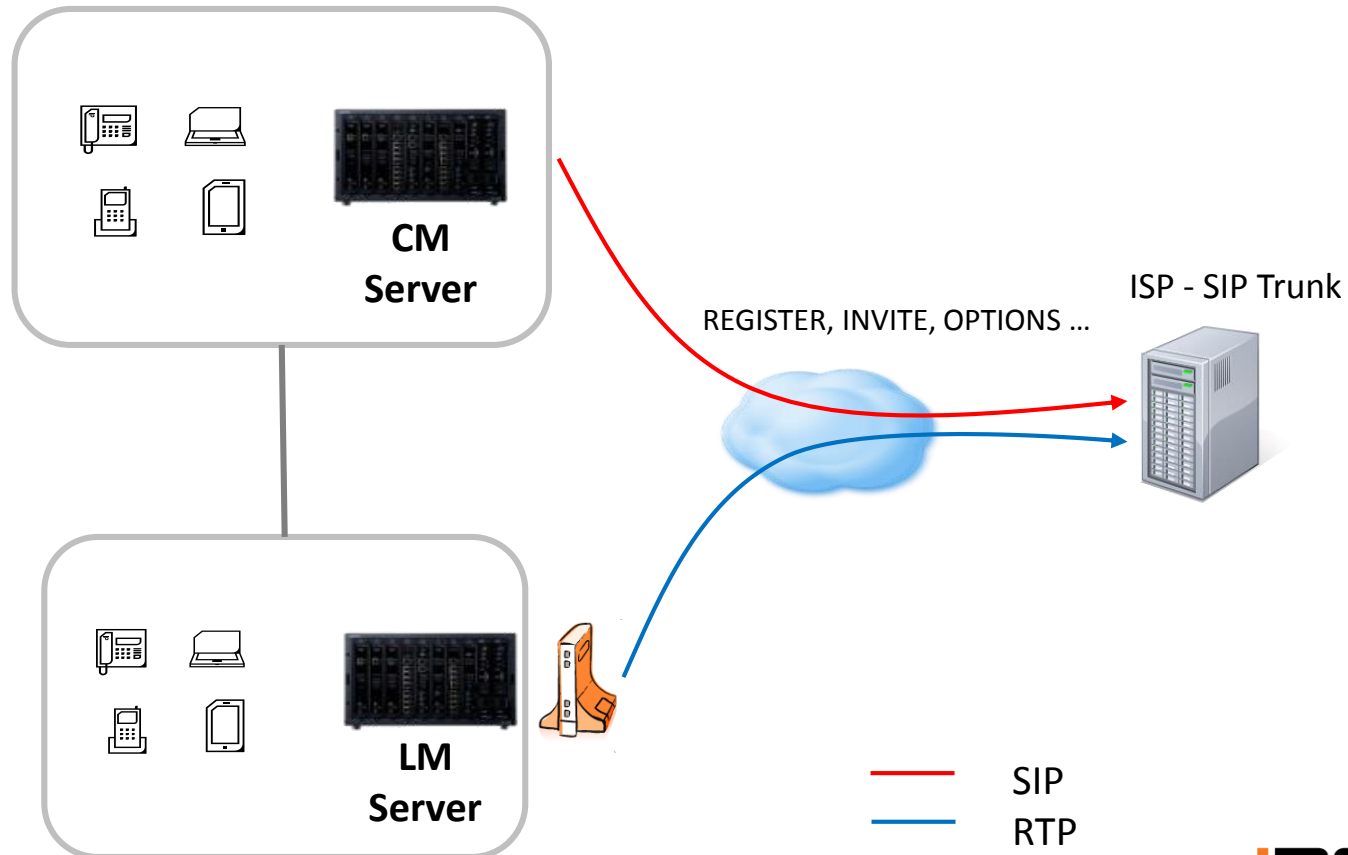
SIP ports are forwarded to UCP. All SIP related ports are assigned to UCP to forward SIP packets to UCP.

RTP packets should be handled in VOIP gateway.

Specific Configurations

SIP Service in T-NET

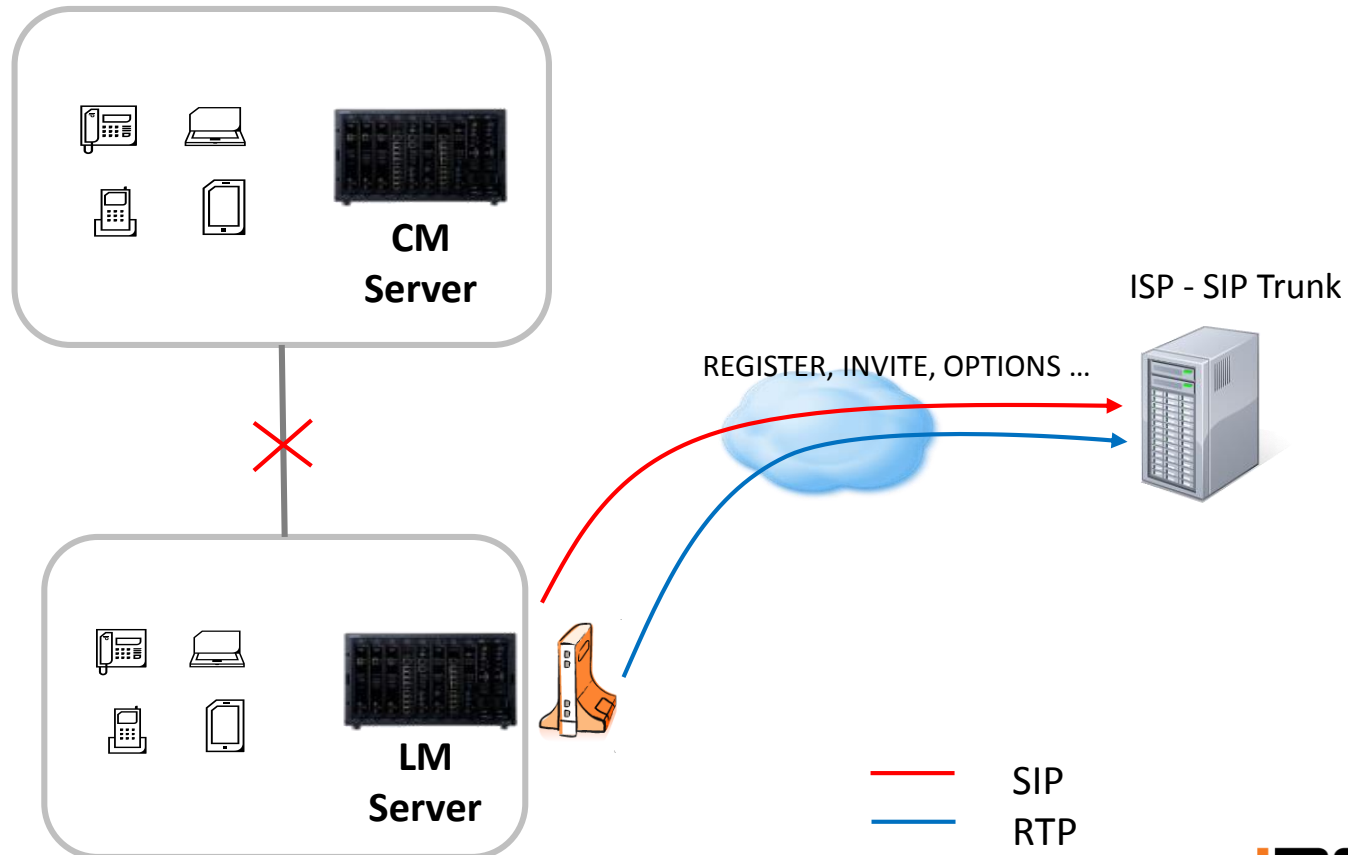
- SIP Trunk is used in T-NET configuration.
 - LM VOIU/VOIM is controlled by CM Server.
 - After connection, REGISTER will be sent to server by **CM server**.



Specific Configurations

SIP Service in T-NET

- SIP Trunk is used in T-NET configuration.
 - LM VOIU/VOIM is controlled by LM Server by disconnection.
 - After disconnection, REGISTER will be sent to server by **LM server**.



Specific Configurations

SIP Service in T-NET

- Configure T-NET case



Trace related with SIP

SIP Raw Level Trace

- Print SIP message before sending to network and receiving from network.
 - Trace is same as Wireshark trace.
 - Mon>t s fsipm → Full SIP Message

```
[ 25/09/17 10:07:51 ]+++++  
Sent 1085 Bytes to 150.150.131.207:5060 by UDP (SendEv)  
-----  
INVITE sip:1000@150.150.131.207 SIP/2.0  
From: "1018"<sip:1018@150.150.150.95>;tag=4e705130-5f969696-13c4-65014-257cf-333808ef-257cf  
To: <sip:1000@150.150.131.207>  
Call-ID: 4e93ccb8-5f969696-13c4-65014-257cf-3073a139-257cf  
CSeq: 1 INVITE  
Via: SIP/2.0/UDP 150.150.150.95:5060;rport;branch=z9hG4bK-257cf-9270161-3b35064c-4e413638  
Max-Forwards: 70  
Allow: INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,REFER,SUBSCRIBE,NOTIFY,MESSAGE,INFO,PRACK,UPDATE  
Supported: replaces,UPDATE,INFO  
P-Asserted-Identity: "1018"<sip:1018@150.150.150.95>  
Remote-Party-ID: <sip:1018@150.150.150.95>; party=calling; privacy=off; screen=yes  
Privacy: none  
User-Agent: Ericsson-LG Enterprise iPECS-UCP  
Contact: "1018"<sip:1018@150.150.150.95>;transport=UDP  
Content-Type: application/sdp  
Content-Length: 277
```

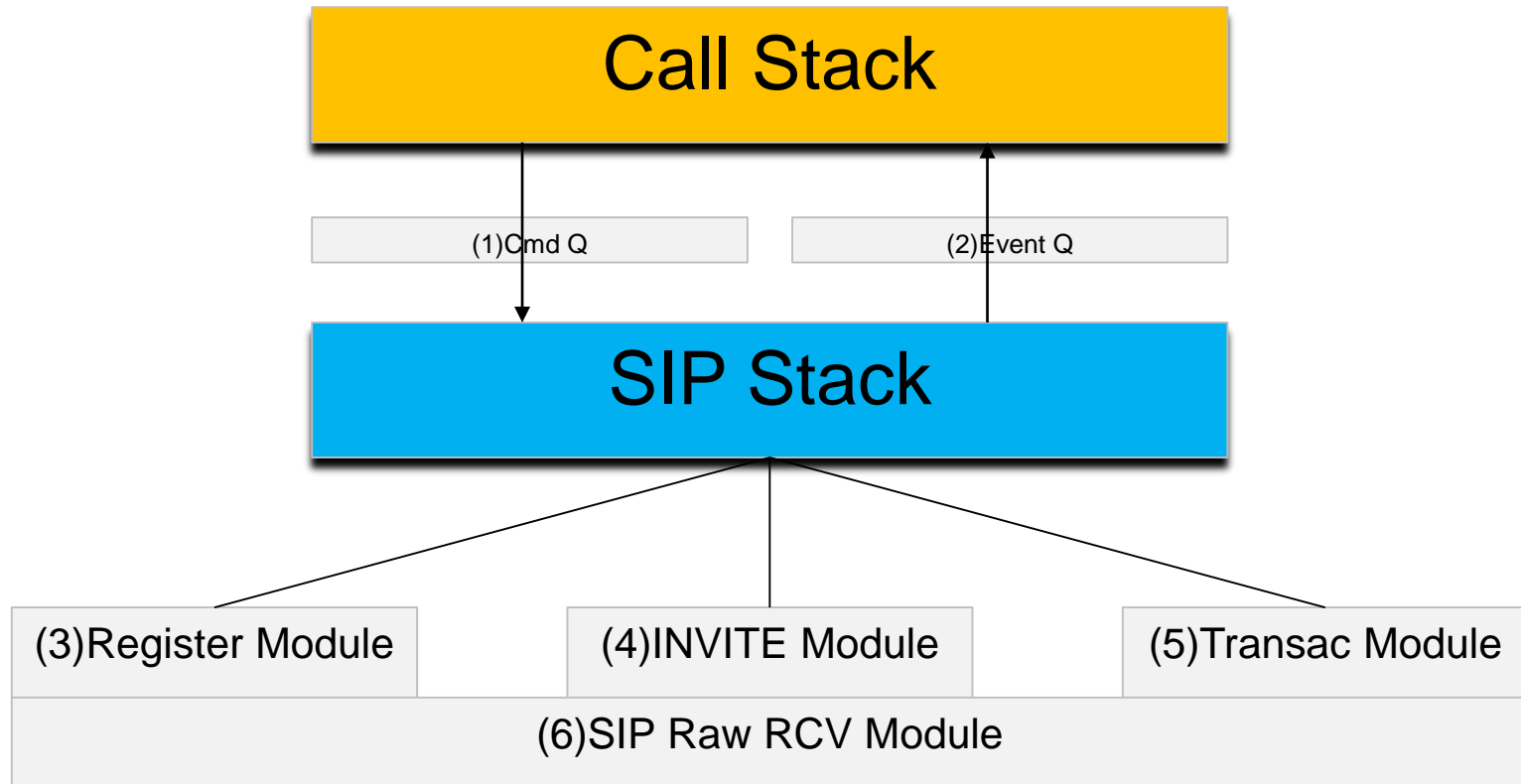
Time	Source	Destination	Protocol	Length	Info
298 2017-09-25 10:16:36.257898	150.150.150.95	150.150.131.207	SIP/SDP	1127	Request: INVITE sip:1000@150.150.131.207
315 2017-09-25 10:16:36.371366	150.150.131.207	150.150.150.95	SIP	571	Status: 100 Trying
348 2017-09-25 10:16:36.472330	150.150.131.207	150.150.150.95	SIP/SDP	914	Status: 183 Session Progress

```
User Datagram Protocol, Src Port: 5060, Dst Port: 5060  
Session Initiation Protocol (INVITE)  
> Request-Line: INVITE sip:1000@150.150.131.207 SIP/2.0  
v Message Header  
> From: "1018"<sip:1018@150.150.150.95>;tag=4e705130-5f969696-13c4-65014-257cf-333808ef-257cf  
> To: <sip:1000@150.150.131.207>  
Call-ID: 4e93ccb8-5f969696-13c4-65014-257cf-3073a139-257cf  
> CSeq: 1 INVITE  
> Via: SIP/2.0/UDP 150.150.150.95:5060;rport;branch=z9hG4bK-257cf-9270161-3b35064c-4e413638  
Max-Forwards: 70  
Allow: INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,REFER,SUBSCRIBE,NOTIFY,MESSAGE,INFO,PRACK,UPDATE  
Supported: replaces,UPDATE,INFO  
> P-Asserted-Identity: "1018"<sip:1018@150.150.150.95>  
> Remote-Party-ID: <sip:1018@150.150.150.95>; party=calling; privacy=off; screen=yes  
Privacy: none  
User-Agent: Ericsson-LG Enterprise iPECS-UCP UCP100 2.2.24  
> Contact: "1018"<sip:1018@150.150.150.95:5060;transport=UDP>  
Content-Type: application/sdp  
Content-Length: 277
```

Same Call-ID

Trace related with SIP

Structure of SIP Call Flow



Mon>t s sip → Trace related with SIP handling

Trace related with SIP

Structure of SIP Call Flow

- (1) Command from Call Stack
 - [Sipm_CallMsgHandler] CALL ---> SIPM (.....)

- (2) Event to Call Stack
 - <SIPM Msg> : MSG(SIP_INVITE_MSG) :Request
 - <SIPM Msg> : MSG(SIP_RESPONSE_MSG) :Response

- (3) Register Call Back Function
 - [Sipm_SipEvRegState] : Status control
 - [Sipm_SipEvRegMsgReceive] : Hooking for receiving
 - [Sipm_SipEvRegMsgSend] : Hooking for sending

- (4) Invite Call Back Function
 - [Sipm_SipEvCallState] : Status control
 - [Sipm_SipEvCallMsgReceive] : Hooking for receiving
 - [Sipm_SipEvCallMsgSend] : Hooking for sending

- (5) Transc Call Back Function
 - [Sipm_SipEvTransState] : Status control
 - [Sipm_SipEvTransMsgReceive] : Hooking for receiving
 - [Sipm_SipEvTransMsgSend] : Hooking for sending

- (6) Raw level Hooking for receiving
 - [Sipm_SipTransportMsgReceivedExt] : Hooking Used for PGM210-IP Auth

Trace related with SIP

Structure of SIP Call Flow – Outgoing INVITE

- Check [CallIdx:8] for one call.

```
(1) [CallIdx:8][Sipm_CallMsgHandler] CALL ----> SIPM (INVITE)
(2) [CallIdx:8][Sipm_SipEvCallMsgSend] SIPM ----> INVITE
    → INVITE
(3) [CallIdx:8][Sipm_SipEvCallState(s:01)] OUTGOING - Inviting(reason:LOCAL_INVITING)
    ←100_TRYING
(4) [CallIdx:8][Sipm_SipEvCallMsgReceive] SIPM <--- 100(Method:INVITE(0), reason:Trying)
(5) <SIPM Msg> : MSG(SIP_RESPONSE_MSG), Evt type(SIPTRUNK) type(SIPTRUNK) GWnum(13) phy_num(7)
(6) [CallIdx:8][Sipm_SipEvCallState(s:13)] OUTGOING - Proceeding(reason:NOTHING_ALL)
    ←407_AUTHEN
(7) [CallIdx:8][Sipm_SipEvCallMsgReceive] SIPM <--- 407(Method:INVITE(0), reason:Proxy Authentication Required)
(8) [CallIdx:8][Sipm_SipEvCallMsgSend] SIPM ----> ACK
    → ACK
(9) [CallIdx:8][Sipm_SipEvCallState(s:03)] OUTGOING - Unauthenticated(reason:AUTH_NEEDED)
(10) [CallIdx:8][Sipm_SipEvCallMsgSend] SIPM ----> INVITE
    → INVITE
(11) [CallIdx:8][Sipm_SipEvCallState(s:01)] OUTGOING - Inviting(reason:LOCAL_INVITING)
    ←100_TRYING
(12) [CallIdx:8][Sipm_SipEvCallMsgReceive] SIPM <--- 100(Method:INVITE(0), reason:Trying)
(13) [CallIdx:8][Sipm_SipEvCallState(s:13)] OUTGOING - Proceeding(reason:NOTHING_ALL)
(14) <SIPM Msg> : MSG(SIP_RESPONSE_MSG), Evt type(SIPTRUNK) type(SIPTRUNK) GWnum(13) phy_num(7)
    ←180_RINGING
(15) [CallIdx:8][Sipm_SipEvCallMsgReceive] SIPM <--- 180(Method:INVITE(0), reason:Ringing)
(16) <SIPM Msg> : MSG(SIP_RESPONSE_MSG), Evt type(SIPTRUNK) type(SIPTRUNK) GWnum(13) phy_num(7)
    ← 200_OK
(17) [CallIdx:8][Sipm_SipEvCallMsgReceive] SIPM <--- 200(Method:INVITE(0), reason:OK)
(18) [CallIdx:8][Sipm_SipEvCallState(s:10)] OUTGOING - RemoteAccepted(reason:REMOTE_ACCEPTED)
(19) [CallIdx:8][Sipm_SipEvCallMsgSend] SIPM ----> ACK
    → ACK
(20) [CallIdx:8][Sipm_SipEvCallState(s:06)] OUTGOING - Connected(reason:ACK_SENT)
(21) <SIPM Msg> : MSG(SIP_RESPONSE_MSG), Evt type(SIPTRUNK) type(SIPTRUNK) GWnum(13) phy_num(7)
```

Trace related with SIP

Structure of SIP Call Flow – Outgoing INVITE Failure

- Check [CallIdx:11] for one call.

```
(1) [CallIdx:11][Sipm_CallMsgHandler] CALL ---> SIPM (INVITE)
(2) [CallIdx:11][Sipm_SipEvCallMsgSend] SIPM ---> INVITE
    → INVITE
(3) [CallIdx:11][Sipm_SipEvCallState(s:01)] OUTGOING - Inviting(reason:LOCAL_INVITING)
    ←100_TRYING
(4) [CallIdx:11][Sipm_SipEvCallMsgReceive] SIPM <--- 100(Method:INVITE(0), reason:Trying)
(5) <SIPM Msg> : MSG(SIP_RESPONSE_MSG), Evt type(SIPTRUNK) type(SIPTRUNK) GWnum(13) phy_num(11)
(6) [CallIdx:11][Sipm_SipEvCallState(s:13)] OUTGOING - Proceeding(reason:NOTHING_ALL)
    ←407_AUTHEN
(7) [CallIdx:11][Sipm_SipEvCallMsgReceive] SIPM <--- 407(Method:INVITE(0), reason:Proxy Authentication Required)
(8) [CallIdx:11][Sipm_SipEvCallMsgSend] SIPM ---> ACK
    → ACK
(9) [CallIdx:11][Sipm_SipEvCallState(s:03)] OUTGOING - Unauthenticated(reason:AUTH_NEEDED)
(10) [CallIdx:11][Sipm_SipEvCallMsgSend] SIPM ---> INVITE
    → INVITE
(11) [CallIdx:11][Sipm_SipEvCallState(s:01)] OUTGOING - Inviting(reason:LOCAL_INVITING)
    ←100_TRYING
(12) [CallIdx:11][Sipm_SipEvCallMsgReceive] SIPM <--- 100(Method:INVITE(0), reason:Trying)
(13) [CallIdx:11][Sipm_SipEvCallState(s:13)] OUTGOING - Proceeding(reason:NOTHING_ALL)
(14) <SIPM Msg> : MSG(SIP_RESPONSE_MSG), Evt type(SIPTRUNK) type(SIPTRUNK) GWnum(13) phy_num(11)
    ←404_NOT_FOUND
(15) [CallIdx:11][Sipm_SipEvCallMsgReceive] SIPM <--- 404(Method:INVITE(0), reason:Not Found)
(16) [CallIdx:11][Sipm_SipEvCallMsgSend] SIPM ---> ACK
    →ACK
(17) <SIPM Msg> : MSG(SIP_RESPONSE_MSG), Evt type(SIPTRUNK) type(SIPTRUNK) GWnum(13) phy_num(11)
(18) [CallIdx:11][Sipm_SipEvCallState(s:07)] OUTGOING - Disconnected(reason:REQUEST_FAILURE)
(19) [[CallIdx:11][Sipm_SipEvCallState(s:09)] OUTGOING - Terminated(reason:CALL_TERMINATED)
(20) <SIPM Msg> : MSG(SIP_BYE_MSG), Evt type(SIPTRUNK) type(SIPTRUNK) GWnum(13) phy_num(11)
(21) <SIPM Msg> : MSG(SIP_RLS_OK_MSG), Evt type(SIPTRUNK) type(SIPTRUNK) GWnum(13) phy_num(11)
```

Trace related with SIP

Structure of SIP Call Flow – REGISTER

- Check [RegIdx:1] for one register.
 - (1) [RegIdx:1][Sipm_SipEvRegMsgSend] SIPM ---> REGISTER
→ REGISTER
 - (2) [RegIdx:1][Sipm_SipEvRegState] (s:2) - Registering (reason:USER_REQUEST)
← 401 Unauth
 - (3) [RegIdx:1][Sipm_SipEvRegMsgReceive] SIPM <--- 401(Unauthorized)
 - (4) [RegIdx:1][Sipm_SipEvRegState] (s:4) - Unauthenticated (reason:RESPONSE_UNAUTHENTICATED_RECVD)
 - (5) [RegIdx:1][Sipm_SipEvRegMsgSend] SIPM ---> REGISTER
→ REGISTER
 - (6) [RegIdx:1][Sipm_SipEvRegState] (s:2) - Registering (reason:USER_REQUEST)
← 200 OK
 - (7) [RegIdx:1][Sipm_SipEvRegMsgReceive] SIPM <--- 200(OK)
 - (8) [RegIdx:1][Sipm_SipEvRegState] (s:5) - Registered (reason:RESPONSE_SUCCESSFUL_RECVD)

Trace related with SIP

Get the trace for each cases.

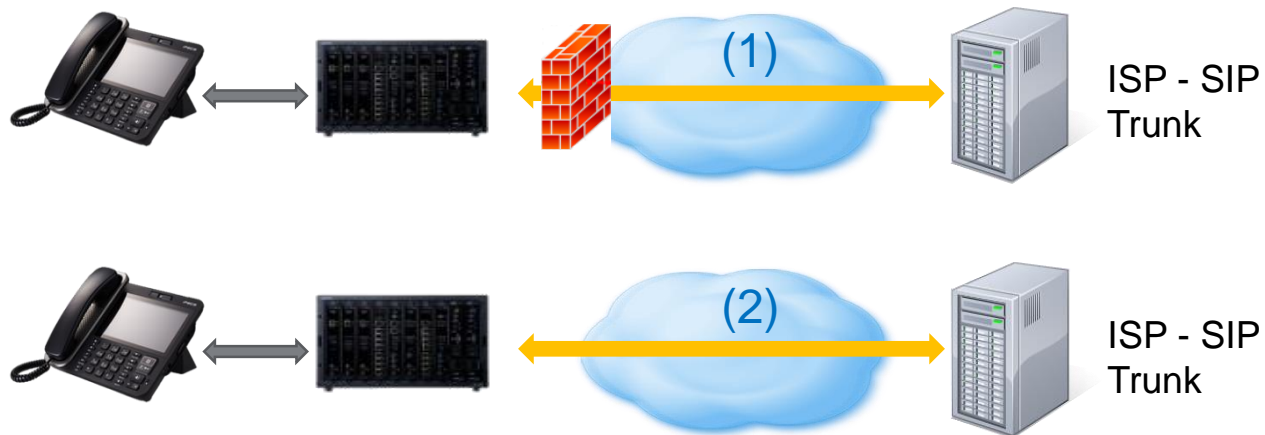
- Send SIP REGISTER and get the trace.
- Send SIP INVITE and get the trace.



Trouble Shooting

Network Configuration

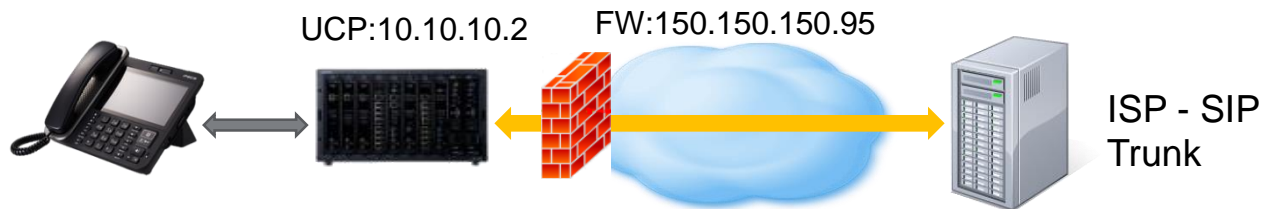
- Check if F/W device is installed and where F/W device is located.
 - (1) Server is located outside of F/W - F/W IP must be used or SIP ALG feature is needed in F/W .
 - (2) Server is located inside of F/W or connected using public IP – Local IP must be used.



Trouble Shooting

Network Configuration

- Server is located outside of F/W
 - Is SIP ALG feature used in F/W device?
 - If SIP ALG feature malfunctions, mute problem happens.
 - SIP ALG must change ip address in SIP message – contact, via, media ip address.
 - SIP ALG must relay SIP message and RTP packets.

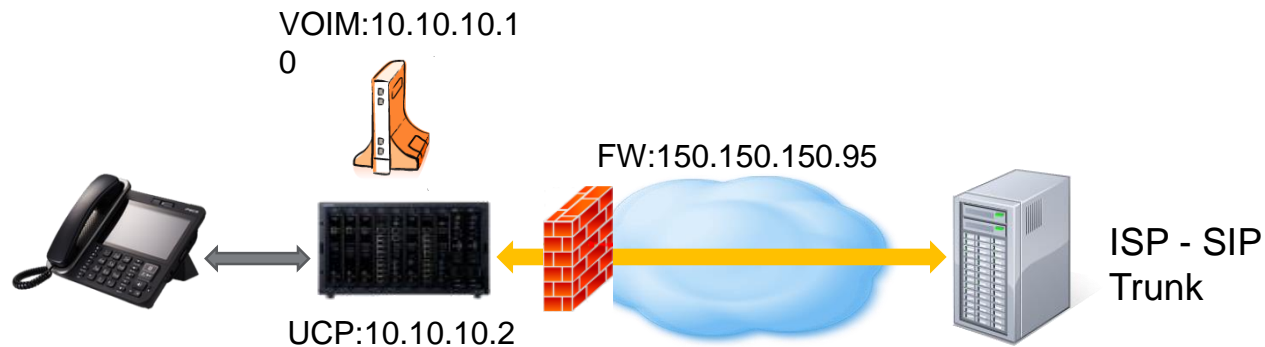


SIP ALG has own Table to forward packet and change IP address

Trouble Shooting

Network Configuration

- Server is located outside of F/W
 - Is it possible to disable SIP ALG feature?
 - Some F/W can not disable SIP ALG feature.
 - Is it possible to set Port Forwarding?





Local IP:port	Public IP:port	
10.10.10.2:5060	150.150.150.95:5060	SIP signaling port
10.10.10.10:10000 ~10048	150.150.150.95:10000 ~10048	RTP port

Port forwarding Table

Trouble Shooting

SIP Registration Log

- Maintenance – Trace – SIP RegUnreg Log View.
 - SENDFAIL(6) : No Response from server for REGISTER message.
 - Check network problem or server side.
 - FAIL(6) – 404 : “404 Not Found” from server for REGISTER message.
 - Check ID and password with server.
 - REG(5) : “200OK” from server for REGISTER message.
 - System is successfully registered.


System Information	SIP RegUnreg Log View  
SIP RegUnreg Log	
18 Sep 2017 11:22:10 IP:150.150.131.207 ID:1018@150.150.150.95 SENDFAIL(6)	
18 Sep 2017 11:22:36 IP:150.150.131.207 ID:1018@150.150.150.95 SENDFAIL(6)	
18 Sep 2017 11:26:14 IP:sipconnect.qsc.de ID:1018@150.150.150.95 SENDFAIL(6)	
22 Sep 2017 16:09:03 IP:150.150.131.207 ID:12345678@150.150.150.95 FAIL(6)-404	
22 Sep 2017 16:14:44 IP:150.150.131.207 ID:1018@150.150.150.95 REG(5)	

Trouble Shooting

SIP Registration Log

- If you see the below case, we can say that network or server is unstable in that time.
 - SENDFAIL(7)
 - FAIL(7)
 - REG(5)
 - ...

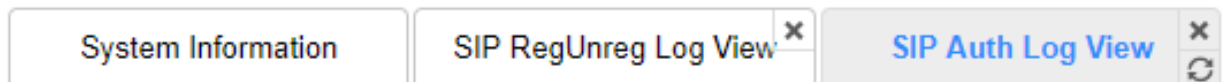
- 1: TERMINATED
- 2: REGISTERING
- 3: REDIRECTED
- 4: UNAUTHENTICATED
- 5: REGISTERED
- 6: FAILED
- 7: SEND_FAILURE

System Information		SIP RegUnreg Log View 		
SIP RegUnreg Log				
22 Sep 2017 16:25:27	IP:150.150.131.207	ID:1018@150.150.150.95	REG(5)	Reg
22 Sep 2017 16:26:26	IP:150.150.131.207	ID:1018@150.150.150.95	SENDFAIL(7)	Reg Fail
22 Sep 2017 16:26:26	IP:150.150.131.207	ID:1018@150.150.150.95	FAIL(7)	Reg
22 Sep 2017 16:27:17	IP:150.150.131.207	ID:1018@150.150.150.95	REG(5)	Reg
22 Sep 2017 16:28:16	IP:150.150.131.207	ID:1018@150.150.150.95	SENDFAIL(7)	Reg Fail
22 Sep 2017 16:28:16	IP:150.150.131.207	ID:1018@150.150.150.95	FAIL(7)	Reg
22 Sep 2017 16:29:07	IP:150.150.131.207	ID:1018@150.150.150.95	REG(5)	Reg

Trouble Shooting

SIP Authentication Log

- Maintenance – Trace – SIP Auth Log View.
 - Data Time IP ID SIP_Method
 - 22 Sep 2017 17:20:00 IP:66.23.129.253 ID:0709235149 INVITE
 - 22 Sep 2017 17:21:16 IP:103.26.173.4 ID:0734560650 INVITE
- If there are many logs from unknown Ips, consider **hacking trial**.
 - For more information for security, refer security session.



System Information SIP RegUnreg Log View SIP Auth Log View

SIP Auth Log	
22 Sep 2017 17:19:09 IP:150.150.131.146 ID:	INVITE
22 Sep 2017 17:20:00 IP:66.23.129.253 ID:0709235149	INVITE
22 Sep 2017 17:21:16 IP:103.26.173.4 ID:0734560650	INVITE

Trouble Shooting

SIP call disconnection after 30 seconds

- SIP stack will disconnect incoming call if final ACK is not received.
 - This kind of problem is related with Contact IP address in 200OK contact header.
- Check which ip address must be used in your configuration.

Contact Header Rule

	PGM132-USE Board IP for SIP	PGM132-Firewall IP Address(VOIM)	PGM133-Firewall IP Apply	PGM102-Firewall IP Address	Contact IP Address
Dual Broadband case	O	O	O	Don't care	VOIM Firewall IP
	O	Other Cases			VOIM Local IP
Normal firewall case	X	Don't care	O	O	UCP Firewall IP
	X	Other Cases			UCP Local IP

Trouble Shooting

SIP call disconnection after 30 seconds

- UCP IP is 10.180.240.220 and VOIM IP is 10.180.240.228.
- Customer set PGM132 – USE Board IP for SIP for VOIM (Turn off USE Board IP)
- Contact IP has VOIM Local IP.

No.	Time	Source	Destination	Protocol	Length	Info
259	2017-08-18 23:16:25.176433	10.4.254.14	10.180.240.220	SIP/SDP	1347	Request: INVITE sip:8773@10.180.240.220;user=phone;transport=tcp
265	2017-08-18 23:16:25.235577	10.180.240.220	10.4.254.14	SIP	521	Status: 100 Trying
266	2017-08-18 23:16:25.235732	10.180.240.220	10.4.254.14	SIP	607	Status: 180 Ringing
338	2017-08-18 23:16:35.856782	10.180.240.220	10.4.254.14	SIP/SDP	878	Status: 200 OK
343	2017-08-18 23:16:36.357518	10.180.240.220	10.4.254.14	SIP/SDP	878	Status: 200 OK
345	2017-08-18 23:16:37.357836	10.180.240.220	10.4.254.14	SIP/SDP	878	Status: 200 OK
349	2017-08-18 23:16:39.358197	10.180.240.220	10.4.254.14	SIP/SDP	878	Status: 200 OK
367	2017-08-18 23:16:43.358437	10.180.240.220	10.4.254.14	SIP/SDP	878	Status: 200 OK
401	2017-08-18 23:16:47.358732	10.180.240.220	10.4.254.14	SIP/SDP	878	Status: 200 OK
425	2017-08-18 23:16:51.359043	10.180.240.220	10.4.254.14	SIP/SDP	878	Status: 200 OK
434	2017-08-18 23:16:55.359360	10.180.240.220	10.4.254.14	SIP/SDP	878	Status: 200 OK
462	2017-08-18 23:16:59.359702	10.180.240.220	10.4.254.14	SIP/SDP	878	Status: 200 OK
489	2017-08-18 23:17:03.360008	10.180.240.220	10.4.254.14	SIP/SDP	878	Status: 200 OK
512	2017-08-18 23:17:07.360310	10.180.240.220	10.4.254.14	SIP/SDP	878	Status: 200 OK
516	2017-08-18 23:17:07.857244	10.180.240.220	10.4.254.14	SIP	590	[TCP Previous segment not captured] Request: BYE sip:5731@10.4.254.14
518	2017-08-18 23:17:07.888729	10.4.254.14	10.180.240.220	SIP	501	Status: 200 OK

Session Initiation Protocol (200)

> Status-Line: SIP/2.0 200 OK

Message Header

> From: <sip:5731@lim1.MX-ONE;user=phone>;tag=9051800c

> To: <sip:8773@10.180.240.220;user=phone>;tag=4e835dc8-dcf0b40a-13c4-65014-a9167-66e17340-a9167

Call-ID: 4FJF46BrfuEhnmvKuAFnTQ..

> CSeq: 1 INVITE

> Via: SIP/2.0/TCP 10.4.254.14:5060;rport=59771;branch=z9hG4bK-524287-1---21019d324b6d792c

> Record-Route: <sip:10.4.254.14:5060;transport=tcp;ln>

> Contact: <sip:8773@10.180.240.228:5060;transport=TCP;user=phone>

Allow: INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,REFER,SUBSCRIBE,NOTIFY,MESSAGE,INFO,PRACK,UPDATE

Supported: replaces,UPDATE,INFO

User-Agent: Ericsson-LG Enterprise iPECS-UCP UCP600 2.1.42

Trouble Shooting

SIP call has one-way mute problem.

- When SIP ALG feature is set in F/W, VOIM Local IP can be used.
- When Port Forwarding rule is used, VOIM Firewall IP must be used.
- If user has mute problem even RTP IP is right, Wireshark trace in front of UCP and VOIM will be helpful to find error.
 - Sometimes F/W device blocks RTP packet from server side.

SDP IP rule

PGM132-Firewall IP Address(VOIM)	PGM133-Firewall IP Apply	SDP IP Address
O	O	VOIM Firewall IP
Don't care	X	VOIM Local IP

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