

iPECS eMG80 System SIP

(iPECS-eMG80_SW-TRA-01-004)

11 July, 2013

REVISION HISTORY

ISSUE	DATE	DESCRIPTION OF CHANGES
0.1	09-JUL-13	Preliminary release
0.2	09-July-13	Domestic Training

Copyright© 2013 Ericsson-LG Co. Ltd. All Rights Reserved

This material is copyrighted by Ericsson-LG Co. Ltd. Any unauthorized reproductions, use or disclosure of this material, or any part thereof, is strictly prohibited and is a violation of Copyright Laws. Ericsson-LG reserves the right to make changes in specifications at any time without notice. The information furnished by Ericsson-LG in this material is believed to be accurate and reliable, but is not warranted to be true in all cases.

Ericsson-LG and iPECS-MG are trademarks of Ericsson-LG Co. Ltd.

Table of Contents

- **SIP extension**
- **SIP trunk**

SIP Extension Programming

Programming in iPECS Web Admin

- Web Admin PGM 443 Station User Login
- Assign ID / Password and desired Station number

Index : [1-50]									
Index	Registered Number	Linked	Version	ID	Password	Zone	Desired Number	Nation	Language
1				150	abcd1234	1	150	Australia ▼	English ▼
2									

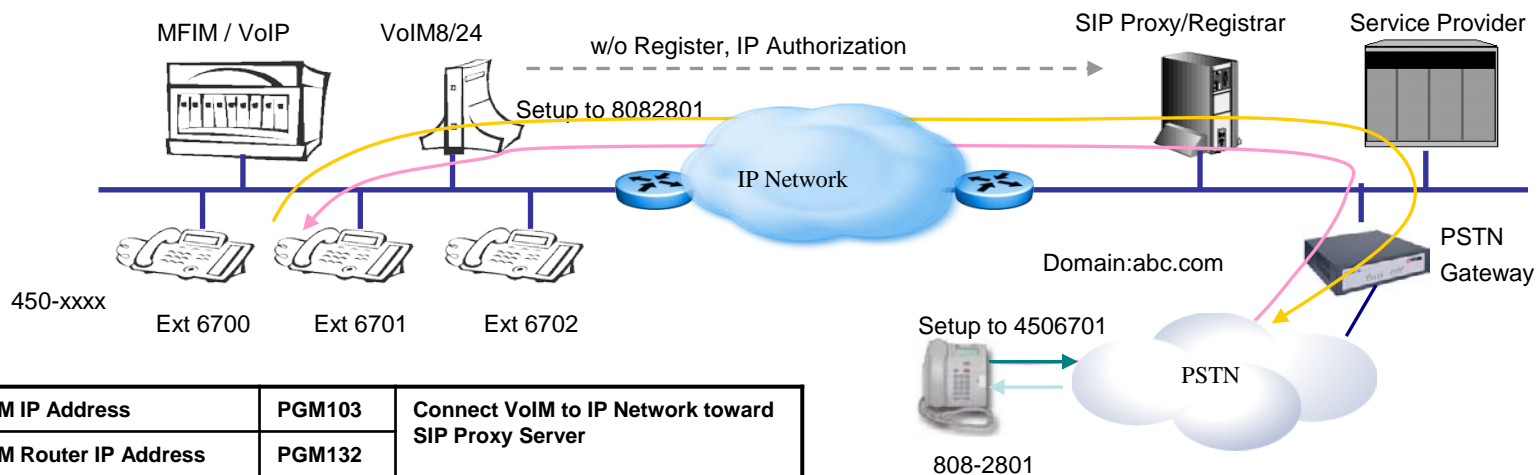
Programming in SIP Extension

- Programming in SIP Extension is required for the registration according to it's user manual
 - ➔ Proxy address & Domain name : IP Address of MFIM
 - ➔ Expiry timer : 60sec (Recommend)

SIP Trunk

Example & Programming (1)

 w/o any authorization ID between IP PBX and SIP Service Provider



VoIM IP Address	PGM103	Connect VoIM to IP Network toward SIP Proxy Server
VoIM Router IP Address	PGM132	
CO Type (PGM 140)	PGM140	DID / Normal
CO Group	PGM141	01~20/72
CO VOIP Mode		SIP
SIP User Table Index	PGM111	0(N/A)
N/A	PGM126	N/A
CLI Programming	PGM114	CLI Station Number (ex. '67xx')
	PGM143	Type of Number for Calling Party Info : National/ Subscriber
		COLP/CLIP Table Index : 0 ~
	PGM201	CLIP/COLP Table (ex. '450')
Incoming Call Routing (PGM144, if CO Type is 'Normal')	PGM143	DID Remove Number (ex. 3)
	PGM145	DID Conversion Type, Number of Digits Expected, Digit Mask

Proxy Server Address	PGM133 - common	IP Address of SIP Proxy Server
1st / 2nd DNS Address		IP Address of DNS Server
Use Outbound Proxy		'ON' if there is no DNS Server
Domain		IP or Name of Domain
Proxy Server UDP Port		if empty (default: 5060)
Local Server UDP Port		if empty (default: 5060)
FailOver PSTN		..
External CODEC Priority		711-u : single CODEC
		None : follow system CODEC
		711-u : Use single codec only 729 'ON' or 'OFF' 723.1

SIP Trunk

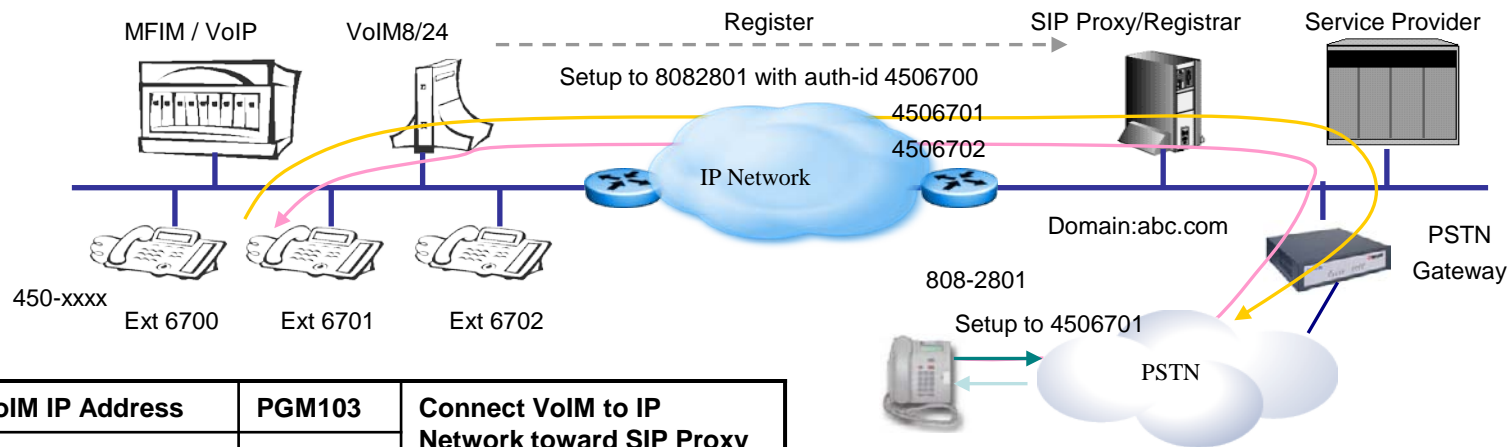
Example & Programming (1)

PGM133 - ID Presentation Option	ID Usage	P-Asserted-Identity		NOT USE / USE
		Remote-Party-ID		NOT USE / USE
	ID Individuality	From ID		Extension Outgoing-CLI
		P-Asserted-Identity		
		Contact ID		Extension Outgoing-CLI
		Remote-Party-ID		
	Offnet Call Route ID Transit	CO to Offnet Direct Call Route	From/Contact ID	ORG
			P-Asserted-Identity	
			Remote-Party-ID	
		Offnet Call Forward by Station	From/Contact ID	ORG
			P-Asserted-Identity	
			Remote-Party-ID	
		Mobile Extension External Call	From/Contact ID	ORG
			P-Asserted-Identity	
			Remote-Party-ID	
		Fixed Table Assignment		N/A

SIP Trunk

Example & Programming (2)

w/ individual (multiple) authorization ID (PSTN Number)



VoIM IP Address	PGM103	Connect VoIM to IP Network toward SIP Proxy Server
VoIM Router IP Address	PGM132	
CO Type (PGM 140)	PGM140	DID / Normal
CO Group	PGM141	01~20/72
CO VOIP Mode		SIP
SIP User Table Index	PGM111	1 ~
SIP User Table	PGM126	Program the Attributes
CLI Programming	PGM111	SIP User Table Index
	PGM133	ID Presentation Option
Incoming Call Routing	PGM126	ID ASSIGNED STATION
	Ring Route Type	RING ASSIGNMENT
		DID CONVERSION

Registration User ID	4506701@abc.com
Authentication User ID	Presented by Service Provider
Authentication User Password	
Contact Number	4506701
User ID Register	Provision / Register
Authorized Representative ID Table Index	0(N/A)
User ID Usage	ON
Ring Route Type	One of selection
DID Conversion Type : one of type 0/1/2	
Number of Digits Expected : 4	
DID Digit Mask : ****	

SIP Trunk

Example & Programming (2)

	ID Usage	P-Asserted-Identity		NOT USE / USE	
		Remote-Party-ID		NOT USE / USE	
	ID Individuality	From ID		Extension SIP-User-ID-Table	
		P-Asserted-Identity			
		Contact ID		Extension SIP-User-ID-Table	
		Remote-Party-ID			
		CO to Offnet Direct Call Route	From/Contact ID	SYS ATD ORG Fixed Table	
			P-Asserted-Identity		
			Remote-Party-ID		
		Offnet Call Forward by Station	From/Contact ID	EXT ORG Fixed Table	
			P-Asserted-Identity		
			Remote-Party-ID		
		Mobile Extension External Call	From/Contact ID	EXT ORG Fixed Table	
			P-Asserted-Identity		
			Remote-Party-ID		
The Others (Proxy IP, Domain ...)		Same As PGM 133 - Common			

SIP Trunk

Example & Programming (3)

w/ single authorization (private) ID and multiple (public) PSTN Number

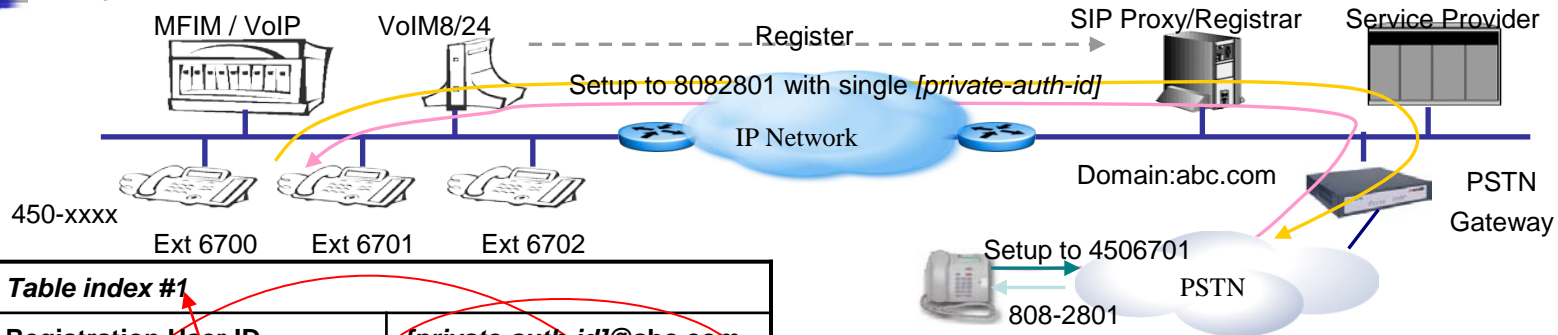


Table index #1

Registration User ID	[private-auth-id]@abc.com
Authentication User ID	ID/PWD : Presented by
Authentication User Password	Service Provider
User ID Register	Provision / Register
Authorized Representative ID Table Index	0(N/A)
User ID Usage	ON

Table index #2 ~

Registration User ID	4506701@abc.com
Authentication User ID	N/A
Authentication User Password	
Contact Number	4506701
User ID Register	Provision
Authorized Representative ID Table Index	1
User ID Usage	ON

VoIM IP Address	PGM103	Connect VoIM to IP Network toward SIP Proxy Server
VoIM Router IP Address	PGM132	
CO Type (PGM 140)	PGM140	DID / Normal
CO Group	PGM141	01~20/72
CO VOIP Mode		SIP
SIP User Table Index	PGM111	1 ~
SIP User Table	PGM126	Program the Attributes
CLI Programming	PGM111	SIP User Table Index
	PGM133	ID Presentation Option
Incoming Call Routing	PGM126 Ring Route Type	ID ASSIGNED STATION
		RING ASSIGNMENT
		DID CONVERSION

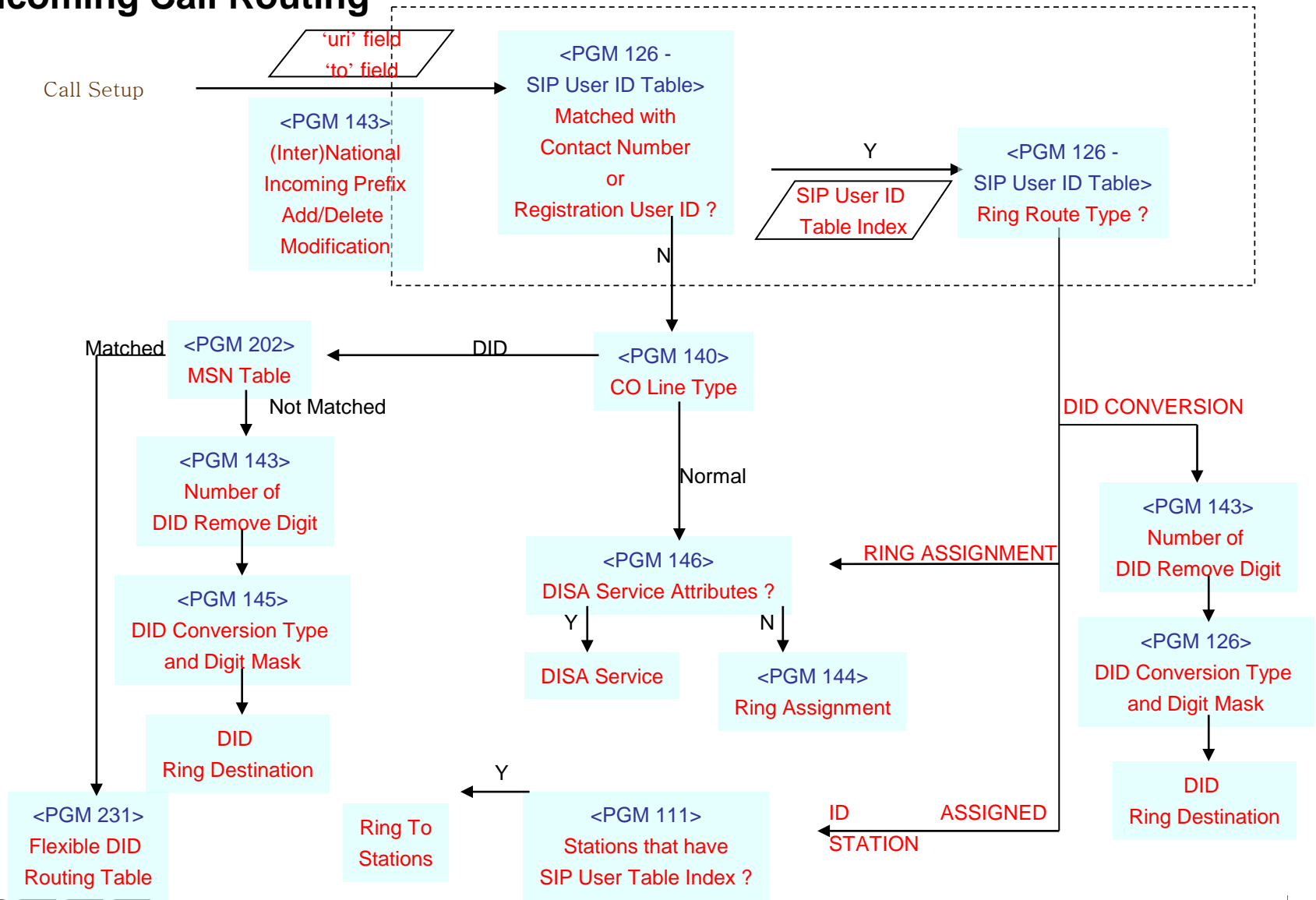
SIP Trunk

Example & Programming (3)

PGM133 - ID Presentation Option	ID Usage	P-Asserted-Identity		USE
		Remote-Party-ID		NOT USE / USE
	ID Individuality	From ID		Extension SIP-User-ID-Table
		P-Asserted-Identity		Authorized Representative ID
		Contact ID		
		Remote-Party-ID		Extension SIP-User-ID-Table
	Offnet Call Route ID Transit	CO to Offnet Direct Call Route	From/Contact ID	ORG
			P-Asserted-Identity	Fixed Table
			Remote-Party-ID	ORG
		Offnet Call Forward by Station	From/Contact ID	ORG
			P-Asserted-Identity	Fixed Table
			Remote-Party-ID	ORG
		Mobile Extension External Call	From/Contact ID	ORG
			P-Asserted-Identity	Fixed Table
			Remote-Party-ID	ORG
		Fixed Table Assignment		1
The Others (Proxy IP, Domain ...)		Same As PGM 133 - Common		


SIP Trunk

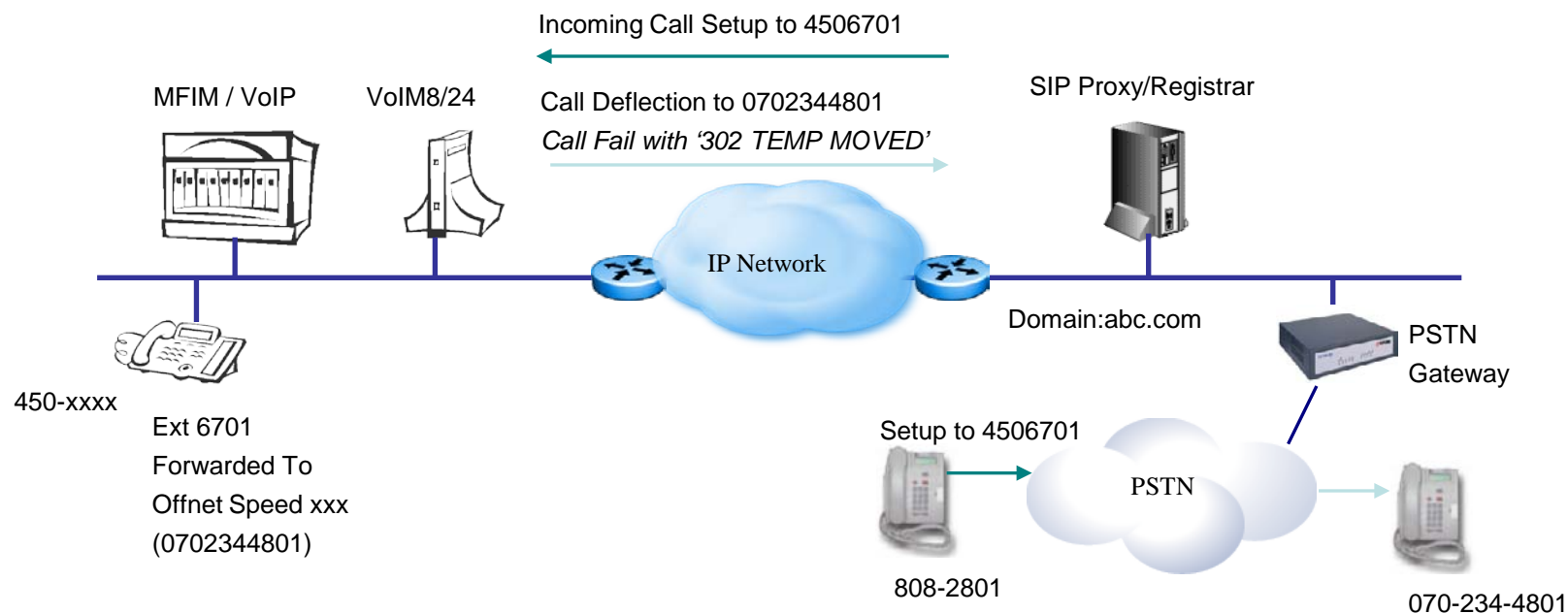
Incoming Call Routing



SIP Trunk

Call Deflection

 SIP / H323 Simultaneous Utilization



Station Call Forward		PGM 143 - ISDN-SS CD/CR(CFU)
PGM 120 Preset Call Forward	Manual Call Forward	Call Deflection

SIP Trunk

Think About



SIP / H323 Simultaneous Utilization

PGM 141	CO VOIP Mode : SIP
PGM 145	DID Conversion Type : Use 'as is'
SIP Out-bound Call	OK
SIP In-bound Call	Should be considered. Refer to Comment below
H323 Networking Out-bound Call	OK
H323 Networking In-bound Call	OK (automatic by NET Numbering)

Comment	<p>If this VoIP is utilized for Distributed Networking channel, it will be H323 mode implicitly and automatically when NET call is made or income, even though its mode is explicitly set to SIP. It say that mode 'SIP' is for SIP Trunk & H323 Networking simultaneously. In this case of simultaneous utilization, incoming SIP call routing should be more considered because DID Conversion Type for NET should be Use 'as is'. For example, when DDDD-XXXX is called to from SIP Trunc, DDDD should be removed by PGM143(DID remove digit) and ring to station XXXX. Another method is utilization of PGM126 SIP User ID table – ID Usage 'ON' and Ring Route Type for SIP incoming call routing.</p>
----------------	--

SIP Trunk

Support [RFC xxxx] Specifications

<i>Supported by Call Processing and SIP G/W Module</i>			
RFC 3261	Session Initiation Protocol (SIP)		
RFC 3665	SIP Basic Call Flow Examples		
RFC 2327	Session Description Protocol (SDP)		
RFC 3264	An Offer/Answer Method with SDP		
RFC 2617	HTTP Authentication: Basic and Digest Access Authentication		
RFC 2833	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals		
RFC 3323	A Privacy Mechanism for the SIP		Privacy
RFC 3325	Private Extensions to the SIP for Asserted Identity within Trusted Networks		P-Asserted-Identity
RFC 2976	The SIP INFO Method		
RFC 4028	Session Timers in the SIP		Session-Expires, Min-SE
RFC 3311	The SIP UPDATE Metho		UPDATE
RFC 3262	Reliability of Provisional Responses in the SIP (PRACK)		PRACK
<i>Others</i>			
RFC 3265	SIP Specific Event Notification	SUBSCRIBE, NOTIFY	Partially
RFC 3428	SIP Extension for Instant Messaging	MESSAGE	SMS only
RFC 3515	SIP Refer Method	REFER	Supported for SIP Extension Only ; Call Transfer on SIP Trunc is Performed by System internally

SIP Trunk

Attributes (1)

CO Type	CO Line	DID or Normal	PGM 140
CO Group : 'xx'	CO Line	01~20/72	PGM 141
CO VOIP Mode	CO Line	SIP / COMMON / H323	PGM 141
SIP User Table Index	Station	0(N/A) or 1 ~	PGM 111
Registration User ID	Table Index	[Number]@[Domain:port] Ex. 0704504114@150.150.150.85:5060 0704504114@150.150.150.85 0704504114@sp.co.kr.85:5060 0704504114@sp.co.kr	PGM126
Authentication User ID		Authorization ID	
Authentication User Password		Authorization Password	
Contact Number		[Number] Ex. 0704504114	
Contact Display Name		[Name]	
Asc Station Number		N/A	
User ID Register		Provision / Register	
Authorized Representative ID Table Index		0(N/A) or 1 ~	
User ID Usage		OFF/ON (should be ON for in/out call using this tale)	
Ring Route Type		ID ASSIGNED STATION / RING ASSIGNMENT / DID CONVERSION	
DID Conversion Type		DID Conv. / Use 'as is' / Modification by flexible DID table	
Number of Digits (2-4) Expected from DID Circuit		3 or 4	
DID Digit Mask (4 digits: *,#,0-9)		#*** or ****	
SMS Received Station Number		[Station Number]	

SIP Trunk

Attributes (2)

Soft Switch Type	Normal or Specific SIP Server or Service Provider Type	PGM 133 - Slot seq. of VoIM
Proxy Server Address	IP or DNS Name of SIP Service Provider Server	
Primary DNS Address	IP of 1 st DNS resolution server	
Secondary DNS Address	IP of 2 nd DNS resolution server	
Use Outbound Proxy	USE / NOT USE	
Connection Mode	UDP / TCP / TLS	
Caller Name Service	NOT USE / USE (local display : PGM114 CLI Name Display)	
181 Being Forwarded	NOT USE / USE (refer to PGM143 : ISDN-SS CD/CR(CFU) – Call Deflection (Moved Temp 302)	
100rel support	OFF/ON : PRACK Use or Not	
Use single codec only	OFF / ON	
Use rport method	OFF / ON	
Domain	IP or DNS Name of SIP Service Provider Server	
Invite Acceptance	Domain Only / From All	
Contact Address Domain	SIP GW Addr / Server Domain	
Proxy Registration Timer	Registration Frequency (normally 3600 or less) from VoIM to SIP Proxy Server	
Proxy Server UDP Port	When 'Connection Mode' is UDP : Server SIP UDP port#	
Proxy Server TCP Port	When 'Connection Mode' is TCP : Server SIP TCP port#	
Proxy Server TLS Port	When 'Connection Mode' is TLS : Server SIP TLS port#	
Local Server UDP Port	When 'Connection Mode' is UDP : VoIM SIP UDP port#	
Local Server TCP Port	When 'Connection Mode' is TCP : VoIM SIP TCP port#	
Local Server TLS Port	When 'Connection Mode' is TLS : VoIM SIP TLS port#	
Registration UID Range	[Empty] for non-registration or Index range of SIP User Table with User ID Register 'Register', And User ID Usage 'ON' in PGM 126.	
DTMF Type	INBAND / 2833 / INFO(type: SIMPLE, CISCO, TELEPHONE Event, Nortel Networks)	
SIPEXT Registration Usage	OFF / ON	
SIPEXT Registration Interval	Registration Frequency (normally 3600 or less) from SIP Terminal to VoIM	

SIP Trunk

Attributes (3)

ID - Presentation Option	ID Usage	P-Asserted-Identity		NOT USE / USE	PGM 133 - Slot seq. of VoIM
		Remote-Party-ID		NOT USE / USE	
	ID Individuality	From ID		Extension SIP-User-ID-Table Extension Outgoing-CLI Authorized Representative ID	
		P-Asserted-Identity			
		Contact ID		Extension SIP-User-ID-Table Extension Outgoing-CLI	
		Remote-Party-ID			
	Offnet Call Route ID Transit	CO to Offnet Direct Call Route	From/Contact ID	SYS ATD ORG Fixed Table	
			P-Asserted-Identity		
			Remote-Party-ID		
		Offnet Call Forward by Station	From/Contact ID	EXT ORG Fixed Table	
			P-Asserted-Identity		
			Remote-Party-ID		
		Mobile Extension External Call	From/Contact ID	EXT ORG Fixed Table	
			P-Asserted-Identity		
			Remote-Party-ID		
		Fixed Table Assignment		Index of SIP User Table which will be representative ID table for authorization	
External CODEC Priority Configuration		1 st priority	g.711-u g.711-a g.729 g.729-a g.723.1 none		
		2 nd priority			
		3 rd priority			
		4 th priority			
		5 th priority			

SIP Trunk

Attributes (4)

SIP Call Setup FailOver Option	Call Setup No Response Time		no response timer after send outgoing setup message to SIP proxy server - 0 or [Empty] : do not use 'no response timer' 3~10 : wait for 3 to 10 second	PGM 133 - Slot seq. of VoIM
	FailOver CO Group Number		Case #1 - SIP CO line is on connected/alive state : after no response time, setup message will be re-sent using this failover CO line group Case #2 - SIP CO line is on disconnected/OOS state : setup message will be sent using this failover CO line group	
URI Formatting and Rules	General Formatting	To Field Method	sip:method To: <sip:[Number]@[Service Provider Domain Name];user=phone>	
			'tel:method' To: <tel:+[Number]>	
		Numbering Format	Local [Number]@[Service Provider Domain Name]	
			Global(+E164) +[E.164 Address]@[Service Provider Domain Name] E.164 Address : Nation Code & Area Code	
		Local: include Area Code	NO / YES If 'Numbering Format' is Local, Area Code (PGM 143) is automatically inserted or not to 'To' [Number] user dial '8701234', area code is '042' uri is, 0428701234@[Service Provider Domain Name]	
		Global: include phone-context	NO / YES If 'Numbering Format' is Global and 'To Field Method' is tel:method, 'phone-context' is automatically added by following or not. user dial '0011428701234' uri is, tel:0011428701234;phone-context=+82	

SIP Trunk

Attributes (5)

URI Fromatting and Rules	Specific Fromatting by Conversion	Numbering Case #1: From (4 dgt) >		> To (6 dgt)	[User Dial]	[Result]
		Example	0	+82	0314504639	+ 82314504639
			00	+	0082314504639	+ 82314504639
			1588	1588	15886724	15886724
			031		0314504639	4504639

iPECS

Your Communications Solution