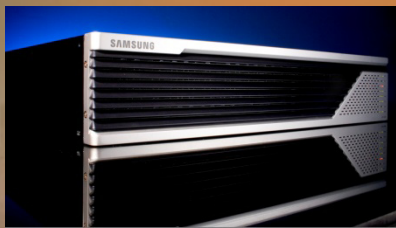


Voice Configuration - SIP Trunking Gateway



- ❶ iBG1003 VoIP Feature Overview
- ❷ VoIP Gateway Administration
 - host, bind, sip-ua, sip-server, registrar, sip-connect, peer-monitor, relay-server
 - Voice Services (SIP), Voice Classes (Codec)
- ❸ Understanding Voice Interfaces
 - Voice Port, Dial Peer, Module, Trunk-group
- ❹ Analog Line Administration - FXS
- ❺ Analog Trunk Administration – FXO, DID
- ❻ Digital Trunk Administration - Interface bundle ISDN

iBG1003 VoIP Gateway Overview

● Voice Signaling & Interfaces

- Analog FXS (loop-start, ground-start) signaling
- Analog DID (Direct Inward Dial)
- Analog FXO (loop-start, ground-start) signaling
- BRI/T1/E1 ISDN BRI/PRI Q.931 User Side/Network Side

● VoIP Protocols

- SIP UA
 - Analog UA registration
 - HTTP Digest authentication, Challenge response, TLS
 - Demand Testing (OPTION Keepalive)
- SIP Outbound Proxy (stateless outbound proxy)
- RTP, RTCP

● Call Features

- Blind Transfer with transfer Recall
- Attended Transfer
- 3way conference
- CLID, CNID

● Voice QoS

- Classification & Marking - IP Precedence, DSCP, RTP Port range
- Low Latency Queuing
- Gain/Volume control

● DSP Features

- CODEC : G.711, G723.1, G.726, G.729
- Echo Canceller - G.168 (128ms)
- Jitter buffer – Max 200ms
- Comfort Noise Generation (CNG), Voice Activity Detection (VAD)
- Tone - MF, DTMF detection/ generation

● System Service Features

- FAX support : T.38 Fax Relay, G.711 pass-through
- Dial Plan
 - Secondary Dial Tone
 - Short/Long Inter digit Timer
- DTMF Relay - RFC2833, Queued-digits
- Early Media Handling

● Management

- Command Line Interface
- SNMP MIB management (get/set)
- Call Log / Statistics
- Alarm/Event/Debug Logging, Remote logging

VoIP Gateway Administration

● Voip-gateway

|-- host

|-- bind

|-- rtp, rtcp

|-- ip-qos

|-- sip-ua

|-- shutdown



● Voice class

|-- codec codec-preference

● Voice service

|-- codec-list

|-- sip

|-- voip

| |-- fax

| | |-- dsp

| | |-- vad

|-- voice

| |-- locale



- Command used to bind the source address for media and control packets to the IP address of a specific interface.
 - Caution
 - voip-gateway should be **shutdown forced before changing the binding interface.**
 - Syntax
 - [no] bind media interface {**ethernet | bundle | loopback | vlan**} <interface-id>
 - Example
 - < voip-gateway>
 - iBG1003/configure/voip-gateway# shutdown forced
 - iBG1003/configure/voip-gateway# bind media interface ethernet 0/4
 - iBG1003/configure/voip-gateway# bind control interface ethernet 0/4
 - iBG1003/configure/voip-gateway# no shutdown
 - <relay-server>
 - iBG1003/configure/relay-server# shutdown
 - iBG1003/configure/relay-server# bind control interface ethernet 0/4

- Command used to specify the domain name to be used in iBG.
 - Syntax
 - [no] host domain-name <domain>
 - Example
 - iBG1003/configure/voip-gateway# host domain-name company.com

shutdown / no shutdown (voip-gateway)



- Command used to start/stop the VoIP gateway.
- 'shutdown forced' command not only stops the VoIP gateway but also deregisters all the stations and unbinds a VoIP interface.

- Syntax

- [no] shutdown [forced]

- Example

- stop VoIP service
iBG1003/configure/voip-gateway# shutdown
 - stop VoIP service and deregister all stations
iBG1003/configure/voip-gateway# shutdown forced
 - start VoIP service
iBG1003/configure/voip-gateway# no shutdown

Set up the VoIP Gateway



● Basic configuration for VoIP Gateway

1. Setup an IP address of iBG VoIP Gateway

- iBG1003# conf t
- iBG1003/configure# voip-gateway
- iBG1003/configure/voip-gateway# **bind media interface ethernet 0/4**
- iBG1003/configure/voip-gateway# **bind control interface ethernet 0/2**

2. Setup an domain name of iBG VoIP Gateway

- iBG1003/configure/voip-gateway# **host domain-name company.com**

3. Start VoIP Service

- iBG1003/configure/voip-gateway# **no shutdown**

show voip gateway



● Command to use show VoIP configuration information and status

● Syntax

- show voip gateway

```
iBG1003# show voip gateway
VoIP Gateway Status
Gateway Admin Status      : UP
Gateway Operation Status  : Stand-alone mode (Server Ready)
  Server type : Generic SIP(RFC3261) [ SIPconnect, Option1 ]
  Registrar   : sip-server
  SIP-server   : ipv4:90.90.13.201 UDP

Gateway IP address
  Binding status : ethernet 0/0, ethernet 0/0
  Control IP address : ipv4:90.90.13.82
  Media IP address  : ipv4:90.90.13.82

Gateway Accounting
  RADIUS   : DISABLED
  SYSLOG   : DISABLED
Default domain name : samsung.com

VoIP Protocol status
  VoIP service : ENABLED
  SIP service  : ENABLED
  H.323 service : DISABLED

VoIP Media configuration
  QoS Media : ef
  QoS Signal: ef
  RTP Start Port: 16384, Range: 512
  RTCP Interval : 5 (1-10)
```

DISABLED when host ip-address is used

sip-ua commands



- sip-server
- registrar
- sip-connect
- authentication



- Command to set SIP-Server address. You can also set transport type, URI type and secondary.
- Syntax
 - [no] sip-server ip-address {ipv4: | ipv6: | dns:} [transport { udp | tcp | tls }] [uri { sip | sips }] [secondary]
- Example
 - iBG1003/configure/voip-gateway/sip-ua# sip-server ipv4:90.90.13.201 transport udp secondary



- Command to set Registrar-Server. You can also set transport type, URI type, expires and retry timer

- Syntax

- [no] registrar ip-address { ipv4: | ipv6: | dns: } [transport { udp | tcp | tls }] [uri {sip | sips}] expires <value> retry <value>

- Example

- iBG1003/configure/voip-gateway/sip-ua# registrar ip-address ipv4:90.90.13.201 transport udp expires 1800

● Command to set representative username for sip-connect.

● Syntax

- [no] sip-connect [public-identity <username>] [type { option1 | option2 }]

● Example

- iBG1003/configure/voip-gateway/sip-ua# sip-connect public-identity 3300

- Command to specify the user name, password and realm to be used in REGISTER for registration in a Call-server. This command is for global configuration of authentication. When there's no authentication information on each dial-peer, then it is used.

- Syntax

- [no] authentication username <global-username> password <global-password> [realm <global-realm>]

- Example

- iBG1003/configure/voip-gateway/sip-ua# authentication username sip1 password sip123

Relay-server commands



- sip-server
- registrar
 - Reference : sip-ua command

Verifying and Debugging SIP Features



- This section describes how to use the show command and debug command to verify and troubleshoot the SIP feature of iBG.
 - show sip-ua parameters
 - show sip-ua maps
 - show sip-ua registration
 - show sip-ua calls
 - debug sip dump event
 - debug sip dump message



● Show reason mapping between PSTN and SIP

```
iBG1003# show sip-ua maps pstn-sip
```

| PSTN-Cause | Configured SIP-Status | Default SIP-Status |
|------------|--------------------------|-----------------------|
| 1 | 404 | 404 |
| 2 | 404 | 404 |
| 3 | 404 | 404 |
| 4 | 500 | 500 |
| 5 | 500 | 500 |
| 6 | 500 | 500 |
| 7 | 500 | 500 |
| 8 | 500 | 500 |
| 9 | 500 | 500 |
| 16 | 200 | 200 |

→ Q.850 clear cause 1 (Unallocated number)
maps to 404 Not Found

```
iBG1003# show sip-ua maps sip-pstn
```

| SIP-Status | Configured PSTN-Cause | Default PSTN-Cause |
|------------|--------------------------|-----------------------|
| 400 | 127 | 127 |
| 401 | 57 | 57 |
| 402 | 21 | 21 |
| 403 | 57 | 57 |
| 404 | 1 | 1 |
| 405 | 127 | 127 |
| 406 | 127 | 127 |

→ 403 Fobidden maps to Q.850 clear cause 21
(Call rejected)

show sip-ua registration



- Show registration information of FXS stations of iBG.

tag

iBG1003# show sip-ua registration

| ID | DEST-PATTERN | EXPIRES | STATUS | PORT | AUTHENTICATION |
|------|--------------|---------|--------|-------|----------------|
| 3200 | 3200 | 3600 | yes | 0/0/2 | 3200[****] |
| 3201 | 3201 | 3600 | yes | 0/0/3 | 3201[****] |
| 3100 | 3100 | 3600 | yes | 2/0 | 3100[****] |
| 3101 | 3101 | 3600 | yes | 2/1 | 3101[****] |
| 3102 | 3102 | 3600 | yes | 2/2 | 3102[****] |
| 3103 | 3103 | 3600 | yes | 2/3 | 3103[****] |
| 3104 | 3104 | 3600 | yes | 2/4 | 3104[****] |
| 3105 | 3105 | 3600 | yes | 2/5 | 3105[****] |
| 3106 | 3106 | 3600 | yes | 2/6 | 3106[****] |
| 3107 | 3107 | 3600 | yes | 2/7 | 3107[****] |
| 3108 | 3108 | 3600 | yes | 2/8 | 3108[****] |
| 3109 | 3109 | 3600 | yes | 2/9 | 3109[****] |
| 3110 | 3110 | 3600 | yes | 2/10 | 3110[****] |
| 3111 | 3111 | 3600 | yes | 2/11 | 3111[****] |

Authentication
(user name and password)
information, passwords are
not shown.

Number of 14/total(14) are registered

Destination-pattern
from dial-peer

Expires timer

Registered (yes) or no

Voice-port number of the station

show sip-ua connections

- Show SIP connection information of iBG.
 - UAS listening socket information appears.
 - When iBG has a connection with other peers, it appears

```
iBG1003# show sip-ua connections
```

```
UDP CONNECTION INFO
```

| No. | Local Address | Remote Address | Status |
|-----|---------------------|----------------|--------|
| 0. | 172.19.30.254 :5060 | - | Listen |

```
TOTAL NUMBER : 1
```

```
TCP CONNECTION INFO
```

| No. | Local Address | Remote Address | Status |
|-----|---------------------|----------------|----------|
| 0. | 172.19.30.254 :5060 | - | : Listen |

```
TOTAL NUMBER : 1
```

```
TLS CONNECTION INFO
```

| No. | Local Address | Remote Address | Status | Algorithm |
|-----|---------------------|----------------|--------|-----------|
| 0. | 172.19.30.254 :5061 | - | Listen | |

```
TOTAL NUMBER : 1
```



● Show detailed sip calls information

Context ID (Internal Value) **Call-ID**

```
iBG1003# show sip-ua calls
```

SIP-CALL CTXID : 40

- SIP-CALLID : AAu_0BqnCqccVgAAAAAAA@172.19.30.254
- CALL-STATES : PIC_T_Active
- FROM : <sip:3201@company.com>;tag=ubigate0000...000040
- TO : <sip:3002@company.com>;tag=80a2fa4da1336543d56f6000
- Request URI : [1] 3002@company.com:0
- SOURCE ADDRESS (LocUri) : udp 3201@172.19.30.254:5060
- DESTINATION ADDRESS (RemoteUri) : tcp @172.19.30.44:0
- MEDIA INFO
 - PAYLOAD ID(CODEC) : g711u
 - RTP LOCAL ADDRESS : 172.19.30.202:34008
 - RTP REMOTE ADDRESS : 5.1.1.1:16470
 - DTMF PAYLOAD TYPE : 127

TOTAL NUMBER : 1

SDP information **Total number of calls** **Local/Remote Target (Contact)** **From/To header**

debug sip dump event

- Dump received/transmitted SIP messages (request-line only)

```
iBG1003# debug sip dump req
```

```
SIP:Tx ---> 172.19.30.20/5060 INVITE sip:3201@company.com;avaya-cm-fnu=off-hook SIP/2.0
SIP:Rx <--- 172.19.30.20/32771 SIP/2.0 100 Trying
SIP:Rx <--- 172.19.30.20/32771 SIP/2.0 407 Proxy Authentication Required
SIP:Tx ---> 172.19.30.20/5060 ACK sip:3201@company.com;avaya-cm-fnu=off-hook SIP/2.0
SIP:Tx ---> 172.19.30.20/5060 INVITE sip:3201@company.com;avaya-cm-fnu=off-hook SIP/2.0
SIP:Rx <--- 172.19.30.20/32771 SIP/2.0 100 Trying
SIP:Rx <--- 172.19.30.20/32771 SIP/2.0 183 Session Progress
SIP:Tx ---> 172.19.30.20/5060 PRACK sip:3201@172.19.30.44;transport=tcp SIP/2.0
SIP:Rx <--- 172.19.30.20/32771 SIP/2.0 407 Proxy Authentication Required
SIP:Tx ---> 172.19.30.20/5060 PRACK sip:3201@172.19.30.44;transport=tcp SIP/2.0
SIP:Rx <--- 172.19.30.20/32771 SIP/2.0 200 OK
SIP:Tx ---> 172.19.30.20/5060 CANCEL sip:3201@company.com;avaya-cm-fnu=off-hook SIP/2.0
SIP:Rx <--- 172.19.30.20/32771 SIP/2.0 200 OK
SIP:Rx <--- 172.19.30.20/32771 SIP/2.0 487 Request Terminated
SIP:Tx ---> 172.19.30.20/5060 ACK sip:3201@company.com;avaya-cm-fnu=off-hook SIP/2.0
```

debug sip dump message



● Dump received/transmitted SIP full messages

```
iBG1003# debug sip dump req
```

```
=====
15:50:36.290 [UA] UDP SIP Packet(730bytes) Sent ---> 12.3.177.202/5060
INVITE sip:205@12.3.177.202 SIP/2.0
Via: SIP/2.0/UDP 12.3.177.85:5060;branch=z9hG4bKhsig0000000133AARs0OTMBMwAQgAAAAAAA
From: <sip:2500@12.3.177.85>;tag=971885ad-N690-7231429
To: <sip:205@12.3.177.202>
Call-ID: AARs0OTMBMwAQQAAAAAAA-63f7-00001c@12.3.177.85
CSeq: 1 INVITE
Contact: <sip:2500@12.3.177.85:5060>
P-Asserted-Identity: <sip:2500@12.3.177.85>
Max-Forwards: 70
Allow: ACK,BYE,CANCEL,INFO,INVITE,MESSAGE,NOTIFY,OPTIONS,PRACK,REFER,SUBSCRIBE,UPDATE
Supported: 100rel, replaces
User-Agent: Samsung-iBG-SIPUA/2.8.0.1
Content-Type: application/sdp
Content-Length: 140

v=0
o=2500 1 1 IN IP4 12.3.177.85
s=-
c=IN IP4 12.3.177.85
t=0 0
m=audio 16628 RTP/AVP 18
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
```


Voice Class Mode Command



● Codec

- Command used to create voice class codec or to enter the voice class codec configuration mode.
- Syntax
 - [no] voice class codec <codec-list-id>
 - codec-list-id: Unique number to identify codec voice class. Numbers from 1 to 10000 are available.
- Example
 - iBG1003/configure# voice class codec 1

- Command used to set codec type and packet interval size with preference in codec list.

- Syntax

- [no] codec-preference <preference> {g711alaw | g711ulaw | g723.1 5.3 | g723.1 6.3 | g726 | g729} <size>
- preference: Codec preference value in codec-list. Can set the value from 1 to 6. Can set a total of 6 codecs.
- size: Interval time between voice packets. Can set the value from 10/20/30/40/50/60 except G.723.1(30/60ms)

- Example

```
iBG1003/configure/voice/class/codec 1# codec-preference 1  
g711alaw 20
```

Voice Services Mode Commands



- voip codec-list
- sip

- Specify a codec class number to using it as global codec set

- Syntax

- [no] voice service codec-list <codec class number>

- Example

- Define codec-list

iBG1003/configure# **voice class codec 1**

iBG1003/configure/voice/class/codec **1**# codec-preference 1 g711alaw 20

iBG1003/configure/voice/class/codec **1**# codec-preference 2 g711ulaw 20

iBG1003/configure/voice/class/codec **1**# codec-preference 3 g723 20

iBG1003/configure# **voice class codec 2**

iBG1003/configure/voice/class/codec **2**# codec-preference 1 g711ulaw 60

iBG1003/configure/voice/class/codec **2**# codec-preference 2 g711alaw 60

iBG1003/configure/voice/class/codec **2**# codec-preference 3 g729 20

- Apply codec-list to voip

iBG1003/configure#voice service voip

iBG1003/configure/voice service/voip# **codec-list 1**

voice service sip commands



- dtmf-relay
- no-answer-timer
- local-port



- You can use dtmf-relay command in a SIP service configuration mode to specify the method of sending (relaying) a dual tone multi-frequency (DTMF) tone.
 - Syntax
 - [no] dtmf-relay { inband | rtp-nte | sip-notify | sip-info | queued-digits }
 - default: rtp-nte (rfc2833)
 - Example
 - The following example is the procedure of making a setting to use SIP NOTIFY message as a DTMF relay method.
iBG1003/configure# voice service sip
iBG1003/configure/voice/service/sip# dtmf-relay sip-notify

- If you want to change the timer value specifying the time between 1xx response and up to 2xx response message in Session Initiation Protocol (SIP), use the no-answer-timer command in a SIP service configuration mode. Use “no” command to set it to the default value.

- Syntax

- [no] no-answer-timer <seconds>
- timer range: 5 ~ 180 sec, default: 120 secs

- Example

- The following is used to set no answer timer to 100.
iBG1003/configure# voice service sip
iBG1003/configure/voice/service/sip# no-answer-timer 100

- Use this command in the SIP service configuration mode to change local port numbers in UDP, TCP, and TLS of SIP-UA or Outbound Proxy (SIP Relay server).
- The ports indicate local ports – source ports that are bound to SIP-UA or Outbound Proxy server.
- To restore the system default, use the 'no' form of this command.

- Syntax

- [no] local-port {sip-ua [udp <port-number> | tcp <port-number> | tls <port-number>] | sip-server [udp <port-number> | tcp <port-number> | tls <port-number>]}

- Example

- The following example shows how to change SIP-UA's UDP, TCP, and TLS ports to 5070, 5070, and 5071, respectively .
iBG1003/configure# voice service sip
iBG1003/configure/voice/service/sip# local-port sip-ua udp 5070 tcp 5070 tls 5071

show voice dsp



- Command to show the current status of DSP including allocated voice channels.

```
iBG1003# show voice dsp
```

D S P C O N F I G U R A T I O N

```
+ IMAGE-PROFILE: /cf0/dsp_1_0_1.bin
+ IMAGE-VERSION: v1.0.1
+ CORE-VERSION: Test.5.02
```

DSP Image version

```
+ IP-ADDRESS: 5.1.1.1
+ BASE-UDP-PORT: 16384
+ UDP-PORT-RANGE: 512
+ ACML: 137/138 CHANNELS AVAILABLE
```

| TYPE | DSPCH | CODEC | STATE | VOICE PORT | TS | PACKET TX/RX-PAK-CNT |
|------|-------|------------|-------|---------------|----|-------------------------|
| M520 | 046 | { g.711a } | BUSY | 0/0/3 | -- | 80/143 |

Show used channel
allocated to DSP

```
+ TOTAL ACTIVES: 1
```

show voip rtp connections



- Command used to show voip rtp connections established on the system .

- Example

iBG# show voip rtp connections

Capability Direction
sr: SendRecv
ro: RecvOnly
so: SendOnly

| iBG# show voip rtp connections | | | | | | | | | |
|--------------------------------|---------|---------------|-------|-------------|---------|---------------|-------|--------|------|
| ===== | | | | | | | | | |
| VOIP RTP CONNECTIONS | | | | | | | | | |
| ===== | | | | | | | | | |
| L o c a l | | | | R e m o t e | | | | | |
| ----- | | | | | | | | | |
| No. | Call-ID | IP-Address | Port | | Call-ID | IP-Address | Port | | |
| 1 | 17 | 172.19.30.254 | 16400 | (3201) | 18 | 172.19.30.254 | 16402 | (3200) | [sr] |
| 2 | 20 | 172.19.30.254 | 16402 | (3201) | 19 | 172.19.30.254 | 16400 | () | [sr] |

Call Context Id
(Internal Value)

Calling Number

Called Number

show voice dsp misc



● Command to display detailed DSP information.

● Syntax

- show voice dsp [misc | pkt | vop]

Miscellaneous information of DSP

```
iBG1003# show voice dsp misc
```

| CH# | FAX | FAX RATE | T38 ECM | DISC SUPV | CN 3389 | CRYPTO | LOCAL IP-ADDR/PORT | REMOTE IP-ADDR/PORT |
|-----|-----|----------|---------|-----------|---------|--------|--------------------|---------------------|
| 054 | off | 9600 | off | off | off | off | 5010101h/16492 | AC131ECAh/34008 |

```
iBG1003# show voice dsp pkt
```

Packet related information of DSP

| CH# | TIME | DELAY CUR/MIN/MAX | TX-PKT VCE/ SIG/ CNO | RX-PKT VCE/ SIG/ CNO/TOCT | RX-PKT LS/ OS/ LT/ PJ |
|-----|------|-------------------|----------------------|---------------------------|-----------------------|
| 055 | 9 | 45/ 45/ 45 | 75/ 27/ 27 | 435/ 0/ 0/69.6 | 0/ 0/ 0/ 1 |

```
iBG1003# show voice dsp vop
```

Voice channel option related information of DSP

| CH# | PTIME | VAD | CNG | DTMFPT TX/ RX | DYN-JIT-BUF INIT/MIN/MAX | ECHO-CANCEL EC/NLP/TAIL | IP DSCP | GAIN IN/OUT |
|-----|-------|-----|-----|---------------|--------------------------|-------------------------|---------|-------------|
| 55 | 20 | off | on | 127/127 | 20/ 20/200 | on/ on/ 48 | 88h | 0/ 0 |
| 18 | 20 | off | on | 127/127 | 20/ 20/200 | on/ on/ 48 | 88h | 0/ 0 |
| 19 | 20 | off | off | INB/INB | 20/ 20/200 | on/ on/ 48 | 00h | 0/ 0 |

Voice channel option related information of DSP

Voice Port Administration

Analog Voice Port – FXS/FXO



- Analog voice port is used to transmit voice and call signal between a packet-based network and the existing telephony network. The configuration of analog voice port is automatically perceived when the system boots up, and it works in a default configuration.
 - **FXS - Foreign eXchange Subscriber interface**
 - FXS(the plug on the wall) delivers POTS service from the local phone company's Central Office (CO) and must be connected to subscriber equipment (telephones, modems, and fax machines). In other words an FXS interface points to the subscriber.
 - **FXO - Foreign eXchange Office interface**
 - FXO (the plug on the phone) receives POTS Service, typically from a Central Office of the Public Switched Telephone Network (PSTN). In other words an FXO interface points to the Telco office. A standard RJ-11 modular telephone cable connects the FXO voice interface card to the PSTN or PBX through a telephone wall outlet.

Analog Voice Port – DID



- Direct Inward Dialing (DID), which is the service proposed by Telephone service providers, is a service to allow a caller to dial to extension of PBX without help of an attendant. DID enables every extension of a PBX to work as if every extension is directly connected to PSTN.
 - **Start Signal**
 - Immediate Start
 - wink-Start
 - Delay Dial
- Restrictions of DID Trunk
 - Dial tone is not present.
 - Outgoing calls are not allowed. If an outgoing call is attempted, the caller will get a fast busy signal.

Digital Voice Port – ISDN BRI/PRI port



- Digital voice ports are automatically generated when the 'activate' command is executed after the 'incoming-voice' command has been executed for an isdn interface.
- Using trunk-group, one can group multiple digital ports.
- ISDN BRI/PRI
 - BRI = voice-port 0/0/0, 01~02 channels are displayed.
 - PRI = voice-port 0/0/0:D, 01~23 or 01~31 channels are displayed.

voice-port commands summary



- voice-port
 slto/[subslot]/port[:D]
 - |-- signal
 - |-- dial-type
 - |-- description
 - |-- ring
 - |-- echo-cancel
 - |-- input
 - |-- output
 - |-- impedance
 - |-- non-linear
 - |-- comfort-noise
 - |-- timeouts
 - |-- playout-delay
 - |-- supervisory
 - |-- caller-id
 - |-- station
 - |-- mwi
 - |-- battery-reversal
 - |-- trunk-group
 - |-- bearer-cap
 - |-- shutdown

- To specify the type of signaling for a voice port, use the 'signal' command in voice-port configuration mode. To restore the default setting, use the no form of this command.
- Syntax
 - [no] signal {did | ground-start | loop-start},
 - 'signal did' command is supported only for FXS card
- Example
 - The following example configures ground-start signaling as the signaling type for a voice port, which means that both sides of a connection can place a call and hang up.
iBG1003/configure# voice-port 0/2/2
iBG1003/configure/voice-port (0/2/2)# signal loop-start

- Command used to allow the sending or receiving of caller-ID information. Use the 'caller-id enable' command in voice-port configuration mode at the sending FXS voice port or the receiving FXO voice port.

- Syntax

- [no] caller-id enable {type1 | type2}

- Example

- iBG1003/configure/voice-port (0/0/0)# caller-id enable type1

● Command used to set caller-id signaling method.

- Syntax

- caller-id type {fsk | dtmf}

- Example

iBG1003/configure/voice-port (0/0/0)# caller-id type fsk

show voice port summary (analog)



| | | |
|-----------------------|--|---------------------|
| Shutdown | down | down |
| Idle | on-hook | Idle |
| Busy (Origin Side) | off-hook tone_rb, tone_busy, tone_congest, tone_dial | idle |
| Busy (Term side) | Idle | ringing off-hook |

Shutdown
/no shutdown

Interface
up/down

iBG1003# show voice port summary

| PORT | CH | SIG-TYPE | ADMIN | OPER | IN STATUS | OUT STATUS | EC |
|-------|----|------------|-------|------|-----------|------------|----|
| 0/0/0 | -- | fxs-ls | up | up | on-hook | idle | y |
| 0/0/1 | -- | fxs-ls | up | up | on-hook | idle | y |
| 0/0/2 | -- | fxs-ls | up | up | on-hook | idle | y |
| 0/0/3 | -- | fxs-ls | up | up | on-hook | idle | y |
| 2/0 | -- | fxs-did-im | up | up | idle | idle | y |
| 2/1 | -- | fxs-did-wk | up | up | idle | idle | y |
| 2/2 | -- | fxs-did-dl | up | up | idle | idle | y |

Echo Cancel
Enabled

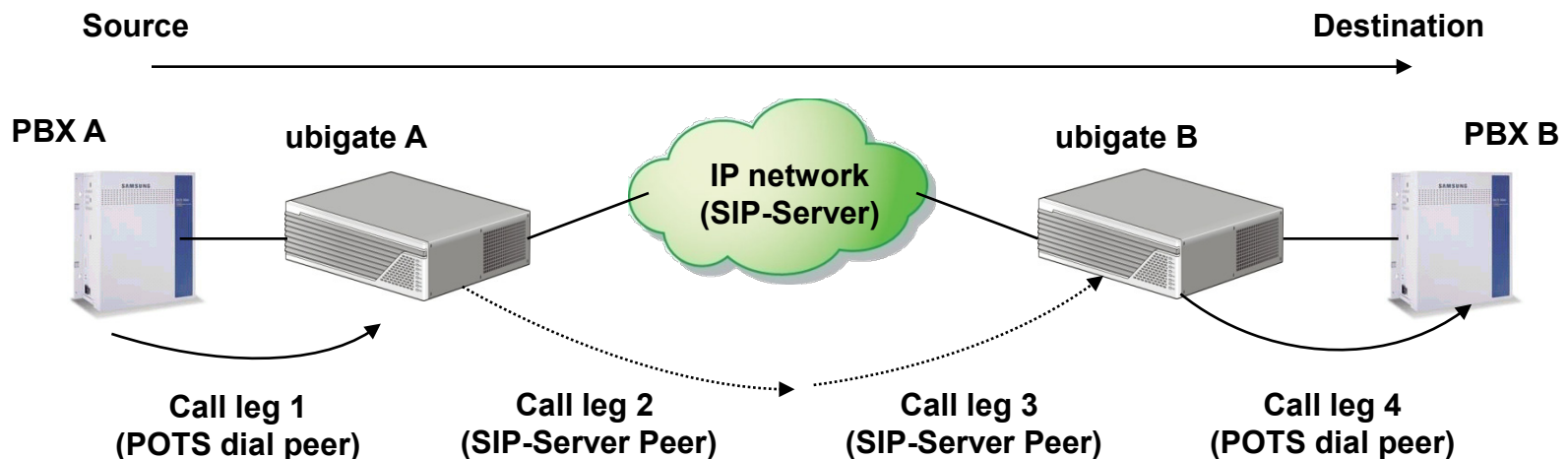
Idle or states of each signal

Dial-Peer Administration

Dial-Peer concept



- Configuring dial peer is an important task for establishing dial plan and transmitting voice on packet network. Using dial peer, the system makes a decision on how to connect a call and identifies origination or termination of a call.
- A call leg is a logical connection between any devices on the path of a voice call, such as gateway and telephony device. In below figure, a voice call consists of four call legs, two call legs for ubigate A and two call legs for ubigate B.



Inbound Dial-Peer



● Inbound Dial Peer

- Inbound dial peer is the origination of a voice call such as a trunk and a subscriber.

● Outbound Dial Peer

- Outbound dial peer is the termination of a voice call such as a trunk and a subscriber.
- Outbound dial peer is determined by matching the destination pattern of outbound dial peer and the called number.

dial-peer command summary



● dial-peer voice pots <tag>

|-- authentication

|-- destination-pattern

|-- register e164

|-- port

|-- trunkgroup-label

|-- trunk-group

|-- direct-inward-dial

|-- description

|-- shutdown

Express Setup for Analog Line

- Use 'authentication' command in pots dial-peer configuration mode to enter information for SIP digest authentication per dial-peer. It is used when a dial peer or a subscriber registers with the SIP server.

- **syntax :**

- authentication <username> <passwd>

- **parameters:**

- username: string
 - passwd: any number of strings

- **example :**

```
iBG/configure# dial-peer voice pots 2
```

```
iBG/configure/dial-peer/voice/pots 2# authentication test1 *****
```

- **cautions:**

- *Passwords are displayed in an encrypted form if the 'secure_passwords' is configured.*

- This command is used to set prefix or full E164 number in dial-peer. To cancel, use no command.

- **Syntax :**

- [no] destination-pattern < string >

- **Example :**

iBG/configure/dial-peer/voice/pots 10# destination-pattern 3244



- This is the command that associates a specific voice port with a dial peer. To delete from dial peer, use no form of this command.

- **Syntax:**

- [no] port {slot /[sub-slot]/port | slot/[subslot]/port:D}

- **Example:**

- Assigns port 0/0/1 to pots dial peer 10.
iBG/configure# dial-peer voice pots 10
iBG/configure/dial-peer/voice/pots 10# port 0/0/1

- **Caution:**

- It is not needed to associate a port with a dial peer when you configure a trunk-group.

- To register dial-peer with a SIP server, use 'register' command in the pots dial-peer configuration mode. Register digits string set in destination-pattern is used as username during registration.

- **Syntax :**

- [no] register e164

- **Caution:**

- If you do not configure this command for a dial-peer, it is not registered with the SIP server.

- **Example :**

- iBG/configuration# dial-peer voice pots 1

- iBG/configuration/dial-peer/voice/pots 1# register e164

show dial-peer voice summary



● show dial-peer voice { all | num <num> | summary }

dial-peer tag

Shutdown/no shutdown

```
iBG1003# show dial-peer voice summary
```

| ID | TYPE | DEST-PATTERN | PORT | ADMIN | OPER | OUT-STAT |
|------|------|--------------|----------|-------|------|----------|
| 3100 | POTS | 3100 | 2/0 | up | up | up |
| 3101 | POTS | 3101 | 2/1 | up | up | up |
| 3102 | POTS | 3102 | 2/2 | up | up | up |
| 3103 | POTS | 3103 | 2/3 | up | up | up |
| 3121 | POTS | 3121 | 2/21 | up | up | up |
| 3122 | POTS | 3122 | 2/22 | up | up | up |
| 3123 | POTS | 3123 | 2/23 | up | up | up |
| 3200 | POTS | 3120 | 0/0/2 | up | up | up |
| 3201 | POTS | 3201 | 0/0/3 | up | up | up |
| 3333 | POTS | trunkgroupA | TG:2 | up | up | up |
| | | | lincroft | | | |
| | | | denver | | | |

State "down" in case that

- Trunk Alarm,
- Busyout,
- Admin shutdown,
- port unconfigured,
- TG unconfigured

Trunkgroup
domain

Physical multiple
trunk-group

Analog line:

- destination-pattern and voice-port must be configured.

Trunk:

- trunkgroup-label and voice-port (or Trunk-group) must be configured.

Module Administration

module command summary



● |-- module

```
| |-- t1
| | |-- framing
| | |-- linecode
| | |-- yellow_alarm
| | |-- clock_source
| | |-- enable
| | |-- alarms
| | | |-- thresholds
| | | | |-- user
| | | |-- hierarchy
| | |-- linemode
| | | |-- csu
| | | |-- dsx
| | |-- circuitId
| | |-- contactInfo
| | |-- description
| | |-- name
| | |-- loopback_framing
```

● |-- module

```
| |-- e1
| | |-- framing
| | |-- linecode
| | |-- yellow_alarm
| | |-- clock_source
| | |-- enable
| | |-- alarms
| | | |-- thresholds
| | | | |-- user
| | | |-- hierarchy
| | |-- linemode
| | |-- circuitId
| | |-- contactInfo
| | |-- description
| | |-- name
```

- This command accesses next-level commands for selecting an T1/E1 link for configuration.

- Syntax

- module t1/e1 <slot/subslot/port>
number : <slot/[subslot]/port> Number of the T1/E1 WAN

- Example

iBG1003/configure# module e1 0/0/0

iBG1003/configure# module t1 0/1

- This command specifies a network timing source for an T1/E1 link.
 - Syntax
 - clock_source { internal | line }
 - internal: Router system's internal clock (default).
 - line: Clock recovered from the incoming T1/E1 signal (loop timing).
 - Example

iBG1003/configure/module/e1 0/0/0# clock_source internal

 - If iBG1003's Reference Clock Source is unique, then "clock_source" command is set to default value (clock_source internal).
 - If iBG1003's Reference Clock Sources are dual, then "clock_source" command set to "line" (clock_source line).

show module configuration

```
iBG1003# 84# show module configuration e1 0/0/0
E1 0/0/0 is ENABLED
Alarm Hierarchy: TRUE,
Yellow Alarm: GENERATE & DETECT
Framing:CRC, LineCode:HDB3, ClockSource:INT, LBO:long haul
CIRCUIT-ID : Not Configured ,CONTACT-INFO : Not Configured ,
DESCRIPTION : Not Configured , LINK NAME : Not Configured ,
```

Line Status:

```
  RLOS:OFF   RAIS:OFF   RLOF:OFF   RRAI:OFF   TAIS:OFF
  TRAI:OFF   TPtrn:OFF   Loop:OFF
```

Timeslot Map:

1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16

| | | | | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| — | — | — | — | — | — | — | — | — | — | — | — | — | — | — | — |

17 18 19 20 21 22 23 24 25 26 27 28 29 30 31

| | | | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| — | — | — | — | — | — | — | — | — | — | — | — | — | — | — |

Mode : VOICE

Analog Line Administration

1. Analog FXS Configuration

- iBG1003/configure# voice-port 0/2/2
 - shutdown
 - signal *loop-start*
 - cptone *locale* (ex. cptone *us*)
 - timeouts timer-name <*timer-values*>
 - timing hookflash-in <*timer-value*>
 - no shutdown
 - exit
- dial-peer voice pots tag
 - destination-pattern 3002
 - port 0/2/2
 - register e164

- Verify that all of ip equipments are working correctly – ping, telnet etc.,
- Setup voice port and dial-peer for each line
- Setup voip gateway and run the voice service start
- Verify that registration is successful
- Call can be made between any combination of endpoints
- Debugging

Analog Trunk Administration

1. Analog FXO Trunk Configuration
2. Analog DID Trunk Configuration

- iBG1003/configure# voice-port 0/2/2
 - shutdown
 - signal { *loop-start* | *ground-start* }
 - dial-type { *dtmf* | *mf* }
 - timing digit *timer-value*
 - timing guard-out *timer-value*
 - no shutdown
 - exit
- dial-peer voice pots tag
 - port 0/2/2

- iBG1003/configure# voice-port 0/2/2
 - shutdown
 - signal did { *immediate* | *wink-start* | *delay-dial* }
 - timing wink-wait *timer-value*
 - timing wink-duration *timer-value*
 - timing delay-duration *timer-value*
 - timing delay-start *timer-value*
 - no shutdown
 - exit
- dial-peer voice pots tag
 - port 0/2/2

Digital Trunk Administration

1. ISDN Interface Administration

Digital trunk related command summary



● Interface bundle <bundle_name>

- link

- |-- bri
 - |-- pri_t1
 - |-- pri_e1

- isdn

- |-- **switch-type**
 - |-- **side**
 - |-- **incoming-voice**
 - |-- **activate**

Express Setup of interface
bundle isdn.

● Interface bundle <bundle_name>

- isdn

- |-- spid1
 - |-- spid2
 - |-- disconnect-cause
 - |-- idle-timeout
 - |-- connect-delay
 - |-- callednum
 - |-- caller
 - |-- answer1
 - |-- answer2
 - |-- keep-alive
 - |-- tei-mode
 - |-- tei-value
 - |-- overlap-receiving
 - |-- calling-number
 - |-- send-alerting
 - |-- progress
 - |-- sending-complete
 - |-- timer
 - | |-- t309
 - | |-- t301

- This command adds or reconfigures WAN bundle interfaces on iBG routers.

- **syntax:**

- [no] interface bundle bundle_name<bundle_name>
- “no” form of this command will remove the bundle

- **example:**

- iBG1003/configure# **interface bundle pri0**

interface bundle link BRI/T1/E1



- This command assigns one or more BRI/E1/T1 links to a WAN/Voice bundle

- **syntax:**

- [no] link <bri | pri_t1 | pri_e1 > port_number <number : timeslot>
- Range:
 - E1 :1-31, T1:1-23, BRI:2

- **example:**

```
iBG1003/configure/interface/bundle pri0# link bri 0/1/1:2  
iBG1003/configure/interface/bundle pri0# link pri_e1 0/1/1  
iBG1003/configure/interface/bundle pri0# link pri_t1 0/2/1:1-23
```

interface bundle isdn side



- Configures the ISDN interface side. The two sides of ISDN interface are network side (NET) and user side (USR).

- **syntax:**

- side side_type < type >
- “no” form of this command does not exist.

- **example:**

- iBG1003/configure/interface/bundle pri1/isdn# **side NET**

interface bundle isdn switch-type



- Use this command to configure the switch type

- **syntax:**

- switch-type switch_type_name <name>
- “no” form of this command does not exist

- **example:**

iBG1003/configure/interface/bundle pri1/isdn# **switch-type primary-euro**

iBG1003/configure/interface/bundle pri1/isdn# **switch-type basic-euro**

interface bundle isdn incoming-voice



- Configures the ISDN port for incoming voice calls.

- **syntax:**

- [no] incoming-voice
- “no” form of this command disables voice on this port

- **example:**

iBG1003/configure/interface/bundle pri1/isdn# **incoming-voice**

interface bundle isdn activate



- This command activates the ISDN services. If it is configured, voice-ports are generated on this link on a channel basis.

- **syntax:**

- [no] activate
- “no” form of this command will set the isdn deactivate

- **example:**

iBG1003/configure/interface/bundle pri1/isdn# **activate**

show isdn interfaces

- Displays all ISDN interface information

- **syntax:**

- show isdn interfaces

```
iBG1003# show isdn interfaces

ISDN Information: pri0
-----
caller                18001111212
answer1               -
answer2               -
called-number         987654
spid1                 -
spid2                 -
idle-timeout          5
connect delay         15
keep-alive             10000
disconnect-cause      17
switch-type           primary-5ess
tei-mode              point-to-point
calling-number         1234567
```



● Enable all debug messages for isdn messages

● syntax:

- [no] debug isdn all bundle-name<bundle_name>
- “no” form of this command will disable this feature.

```
iBG1003# # show debug isdn pri0
```

| Debug level | Status |
|----------------------|----------|
| ===== | ===== |
| Debug cc | Disabled |
| Debug q921 | Disabled |
| Debug q921 Timers | Disabled |
| Debug q931 | Disabled |
| Debug q931 timers | Disabled |
| Debug Physical layer | Disabled |
| Debug datapath | Disabled |

debug isdn
all

```
iBG1003# # show debug isdn pri0
```

| Debug level | Status |
|----------------------|---------|
| ===== | ===== |
| Debug cc | Enabled |
| Debug q921 | Enabled |
| Debug q921 Timers | Enabled |
| Debug q931 | Enabled |
| Debug q931 timers | Enabled |
| Debug Physical layer | Enabled |
| Debug datapath | Enabled |

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