



## Troubleshooting Guide

### Troubleshooting ISDN PRI in AOS

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This troubleshooting guide is intended to help troubleshoot Intergraded Services Digital Network (ISDN) Primary Rate Interface (PRI), using the ADTRAN Operating System (AOS) Command Line Interface (CLI) on IP Business Gateways (IPBGs), but does not include details about the ISDN technology, or configuring the IPBG for initial voice setup.

This guide contains the following sections:

- [IPBG Product and Design Considerations](#)
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For further reference, the following documents can be accessed from ADTRAN's Support Community (<https://supportforums.adtran.com>) for setup, design, and configuration examples:

- [Total Access 900/900e PRI Trunk Quick Configuration Guide \(Article #1957\)](#)
- [PRI Signaling for DSX-1 \(Article #3396\)](#)
- [Configuring the Total Access 900 Series ISDN PRI Interface \(Article #1538\)](#)
- [Caller ID Delivery in AOS Voice Devices \(Article #2301\)](#)
- [Blocking Outbound Caller ID in AOS Voice Devices \(Article #2312\)](#)
- [SIP to PRI Sample Configuration \(Article #3371\)](#)

## IPBG Product and Design Considerations

There are several different product platforms in the IPBG series. If you are unsure what product to choose, it is recommended you consult your ADTRAN reseller (<http://adtran.com/wheretobuy>) or ADTRAN's EN Applications Engineering department ([http://adtran.com/web/page/portal/Adtran/waverunner\\_page\\_support\\_presalesdesignasst](http://adtran.com/web/page/portal/Adtran/waverunner_page_support_presalesdesignasst)) for assistance with product selection.

For a detailed list of supported features for each series of IPBGs, including maximum PRI trunks and maximum concurrent TDM to IP calls, use the AOS Feature Matrix: <http://kb.adtran.com/article.aspx?article=2272&p=2>

## Physical Connectivity Considerations

The Total Access (TA) 900 and 900e series of IPBGs do not support the termination of a TELCO ISDN circuit into the DSX port, because the DSX port does not have the appropriate T1 surge protection required to protect the IPBG and the TELCO's equipment from electrical damage.

The interface type, on the IPBG and the device to which you are connecting the PRI trunk to, will need to be considered in order to select the appropriate cabling type. A straight through cable is used between differing interface pinouts, while a T1 crossover cable is used between interfaces with the same pinout. If the connected device has a non standard interface pinout, a custom cable will need to be crafted to line the transmit and receive pairs up correctly. Below you can find a chart indicating which ports on the various IPBGs can be used to terminate a voice trunk and its interface type (figure 1.), and a chart labeling the DS1 interface pinout (figure 2.) and DSX interface pinout (Figure 3.).

IPBG Voice Port Interface Type

Platform	Voice T1 Port(s)	Interface Type
TA 900 Series	T1 0/2	DSX
TA 900 Series ADSL	T1 0/1	DSX
TA 900e Series	T1 0/3 and T1 0/4	DS1
NetVanta 640	T1 0/1 – T1 0/4	DS1
NetVanta 6240	T1 0/1 – T1 0/4	DS1
NetVanta 6310	T1 0/1	DSX*
NetVanta 6355**	T1 1/1 or T1 2/1	DS1

Figure 1.

\* Port is pinned out as DSX but is actually a DS1 with the necessary surge protection to connect to the PSTN

\*\* Has to be a T1/PRI VIM and not a standard T1 NIM

**DS1 Pin-out**

<b>Pin</b>	<b>Name</b>	<b>Description</b>
1	R1	Receive data from the network (Ring 1)
2	T1	Receive data from the network (Tip 1)
3	-	Unused
4	R	Transmit data toward the network (Ring)
5	T	Transmit data toward the network (Tip)
6-8	-	Unused

Figure 2.

**DSX Pin-out**

<b>Pin</b>	<b>Name</b>	<b>Description</b>
1	R	Transmit data toward the network (Ring)
2	T	Transmit data toward the network (Tip)
3	-	Unused
4	R1	Receive data from the network (Ring 1)
5	T1	Receive data from the network (Tip 1)
6-8	-	Unused

Figure 3.

## Reading an ISDN Message

An ISDN PRI communicates with an IPBG using the Q.931 message protocol. To view the Q.931 messages within the IPBGs, ISDN debug commands initiate a protocol analyzer which interprets the Q.931 messages into a more “human friendly” readable format. In the CLI, use the debug command **debug isdn L2-formatted** to see ISDN Q.931 messages. A detailed explanation of Q.931 messages are covered in the [Call Flow for ISDN PRI](#) section.

## Troubleshooting PRI D-channel Establishment

Access to the IPBG is required in order to troubleshoot problems establishing an ISDN PRI D-channel. Troubleshooting can be performed using the CLI.

The output below is from the **debug isdn l2-formatted** command. This debug information shows the ISDN/Q.931 messages between an IPBG, acting in the Network (NT) role, and a PBX in User role (TE):

The **SABME** message is **sent** by the IPBG to establish the ISDN PRI D-channel.

```
ISDN.L2_FMT PRI 1 =====
ISDN.L2_FMT PRI 1 Sent Sapi:00 C/R:C Tei:00 Ctl: SABME P:1
ISDN.L2_FMT PRI 1 =====
```

The remote device responds with a **UA** message to acknowledge the SABME. This message is received (**Recd**) by the IPBG.

```
ISDN.L2_FMT PRI 1 =====
ISDN.L2_FMT PRI 1 Recd Sapi:00 C/R:C Tei:00 Ctl: UA F:1
ISDN.L2_FMT PRI 1 =====
```

After reception of this message, the ISDN PRI transitions to the “**UP**” status. Issue the **show interface pri 1** command to verify the status of the PRI interface on the IPBG. Further, there should be a “**D**” at the end of the channel status section output indicating the ISDN PRI D-channel is “UP.”

IPBG# **show interface pri 1**

pri 1 is **UP**

Description: PRI INTERFACE

Switch protocol: National ISDN 2

Signaling role: network (NT) b-channel-restarts disabled

Ringling signal: calling TE provides audible ringing

Calling-party override: disabled  
Calling-party presentation: allowed  
Calling-party number: (no number configured)  
digits transferred all  
ISDN name-delivery: disabled  
progress indicator #8 in alerting message: enabled  
progress indicator #2 in connect message: disabled  
progress indicator #1 in setup message: disabled  
progress indicator #3 in setup message: disabled  
progress indicator location: public  
resource-selection: circular descending  
TBCT: disabled  
area code:  
Transmission of redirecting numbers is enabled  
Redirecting numbers: sent as received  
Connected interface: t1 0/2 tdm-group 1  
Channel status 123456789012345678901234

.....D

Legend: - = Unallocated    . = Inactive  
A = Active B channel    B = Backup D channel  
D = Active D channel    M = Maintenance  
R = Restart

1220 packets input, 4880 bytes, 0 no buffer  
0 runts, 0 giants, 0 throttles  
0 input errors, 0 CRC, 0 frame  
0 abort, 0 discards, 0 overruns  
1220 packets output, 4880 bytes, 0 underruns

### Common Issues

If the PRI interface is in a “DOWN” state verify the following settings:

- Make sure the T1 interface is “UP.” (To view the T1 status issue the **show interface t1 0/x** command, where “X” represents the T1 voice port configured to connect to the ISDN PRI.)

IPBG# **show interface t1 0/2**

!

t1 0/2 is **UP**

**Receiver has no alarms**

T1 coding is B8ZS, framing is ESF

FDL type is ANSI

Line build-out is 0dB

No remote loopbacks, No network loopbacks

Acceptance of remote loopback requests enabled

Tx Alarm Enable: rai  
Last clearing of "show interface" counters 05:09:02  
loss of frame : 0  
loss of signal : 0  
AIS alarm : 0  
Remote alarm : 0

DS0 Status: 123456789012345678901234  
XXXXXXXXXXXXXXXXXXXXXXXXXXXX  
Status Legend: '-' = DS0 is not allocated  
'X' = DS0 is allocated (nailed)

Signaling Bit Status: 123456789012345678901234  
RxA: 100000100000001000000010  
RxB: 000000101000001000000010  
  
TxA: 100000111000001000000011  
TxB: 100000100000001010000010  
123456789012345678901234

Line Status: -- No Alarms --

5 minute input rate 0 bits/sec, 0 packets/sec  
5 minute output rate 0 bits/sec, 0 packets/sec  
Current Performance Statistics:  
0 Errored Seconds, 0 Bursty Errored Seconds  
0 Severely Errored Seconds, 0 Severely Errored Frame Seconds  
0 Unavailable Seconds, 0 Path Code Violations  
0 Line Code Violations, 0 Controlled Slip Seconds  
0 Line Errored Seconds, 0 Degraded Minutes

TDM group 1, line protocol is not set  
Encapsulation is not set

- Verify the T1 cable is physically connected to the proper voice port on the IPBG, and it is the correct type of cable (see the section on [Physical Connectivity Considerations](#)).
- Verify the T1 timing is configured correctly, where one device is the master, and the other is the slave.
- Ensure the T1 interface is enabled with the “**no shutdown**” command, and has the correct settings: line framing, line coding, and the proper **number of channels** are configured. (To view the running configuration of the T1 interface, issue the **show run interface t1 0/x** command, where “X” represents the T1 voice port configured to connect to the ISDN PRI.)

```
IPBG# show run interface t1 0/2
!  
interface t1 0/2  
  tdm-group 1 timeslots 1-24 speed 64  
  no shutdown
```

*\*Note: ESF line framing and B8ZS line coding are default settings and will not show in the running configuration.*

*\*Note: Channel 24 must be included when specifying timeslots because it is the dedicated channel that carries the PRI signaling.*

- Verify the PRI interface is enabled with the “**no shutdown**” command, and the PRI is **connected to the proper T1 interface**. (To view the running configuration of the PRI interface, issue the **show run interface pri** command.)

```
IPBG# show run interface pri
!  
interface pri 1  
  description PRI INTERFACE  
  connect t1 0/2 tdm-group 1  
  role network b-channel-restarts disable  
  no shutdown
```

- Make sure the ISDN PRI B-channel **resource-selection** on the IPBG’s ISDN voice trunk corresponds to that setting on the PBX (The default setting on the IPBG is **resource-selection circular descending**). (To view the running configuration of the ISDN trunk, issue the **show run voice trunk** command.)

```
IPBG# show run voice trunk
!  
voice trunk T02 type isdn  
  resource-selection circular descending  
  caller-id-override number-inbound 9045558820  
  connect isdn-group 1  
  rtp delay-mode adaptive  
  rtp dtmf-relay inband  
  codec-group DEFAULT
```

- Verify the PRI role on the IPBG and PBX. The **role** command is used to configure the interface protocol the PRI will use. When connecting to a PBX, typically the IPBG is configured in a PRI **network** role (**network b-channel-restarts disable** is the default configuration for IPBGs), and the PBX acts as the user. (To modify the role use the following command: (config-pri x) **role**

[network | network b-channel-restarts disable | network b-channel-restarts enable | user)

After the above settings have been verified, if **SABME** messages are continuously sent from the IPBG and no ISDN messages are received from the other device, then the other device's settings need to be checked.

```
ISDN.L2_FMT PRI 1 =====
ISDN.L2_FMT PRI 1 Sent Sapi:00 C/R:C Tei:00 Ctl: SABME P:1
ISDN.L2_FMT PRI 1 =====
ISDN.L2_FMT PRI 1 Sent Sapi:00 C/R:C Tei:00 Ctl: SABME P:1
ISDN.L2_FMT PRI 1 =====
ISDN.L2_FMT PRI 1 Sent Sapi:00 C/R:C Tei:00 Ctl: SABME P:1
ISDN.L2_FMT PRI 1 =====
ISDN.L2_FMT PRI 1 Sent Sapi:00 C/R:C Tei:00 Ctl: SABME P:1
ISDN.L2_FMT PRI 1 =====
```

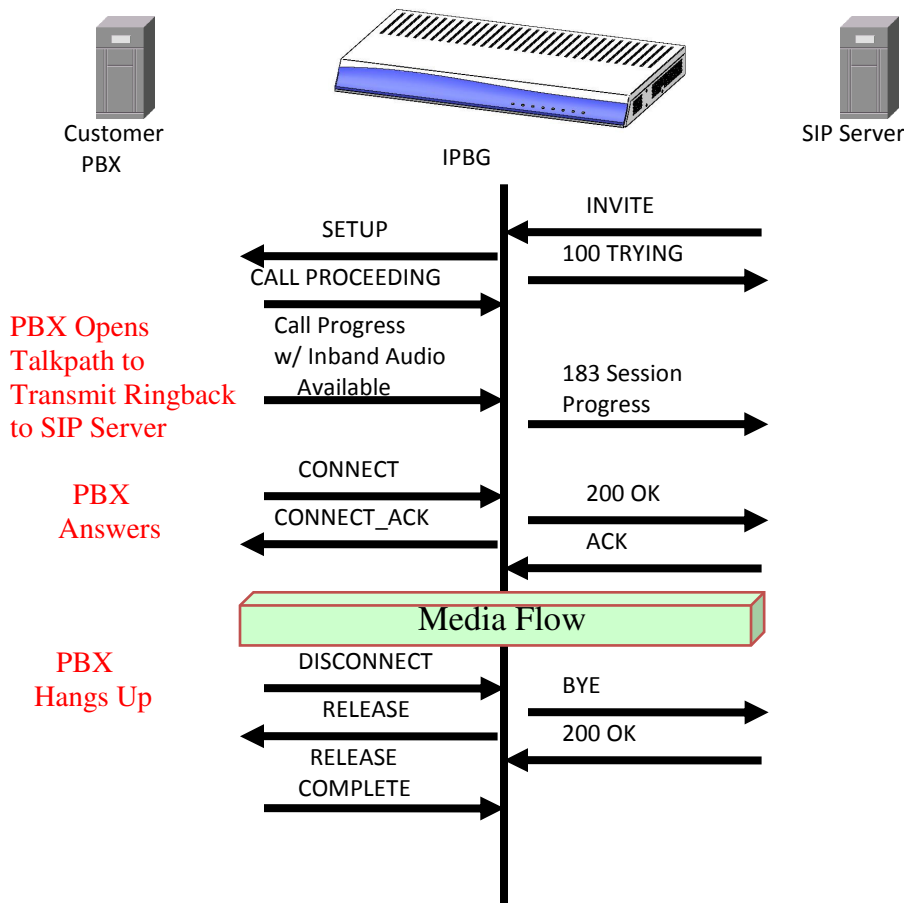


## Call Flow for ISDN PRI

### Example 1.

The debug output below shows the Q.931 messages for a call that will be routed out the PRI interface on the IPBG toward a PBX (PBX Provides Ringback). This call was initiated from the PSTN and send to the IPBG from the SIP server:

*\*Note: Further call flow scenarios are shown in the diagrams in the [Appendix](#) (Examples 2-4.), but the ISDN Q.931 debug information is not.*



When **debug isdn l2-formatted** is enabled, the line labeled “M – XX” will indicate the type of message being passed. You can trace a single ISDN call with a **call reference value (CRV)**, which the IPBG records in hex. All transmitted (**Sent**) messages (Sent from the IPBG) in the call leg in the example will include a **CRV** of 0009. All received (**Recd**) messages will have a **CRV** of 8009. The last 3 characters of the **CRV** (009 in the example) should always be identical, so searching for messages including these characters is commonly the most efficient way to quickly track a single call.

The **SETUP** message below was transmitted (**Sent**) from the IPBG. Notice the **Calling Party Number** and **Called Party Number** in this initial **SETUP** message. Proper formatting of the Number Type, Number Plan, and manipulation of these fields with the ISDN number templates can affect call completion.

```

ISDN.L2_FMT PRI 1 =====
ISDN.L2_FMT PRI 1 Sent = Sapi:00 C/R:C Tei:00
ISDN.L2_FMT PRI 1      Ctl:INFO  Ns:33 Nr:8
ISDN.L2_FMT PRI 1      Prot:08 CRL:2 CRV:0009
ISDN.L2_FMT PRI 1      M - 05 SETUP
ISDN.L2_FMT PRI 1      IE - 04 BEARER CAPABILITY Len=3
ISDN.L2_FMT PRI 1          80 Xfer Cap.:SPEECH
ISDN.L2_FMT PRI 1          90 Xfer Rate:64k
ISDN.L2_FMT PRI 1          A2 Layer 1:u-Law
ISDN.L2_FMT PRI 1      IE - 18 CHANNEL ID      Len=3
ISDN.L2_FMT PRI 1          A1 Primary Rate
ISDN.L2_FMT PRI 1          Intfc ID:IMPLICIT
ISDN.L2_FMT PRI 1          Pref/Excl:PREFERRED
ISDN.L2_FMT PRI 1          D-Chan Indicated:NO
ISDN.L2_FMT PRI 1          Chan. Sel:FOLLOWS
ISDN.L2_FMT PRI 1          83 Numb/Map:NUMBER
ISDN.L2_FMT PRI 1          81 Channel:1
ISDN.L2_FMT PRI 1      IE - 1C FACILITY      Len=14
ISDN.L2_FMT PRI 1          Calling Name: not available
ISDN.L2_FMT PRI 1      IE - 6C CALLING PARTY # Len=12
ISDN.L2_FMT PRI 1          00 Numb. Type:UNKNOWN
ISDN.L2_FMT PRI 1          Numb. Plan:UNKNOWN
ISDN.L2_FMT PRI 1          80 Presentation:ALLOWED
ISDN.L2_FMT PRI 1          Ph.# 2395553300
ISDN.L2_FMT PRI 1      IE - 70 CALLED PARTY # Len=11
ISDN.L2_FMT PRI 1          80 Numb. Type:UNKNOWN
ISDN.L2_FMT PRI 1          Numb. Plan:UNKNOWN
ISDN.L2_FMT PRI 1          Ph.# 4421
ISDN.L2_FMT PRI 1 =====
    
```

The **CALL\_PROC** message below was received (**Recd**) from the device on the other end of the PRI. This message indicates that the other device has started to process the call. This is indicated because the IPBG received (**Recd**) this message.

```

ISDN.L2_FMT PRI 1 =====
ISDN.L2_FMT PRI 1 Recd Sapi:00 C/R:R Tei:00 INFO Ns:26 Nr:18 P:0
ISDN.L2_FMT PRI 1      Prot:08 CRL:2 CRV:8009
    
```

```

ISDN.L2_FMT PRI 1 M - 02 CALL_PROC
ISDN.L2_FMT PRI 1 IE - 18 CHANNEL ID Len=3
ISDN.L2_FMT PRI 1 A9 Primary Rate
ISDN.L2_FMT PRI 1 Intfc ID:IMPLICIT
ISDN.L2_FMT PRI 1 Pref/Excl:EXCLUSIVE
ISDN.L2_FMT PRI 1 D-Chan Indicated:NO
ISDN.L2_FMT PRI 1 Chan. Sel:FOLLOWS
ISDN.L2_FMT PRI 1 83 Numb/Map:NUMBER
ISDN.L2_FMT PRI 1 8f Channel:15
ISDN.L2_FMT PRI 1 =====
  
```

The next message received (**Recd**) from the other device is the **PROGRESS** message. This particular message indicates that the call is being connected and that there is audio available over the talk path. This audio could be the ringing tone, or it could be other call progress tones, such as a busy signal.

```

ISDN.L2_FMT PRI 1 =====
ISDN.L2_FMT PRI 1 Recd Sapi:00 C/R:R Tei:00 INFO Ns:27 Nr:18 P:0
ISDN.L2_FMT PRI 1 Prot:08 CRL:2 CRV:8009
ISDN.L2_FMT PRI 1 M - 03 PROGRESS
ISDN.L2_FMT PRI 1 IE - 1E PROGRESS INDICATOR Len=2
ISDN.L2_FMT PRI 1 80 Location:U
ISDN.L2_FMT PRI 1 88 Description:INBAND AUDIO AVAIL
ISDN.L2_FMT PRI 1 =====
  
```

The **CONNECT** message received (**Recd**) below indicates that the called party has answered the call.

```

ISDN.L2_FMT PRI 1 =====
ISDN.L2_FMT PRI 1 Recd Sapi:00 C/R:R Tei:00 INFO Ns:28 Nr:18 P:0
ISDN.L2_FMT PRI 1 Prot:08 CRL:2 CRV:8009
ISDN.L2_FMT PRI 1 M - 07 CONNECT
ISDN.L2_FMT PRI 1 =====
  
```

The **CONNECT\_ACK** message is sent to acknowledge that the IPBG received the CONNECT message from the other device.

```

ISDN.L2_FMT PRI 1 =====
ISDN.L2_FMT PRI 1 Sent Sapi:00 C/R:C Tei:00 INFO Ns:18 Nr:29 P:0
ISDN.L2_FMT PRI 1 Prot:08 CRL:2 CRV:0009
  
```

```
ISDN.L2_FMT PRI 1 M - 0F CONNECT_ACK
ISDN.L2_FMT PRI 1 =====
```

The message below indicates that the end user or the network has requested that the call be torn down. That request is seen here in the **DISCONNECT** message. You will typically see this message after the called party/calling party has ended the call.

```
ISDN.L2_FMT PRI 1 =====
ISDN.L2_FMT PRI 1 Recd Sapi:00 C/R:R Tei:00 INFO Ns:29 Nr:19 P:0
ISDN.L2_FMT PRI 1 Prot:08 CRL:2 CRV:8009
ISDN.L2_FMT PRI 1 M - 45 DISCONNECT
ISDN.L2_FMT PRI 1 IE - 08 CAUSE Len=2
ISDN.L2_FMT PRI 1 80 Location:U
ISDN.L2_FMT PRI 1 90 Cause:16 (NORMAL_CLEARING)
ISDN.L2_FMT PRI 1 =====
```

The IPBG sends the **RELEASE** message to indicate that it has released the channel and ended the call.

```
ISDN.L2_FMT PRI 1 =====
ISDN.L2_FMT PRI 1 Sent Sapi:00 C/R:C Tei:00 INFO Ns:19 Nr:30 P:0
ISDN.L2_FMT PRI 1 Prot:08 CRL:2 CRV:0009
ISDN.L2_FMT PRI 1 M - 4D RELEASE
ISDN.L2_FMT PRI 1 =====
```

The last message to be seen is the **RELEASE\_CMP** message that indicates that the previous RELEASE message was received and that the releasing of the channel and call termination process has completed.

```
ISDN.L2_FMT PRI 1 =====
ISDN.L2_FMT PRI 1 Recd Sapi:00 C/R:R Tei:00 INFO Ns:30 Nr:20 P:0
ISDN.L2_FMT PRI 1 Prot:08 CRL:2 CRV:8009
ISDN.L2_FMT PRI 1 M - 5A RELEASE_CMP
ISDN.L2_FMT PRI 1 =====
```

## IPBG Caller-ID Issues (Including 911)

Access to the IPBG is required in order to troubleshoot problems with receiving and/or transmitting caller-ID. Troubleshooting can be performed using either the CLI or the GUI.

### Common Issues

The most common caller-ID issues:

- a) Inbound users are not receiving caller-ID number or name. This is most commonly an issue with the SIP server or the customer PBX, but could also be caused by a configuration issue on the IPBG.
- b) Inbound users are not receiving caller-ID name, but they are receiving caller-ID number. This is most commonly a configuration issue on the SIP server, the IPBG, or the customer PBX.
- c) Outbound calls are not showing the user's caller-ID. This is most commonly a configuration issue on the IPBG or PBX.
- d) Outbound 911 calls are not showing the correct caller-ID. This is most commonly a configuration issue on the IPBG or PBX.

### Configuration Verification

#### *a) No inbound caller-ID*

On inbound calls, the IPBG will always automatically pass a caller-ID number that is received from the SIP server. (See the debugging section to confirm the receipt of caller-ID). The only time this could change is if the SIP trunk on the IPBG is configured to modify the caller-ID. As long as there is not a **caller-id-override number-inbound** command on the SIP trunk (T01), the IPBG will not override the caller-ID before it is transmitted to the PBX.

To view the voice trunk configuration from the CLI, use the command **show run voice trunk**.

```
IPBG# show run voice trunk
!
voice trunk T01 type sip
caller-id-override emergency-outbound 9045558820
```

```

match dnis "XXX5551212" replace ani "9045558820"
sip-server primary sip.server.net
registrar primary sip.server.net
registrar expire-time 600
registrar threshold percentage 25
authentication username "9045558820" password encrypted
"3b2f0cc22317417a711eacefe74ad98224f469d1682ad4f0cfd445dd4a1ef3ea9b0a"
register 9045558820
codec-group DEFAULT
default-ring-cadence internal
!
voice trunk T02 type isdn
resource-selection circular descending
caller-id-override number-inbound 9045558820
connect isdn-group 1
rtp delay-mode adaptive
rtp dtmf-relay inband
codec-group DEFAULT
    
```

In the GUI, this can be verified from the **Voice>Trunk Accounts** page

**Add / Modify / Delete Trunk Accounts**

Use this page to add and configure trunk accounts.

**Add a New Trunk Account**

Trunk Name:

Type:

**Modify/Delete Trunk Account**

Click on a name to edit that trunk's settings.

Trunk Name	ID	Type	Supervision
<a href="#">&lt;No Trunk Name Set&gt;</a>	T01	SIP	SIP
<a href="#">&lt;No Trunk Name Set&gt;</a>	T02	ISDN	ISDN

Browse to the **SIP Trunk** page.

**Edit SIP Trunk**

Use this screen to modify the SIP Trunk configuration.

**Trunk Account Information**

Trunk ID: T01

Type: SIP

Trunk Name:

Reject External:

Max Number Calls: 42

Emergency Caller ID Override: 9045555889 Use Match-Substitution:

Inbound Caller ID Override:

Inbound Caller ID Override Method: Always

Under **Trunk Account Information**, the **Inbound Caller ID Override** box should be blank; otherwise the IPBG will override caller-ID on inbound calls sent to the PBX.

*b) No inbound caller-ID (name only)*

If the user is receiving caller-ID number, but not caller-ID name, the IPBG's PRI configuration should be checked.

From the CLI, check the configuration of the PRI interface with the command **show run interface pri**.

```
IPBG# show run interface pri
!
interface pri 1
  isdn name-delivery setup
  calling-party name-facility-timeout 0
  connect t1 0/3 tdm-group 1
  digits-transferred 4
  role network b-channel-restarts disable
  no shutdown
```

In the example above, the line **isdn name-delivery setup** indicates the method the IPBG is using to transmit caller-ID name to the PBX on the PRI. There are three options for transmitting caller-ID name (display, proceeding, and setup):

```
IPBG(config-pri 1)# isdn name-delivery
display          - Deliver calling party name in setup msg (Display IE)
```

proceeding	- Deliver calling party name after proceeding msg
setup	- Deliver calling party name in setup msg (Facility IE)

If the particular method configured is not working properly, the PBX may be looking to receive caller-ID name via a different method. It is recommended that the inbound call received from the SIP server has been verified as transmitting caller-ID name before adjusting this setting (see the debugging section later in this guide).

*\*Note: By default, the command **no isdn name-delivery** is configured on the PRI interface and will not pass caller-ID name.*

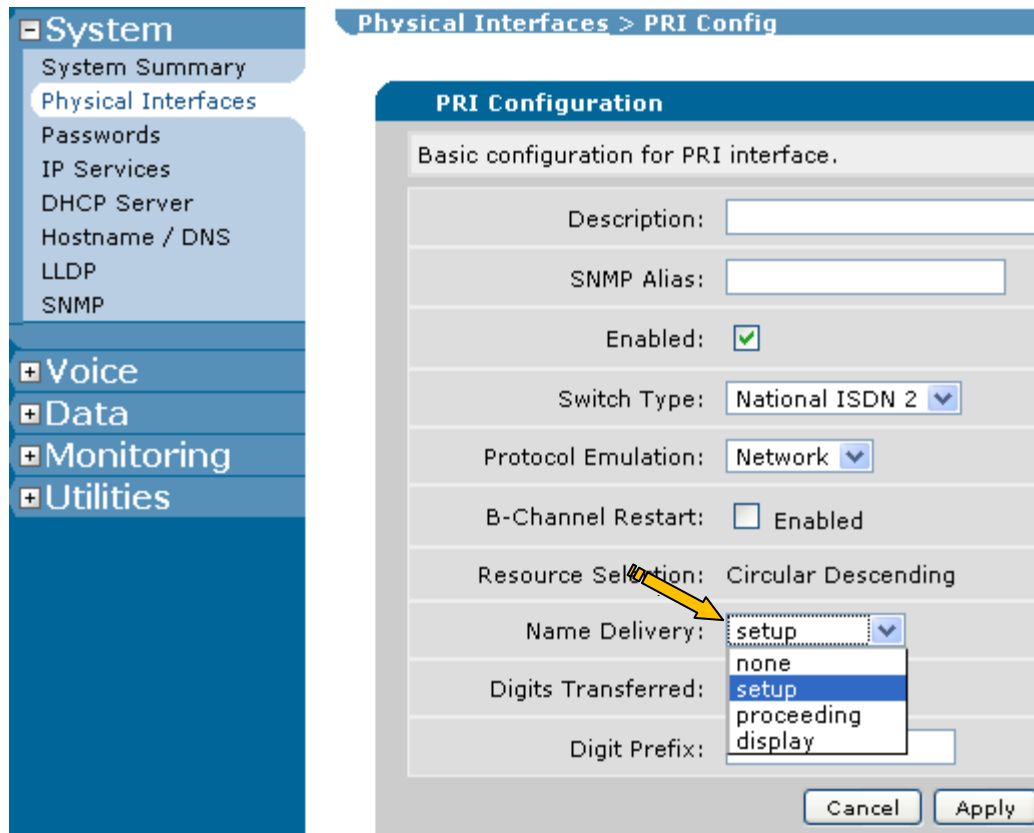
In the GUI, the same option can be checked from the **System>Physical Interfaces>PRI** page

The screenshot shows the 'Physical Interfaces' configuration page. The left sidebar has a menu with 'System' expanded and 'Physical Interfaces' selected. The main content area has a title 'Physical Interfaces' and a description: 'This is a list of all the physical interfaces connected via a plug-in module. View its name.' Below this is a table with two columns: 'Name' and 'Logical Interface'. The table lists various interfaces, with 't1 0/3' having 'pri 1' as its logical interface. A yellow arrow points to the 'pri 1' text.

Name	Logical Interface
<a href="#">t1 0/1</a>	none
<a href="#">t1 0/2</a>	none
<a href="#">t1 0/3</a>	<a href="#">pri 1</a>
<a href="#">t1 0/4</a>	none
<a href="#">eth 0/1</a>	none
<a href="#">eth 0/2</a>	none
<a href="#">fxo 0/0</a>	none
<a href="#">fxs 0/1</a>	none
<a href="#">fxs 0/2</a>	none
<a href="#">fxs 0/3</a>	none
<a href="#">fxs 0/4</a>	none
<a href="#">fxs 0/5</a>	none
<a href="#">fxs 0/6</a>	none
<a href="#">fxs 0/7</a>	none
<a href="#">fxs 0/8</a>	none

Click on the **PRI** under the **Logical Interface** column to go to the PRI configuration page.





From this page, the caller-ID **Name Delivery** option can be checked or changed. If the particular method configured is not working properly, the PBX may be looking to receive caller-ID name via a different method. It is recommended that the inbound call received from the SIP server has been verified as transmitting caller-ID name before adjusting this setting (see the debugging section later in this guide).

*c) Outbound Caller-ID (non 911 calls)*

If outbound caller-ID is not being transmitted properly, the caller-ID settings on the IPBG can be verified from the CLI with the command **show run voice trunk**.

```
IPBG# show run voice trunk
!
voice trunk T01 type sip
caller-id-override emergency-outbound 9045558820
match dnis "XXX5551212" replace ani "9045558820"
sip-server primary sip.server.net
registrar primary sip.server.net
registrar expire-time 600
registrar threshold percentage 25
authentication username "9045558820" password encrypted
"3b2f0cc22317417a711eacefe74ad98224f469d1682ad4f0cfd445dd4a1ef3ea9b0a"
```

```
register 9045558820
codec-group DEFAULT
default-ring-cadence internal
!
voice trunk T02 type isdn
resource-selection circular descending
caller-id-override number-inbound 9045558820
connect isdn-group 1
rtp delay-mode adaptive
rtp dtmf-relay inband
codec-group DEFAULT
```

The highlighted commands indicate that the IPBG is overriding the caller-ID that is being received from the PBX.

*\*Note: These commands are not necessary if the customer is transmitting their own caller-ID number from the PBX.*

In the example output above, the **caller-id-override number-inbound** command on the ISDN trunk changes all caller-ID received from the PRI to the number specified in the command (9045558820 in the example). Every call received from the PBX, regardless of the caller-ID transmitted from the PBX itself, will now show the override number. This override is done as soon as the call is received by the IPBG.

The **match dnis “x” replace ani “y”** commands take action as the call is routed out the trunk. If the customer dials a number that matches one of the “DNIS” templates, the caller-ID for that call will be changed to the value specified in the command. This substitution is performed as one of the last actions of the IPBG before the call is routed out the SIP trunk, so this command will take precedence over the **caller-id-override number-inbound** command.

For example, if the user were to send a call to the IPBG, via the PRI with a DNIS (dialed number) of “963-555-1212,” the following would happen:

- 1) The **caller-id-override number-inbound** command would change the caller-ID on this call to 904-555-8820, regardless of the caller-ID received from the PBX.
- 2) The caller-ID would then be overridden a second time when the call is routed out the SIP trunk, as the dialed number (DNIS) would match the **match dnis XXX5551212 replace ani 9045558820** command. “X” is a wildcard, which matches any digit 1-9. In this case, the “match” command would not actually make any changes, since both commands were overriding with the same number.

*\*Note: Any “match/replace” or “match/substitution” commands will take precedence over the “caller-id-override” command as the “match” commands are one of the last things performed before the call is routed out of the IPBG.*

In the GUI, these commands can be verified from the **Voice>Trunk Accounts** page.

**Voice**

- Stations
  - User Accounts
  - Ring Groups
- Trunks**
  - Trunk Accounts
  - Trunk Groups
- System Setup
  - Classes of Service
  - Dial Plan
  - ISDN Num Templates
  - Codec Lists
  - Call Coverage Lists
  - System Parameters
  - Local SIP Server
  - Local SIP Proxy
  - SIP Client Locations
  - VoIP Settings
  - Email Alerts

**Add / Modify / Delete Trunk Accounts**

Use this page to add and configure trunk accounts.

**Add a New Trunk Account**

Trunk Name:

Type:

**Modify/Delete Trunk Account**

Click on a name to edit that trunk's settings.

Trunk Name	ID	Type	Supervision
<a href="#">&lt;No Trunk Name Set&gt;</a>	T01	SIP	SIP
<a href="#">&lt;No Trunk Name Set&gt;</a>	T02	ISDN	ISDN

Click on the link for the **ISDN Trunk** to verify the caller-id-override that will be performed on calls received from the PRI.

**Edit Trunk**

Use this dialog to modify the Trunk Account configuration.

**Trunk Account Information**

Trunk ID: T02

Type: ISDN

Supervision: ISDN

Trunk Name:

Reject External:

Resource Selection:

Emergency Caller ID Override:  Use Match-Substitution:

Inbound Caller ID Override:

Inbound Caller ID Override Method:

*\*Note: This command is not necessary if the PBX is sending its own caller-ID, as long as it is sending the caller-ID number the SIP server is expecting.*

From the **Voice>Trunk Accounts** page, click on the **SIP Trunk** in order to verify any “match/sub” and “match/replace” commands configured on the **DNIS:ANI Replacement** tab.

**Add New DNIS:ANI Replacement**

Match DNIS Template:  20 characters max.

ANI Replacement:  20 characters max.

ANI Name:  20 characters max.

**View/Modify DNIS:ANI Replacement Entries**

DNIS:ANI Replacement entries are evaluated in the order displayed here. The first template that matches will be used, so make sure you have the templates in the desired order (usually, more specific templates first). HINT: Click on an existing replacement entry to use it as a template for a new entry.

Move	DNIS Match	ANI Replacement	ANI Name
	XXX5551212	9045558820	

This trunk is modifying the ANI (caller-ID) on any calls that match the DNIS (dialed number) formats specified in the entries. For example, if the user were to dial “963-555-1212,” the caller-ID on the call would be changed to 904-555-5889. “X” is a wildcard that matches any digit 1-9.

In this example, the caller-id override on the ISDN trunk and the ANI replacements are both replacing the caller-ID with the same number; therefore, the ANI replacements are not necessary. This configuration will not cause any adverse effects.

#### d) Outbound Caller-ID for 911

Caller-ID on 911 calls can be overridden separately from other calls if necessary. If an override is necessary, from the CLI, this can be configured on the outbound trunk, which is the SIP trunk in most scenarios.

*\*Note: If the caller-ID received from the PBX or the caller-ID being overridden by the IPBG on the non-911 calls is already configured with the same number that should be transmitted on 911 calls, no extra configuration is necessary.*

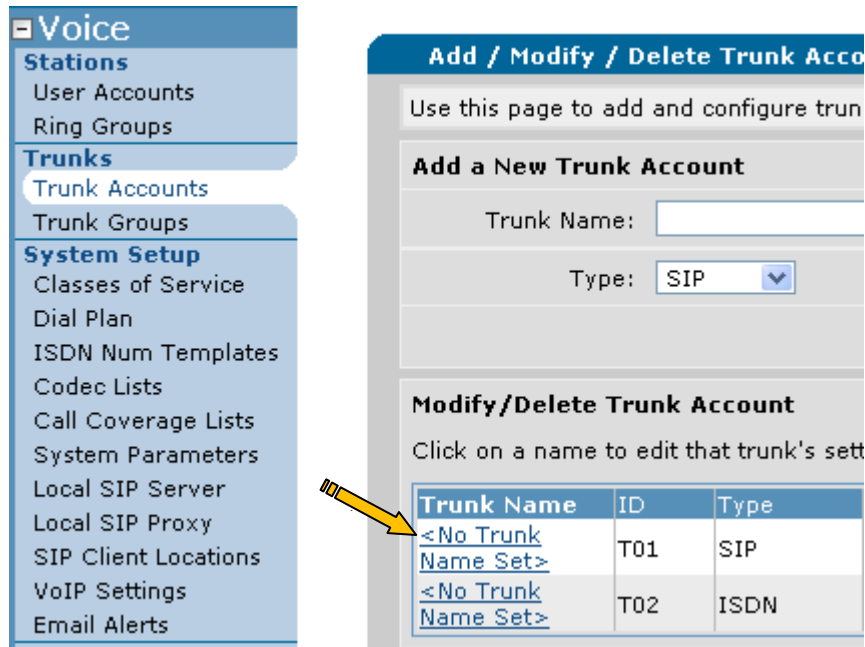
The command **show run voice trunk** will show the current configuration.

```
IPBG# show run voice trunk
!
voice trunk T01 type sip
  caller-id-override emergency-outbound 9045558820
  match dnis "XXX5551212" replace ani "9045558820"
  sip-server primary sip.server.net
  registrar primary sip.server.net
  registrar expire-time 600
  registrar threshold percentage 25
  authentication username "9045558820" password encrypted
  "3b2f0cc22317417a711eacefe74ad98224f469d1682ad4f0cfd445dd4a1ef3ea9b0a"
  register 9045558820
  codec-group DEFAULT
  default-ring-cadence internal
!
voice trunk T02 type isdn
  resource-selection circular descending
  caller-id-override number-inbound 9045558820
  connect isdn-group 1
  rtp delay-mode adaptive
  rtp dtmf-relay inband
  codec-group DEFAULT
```

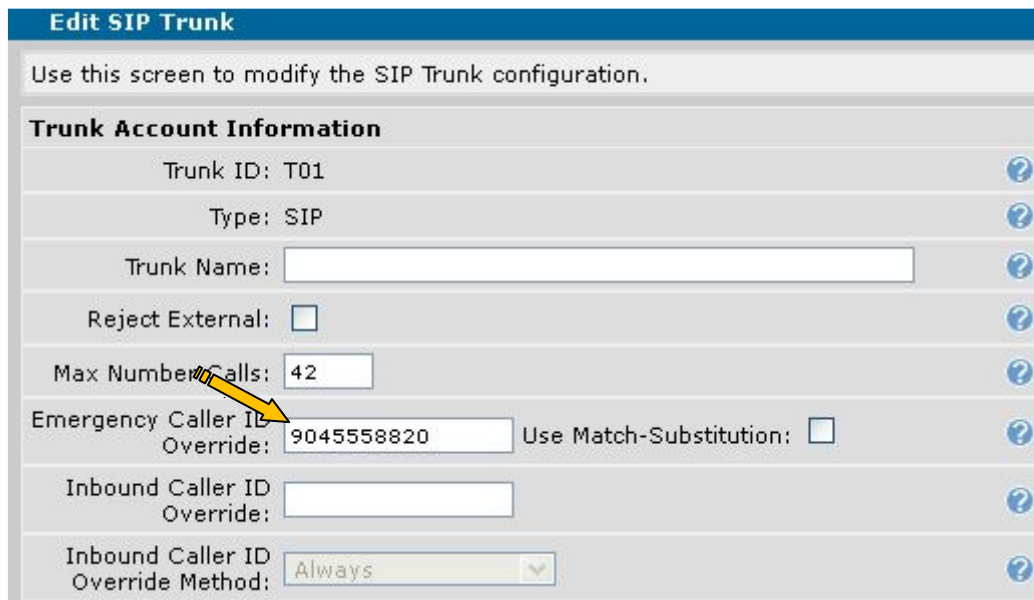
In the above output, the command **caller-id-override emergency-outbound** will override all emergency calls transmitted out this trunk. All 911 calls are considered emergency calls.

*\*Note: This command is configured on the OUTBOUND trunk while the **caller-id-override number-inbound** configuration command is done on the INBOUND trunk.*

In the GUI, this option can be verified by browsing to the **Voice>Trunk Accounts** page. Then navigate to the **SIP Trunk** page.



From here, the **Emergency Caller ID Override** can be verified or configured.



The **Emergency Caller ID Override** will override all emergency calls transmitted out this trunk. All 911 calls are considered emergency calls.

*\*Note: This command is configured on the outbound trunk while the Inbound Caller ID Override is done on the inbound trunk.*

Debugging

Call signaling can be monitored via the IPBG CLI. The following debugs will provide context for a caller-ID issue:

- **debug isdn l2-formatted** – displays the Q.931/PRI signaling negotiated between the IPBG and the customer PBX
- **debug sip stack message** – displays the SIP signaling negotiated between the IPBG and the SIP server

Enabling both of these debugs at the same time will provide quite a bit of output so it may be simpler to verify the ISDN PRI signaling coming from the customer PBX first with only **debug isdn l2-formatted** enabled. All debugs can be disabled from the CLI by entering the command **undebug all** or “**u a**” for short. Once the debugs are enabled, a test call can be placed.

a) *Outbound Caller-ID Example*

*Sample ISDN Debug*

```

ISDN.L2_FMT PRI 2 Recd = Sapi:00 C/R:R Tei:00
ISDN.L2_FMT PRI 2 Ctl:INFO Ns:2 Nr:20
ISDN.L2_FMT PRI 2 Prot:08 CRL:2 CRV:00DE
ISDN.L2_FMT PRI 2 M - 05 SETUP
ISDN.L2_FMT PRI 2 IE - 04 BEARER CAPABILITY Len=3
ISDN.L2_FMT PRI 2 80 Xfer Cap.:SPEECH
ISDN.L2_FMT PRI 2 90 Xfer Rate:64k
ISDN.L2_FMT PRI 2 A2 Layer 1:u-Law
ISDN.L2_FMT PRI 2 IE - 18 CHANNEL ID Len=3
ISDN.L2_FMT PRI 2 A9 Primary Rate
ISDN.L2_FMT PRI 2 Intfc ID:IMPLICIT
ISDN.L2_FMT PRI 2 Pref/Excl:EXCLUSIVE
ISDN.L2_FMT PRI 2 D-Chan Indicated:NO
ISDN.L2_FMT PRI 2 Chan. Sel:FOLLOWS
ISDN.L2_FMT PRI 2 83 Numb/Map:NUMBER
ISDN.L2_FMT PRI 2 83 Channel:3
ISDN.L2_FMT PRI 2 IE - 6C CALLING PARTY # Len=11
ISDN.L2_FMT PRI 2 A1 Numb. Type:NATIONAL
ISDN.L2_FMT PRI 2 Numb. Plan:ISDN/Telephony
ISDN.L2_FMT PRI 2 Ph.# 8035559791
ISDN.L2_FMT PRI 2 IE - 70 CALLED PARTY # Len=12
ISDN.L2_FMT PRI 2 80 Numb. Type:UNKNOWN
ISDN.L2_FMT PRI 2 Numb. Plan:UNKNOWN
ISDN.L2_FMT PRI 2 Ph.# 2705550151
    
```

Notice the **Calling Party #** field in this received **SETUP** message. This indicates that this PBX sent a call to the IPBG from 803-555-9791.

To confirm the actual caller-ID transmitted to the SIP server, a SIP debug will be necessary.

#### Sample SIP Debug

```
SIP.STACK MSG Tx: UDP src=63.230.36.142:5060 dst=209.3.124.215:5060
SIP.STACK MSG INVITE sip:2705550151@209.3.124.215:5060 SIP/2.0
SIP.STACK MSG From:
<sip:8035559791@63.230.36.142:5060;transport=UDP>;tag=2d5fab0-0-13c4-1fdb73-
7659a33b-1fdb73
SIP.STACK MSG To: <sip:2705550151@209.3.124.215:5060>
SIP.STACK MSG Call-ID: 2da0850-0-13c4-1fdb73-9feed2b-
1fdb73@63.230.36.142
SIP.STACK MSG CSeq: 1 INVITE
SIP.STACK MSG Via: SIP/2.0/UDP 63.230.36.142:5060;branch=z9hG4bK-
1fdb73-7c713c5a-18389995
SIP.STACK MSG Max-Forwards: 70
SIP.STACK MSG Supported: 100rel,replaces
SIP.STACK MSG Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY,
OPTIONS, PRACK, REFER, REGISTER
SIP.STACK MSG User-Agent: ADTRAN_Total_Access_908e
SIP.STACK MSG Contact:
<sip:8035559791@63.230.36.142:5060;transport=UDP>
SIP.STACK MSG Content-Type: application/SDP
SIP.STACK MSG Content-Length: 186
SIP.STACK MSG
SIP.STACK MSG v=0
SIP.STACK MSG o=- 1259859813 1259859813 IN IP4 63.230.36.142
SIP.STACK MSG s=-
SIP.STACK MSG c=IN IP4 63.230.36.142
SIP.STACK MSG t=0 0
SIP.STACK MSG m=audio 10120 RTP/AVP 18
SIP.STACK MSG a=rtpmap:18 G729/8000
SIP.STACK MSG a=fmtp:18 annexb=no
SIP.STACK MSG a=silenceSupp:off - - - -
SIP.STACK MSG
```

The **FROM** field in this **INVITE** message indicates the caller-ID being sent to the SIP server is 803-555-9791.

#### b) Inbound Caller-ID Example



*Sample SIP Debug*

```

SIP.STACK MSG Rx: UDP src=10.2.0.40:5060 dst=10.100.110.15:5060
SIP.STACK MSG INVITE sip:2395554421@10.100.110.15:5060;transport=udp
SIP/2.0
SIP.STACK MSG Via: SIP/2.0/UDP 10.2.0.40:5060;branch=z9hG4bK-
e597e1c0f8b29-10.2.0.40-1
SIP.STACK MSG Allow-Events: message-summary, refer, dialog, presence, call-
info
SIP.STACK MSG Max-Forwards: 70
SIP.STACK MSG Call-ID: B379104E@10.2.0.40
SIP.STACK MSG From :
< sip: 2395553300 @10.2.0.40:5060>;tag=10.2.0.40+1+187c74+3838a652;
SIP.STACK MSG To: < sip:2395554421@10.100.110.15>
SIP.STACK MSG CSeq: 478099227 INVITE
SIP.STACK MSG Expires: 180
SIP.STACK MSG Supported: 100rel
SIP.STACK MSG Content-Length: 119
SIP.STACK MSG Content-Type: application/sdp
SIP.STACK MSG Contact: < sip:2395553300@10.2.0.40:5060>;isup-oli=00
SIP.STACK MSG P-Asserted-Identity: < sip:2395553300@10.2.0.40:5060>
SIP.STACK MSG v=0
SIP.STACK MSG o=- 1837201869 1837201869 IN IP4 10.2.0.6
SIP.STACK MSG s=-
SIP.STACK MSG c=IN IP4 10.2.0.6
SIP.STACK MSG t=0 0
SIP.STACK MSG m=audio 36456 RTP/AVP 18 0
SIP.STACK MSG a=ptime:20

```

The **FROM** field in this **INVITE** message indicates the call is being received from 239-555-3300, which will be transmitted to the PBX as caller-ID. Notice the section of the **FROM** field between the “:” and the “<” is empty, which indicates that the SIP server did not send a **caller-ID name**.

The ISDN debug will confirm the actual message being sent to the PBX.

*Sample ISDN Debug*

```

ISDN.L2_FMT PRI 1 Sent = Sapi:00 C/R:C Tei:00
ISDN.L2_FMT PRI 1 Ctl:INFO Ns:33 Nr:8
ISDN.L2_FMT PRI 1 Prot:08 CRL:2 CRV:7BA8
ISDN.L2_FMT PRI 1 M - 05 SETUP
ISDN.L2_FMT PRI 1 IE - 04 BEARER CAPABILITY Len=3
ISDN.L2_FMT PRI 1 80 Xfer Cap.:SPEECH
ISDN.L2_FMT PRI 1 90 Xfer Rate:64k

```

ISDN.L2_FMT PRI 1	A2 Layer 1:u-Law	
ISDN.L2_FMT PRI 1	IE - 18 CHANNEL ID	Len=3
ISDN.L2_FMT PRI 1	A1 Primary Rate	
ISDN.L2_FMT PRI 1	Intfc ID:IMPLICIT	
ISDN.L2_FMT PRI 1	Pref/Excl:PREFERRED	
ISDN.L2_FMT PRI 1	D-Chan Indicated:NO	
ISDN.L2_FMT PRI 1	Chan. Sel:FOLLOWS	
ISDN.L2_FMT PRI 1	83 Numb/Map:NUMBER	
ISDN.L2_FMT PRI 1	81 Channel:1	
ISDN.L2_FMT PRI 1	IE - 1C FACILITY	Len=14
ISDN.L2_FMT PRI 1	Calling Name: not available	
ISDN.L2_FMT PRI 1	IE - 6C CALLING PARTY #	Len=12
ISDN.L2_FMT PRI 1	00 Numb. Type:UNKNOWN	
ISDN.L2_FMT PRI 1	Numb. Plan:UNKNOWN	
ISDN.L2_FMT PRI 1	80 Presentation:ALLOWED	
ISDN.L2_FMT PRI 1	Ph.# 2395553300	
ISDN.L2_FMT PRI 1	IE - 70 CALLED PARTY #	Len=11
ISDN.L2_FMT PRI 1	80 Numb. Type:UNKNOWN	
ISDN.L2_FMT PRI 1	Numb. Plan:UNKNOWN	
ISDN.L2_FMT PRI 1	Ph.# 4421	

Notice the **Calling Party #** field in this transmitted **SETUP** message. This indicates the IPBG sent a call to the PBX from 239-555-3300. The **calling name** received on this particular call was unavailable, so the PBX will see “not available” as the caller-ID name.

## Cannot Receive Inbound Calls

This section requires access to the IPBG in order to troubleshoot the inability to receive inbound calls from the PSTN. This section assumes that outbound calls are working correctly.

### Common Issues

The most common causes for inbound calling failures are:

- a) The number being dialed is an incorrect number – confirm the number being dialed matches the numbers that should be routed to the IPBG.
- b) The SIP server is not properly configured to route calls to the IPBG.
- c) The local PBX is expecting a different number of called party digits than the number the IPBG is configured to send. Confirm the number of digits the PBX should be receiving and then verify the configuration on the IPBG.

### Configuration Verification

Inbound calls may not route properly if the IPBG is not configured properly. Based on the fact that outbound calls work properly, it can be assumed the IPBG's trunks are configured properly, but it may be possible the grouped-trunks are not. This configuration can be checked via the CLI or the GUI.

From the CLI, use the command **show run voice grouped-trunk**

```
IPBG# show run voice grouped-trunk
!  
voice grouped-trunk SIP  
  no description  
  trunk T01  
  accept $ cost 0  
!  
voice grouped-trunk PRI  
  no description  
  trunk T02  
  accept $ cost 0
```

Verify all sections bolded in the example above are included in the configuration, otherwise inbound calls will not route to the customer PBX.

In the GUI, you can browse to the **Voice>Trunk Groups** page.

**Add / Modify / Delete Trunk Groups**

Use this page to add and configure trunk groups.

**Add a New Trunk Group**

Group Name:  *Enter a name for this group.*

**Modify/Delete Trunk Group**

This is a description of this list

Trunk Group	Description	
<a href="#">SIP</a>		<input type="button" value="Delete"/>
<a href="#">PRI</a>		<input type="button" value="Delete"/>

From there, verify the **PRI Trunk Group** is built and then click on it.

**Edit Trunk Group 'PRI'**

Basic configuration for a Trunk Group. Click 'Apply' when done.

**Trunk Group Information**

Trunk Group Name: PRI

Description:

Resource Selection:

**Trunk Group Members**

Below is a list of [Trunk Accounts](#) that are being used in this Trunk Group.

Trunk Account	ID	Type	Supervision	
<a href="#">&lt;No Description Set&gt;</a>	T02	ISDN	ISDN	<input type="button" value="Delete"/>

Verify trunk **T02** is added as a member of the PRI trunk group.

**Detailed View - Permit/Restriction Call Templates**

Permit Template	Cost
\$	Low (0)

**Restriction Template**

There are no configured Restriction Templates

Scroll down to the bottom of the page and click the “+” symbol next to “Detailed View.” Verify there is a permit template of “\$.”

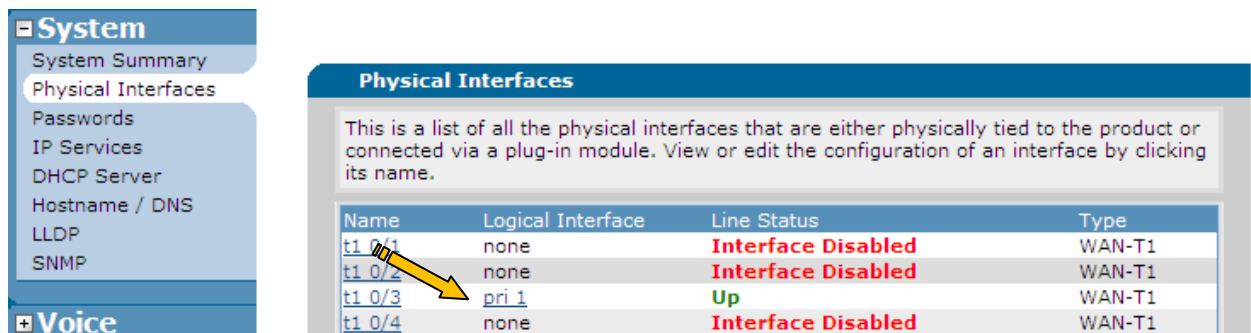
It may also be helpful to verify the number of digits being passed on the PRI towards the PBX. In this example, the PRI is configured to pass 4 digits to the customer PBX.

*\*Note: By default, the PRI interface is configured to **digits transferred all**. This configuration command will not show in the running-configuration.*

From the CLI, use the command **show run interface pri**.

```
IPBG# show run interface pri
!  
interface pri 1  
  isdn name-delivery setup  
  calling-party name-facility-timeout 0  
  connect t1 0/3 tdm-group 1  
  digits-transferred 4  
  role network b-channel-restarts disable  
  no shutdown
```

In the GUI, you can check this option by browsing to the **System>Physical Interfaces>pri 1** page.



The screenshot displays the GUI for configuring physical interfaces. On the left, a sidebar menu shows 'System' expanded, with 'Physical Interfaces' selected. The main content area is titled 'Physical Interfaces' and contains a table with the following data:

Name	Logical Interface	Line Status	Type
t1 0/1	none	Interface Disabled	WAN-T1
t1 0/2	none	Interface Disabled	WAN-T1
t1 0/3	pri 1	Up	WAN-T1
t1 0/4	none	Interface Disabled	WAN-T1

A yellow arrow points to the 'pri 1' entry in the 'Logical Interface' column of the 't1 0/3' row.

Click on **pri 1** interface from the **Logical Interface** column to be taken to the **PRI Configuration** page.

This example configuration shows the PRI configured to send 4 digits to the PBX on inbound calls from the PSTN.

If the configuration has been verified, a CLI debug may be necessary.

### Debugging

Call signaling can be monitored via the IPBG CLI. The following debugs will provide context for an inbound call attempt:

- **debug sip stack message** – displays the SIP signaling negotiated between the IPBG and the SIP server
- **debug isdn l2-formatted** – displays the Q.931/PRI signaling negotiated between the IPBG and the customer PBX

Enabling both of these debugs at the same time will provide quite a bit of output so it may be simpler to verify the signaling coming from the SIP server first with only **debug sip stack message** enabled. All debugs can be disabled from the CLI by entering the command **undebug all** or “**u a**” for short. Once the debugs are enabled, a test call can be placed.

### *Sample SIP Debug*

```
SIP.STACK MSG Rx: UDP src=10.2.0.40:5060 dst=10.100.110.15:5060
SIP.STACK MSG INVITE sip:2395554421@10.100.110.15:5060;transport=udp
```

```

SIP/2.0
SIP.STACK MSG      Via: SIP/2.0/UDP 10.2.0.40:5060;branch=z9hG4bK-
e597e1c0f8b29-10.2.0.40-1
SIP.STACK MSG      Allow-Events: message-summary, refer, dialog, presence, call-
info
SIP.STACK MSG      Max-Forwards: 70
SIP.STACK MSG      Call-ID: B379104E@10.2.0.40
SIP.STACK MSG      From:
<sip: 2395553300 @10.2.0.40:5060>;tag=10.2.0.40+1+187c74+3838a652;
SIP.STACK MSG      To: <sip: 2395554421 @10.100.110.15>
SIP.STACK MSG      CSeq: 478099227 INVITE
SIP.STACK MSG      Expires: 180
SIP.STACK MSG      Supported: 100rel
SIP.STACK MSG      Content-Length: 119
SIP.STACK MSG      Content-Type: application/sdp
SIP.STACK MSG      Contact: <sip:2395553300@10.2.0.40:5060>;isup-oli=00
SIP.STACK MSG      P-Asserted-Identity: <sip:2395553300@10.2.0.40:5060>
SIP.STACK MSG      v=0
SIP.STACK MSG      o=- 1837201869 1837201869 IN IP4 10.2.0.6
SIP.STACK MSG      s=-
SIP.STACK MSG      c=IN IP4 10.2.0.6
SIP.STACK MSG      t=0 0
SIP.STACK MSG      m=audio 36456 RTP/AVP 18 0
SIP.STACK MSG      a=ptime:20

```

Notice the **URI** and **TO** field in this received **INVITE** message indicate the IPBG is receiving a call destined for 239-555-4421. The **FROM** field indicates the call is being received from 239-555-3300.

If an **INVITE** message is never received, and the IPBG's firewall is allowing SIP traffic (typically UDP 5060) through, then the SIP server configuration should be verified, as it does not appear to be sending calls to the IPBG.

If the configuration of the PRI proved correct, the IPBG should then send a call to the PBX.

#### *Sample ISDN Debug*

```

ISDN.L2_FMT PRI 1 Sent = Sapi:00 C/R:C Tei:00
ISDN.L2_FMT PRI 1      Ctl:INFO  Ns:33 Nr:8
ISDN.L2_FMT PRI 1      Prot:08  CRL:2  CRV:7BA8
ISDN.L2_FMT PRI 1      M - 05  SETUP
ISDN.L2_FMT PRI 1      IE - 04 BEARER CAPABILITY  Len=3
ISDN.L2_FMT PRI 1      80 Xfer Cap.:SPEECH
ISDN.L2_FMT PRI 1      90 Xfer Rate:64k

```

ISDN.L2_FMT PRI 1	A2 Layer 1:u-Law
ISDN.L2_FMT PRI 1	IE - 18 CHANNEL ID Len=3
ISDN.L2_FMT PRI 1	A1 Primary Rate
ISDN.L2_FMT PRI 1	Intfc ID:IMPLICIT
ISDN.L2_FMT PRI 1	Pref/Excl:PREFERRED
ISDN.L2_FMT PRI 1	D-Chan Indicated:NO
ISDN.L2_FMT PRI 1	Chan. Sel:FOLLOWS
ISDN.L2_FMT PRI 1	83 Numb/Map:NUMBER
ISDN.L2_FMT PRI 1	81 Channel:1
ISDN.L2_FMT PRI 1	IE - 1C FACILITY Len=14
ISDN.L2_FMT PRI 1	Calling Name: not available
ISDN.L2_FMT PRI 1	IE - 6C CALLING PARTY # Len=12
ISDN.L2_FMT PRI 1	00 Numb. Type:UNKNOWN
ISDN.L2_FMT PRI 1	Numb. Plan:UNKNOWN
ISDN.L2_FMT PRI 1	80 Presentation:ALLOWED
ISDN.L2_FMT PRI 1	Ph.# 2395553300
ISDN.L2_FMT PRI 1	IE - 70 CALLED PARTY # Len=11
ISDN.L2_FMT PRI 1	80 Numb. Type:UNKNOWN
ISDN.L2_FMT PRI 1	Numb. Plan:UNKNOWN
ISDN.L2_FMT PRI 1	Ph.# 4421

Notice the **Calling Party #** and **Called Party #** fields in this transmitted **SETUP** message. This indicates the IPBG sent a call to the PBX from 239-555-3300 to 4421. If the PBX indicated that it did not receive an inbound call, although the debug shows otherwise, have the PBX vendor review the PBX configuration.



## Cannot Make Outbound Calls

This section requires access to the IPBG in order to troubleshoot the inability to make outbound calls to the PSTN. This section assumes that inbound calls are working correctly.

### Common Issues

The most common causes for outbound calling failures are:

- a) The user is dialing an invalid number – have the user confirm the number they are dialing is correct.
- b) The user is dialing an incorrect number of digits for the type of call they are attempting – when this happens, often a message will be played to the user that indicates how they should be dialing. Verify whether or not they are hearing a message or just a busy signal.
- c) The SIP server does not respond to the IPBG's INVITES – After verification the IPBG is transmitting SIP INVITE messages to the SIP server via a SIP debug, but the SIP server never replies, the SIP debug information should be captured and provided to a SIP server engineer for investigation.

If the user has confirmed they are dialing the correct number, and dialing the correct number of digits, the configuration of the IPBG should be checked.

### Configuration Verification

Outbound calls may not route correctly if the IPBG is not configured properly. Based on the fact that calls can be received, it can be assumed that the IPBG's trunks are configured properly, but it may be possible that their grouped-trunks are not. This configuration can be checked via the CLI or the GUI. Verify all sections bolded in the example are included in the configuration, otherwise outbound calls will not route to the SIP server.

From the CLI, use the command **show run voice grouped-trunk**

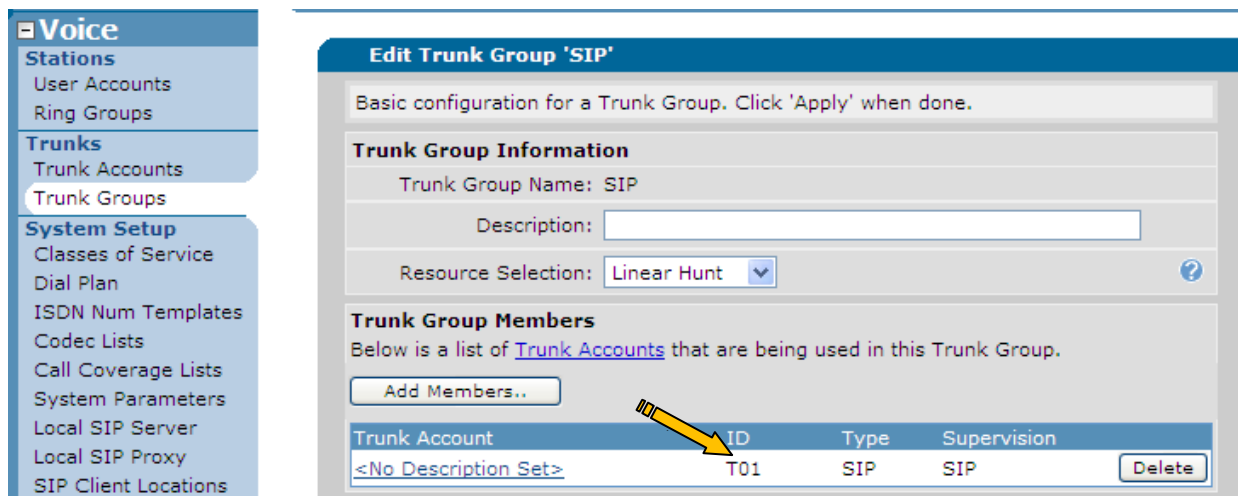
```
IPBG# show run voice grouped-trunk
!  
voice grouped-trunk SIP  
no description  
trunk T01  
accept $ cost 0  
!  
voice grouped-trunk PRI
```

no description  
trunk T02  
accept \$ cost 0

In the GUI, you can browse to the **Voice>Trunk Groups** page.



From there, verify the **SIP Trunk Group** is built and then click on it.



Verify trunk **T01** is added as a member of the **SIP Trunk Group**.



Scroll down to the bottom of the page and click the “+” symbol next to “Detailed View.” Verify there is a permit template of “\$.”

If the configuration has been verified, a CLI debug may be necessary.

### Debugging

Call signaling can be monitored via the IPBG CLI. The following debugs will provide context for an inbound call attempt:

- **debug sip stack message** – displays the SIP signaling negotiated between the IPBG and the SIP server
- **debug isdn i2-formatted** – displays the Q.931/PRI signaling negotiated between the IPBG and the customer PBX

Enabling both of these debugs at the same time will provide quite a bit of output so it may be simpler to verify the signaling coming from the SIP server first with only **debug sip stack message** enabled. All debugs can be disabled from the CLI by entering the command **undebg all** or “**u a**” for short. Once the debugs are enabled, a test call can be placed.

### *Sample ISDN Debug*

```
ISDN.L2_FMT PRI 2 Recd = Sapi:00 C/R:R Tei:00
ISDN.L2_FMT PRI 2 Ctl:INFO Ns:2 Nr:20
ISDN.L2_FMT PRI 2 Prot:08 CRL:2 CRV:00DE
ISDN.L2_FMT PRI 2 M - 05 SETUP
ISDN.L2_FMT PRI 2 IE - 04 BEARER CAPABILITY Len=3
ISDN.L2_FMT PRI 2 80 Xfer Cap.:SPEECH
ISDN.L2_FMT PRI 2 90 Xfer Rate:64k
ISDN.L2_FMT PRI 2 A2 Layer 1:u-Law
ISDN.L2_FMT PRI 2 IE - 18 CHANNEL ID Len=3
ISDN.L2_FMT PRI 2 A9 Primary Rate
ISDN.L2_FMT PRI 2 Intfc ID:IMPLICIT
ISDN.L2_FMT PRI 2 Pref/Excl:EXCLUSIVE
```

```

ISDN.L2_FMT PRI 2      D-Chan Indicated:NO
ISDN.L2_FMT PRI 2      Chan. Sel:FOLLOWS
ISDN.L2_FMT PRI 2      83 Numb/Map:NUMBER
ISDN.L2_FMT PRI 2      83 Channel:3
ISDN.L2_FMT PRI 2      IE - 6C CALLING PARTY # Len=11
ISDN.L2_FMT PRI 2      A1 Numb. Type:NATIONAL
ISDN.L2_FMT PRI 2      Numb. Plan:ISDN/Telephony
ISDN.L2_FMT PRI 2      Ph.# 8035559791
ISDN.L2_FMT PRI 2      IE - 70 CALLED PARTY # Len=12
ISDN.L2_FMT PRI 2      80 Numb. Type:UNKNOWN
ISDN.L2_FMT PRI 2      Numb. Plan:UNKNOWN
ISDN.L2_FMT PRI 2      Ph.# 2705550151

```

Notice the **Calling Party #** and **Called Party #** fields in this received **SETUP** message. This indicates that this PBX sent a call to the IPBG from 803-555-9791 to 270-555-0151. If the call is received and the numbers match what was dialed by the customer, a SIP debug may be necessary.

If the user has indicated that they sent a call, yet the debug did not indicate a received **SETUP** message, the PBX configuration should be verified.

#### Sample SIP Debug

```

SIP.STACK MSG Tx: UDP src=63.230.36.142:5060 dst=209.3.124.215:5060
SIP.STACK MSG INVITE sip: 2705550151 @209.3.124.215:5060 SIP/2.0
SIP.STACK MSG From:
<sip: 8035559791 @63.230.36.142:5060;transport=UDP>;tag=2d5fab0-0-13c4-1fdb73-
7659a33b-1fdb73
SIP.STACK MSG To: <sip: 2705550151 @209.3.124.215:5060>
SIP.STACK MSG Call-ID: 2da0850-0-13c4-1fdb73-9feed2b-
1fdb73@63.230.36.142
SIP.STACK MSG CSeq: 1 INVITE
SIP.STACK MSG Via: SIP/2.0/UDP 63.230.36.142:5060;branch=z9hG4bK-
1fdb73-7c713c5a-18389995
SIP.STACK MSG Max-Forwards: 70
SIP.STACK MSG Supported: 100rel,replaces
SIP.STACK MSG Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY,
OPTIONS, PRACK, REFER, REGISTER
SIP.STACK MSG User-Agent: ADTRAN_Total_Access_908e
SIP.STACK MSG Contact:
<sip:8035559791@63.230.36.142:5060;transport=UDP>
SIP.STACK MSG Content-Type: application/SDP
SIP.STACK MSG Content-Length: 186
SIP.STACK MSG

```

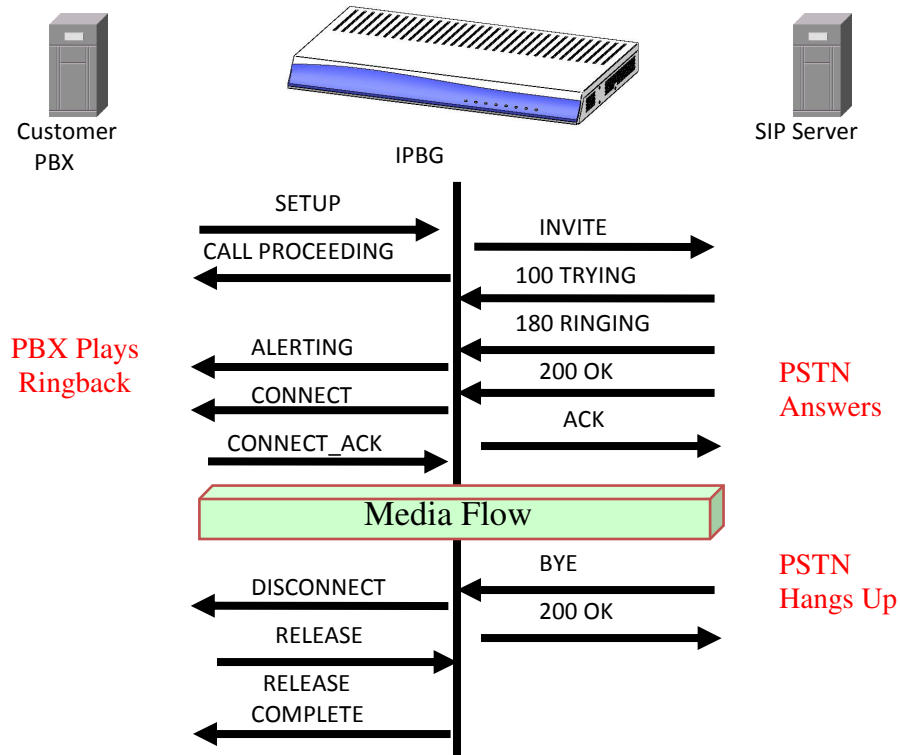
```
SIP.STACK MSG      v=0
SIP.STACK MSG      o=- 1259859813 1259859813 IN IP4 63.230.36.142
SIP.STACK MSG      s=-
SIP.STACK MSG      c=IN IP4 63.230.36.142
SIP.STACK MSG      t=0 0
SIP.STACK MSG      m=audio 10120 RTP/AVP 18
SIP.STACK MSG      a=rtpmap:18 G729/8000
SIP.STACK MSG      a=fmtp:18 annexb=no
SIP.STACK MSG      a=silenceSupp:off - - - -
SIP.STACK MSG
```

Notice the **URI** and **TO** field in this sent **INVITE** message indicate the IPBG is attempting to send this call to 270-555-0151. The **FROM** field indicates the call is being sent from 803-555-9791. The SIP server should respond to the **INVITE** the IPBG sends to it. At this point, if the SIP server does not respond, the SIP debug information should be captured and provided to a SIP server engineer for investigation.

## Appendix

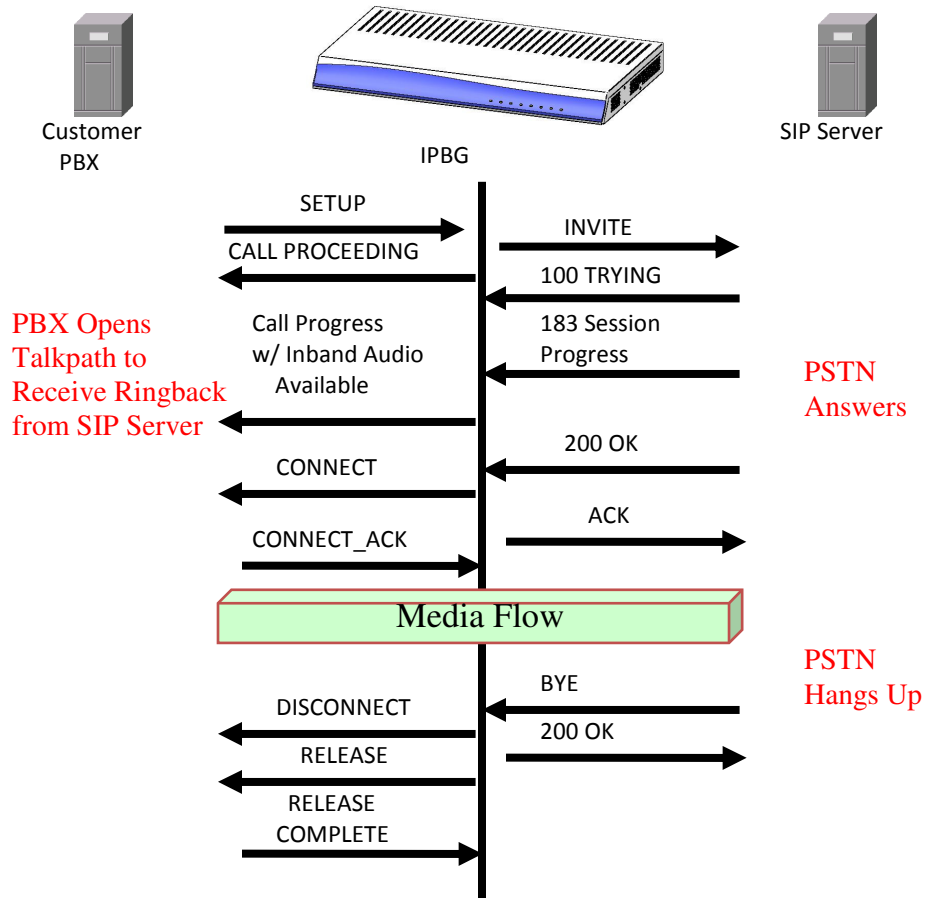
### Example 2.

The diagram below is for a call that is received from the PBX and will be routed out the IPBG's SIP trunk toward the PSTN (PBX Provides Ringback):



Example 3.

In this example, the PBX sends the call to the IPBG via PRI and the SIP Server plays ringback for the caller. However, the 183 Session Progress message could indicate more than just ringback. It could also allow the SIP Server to play a message to the customer, “please dial 1 before this number,” etc.



Example 4.

In this example a call will be routed out the PRI interface on the IPBG toward a PBX (SIP Server Provides Ringback). The call was initiated from the PSTN and sent to the IPBG from the SIP Server:

