

Configuring the Mediatrix® 440x BRI Digital Gateway for Use with the SPA9000 Voice System

This document helps you to configure the Mediatrix® 440x BRI Digital Gateway to provide ISDN-BRI connectivity to your SPA9000 Voice System.

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Network Setup and Physical Installation

This installation process assumes that you already have configured the SPA9000 Voice System. The Mediatrix 440x gateway must be installed in the same local network and in the same VLAN as the SPA9000. If you have any questions about setting up your SPA9000 network, see the documents listed in the Notes at the end of this section.

To initiate installation of Mediatrix 440x gateway, connect the power supply to the power input port and connect the Ethernet network port to the LAN router. The DHCP option on the Mediatrix BRI gateway is active by default, so the gateway automatically gets its IP address from the network DHCP server, if available.

Note: Although DHCP is active by default, it is recommended that you use a static IP address. This step is configured in the following section, [“Configuring the Network Address,”](#) on page 4.

The following figure illustrates a deployment with the Mediatrix 4402 BRI Gateway.

Figure 1 Deployment of the SPA9000 Voice System with a Mediatrix 4402 BRI Gateway



Notes:

- A SPA9000 is required.
- At least one SPA IP phone is required. Several SPA models are available. If you have a wireless network, the WIP310 Wireless-G IP phone also can be connected.
- The SPA9000 provides two ports for the connection of analog phones or fax machines, if needed.

- The optional SPA400 can be used for integrated voice mail service or PSTN access.
- The SLM224P switch and the WRV200 router are recommended equipment for the SPA9000 Voice System.
- Any Mediatrix 440x series BRI Gateway can be used.

Refer to the following sources for more information about your Mediatrix equipment or your SPA9000 Voice System.

- Mediatrix Documentation: The following Mediatrix documentation can be found at www.mediatrix.com.
 - Configuration Notes 0245 Mediatrix 4400 VoIP Gateway PSTN Scenario
 - Mediatrix 4401/4402/4404 (BRI) Software Configuration Guide
 - Mediatrix 4400 Product Brochure
- Cisco SPA9000 Voice System Documentation: The following SPA9000 Voice System documentation can be found at www.cisco.com/go/smallbiz.
 - SPA9000 Voice System Installation and Configuration Guide Using the Setup Wizard
 - SPA9000 Voice System Installation and Configuration Guide - Web-UI (Legacy) Based Product Configuration
 - SPA9000 Voice System Administration Guide

Configuring the Mediatrix 440x Gateway

This section provides procedures on how to configure the Mediatrix 440x BRI Digital Gateway for use in your SPA9000 Voice System. It is recommended that you follow these procedures in the order in which they are presented.

Note: This document describes manual configuration, although the Mediatrix 440x gateway series supports remote provisioning. For more information about the remote provisioning feature, contact Mediatrix.

Logging In to the Mediatrix 440x Gateway Web Interface

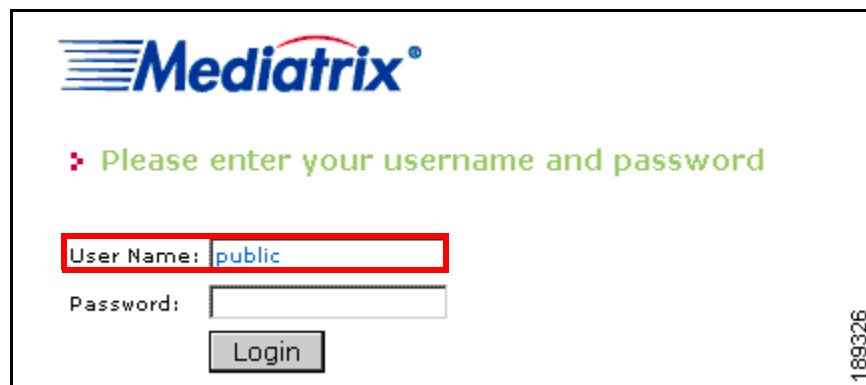
To access the Mediatrix 440x gateway web interface, complete the following steps.

Step 1. Connect a PC to the same LAN as your SPA9000 Voice System.

Check your router's DHCP table to determine the IP address that was assigned automatically when you connected the Gateway. In a web browser, enter the IP address of the Mediatrix gateway. In this example, the IP address is 192.168.1.75.

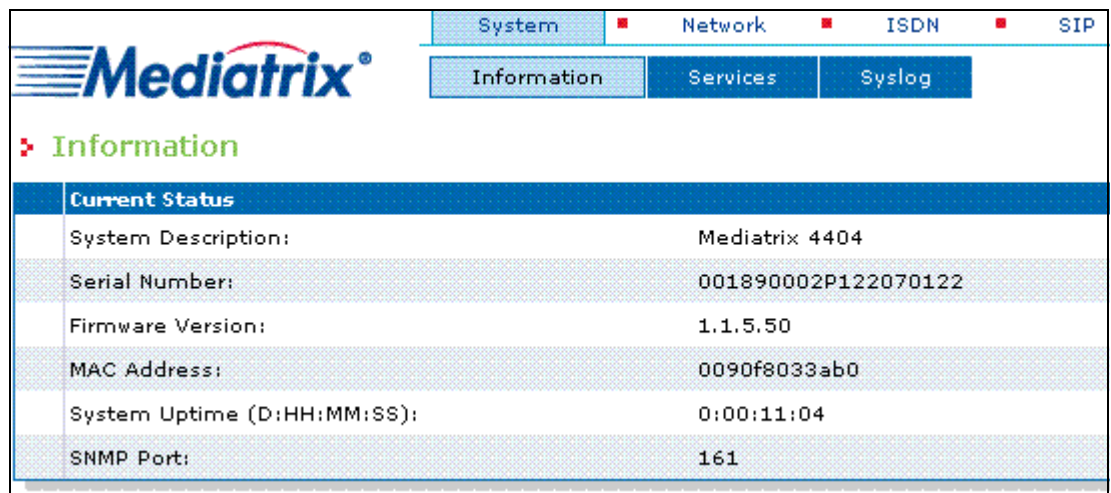
Step 2. When the login window appears, enter a user name in the User Name field. The default user name is **public**.

Step 3. Leave the Password field empty unless this has been previously set. Be sure to set a secure password later to complete the configuration process for your gateway.



The image shows the Mediatrrix login interface. At the top is the Mediatrrix logo. Below it is a green message: "Please enter your username and password". There are two input fields: "User Name:" with the text "public" and "Password:". A red box highlights the "User Name:" field. Below the fields is a "Login" button. On the right side of the login area, the number "189326" is printed vertically.

Step 4. Click the **Login** button. The Main Configuration window appears:



The image shows the Mediatrrix Main Configuration window. The top navigation bar includes tabs for "System", "Network", "ISDN", and "SIP". Below this is a sub-navigation bar with "Information", "Services", and "Syslog". The "Information" tab is selected, and the page title is "Information". Below the title is a table with the following data:

Current Status	
System Description:	Mediatrrix 4404
Serial Number:	001890002P122070122
Firmware Version:	1.1.5.50
MAC Address:	0090f8033ab0
System Uptime (D:HH:MM:SS):	0:00:11:04
SNMP Port:	161

Configuring the Network Address

Although the Mediatrrix 440x BRI Gateway supports DHCP, Cisco recommends that you configure a static IP address for each device in the SPA9000 Voice System. Assigning a static IP address ensures that a device is always reachable by the other devices in the system, even if the DHCP server is restarted and new network addresses are assigned to the DHCP clients.

To set a static IP address on the Mediatrrix 440x gateway, complete the following steps:

Step 1. In the Main Configuration window, click **Network**, and then click **Interfaces**. The Interfaces window appears:

Step 2. Under Interface Configuration, find the Uplink row. Enter the following settings:

- Link: Choose **netwrk** for a network link.
- Connection Type: Choose **Static**.
- Static IP Address: Ttype the static IP address and network mask. This example uses 192.168.1.75 as the static IP address, and 255.255.255.0 (/24) for the network mask.

Interface Configuration				
Interface	Link	Connection Type	Static IP Address	Activation
Rescue	<input type="text"/>	Static	192.168.0.10/24	Disable
Uplink	netwrk	Static	192.168.1.75/24	Enable
<input type="text"/>				

Step 3. To complete the configuration, click **Submit**. The unit reconfigures to its new IP address. To recover access to the Web User Interface, type the new IP address in your web browser.

Configuring the SIP Port for Call Routing to the SPA9000

This process specifies the gateway SIP port and the SIP proxy server address to which the Mediatrix 440x gateway will route incoming ISDN calls and from which it will receive outgoing call requests. In this case, the SIP proxy server is the SPA9000 and the SIP port specified should match the configuration on the SPA9000.

Step 1. In the Main Configuration window, click **SIP** and then click **Gateways**. The Gateways screen appears.

Step 2. In the SIP Port field, type the SIP port number, for example, 5060.

SIP Gateway Configuration			
Gateway Name	Network Interface	SIP Port	SIP Domain
default	Uplink	5060	<input type="text"/>
<input type="text"/>			

Step 3. Click **Submit**.

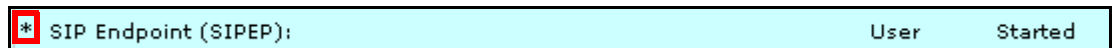
Restarting the Gateway

You need to restart the affected services on the gateway after you make configuration changes. A message appears near the top of the screen when this operation is required. This message includes a link to the Services table, which you can use to restart the specified services.

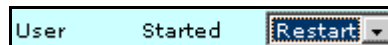


To restart the Mediatrix 440x gateway services after making configuration changes, complete the following steps:

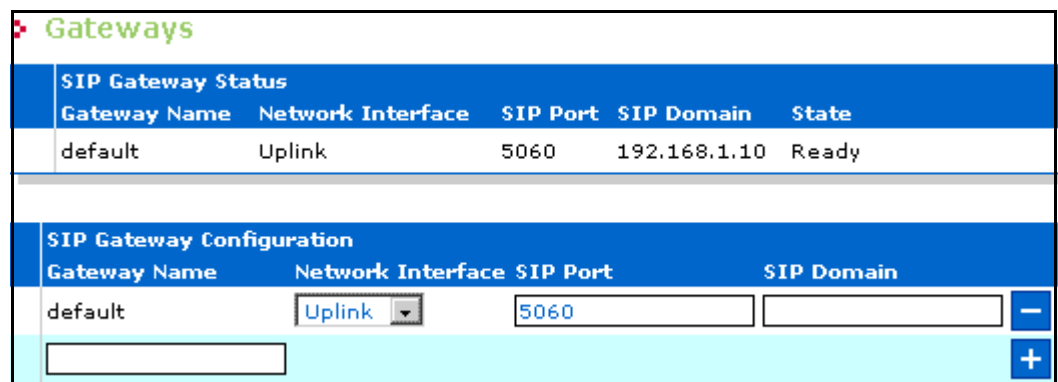
- Step 1.** Click the **Services** link in the message near the top of the page. The Services window appears.
- Step 2.** Look for an asterisk (*) next to the name of the services that need to be restarted. Refer to the following example.



- Step 3.** Choose **Restart** from the Action drop-down list.



- Step 4.** After the service restarts, the correct SIP port appears in the Gateways window.



Configuring Communication to the SPA9000 SIP Proxy Server

You need to configure the communication from the gateway to the SIP proxy server that will be used to route VoIP calls. In this case, the SIP proxy server is the SPA9000.

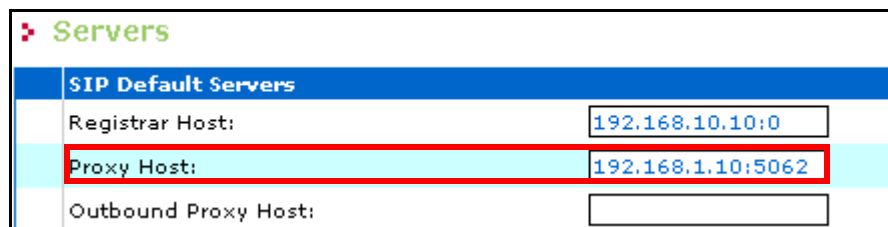
The SPA9000 has four line interfaces that can be configured for ITSP service, PSTN access, or ISDN service. Each line uses a different SIP port, as shown below. You can use any unused line interface for your Mediatrix gateway.

Port	SPA9000 Line Interface Number
5060	Line 1
5061	Line 2
5062	Line 3
5063	Line 4

In the example for the following procedure, the Mediatrix gateway is configured to communicate with a SPA9000 at 192.168.1.10, using port 5062 for Line 3.

To configure communication to the SPA9000 SIP Proxy Server, complete the following steps:

- Step 1.** In the Main Configuration window, click **SIP**, and then click **Servers**. The Servers window appears:
- Step 2.** In the Proxy Host field, enter the correct SPA9000 IP address and SIP port, such as 192.168.1.10:5062.



- Step 3.** Click **Submit**.

Later you will use the SPA9000 web administration server to configure the settings for communication from the SPA9000 to the Mediatrix gateway.

Configuring ISDN Connectivity

After the ISDN line is connected to the BRI1 port (and the BRI2 port, if applicable), complete the following steps to configure ISDN settings:

- Step 1.** In the Main Configuration window, click **ISDN**, and then click **Basic Rate Interface**. The Basic Rate Interface window appears.
- Step 2.** Enter the following settings:
- Endpoint Type: Select **TE** for Terminal Emulation.
 - Connection Type: Select **Point to Multipoint**.

Basic Rate Interface

Select Interface:

Hardware Configuration

Clock Reference (Applies to all interfaces):

Interface Configuration

Endpoint Type:

Connection Type:

Step 3. Click **Submit**

- Step 4.** Restart the ISDN service by completing the following tasks:
- a. Click the **Services** link in the message near the top of the page.
 - b. Find **Integrated Services Digital Network (ISDN)** in the table.
 - c. Choose **Restart** from the Action drop-down list. The service restarts.

Configuring Call Routing

This section describes how to define the gateway function. In the example used in this guide, the Mediatrix gateway routes all incoming ISDN calls from the BRI ports to the SPA9000, and directs call requests from the SPA9000 to ISDN.

This section also describes how to define a hunt group that groups the BRI1 and BRI2 ports on the Mediatrix 4402 gateway.

Note: You do not need to define a hunt group if you are using the Mediatrix 4401 gateway, which has only one BRI port.

To configure call routing, complete the following steps:

- Step 1.** In the Main Configuration window, click the **Telephony** menu option, and then the **Call Routing Config** sub-menu option. The Call Routing Config window appears:
- Step 2.** Click the **+** icon at the bottom right of the **Hunt Index** section. Or, if you are configuring a Mediatrix 4401 gateway, go to [Step 6 on page 9](#).

Step 3. In the Configure Hunt End window, enter the following information:

- Name: Enter a name for the hunt group. In the example, the name is hunt_isdn.
- Destination: Select ISDN-BRI1 and ISDN-BRI2 from the **Suggestion** drop-down menu.

The screenshot shows the 'Call Routing Config' window with a sub-section titled 'Configure Hunt End'. The window is divided into two main columns: 'Value' and 'Suggestion'. The 'Name' field is highlighted with a red box and contains the text 'hunt_isdn'. The 'Destinations' field contains 'isdn-Bri1, isdn-Bri2'. The 'Selection Algorithm' is set to 'Sequential'. The 'Timeout (seconds)' field is set to '0'. The 'Causes' field contains '31, 34, 38, 41, 42, 43, 44, 47'. The 'Suggestion' column features a dropdown menu with 'isdn-Bri1' and 'isdn-Bri2' highlighted by red boxes. Other options in the dropdown include 'isdn-Bri3', 'isdn-Bri4', 'sip-default', 'route-', 'hunt-', and 'Clear'.

Step 4. Click **Submit**.

Next, you will configure the gateway routing criteria. The Mediatrix 440x gateway will route incoming ISDN calls from the specified BRI port to the VoIP interface, and direct outgoing calls to the BRI port.

Step 5. In the Call Routing Config window, click the + icon at the bottom right of the **Route** section. The Configure Route End window appears.

Step 6. To create a route from SIP (sip-default) to the ISDN line(s) (that is, isdn-Bri1 or hunt_isdn), enter the following information:

- Source: Enter **sip-default**.
- Destination: From the Suggestion drop-down list, choose a hunt group, such as **hunt-hunt_isdn**.
—OR— If you are using a Mediatrix 4401, choose **isdn-Bri1**.

Call Routing Config

Configure Route End		
	Value	Suggestion
Source	sip-default	--- Suggestion ---
Properties Criteria	None	
Expression Criteria		--- Suggestion ---
Mappings		--- Suggestion ---
Signaling Properties		--- Suggestion ---
Destination	hunt-hunt_isdn	--- Suggestion ---
Config Status		--- Suggestion --- isdn-Bri1 isdn-Bri2 isdn-Bri3 isdn-Bri4 sip-default hunt-hunt_isdn route- hunt-

Step 7. Click **Submit**.

Step 8. Click **Apply** in the main Call Routing Config window.

Step 9. Restart the affected services by completing the following tasks:

- a. Click the **Services** link in the message near the top of the page.
- b. Find any starred items (*) in the table.
- c. Choose **Restart** from the Action drop-down list. The service restarts.

Configuring DTMF Transport

You must configure DTMF transport in the Mediatrix 440x gateway to guarantee that DTMF tones are properly sent to the SPA9000. DTMF tones are used with Auto Attendants, voice mail, and similar interactive services. DTMF is sent through RTP using the out-of-band method.

To configure DTMF transport, complete the following steps:

- Step 1.** In the Main Configuration window, click the **Telephony** menu option, and then click **CODECS**.
- Step 2.** In the Misc section, under DTMF Transport, select **Out-of-Band using RTP** from the Transport Method drop-down list.

Misc	
Jitter Buffer	
Level:	Normal
Minimum:	30
Maximum:	125
DTMF Transport	
Transport Method:	Out-of-Band using RTP
Payload Type:	96

Step 3. Click **Submit**.

Configuring the SPA9000 to Interoperate with the Mediatrix Gateway

You need to configure the SPA9000 to interoperate with the Mediatrix Gateway. You also need to configure outbound call routing. For example, the SPA9000 will route calls through the ISDN Gateway when a user presses 8 before dialing a telephone number.

Note: Cisco strongly recommends using the SPA9000 Voice System Setup Wizard to configure the SPA9000. This Application Note also provides instructions for using the administration web server, for advanced users.

- [“SPA9000 Configuration Using the SPA9000 Voice System Setup Wizard,”](#) on page 11
- [“SPA9000 Configuration Using the Administration Web Server,”](#) on page 13

SPA9000 Configuration Using the SPA9000 Voice System Setup Wizard

SPA9000 Setup Wizard allows the configuration of SPA9000 Voice Services connected to a Mediatrix® 440x ISDN gateway, automating the configuration of the SPA9000 related parameters.

Note: For information about the SPA9000 Voice System Installation and Configuration using the SPA9000 Setup Wizard, consult the *SPA9000 Voice System Installation and Configuration Guide, Wizard 2.1*.

When configuring **SPA9000 Voice Services**, click on any unused line interface and select **Mediatrix**. An example configuration may be selecting ITSP service on Line 1, PSTN/Voicemail service on Line 2, and ISDN service on Line 3.

Configure SPA9000 Voice Services

SPA9000 supports up to four voice service providers. This Wizard helps you configure 1 ITSP and up to 4 SPA400s on the SPA9000. Line 1 can be assigned to an ITSP for Internet phone calls. Any of the four lines can be assigned to SPA400's for PSTN calls or voice mail. Note that you must register a SPA400 even if you just want to use it for voice mail.

Line 1	<input type="text" value="ITSP"/>
Line 2	<input type="text" value="SPA400 <00183923c59c>"/>
Line 3	<input type="text" value="Mediatrix"/>
Line 4	<input type="text" value="None"/>

In the Configure Mediatrix step, enter the IP Address of the Mediatrix® 440x ISDN gateway. In the following example, the IP address is 192.168.1.75. Other parameters are populated by default parameters ensuring interoperability.

Configure Mediatrix

This is intended only to configure the SPA9000's parameters. For the Mediatrix gateway's configuration please refer to "SPA9000/Mediatrix 440x ISDN gateway configuration guide" available from www.linksys-itsp.com and www.linksys-voip.eu.

Line	<input type="text" value="3"/>
Proxy	<input type="text" value="192.168.1.75"/>
User ID	<input type="text" value="1946"/>
Display Name	<input type="text" value="SPA9000"/>

In the Configure Outbound Call Routes step, select the enter the Prefix (steering digit) that is used to access an ISDN line for an outbound call (8 in this example). You also can select the option for the ISDN lines to become backup lines in case the ITSP service (VoIP) becomes unavailable, by checking the Use as Backup check box. In this example, both PSTN Line 2 (SPA400) and ISDN Line 3 (Mediatrix® 440x) are backup lines for the VoIP service, enhancing reliability.

Note: The Line associated to the Mediatrix® ISDN gateway can be easily identified by the IP address (192.168.1.75) and the User ID used (1946).

Configure Outbound Call Routes

Service Provider 1-4 identifies SPs connected to the SPA9000

1. Select a steering digit [0-9] for each SP
2. Select backup SP

Service Provider	SIP Line	Use as backup	Prefix
Service Provider 1	<input type="text" value="SIP_Line 34700766702001@sip.peoplecall.com"/>	<input type="checkbox"/>	<input type="text" value="0"/>
Service Provider 2	<input type="text" value="SPA400 9000@192.168.1.11"/>	<input checked="" type="checkbox"/>	<input type="text" value="9"/>
Service Provider 3	<input type="text" value="SIP_Line 1946@192.168.1.75"/>	<input checked="" type="checkbox"/>	<input type="text" value="8"/>
Service Provider 4	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>

After the configuration of the SPA9000 using the SPA9000 Voice System Setup Wizard is completed, SPA9000 will update its configuration and connect to the Mediatrix® 440x ISDN Gateway.

SPA9000 Configuration Using the Administration Web Server

To configure the SPA9000 for use with the ISDN Gateway, complete the following steps:

- Step 1.** In your web browser, enter the following URL to connect to the administration web server for the SPA9000: **<spa9000_IP_address/admin/advanced>**
- Step 2.** To configure a SPA9000 line interface for the ISDN Gateway, complete the following tasks:
- a. Click the **Voice** tab.
 - b. Click the **Line 3** tab, or choose any unused line interface. For example, you might configure ITSP service on Line 1, PSTN access through Line 2, and ISDN service on Line 3.
 - c. Scroll down to the **Proxy and Registration** section. Enter the following information:
 - **Proxy:** Enter the Mediatrix 440x gateway IP address and the SIP port that you configured previously for the Mediatrix 440x gateway. In the example, the address is 192.168.1.75 and the port number is 5060
 - **Register:** Select **No**. There is no need to register because this is defined as a peer-to-peer SIP trunk between the SPA9000 and the Mediatrix 440x gateway.
 - **Make Call Without Reg:** Select **Yes**.

- **Ans Call Without Reg:** Select **Yes**.

Proxy and Registration			
Proxy:	192.168.1.75:5060		
Outbound Proxy:			
Use Outbound Proxy:	no	Use OB Proxy In Dialog:	yes
Register:	no	Make Call Without Reg:	yes
Register Expires:	30	Ans Call Without Reg:	yes
Use DNS SRV:	no	DNS SRV Auto Prefix:	no
Proxy Fallback Intvl:	3600	Proxy Redundancy Method:	Normal
Mailbox Status:		Mailbox Subscribe URL:	

Step 3. To set a SIP User ID to identify the line internally, scroll to the **Subscriber Information** section. Enter the following information:

- **Display Name:** Enter a name to identify this line internally.
- **User ID:** Enter a number to identify this line internally.

This example uses Mediatrix as the Display Name and 1946 as the User ID.

Subscriber Information			
Display Name:	Mediatrix	User ID:	1946
Password:		Use Auth ID:	no
Auth ID:		Call Capacity:	unlimited
Contact List:	aa		
Cfwd No Ans Delay:	25		

Step 4. To configure outbound call routing, complete the following tasks:

- Click the **SIP** tab.
- Scroll down to the **PBX Parameters** section, and then enter the dial-out criteria in the **Call Routing Rule** field.

The syntax is illustrated in the following example:

(<:L1>0xx.|<:L2>9xx.|<:L3>8xx.)

This call routing rule specifies the line interface that is used for different dialing sequences. Line 1 (L1) is used when a user dials a number that starts with 0 (0xx.). Line 2 (L2) is used when a user dials a number that starts with 9 (9xx.). Line 3 (L3) is used when a user dials a number that starts with 8 (8xx.).

PBX Parameters			
Proxy Network Interface:	WAN	Proxy Listen Port:	
Multicast Address:	224.168.168.168:6061	Group Page Address:	
Max Expires:	60	Force Media Proxy:	
Proxy Debug Option:	none		
Call Routing Rule:	(<:L1>0xx. <:L2>9xx. <:L3>8xx.)		

Step 5. To configure the dial plan, complete the following tasks:

- a. Click the **SIP** tab.
- b. Scroll down to the **PBX Phone Parameters** section.
- c. Edit the **Phone Dial Plan** to allow users to dial the specified prefix number (for example, 8).

To configure a steering digit, enter the steering digit followed by a comma. For example, to allow 8 to be used as a steering digit for all calls, you would edit the dial plan as shown:

```
(*xx|8,[3469]11|0|00|8,[2-9]xxxxxx|8,1xxx[2-9]xxxxxS0|8,xxxxxxxxxx.)
```

This dial plan allows the following dialing sequences:

- *xx Allows any three-digits number
- 8,[3469]11 Allows the steering digit 8 to be followed by 311, 411, 611, or 911 .
- 0|00 Allows 0 or 00 to dial an operator.
- 8,[2-9]xxxxxx Allows the steering digit 8 to be followed by any seven-digit number beginning with 2 to 9, as in local dialing.
- 8,1xxx[2-9]xxxxxS0 Allows the steering digit 8 to be followed by any 10-digit number beginning with 1, as in long-distance dialing. The timer is set to 0 (S0) to submit the dialed sequence as soon as the tenth digit is dialed.
- 8,xxxxxxxxxx. Allows the steering digit 8 to be followed by any number.

PBX Phone Parameters	
Next Auto User ID:	106
Phone Upgrade Rule:	
Phone Dial Plan:	(*xx 8,[3469]11 0 00 8,[2-9]xxxxxx 8,1xxx[2-9]xxxxxS0 8,xxxx

Step 6. To ensure that the steering digit is removed from the number before it is sent out, complete the following steps:

- a. Click the **Line 3** tab (or another line interface that you configured for your ISDN).
- b. Scroll to the **Dial Plan** section.
- c. Update the dial plan so that the steering digit is removed from the number before it is sent out.

The syntax is illustrated in the following example:

```
(<8:>xx.)
```

The number 8 is removed from the beginning of any sequence that includes at least two digits (xx .)

Dial Plan	
Dial Plan:	(<8:>xx.)

Step 7. Click **Submit All Changes**.

Congratulations! From this point, your system is configured and should be able to receive and originate calls to and from the ISDN.

To review the status of network connectivity and ISDN line(s) and of Mediatrix 440x gateway, check the Network and ISDN Status page respectively.

APPENDIX: SIP Traces

This section provides SIP traces for the following scenarios:

- [“Incoming ISDN Call to the SPA9000,” on page 16](#)
- [“Outgoing ISDN Call from the SPA9000 Internal Endpoint,” on page 24](#)

Incoming ISDN Call to the SPA9000

```
[4]<<192.168.1.75:5060(847)
```

```
[4]<<192.168.1.75:5060(847)
```

```
INVITE sip:914841622@192.168.1.10:5062 SIP/2.0Via: SIP/2.0/UDP
192.168.1.75:5060;branch=z9hG4bKb332447a3Content-Length: 297To:
sip:914841622@192.168.1.10:5062From:
sip:916611146@192.168.1.10:5062;tag=61348d706bce16cCall-ID:
8a9c309d9d1e3eed7e1bb0a641cc361f@192.168.1.10CSeq: 101589334
INVITESupported: timerMin-SE: 1800Session-Expires: 3600Allow: INVITE, ACK,
BYE, CANCEL, REFER, NOTIFYContent-Type: application/sdpContact:
sip:916611146@192.168.1.75:5060Supported: replacesUser-Agent: MxSipApp/
v1.1.3.19 MxSF/v3.2.1.2v=0o=MxSIP 1032267399288319200 1987802900286410546
IN IP4 192.168.1.75s=-c=IN IP4 192.168.1.75t=0 0a=sendrecv=audio 5004
RTP/AVP 0 18 4 8 96 13a=rtpmap:0 PCMU/8000a=rtpmap:18 G729/8000a=rtpmap:4
G723/8000a=rtpmap:8 PCMA/8000a=rtpmap:96 telephone-event/8000a=fmtp:96 0-
15
```

```
[4]->192.168.1.75:5060(360)
```

```
[4]->192.168.1.75:5060(360)
```

```
SIP/2.0 100 TryingTo: sip:914841622@192.168.1.10:5062From:
sip:916611146@192.168.1.10:5062;tag=61348d706bce16cCall-ID:
8a9c309d9d1e3eed7e1bb0a641cc361f@192.168.1.10CSeq: 101589334 INVITEVia:
SIP/2.0/UDP 192.168.1.75:5060;branch=z9hG4bKb332447a3Server: Linksys/
SPA9000-5.1.6(a)Allow-Events: talk, hold, conference, x-spa-ctiContent-
Length: 0
```

```
[pxy]->192.168.1.10:5083(956)
```

```
[pxy]->192.168.1.10:5083(956)
```

```
INVITE sip:aa@192.168.1.10:5083 SIP/2.0Via: SIP/2.0/UDP
192.168.1.10:6060;branch=z9hG4bK-f79922d6From:
<sip:916611146@192.168.1.10:5062>;tag=40b28831d70d554eo4To:
<sip:aa@192.168.1.10:6060>Remote-Party-ID:
<sip:916611146@192.168.1.10:5062>;screen=yes;party=callingCall-ID:
adb9f02d-1cbead82@192.168.1.10CSeq: 101 INVITEMax-Forwards: 70Contact:
"Mediatrix@" <sip:916611146@192.168.1.10:6060>Expires: 240User-Agent:
Linksys/SPA9000-5.1.6(a)Allow-Events: talk, hold, conference, x-spa-
ctiContent-Length: 298Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY,
OPTIONS, REFERSupported: x-sipura, replacesContent-Type: application/
```



```
sdpv=0o=proxy 1032267399288319200 1987802900286410546 IN IP4
192.168.1.75s=-c=IN IP4 192.168.1.10t=0 0a=sendrecvm=audio 16390 RTP/AVP 0
18 4 8 96 13a=rtpmap:0 PCMU/8000a=rtpmap:18 G729/8000a=rtpmap:4 G723/
8000a=rtpmap:8 PCMA/8000a=rtpmap:96 telephone-event/8000a=fmtp:96 0-15

[AA]: aa 1 auto answer delay = 5

[7:0]AAA ALLOC CALL (port=16394)

[7:0]AAA:RTP Rx Up

[pxy]<<127.0.0.1:5083(339)

[pxy]<<127.0.0.1:5083(339)

SIP/2.0 100 TryingTo: <sip:aa@192.168.1.10:6060>From:
<sip:916611146@192.168.1.10:5062>;tag=40b28831d70d554eo4Call-ID: adb9f02d-
1cbead82@192.168.1.10CSeq: 101 INVITEVia: SIP/2.0/UDP
192.168.1.10:6060;branch=z9hG4bK-f79922d6Server: Linksys/SPA9000-
5.1.6(a)Allow-Events: talk, hold, conference, x-spa-ctiContent-Length:
0[pxy]<<127.0.0.1:5083(447)

[pxy]<<127.0.0.1:5083(447)

SIP/2.0 180 RingingTo:
<sip:aa@192.168.1.10:6060>;tag=aca72a2bbcb8b9ei7From:
<sip:916611146@192.168.1.10:5062>;tag=40b28831d70d554eo4Call-ID: adb9f02d-
1cbead82@192.168.1.10CSeq: 101 INVITEVia: SIP/2.0/UDP
192.168.1.10:6060;branch=z9hG4bK-f79922d6Server: Linksys/SPA9000-
5.1.6(a)Remote-Party-ID: Auto Attendant
<sip:aa@192.168.1.10:6060>;screen=yes;party=calledAllow-Events: talk,
hold, conference, x-spa-ctiContent-Length: 0

pri--180-->BCC--180-->pub

pri--180-->BCC--180-->pub

[4]->192.168.1.75:5060(467)

[4]->192.168.1.75:5060(467)

SIP/2.0 180 RingingTo:
sip:914841622@192.168.1.10:5062;tag=346072efc18339aai4From:
sip:916611146@192.168.1.10:5062;tag=61348d706bce16cCall-ID:
8a9c309d9d1e3eed7e1bb0a641cc361f@192.168.1.10CSeq: 101589334 INVITEVia:
SIP/2.0/UDP 192.168.1.75:5060;branch=z9hG4bKb332447a3Server: Linksys/
SPA9000-5.1.6(a)Remote-Party-ID: "Mediatrix®"
<sip:1946@192.168.1.75:5060>;screen=yes;party=calledAllow-Events: talk,
hold, conference, x-spa-ctiContent-Length: 0

[pxy]<<127.0.0.1:5083(869)

[pxy]<<127.0.0.1:5083(869)

SIP/2.0 200 OKTo: <sip:aa@192.168.1.10:6060>;tag=aca72a2bbcb8b9ei7From:
<sip:916611146@192.168.1.10:5062>;tag=40b28831d70d554eo4Call-ID: adb9f02d-
1cbead82@192.168.1.10CSeq: 101 INVITEVia: SIP/2.0/UDP
192.168.1.10:6060;branch=z9hG4bK-f79922d6Contact: Auto Attendant
<sip:aa@192.168.1.10:5083>Server: Linksys/SPA9000-5.1.6(a)Remote-Party-ID:
Auto Attendant <sip:aa@192.168.1.10:6060>;screen=yes;party=calledAllow-
Events: talk, hold, conference, x-spa-ctiContent-Length: 248Allow: ACK,
```

```
BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFERSupported: x-sipura,
replacesContent-Type: application/sdpv=0o=- 62798 62798 IN IP4
192.168.1.10s=-c=IN IP4 192.168.1.10t=0 0m=audio 16394 RTP/AVP 0 100
96a=rtpmap:0 PCMU/8000a=rtpmap:100 NSE/8000a=fmtp:100 192-193a=rtpmap:96
telephone-event/8000a=fmtp:96 0-15a=ptime:30a=sendrecv

[pxy]->192.168.1.10:5083(455)

[pxy]->192.168.1.10:5083(455)

ACK sip:aa@192.168.1.10:5083 SIP/2.0Via: SIP/2.0/UDP
192.168.1.10:6060;branch=z9hG4bK-ee709ed9From:
<sip:916611146@192.168.1.10:5062>;tag=40b28831d70d554eo4To:
<sip:aa@192.168.1.10:6060>;tag=aca72a2bbbc8b9ei7Call-ID: adb9f02d-
1cbead82@192.168.1.10CSeq: 101 ACKMax-Forwards: 70Contact: "Mediatrix@"
<sip:916611146@192.168.1.10:6060>User-Agent: Linksys/SPA9000-
5.1.6(a)Allow-Events: talk, hold, conference, x-spa-ctiContent-Length: 0

CC:Connected

[7:0]AAA:Create AA Object

[7:0]AAA:RTP Tx Up (pt=0->c0a8010a:16390)

[4]->192.168.1.75:5060(902)

[4]->192.168.1.75:5060(902)

SIP/2.0 200 OKTo:
sip:914841622@192.168.1.10:5062;tag=346072efc18339aai4;ref=aaFrom:
sip:916611146@192.168.1.10:5062;tag=61348d706bce16cCall-ID:
8a9c309d9d1e3eed7e1bb0a641cc361f@192.168.1.10CSeq: 101589334 INVITEVia:
SIP/2.0/UDP 192.168.1.75:5060;branch=z9hG4bKb332447a3Contact: "Mediatrix@"
<sip:914841622@192.168.1.10:5062>Server: Linksys/SPA9000-5.1.6(a)Remote-
Party-ID: "Mediatrix@"
<sip:1946@192.168.1.75:5060>;screen=yes;party=calledAllow-Events: talk,
hold, conference, x-spa-ctiContent-Length: 248Allow: ACK, BYE, CANCEL,
INFO, INVITE, NOTIFY, OPTIONS, REFERSupported: x-sipura, replacesContent-
Type: application/sdpv=0o=- 62798 62798 IN IP4 192.168.1.10s=-c=IN IP4
192.168.1.10t=0 0m=audio 16388 RTP/AVP 0 100 96a=rtpmap:0 PCMU/
8000a=rtpmap:100 NSE/8000a=fmtp:100 192-193a=rtpmap:96 telephone-event/
8000a=fmtp:96 0-15a=ptime:30a=sendrecv

[0:0]RTP Rx 1st PKT @16394(3)

[4]<<192.168.1.75:5060(410)

[4]<<192.168.1.75:5060(410)

ACK sip:914841622@192.168.1.10:5062 SIP/2.0Via: SIP/2.0/UDP
192.168.1.75:5060;branch=z9hG4bK1eb32ba63Content-Length: 0To:
sip:914841622@192.168.1.10:5062;tag=346072efc18339aai4From:
sip:916611146@192.168.1.10:5062;tag=61348d706bce16cCall-ID:
8a9c309d9d1e3eed7e1bb0a641cc361f@192.168.1.10CSeq: 101589334 ACKContact:
sip:916611146@192.168.1.75:5060User-Agent: MxSipApp/v1.1.3.19 MxSF/
v3.2.1.2

[2]RegOK. NextReg in 28 (1)

AVT Rx Start Tone: 1
```

```
AVT Rx End Tone
AVT Rx Start Tone: 1
AVT Rx End Tone
AVT Rx Start Tone: 0
AVT Rx End Tone
AVT Rx Start Tone: 11
AVT Rx End Tone
[AA]: xfer to target=110
CC:BlndXferTgt:110
Calling:110@192.168.1.10:6060
[7:0]AAA Rel Call
[7:0]AAA:Destory SAA OBJ
[pxy]<<127.0.0.1:5083(556)
[pxy]<<127.0.0.1:5083(556)
REFER sip:916611146@192.168.1.10:6060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.10:5083;branch=z9hG4bK-2bd7f3bFrom:
<sip:aa@192.168.1.10:6060>;tag=aca72a2bbcb8b9ei7To:
<sip:916611146@192.168.1.10:5062>;tag=40b28831d70d554eo4Referred-By: Auto
Attendant <sip:aa@192.168.1.10:6060>Call-ID: adb9f02d-
lcbead82@192.168.1.10CSeq: 101 REFERMax-Forwards: 70Contact: Auto
Attendant <sip:aa@192.168.1.10:5083>Refer-To:
<sip:110@192.168.1.10:6060>User-Agent: Linksys/SPA9000-5.1.6(a)Allow-
Events: talk, hold, conference, x-spa-ctiContent-Length: 0
pri--REFER-->BCC:target=110@192.168.1.10:6060
pri--REFER-->BCC:target=110@192.168.1.10:6060
[pxy]->192.168.1.169:5060(1037)
[pxy]->192.168.1.169:5060(1037)
INVITE sip:110@192.168.1.169:5060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.10:6060;branch=z9hG4bK-135623b9From:
<sip:916611146@192.168.1.10:5062>;tag=29306e71561lea8abo4To:
<sip:110@192.168.1.10:6060>Remote-Party-ID:
<sip:916611146@192.168.1.10:5062>;screen=yes;party=callingAlert-Info:
ringbackCall-ID: a0dabe69-b49bd913@192.168.1.10CSeq: 101 INVITEMax-
Forwards: 70Contact: "Mediatrix@"
<sip:916611146@192.168.1.10:6060>Expires: 240Referred-By: Auto Attendant
<sip:aa@192.168.1.10:6060>User-Agent: Linksys/SPA9000-5.1.6(a)Allow-
Events: talk, hold, conference, x-spa-ctiContent-Length: 298Allow: ACK,
BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFERSupported: x-sipura,
replacesContent-Type: application/sdpv=0o=proxy 1032267399288319200
1987802900286410546 IN IP4 192.168.1.75s=-c=IN IP4 192.168.1.10t=0
```

```
0a=sendrecvm=audio 16390 RTP/AVP 0 18 4 8 96 13a=rtpmap:0 PCMU/
8000a=rtpmap:18 G729/8000a=rtpmap:4 G723/8000a=rtpmap:8 PCMA/
8000a=rtpmap:96 telephone-event/8000a=fmtp:96 0-15[pxy]-
>127.0.0.1:5083(427)

[pxy]->127.0.0.1:5083(427)

SIP/2.0 202 AcceptedTo:
<sip:916611146@192.168.1.10:5062>;tag=40b28831d70d554eo4From:
<sip:aa@192.168.1.10:6060>;tag=aca72a2bbcb8b9ei7Call-ID: adb9f02d-
1cbead82@192.168.1.10CSeq: 101 REFERVia: SIP/2.0/UDP
192.168.1.10:5083;branch=z9hG4bK-2bd7f3bContact: "Mediatrix@"
<sip:1946@192.168.1.10:6060;transport=udp>Server: Linksys/SPA9000-
5.1.6(a)Allow-Events: talk, hold, conference, x-spa-ctiContent-Length: 0

[pxy]<<192.168.1.169:5060(288)

[pxy]<<192.168.1.169:5060(288)

SIP/2.0 100 TryingTo: <sip:110@192.168.1.10:6060>From:
<sip:916611146@192.168.1.10:5062>;tag=29306e71561ea8abo4Call-ID: a0dabe69-
b49bd913@192.168.1.10CSeq: 101 INVITEVia: SIP/2.0/UDP
192.168.1.10:6060;branch=z9hG4bK-135623b9Server: Linksys/SPA942-
5.1.10Content-Length: 0[pxy]<<192.168.1.169:5060(412)

[pxy]<<192.168.1.169:5060(412)

SIP/2.0 183 Session ProgressTo:
<sip:110@192.168.1.10:6060>;tag=aaceal6d26d547ci0From:
<sip:916611146@192.168.1.10:5062>;tag=29306e71561ea8abo4Call-ID: a0dabe69-
b49bd913@192.168.1.10CSeq: 101 INVITEVia: SIP/2.0/UDP
192.168.1.10:6060;branch=z9hG4bK-135623b9Server: Linksys/SPA942-
5.1.10Remote-Party-ID:
<sip:110@192.168.1.10:6060>;screen=yes;party=calledContent-Length: 0Allow-
Events: dialog

pri--18x-->BCC(conn)

pri--18x-->BCC(conn)

[pxy]<<192.168.1.169:5060(768)

[pxy]<<192.168.1.169:5060(768)

SIP/2.0 200 OKTo: <sip:110@192.168.1.10:6060>;tag=aaceal6d26d547ci0From:
<sip:916611146@192.168.1.10:5062>;tag=29306e71561ea8abo4Call-ID: a0dabe69-
b49bd913@192.168.1.10CSeq: 101 INVITEVia: SIP/2.0/UDP
192.168.1.10:6060;branch=z9hG4bK-135623b9Contact:
<sip:110@192.168.1.169:5060>Server: Linksys/SPA942-5.1.10Remote-Party-ID:
<sip:110@192.168.1.10:6060>;screen=yes;party=calledContent-Length:
203Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER,
SUBSCRIBEAllow-Events: dialogSupported: replacesContent-Type: application/
sdpv=0o=- 62747 62747 IN IP4 192.168.1.169s=-c=IN IP4 192.168.1.169t=0
0m=audio 16480 RTP/AVP 0 96a=rtpmap:0 PCMU/8000a=rtpmap:96 telephone-
event/8000a=fmtp:96 0-15a=ptime:30a=sendrecv

[pxy]->192.168.1.169:5060(457)

[pxy]->192.168.1.169:5060(457)
```

```
ACK sip:110@192.168.1.169:5060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.10:6060;branch=z9hG4bK-f39d492fFrom:
<sip:916611146@192.168.1.10:5062>;tag=29306e71561ea8abo4To:
<sip:110@192.168.1.10:6060>;tag=aaceal6d26d547ci0Call-ID: a0dabe69-
b49bd913@192.168.1.10CSeq: 101 ACKMax-Forwards: 70Contact: "Mediatrix@"
<sip:916611146@192.168.1.10:6060>User-Agent: Linksys/SPA9000-
5.1.6(a)Allow-Events: talk, hold, conference, x-spa-ctiContent-Length: 0

ReferTarget answers BCC INVITE

ReferTarget answers BCC INVITE

pri(ReferTarget)--200-->BCC--reINVITE-->pub

pri(ReferTarget)--200-->BCC--reINVITE-->pub

[4]->192.168.1.75:5060(827)

[4]->192.168.1.75:5060(827)

INVITE sip:916611146@192.168.1.75:5060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.10:5062;branch=z9hG4bK-7625e35From:
<sip:914841622@192.168.1.10:5062>;tag=346072efc18339aai4;ref=110To:
<sip:916611146@192.168.1.10:5062>;tag=61348d706bce16cRemote-Party-ID:
"Mediatrix@" <sip:1946@192.168.1.75:5060>;screen=yes;party=calledCall-ID:
8a9c309d9d1e3eed7e1bb0a641cc361f@192.168.1.10CSeq: 101 INVITEMax-Forwards:
70Contact: "Mediatrix@" <sip:914841622@192.168.1.10:5062>Expires: 30User-
Agent: Linksys/SPA9000-5.1.6(a)Allow-Events: talk, hold, conference, x-
spa-ctiContent-Length: 202Content-Type: application/sdpv=0o=- 62747 62747
IN IP4 192.168.1.169s=-c=IN IP4 192.168.1.10t=0 0m=audio 16388 RTP/AVP 0
96a=rtpmap:0 PCMU/8000a=rtpmap:96 telephone-event/8000a=fmtp:96 0-
15a=ptime:30a=sendrecv

[4]<<192.168.1.75:5060(351)

[4]<<192.168.1.75:5060(351)

SIP/2.0 100 TryingCall-ID:
8a9c309d9d1e3eed7e1bb0a641cc361f@192.168.1.10CSeq: 101 INVITEFrom:
<sip:914841622@192.168.1.10:5062>;tag=346072efc18339aai4;ref=110To:
<sip:916611146@192.168.1.10:5062>;tag=61348d706bce16cVia: SIP/2.0/UDP
192.168.1.10:5062;branch=z9hG4bK-7625e35Content-Length: 0User-Agent:
MxSipApp/v1.1.3.19 MxSF/v3.2.1.2

[4]<<192.168.1.75:5060(748)

[4]<<192.168.1.75:5060(748)

SIP/2.0 200 OKCall-ID: 8a9c309d9d1e3eed7e1bb0a641cc361f@192.168.1.10CSeq:
101 INVITEFrom:
<sip:914841622@192.168.1.10:5062>;tag=346072efc18339aai4;ref=110To:
<sip:916611146@192.168.1.10:5062>;tag=61348d706bce16cVia: SIP/2.0/UDP
192.168.1.10:5062;branch=z9hG4bK-7625e35Content-Length: 220Session-
Expires: 3600;refresher=uasAllow: INVITE, ACK, BYE, CANCEL, REFER,
NOTIFYContent-Type: application/sdpSupported: replacesContact:
sip:916611146@192.168.1.75:5060User-Agent: MxSipApp/v1.1.3.19 MxSF/
v3.2.1.2v=0o=MxSIP 1032267399288319200 1987802900286410547 IN IP4
192.168.1.75s=-c=IN IP4 192.168.1.75t=0 0a=sendrecv m=audio 5004 RTP/AVP 0
96a=rtpmap:0 PCMU/8000a=rtpmap:96 telephone-event/8000a=fmtp:96 0-15
```

```
[4]->192.168.1.75:5060(489)
[4]->192.168.1.75:5060(489)
ACK sip:916611146@192.168.1.75:5060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.10:5062;branch=z9hG4bK-27dddcedFrom:
<sip:914841622@192.168.1.10:5062>;tag=346072efc18339aai4;ref=110To:
<sip:916611146@192.168.1.10:5062>;tag=61348d706bcel6cCall-ID:
8a9c309d9d1e3eed7e1bb0a641cc361f@192.168.1.10CSeq: 101 ACKMax-Forwards:
70Contact: "Mediatrix@" <sip:914841622@192.168.1.10:5062>User-Agent:
Linksys/SPA9000-5.1.6(a)Allow-Events: talk, hold, conference, x-spa-
ctiContent-Length: 0
[pxy]->192.168.1.169:5060(809)
[pxy]->192.168.1.169:5060(809)
INVITE sip:110@192.168.1.169:5060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.10:6060;branch=z9hG4bK-albdad95From:
<sip:916611146@192.168.1.10:5062>;tag=29306e71561ea8abo4To:
<sip:110@192.168.1.10:6060>;tag=aaceal6d26d547ci0Remote-Party-ID:
<sip:916611146@192.168.1.10:5062>;screen=yes;party=callingCall-ID:
a0dabe69-b49bd913@192.168.1.10CSeq: 102 INVITEMax-Forwards: 70Contact:
"Mediatrix@" <sip:916611146@192.168.1.10:6060>Expires: 30User-Agent:
Linksys/SPA9000-5.1.6(a)Allow-Events: talk, hold, conference, x-spa-
ctiContent-Length: 221Content-Type: application/sdpv=0o=proxy
1032267399288319200 1987802900286410547 IN IP4 192.168.1.75s=-c=IN IP4
192.168.1.10t=0 0a=sendrcvm=audio 16390 RTP/AVP 0 96a=rtpmap:0 PCMU/
8000a=rtpmap:96 telephone-event/8000a=fmtp:96 0-15
[pxy]<<192.168.1.169:5060(581)
[pxy]<<192.168.1.169:5060(581)
SIP/2.0 200 OKTo: <sip:110@192.168.1.10:6060>;tag=aaceal6d26d547ci0From:
<sip:916611146@192.168.1.10:5062>;tag=29306e71561ea8abo4Call-ID: a0dabe69-
b49bd913@192.168.1.10CSeq: 102 INVITEVia: SIP/2.0/UDP
192.168.1.10:6060;branch=z9hG4bK-albdad95Contact:
<sip:110@192.168.1.169:5060>Server: Linksys/SPA942-5.1.10Content-Length:
203Content-Type: application/sdpv=0o=- 63056 63056 IN IP4 192.168.1.169s=-
c=IN IP4 192.168.1.169t=0 0m=audio 16480 RTP/AVP 0 96a=rtpmap:0 PCMU/
8000a=rtpmap:96 telephone-event/8000a=fmtp:96 0-15a=ptime:30a=sendrcv
[pxy]->192.168.1.169:5060(457)
[pxy]->192.168.1.169:5060(457)
ACK sip:110@192.168.1.169:5060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.10:6060;branch=z9hG4bK-ad7bd510From:
<sip:916611146@192.168.1.10:5062>;tag=29306e71561ea8abo4To:
<sip:110@192.168.1.10:6060>;tag=aaceal6d26d547ci0Call-ID: a0dabe69-
b49bd913@192.168.1.10CSeq: 102 ACKMax-Forwards: 70Contact: "Mediatrix@"
<sip:916611146@192.168.1.10:6060>User-Agent: Linksys/SPA9000-
5.1.6(a)Allow-Events: talk, hold, conference, x-spa-ctiContent-Length: 0
ReferTarget answers BCC INVITE
ReferTarget answers BCC INVITE
pri(ReferTarget)--200-->BCC--reINVITE-->pub (done)
```

```
pri(ReferTarget)--200-->BCC--reINVITE-->pub (done)

[pxy]<<127.0.0.1:5083(406)

[pxy]<<127.0.0.1:5083(406)

BYE sip:916611146@192.168.1.10:6060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.10:5083;branch=z9hG4bK-4c6ddc2eFrom:
<sip:aa@192.168.1.10:6060>;tag=aca72a2bbcb8b9ei7To:
<sip:916611146@192.168.1.10:5062>;tag=40b28831d70d554eo4Call-ID: adb9f02d-
1cbead82@192.168.1.10CSeq: 102 BYEMax-Forwards: 70User-Agent: Linksys/
SPA9000-5.1.6(a)Allow-Events: talk, hold, conference, x-spa-ctiContent-
Length: 0[pxy]->127.0.0.1:5083(355)

[pxy]->127.0.0.1:5083(355)

SIP/2.0 200 OKTo:
<sip:916611146@192.168.1.10:5062>;tag=40b28831d70d554eo4From:
<sip:aa@192.168.1.10:6060>;tag=aca72a2bbcb8b9ei7Call-ID: adb9f02d-
1cbead82@192.168.1.10CSeq: 102 BYEVia: SIP/2.0/UDP
192.168.1.10:5083;branch=z9hG4bK-4c6ddc2eServer: Linksys/SPA9000-
5.1.6(a)Allow-Events: talk, hold, conference, x-spa-ctiContent-Length: 0

[pxy]<<192.168.1.169:5060(355)

[pxy]<<192.168.1.169:5060(355)

BYE sip:916611146@192.168.1.10:6060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.169:5060;branch=z9hG4bK-a5a0daf6From:
<sip:110@192.168.1.10:6060>;tag=aaceal6d26d547ci0To:
<sip:916611146@192.168.1.10:5062>;tag=29306e71561ea8abo4Call-ID: a0dabe69-
b49bd913@192.168.1.10CSeq: 101 BYEMax-Forwards: 70User-Agent: Linksys/
SPA942-5.1.10Content-Length: 0

private side ends the call

private side ends the call

[pxy]->192.168.1.169:5060(356)

[pxy]->192.168.1.169:5060(356)

SIP/2.0 200 OKTo:
<sip:916611146@192.168.1.10:5062>;tag=29306e71561ea8abo4From:
<sip:110@192.168.1.10:6060>;tag=aaceal6d26d547ci0Call-ID: a0dabe69-
b49bd913@192.168.1.10CSeq: 101 BYEVia: SIP/2.0/UDP
192.168.1.169:5060;branch=z9hG4bK-a5a0daf6Server: Linksys/SPA9000-
5.1.6(a)Allow-Events: talk, hold, conference, x-spa-ctiContent-Length: 0

BCC:Ended

[4]->192.168.1.75:5060(432)

[4]->192.168.1.75:5060(432)

BYE sip:916611146@192.168.1.75:5060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.10:5062;branch=z9hG4bK-967510eFrom:
<sip:914841622@192.168.1.10:5062>;tag=346072efc18339aai4;ref=110To:
<sip:916611146@192.168.1.10:5062>;tag=61348d706bce16cCall-ID:
8a9c309d9d1e3eed7e1bb0a641cc361f@192.168.1.10CSeq: 102 BYEMax-Forwards:
70User-Agent: Linksys/SPA9000-5.1.6(a)Allow-Events: talk, hold,
conference, x-spa-ctiContent-Length: 0
```



```
[4]<<192.168.1.75:5060(365)
[4]<<192.168.1.75:5060(365)
SIP/2.0 200 OKCall-ID: 8a9c309d9d1e3eed7e1bb0a641cc361f@192.168.1.10CSeq:
102 BYEFrom:
<sip:914841622@192.168.1.10:5062>;tag=346072efc18339aai4;ref=110To:
<sip:916611146@192.168.1.10:5062>;tag=61348d706bcel16cVia: SIP/2.0/UDP
192.168.1.10:5062;branch=z9hG4bK-967510eContent-Length: 0Supported:
replacesUser-Agent: MxSipApp/v1.1.3.19 MxSF/v3.2.1.2
```

Outgoing ISDN Call from the SPA9000 Internal Endpoint

```
[pxy]<<192.168.1.169:5060(1070)
[pxy]<<192.168.1.169:5060(1070)
INVITE sip:8648204822@192.168.1.10:6060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.169:5060;branch=z9hG4bK-a9b5b9f7From:
<sip:110@192.168.1.10:6060>;tag=22878a81540c2767o0To:
<sip:8648204822@192.168.1.10:6060>Remote-Party-ID:
<sip:110@192.168.1.10:6060>;screen=yes;party=callingCall-ID: b7719409-
afaa6b7f@192.168.1.169CSeq: 101 INVITEMax-Forwards: 70Contact:
<sip:110@192.168.1.169:5060>;+sip.instance="<00000000-0000-0000-0000-
000E08DAFB1F>"Expires: 240User-Agent: Linksys/SPA942-5.1.10Content-Length:
395Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER,
SUBSCRIBEAllow-Events: dialogSupported: replacesContent-Type: application/
sdpv=0o=- 91509 91509 IN IP4 192.168.1.169s=-c=IN IP4 192.168.1.169t=0
0m=audio 16482 RTP/AVP 0 2 4 8 18 96 97 98 101a=rtpmap:0 PCMU/
8000a=rtpmap:2 G726-32/8000a=rtpmap:4 G723/8000a=rtpmap:8 PCMA/
8000a=rtpmap:18 G729a/8000a=rtpmap:96 G726-40/8000a=rtpmap:97 G726-24/
8000a=rtpmap:98 G726-16/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-
15a=ptime:30a=sendrecv
[BCC]CallRoute:L38648204822
[BCC]CallRoute:L38648204822
[pxy]->192.168.1.169:5060(345)
[pxy]->192.168.1.169:5060(345)
SIP/2.0 100 TryingTo: <sip:8648204822@192.168.1.10:6060>From:
<sip:110@192.168.1.10:6060>;tag=22878a81540c2767o0Call-ID: b7719409-
afaa6b7f@192.168.1.169CSeq: 101 INVITEVia: SIP/2.0/UDP
192.168.1.169:5060;branch=z9hG4bK-a9b5b9f7Server: Linksys/SPA9000-
5.1.6(a)Allow-Events: talk, hold, conference, x-spa-ctiContent-Length: 0
pri-->INVITE-->pub
pri-->INVITE-->pub
Calling:648204822@192.168.1.75:5060
[4]->192.168.1.75:5060(1083)
[4]->192.168.1.75:5060(1083)
```



```
INVITE sip:648204822@192.168.1.75:5060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.10:5062;branch=z9hG4bK-302d0544From: "Mediatrix@"
<sip:1946@192.168.1.75:5060>;tag=1be50f305ef95759o4;ref=110To:
<sip:648204822@192.168.1.75:5060>Remote-Party-ID: "Mediatrix@"
<sip:1946@192.168.1.75:5060>;screen=yes;party=callingCall-ID: 24a3813c-
5308acbd@192.168.1.10CSeq: 101 INVITEMax-Forwards: 70Contact: "Mediatrix@"
<sip:1946@192.168.1.10:5062>Expires: 240User-Agent: Linksys/SPA9000-
5.1.6(a)Allow-Events: talk, hold, conference, x-spa-ctiContent-Length:
394Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFERSupported:
x-sipura, replacesContent-Type: application/sdpv=0o=- 91509 91509 IN IP4
192.168.1.169s=-c=IN IP4 192.168.1.10t=0 0m=audio 16396 RTP/AVP 0 2 4 8 18
96 97 98 101a=rtpmap:0 PCMU/8000a=rtpmap:2 G726-32/8000a=rtpmap:4 G723/
8000a=rtpmap:8 PCMA/8000a=rtpmap:18 G729a/8000a=rtpmap:96 G726-40/
8000a=rtpmap:97 G726-24/8000a=rtpmap:98 G726-16/8000a=rtpmap:101
telephone-event/8000a=fmtp:101 0-15a=ptime:30a=sendrecv
```

```
[4]<<192.168.1.75:5060(344)
```

```
[4]<<192.168.1.75:5060(344)
```

```
SIP/2.0 100 TryingCall-ID: 24a3813c-5308acbd@192.168.1.10CSeq: 101
INVITEFrom: "Mediatrix@"
<sip:1946@192.168.1.75:5060>;tag=1be50f305ef95759o4;ref=110To:
<sip:648204822@192.168.1.75:5060>;tag=c1f2604c716fd1eVia: SIP/2.0/UDP
192.168.1.10:5062;branch=z9hG4bK-302d0544Content-Length: 0User-Agent:
MxSipApp/v1.1.3.19 MxSF/v3.2.1.2
```

```
[4]<<192.168.1.75:5060(725)
```

```
[4]<<192.168.1.75:5060(725)
```

```
SIP/2.0 183 Session ProgressCall-ID: 24a3813c-5308acbd@192.168.1.10CSeq:
101 INVITEFrom: "Mediatrix@"
<sip:1946@192.168.1.75:5060>;tag=1be50f305ef95759o4;ref=110To:
<sip:648204822@192.168.1.75:5060>;tag=c1f2604c716fd1eVia: SIP/2.0/UDP
192.168.1.10:5062;branch=z9hG4bK-302d0544Content-Length: 296Content-Type:
application/sdpContact: sip:648204822@192.168.1.75:5060User-Agent:
MxSipApp/v1.1.3.19 MxSF/v3.2.1.2v=0o=MxSIP 1256160840606035694
47734327189664372 IN IP4 192.168.1.75s=-c=IN IP4 192.168.1.75t=0
0a=sendrecv m=audio 5004 RTP/AVP 0 18 4 8 101a=rtpmap:0 PCMU/8000a=rtpmap:18
G729a/8000a=rtpmap:4 G723/8000a=rtpmap:8 PCMA/8000a=rtpmap:101 telephone-
event/8000a=fmtp:101 0-15
```

```
BCC:Ringback/CallProgress 5
```

```
[pxy]->192.168.1.169:5060(883)
```

```
[pxy]->192.168.1.169:5060(883)
```

```
SIP/2.0 183 Session ProgressTo:
<sip:8648204822@192.168.1.10:6060>;tag=2cce67a3fd752181i4From:
<sip:110@192.168.1.10:6060>;tag=22878a81540c2767o0Call-ID: b7719409-
afaa6b7f@192.168.1.169CSeq: 101 INVITEVia: SIP/2.0/UDP
192.168.1.169:5060;branch=z9hG4bK-a9b5b9f7Server: Linksys/SPA9000-
5.1.6(a)Remote-Party-ID:
<sip:8648204822@192.168.1.10:6060>;screen=yes;party=calledAllow-Events:
talk, hold, conference, x-spa-ctiContent-Length: 297Allow: ACK, BYE,
```

```
CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFERSupported: x-sipura,
replacesContent-Type: application/sdpv=0o=proxy 1256160840606035694
47734327189664372 IN IP4 192.168.1.75s=-c=IN IP4 192.168.1.10t=0
0a=sendrecvm=audio 16398 RTP/AVP 0 18 4 8 101a=rtpmap:0 PCMU/
8000a=rtpmap:18 G729a/8000a=rtpmap:4 G723/8000a=rtpmap:8 PCMA/
8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-15

[4]<<192.168.1.75:5060(716)

[4]<<192.168.1.75:5060(716)

SIP/2.0 180 RingingCall-ID: 24a3813c-5308acbd@192.168.1.10CSeq: 101
INVITEFrom: "Mediatrix@"
<sip:1946@192.168.1.75:5060>;tag=1be50f305ef95759o4;ref=110To:
<sip:648204822@192.168.1.75:5060>;tag=c1f2604c716fd1eVia: SIP/2.0/UDP
192.168.1.10:5062;branch=z9hG4bK-302d0544Content-Length: 296Content-Type:
application/sdpContact: sip:648204822@192.168.1.75:5060User-Agent:
MxSipApp/v1.1.3.19 MxSF/v3.2.1.2v=0o=MxSIP 1256160840606035694
47734327189664372 IN IP4 192.168.1.75s=-c=IN IP4 192.168.1.75t=0
0a=sendrecvm=audio 5004 RTP/AVP 0 18 4 8 101a=rtpmap:0 PCMU/8000a=rtpmap:18
G729a/8000a=rtpmap:4 G723/8000a=rtpmap:8 PCMA/8000a=rtpmap:101 telephone-
event/8000a=fmtp:101 0-15

BCC:Ringback/CallProgress 5

[pxy]->192.168.1.169:5060(883)

[pxy]->192.168.1.169:5060(883)

SIP/2.0 183 Session ProgressTo:
<sip:8648204822@192.168.1.10:6060>;tag=2cce67a3fd752181i4From:
<sip:110@192.168.1.10:6060>;tag=22878a81540c2767o0Call-ID: b7719409-
afaa6b7f@192.168.1.169CSeq: 101 INVITEVia: SIP/2.0/UDP
192.168.1.169:5060;branch=z9hG4bK-a9b5b9f7Server: Linksys/SPA9000-
5.1.6(a)Remote-Party-ID:
<sip:8648204822@192.168.1.10:6060>;screen=yes;party=calledAllow-Events:
talk, hold, conference, x-spa-ctiContent-Length: 297Allow: ACK, BYE,
CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFERSupported: x-sipura,
replacesContent-Type: application/sdpv=0o=proxy 1256160840606035694
47734327189664372 IN IP4 192.168.1.75s=-c=IN IP4 192.168.1.10t=0
0a=sendrecvm=audio 16398 RTP/AVP 0 18 4 8 101a=rtpmap:0 PCMU/
8000a=rtpmap:18 G729a/8000a=rtpmap:4 G723/8000a=rtpmap:8 PCMA/
8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-15

[4]<<192.168.1.75:5060(817)

[4]<<192.168.1.75:5060(817)

SIP/2.0 200 OKCall-ID: 24a3813c-5308acbd@192.168.1.10CSeq: 101 INVITEFrom:
"Mediatrix@"
<sip:1946@192.168.1.75:5060>;tag=1be50f305ef95759o4;ref=110To:
<sip:648204822@192.168.1.75:5060>;tag=c1f2604c716fd1eVia: SIP/2.0/UDP
192.168.1.10:5062;branch=z9hG4bK-302d0544Content-Length: 296Session-
Expires: 3600;refresher=uasAllow: INVITE, ACK, BYE, CANCEL, REFER,
NOTIFYContent-Type: application/sdpSupported: replacesContact:
sip:648204822@192.168.1.75:5060User-Agent: MxSipApp/v1.1.3.19 MxSF/
```

```
v3.2.1.2v=0o=MxSIP 1256160840606035694 47734327189664372 IN IP4
192.168.1.75s=-c=IN IP4 192.168.1.75t=0 0a=sendrecvm=audio 5004 RTP/AVP 0
18 4 8 101a=rtpmap:0 PCMU/8000a=rtpmap:18 G729a/8000a=rtpmap:4 G723/
8000a=rtpmap:8 PCMA/8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-15

[4]->192.168.1.75:5060(476)

[4]->192.168.1.75:5060(476)

ACK sip:648204822@192.168.1.75:5060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.10:5062;branch=z9hG4bK-acc9ba86From: "Mediatrix@"
<sip:1946@192.168.1.75:5060>;tag=1be50f305ef95759o4;ref=110To:
<sip:648204822@192.168.1.75:5060>;tag=clf2604c716fd1eCall-ID: 24a3813c-
5308acbd@192.168.1.10CSeq: 101 ACKMax-Forwards: 70Contact: "Mediatrix@"
<sip:1946@192.168.1.10:5062>User-Agent: Linksys/SPA9000-5.1.6(a)Allow-
Events: talk, hold, conference, x-spa-ctiContent-Length: 0

pub--200-->BCC--200-->pri(from calling)

pub--200-->BCC--200-->pri(from calling)

[pxy]->192.168.1.169:5060(925)

[pxy]->192.168.1.169:5060(925)

SIP/2.0 200 OKTo:
<sip:8648204822@192.168.1.10:6060>;tag=2cce67a3fd752181i4From:
<sip:110@192.168.1.10:6060>;tag=22878a81540c2767o0Call-ID: b7719409-
afaa6b7f@192.168.1.169CSeq: 101 INVITEVia: SIP/2.0/UDP
192.168.1.169:5060;branch=z9hG4bK-a9b5b9f7Contact: "Mediatrix@"
<sip:648204822@192.168.1.10:6060>Server: Linksys/SPA9000-5.1.6(a)Remote-
Party-ID: <sip:8648204822@192.168.1.10:6060>;screen=yes;party=calledAllow-
Events: talk, hold, conference, x-spa-ctiContent-Length: 297Allow: ACK,
BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFERSupported: x-sipura,
replacesContent-Type: application/sdpv=0o=proxy 1256160840606035694
47734327189664372 IN IP4 192.168.1.75s=-c=IN IP4 192.168.1.10t=0
0a=sendrecvm=audio 16398 RTP/AVP 0 18 4 8 101a=rtpmap:0 PCMU/
8000a=rtpmap:18 G729a/8000a=rtpmap:4 G723/8000a=rtpmap:8 PCMA/
8000a=rtpmap:101 telephone-event/8000a=fmtp:101 0-15

[pxy]<<192.168.1.169:5060(476)

[pxy]<<192.168.1.169:5060(476)

ACK sip:648204822@192.168.1.10:6060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.169:5060;branch=z9hG4bK-f8b3d09fFrom:
<sip:110@192.168.1.10:6060>;tag=22878a81540c2767o0To:
<sip:8648204822@192.168.1.10:6060>;tag=2cce67a3fd752181i4Call-ID:
b7719409-afaa6b7f@192.168.1.169CSeq: 101 ACKMax-Forwards: 70Contact:
<sip:110@192.168.1.169:5060>;+sip.instance="<00000000-0000-0000-0000-
00E08DAFB1F>"User-Agent: Linksys/SPA942-5.1.10Content-Length: 0Allow-
Events: dialog

[pxy]<<192.168.1.169:5060(360)

[pxy]<<192.168.1.169:5060(360)
```

```
BYE sip:648204822@192.168.1.10:6060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.169:5060;branch=z9hG4bK-ed786d21From:
<sip:110@192.168.1.10:6060>;tag=22878a81540c2767o0To:
<sip:8648204822@192.168.1.10:6060>;tag=2cce67a3fd752181i4Call-ID:
b7719409-afaa6b7f@192.168.1.169CSeq: 102 BYEMax-Forwards: 70User-Agent:
Linksys/SPA942-5.1.10Content-Length: 0

private side ends the call

private side ends the call

[pxy]->192.168.1.169:5060(361)

[pxy]->192.168.1.169:5060(361)

SIP/2.0 200 OKTo:
<sip:8648204822@192.168.1.10:6060>;tag=2cce67a3fd752181i4From:
<sip:110@192.168.1.10:6060>;tag=22878a81540c2767o0Call-ID: b7719409-
afaa6b7f@192.168.1.169CSeq: 102 BYEVia: SIP/2.0/UDP
192.168.1.169:5060;branch=z9hG4bK-ed786d21Server: Linksys/SPA9000-
5.1.6(a)Allow-Events: talk, hold, conference, x-spa-ctiContent-Length: 0

BCC:Ended

[4]->192.168.1.75:5060(425)

[4]->192.168.1.75:5060(425)

BYE sip:648204822@192.168.1.75:5060 SIP/2.0Via: SIP/2.0/UDP
192.168.1.10:5062;branch=z9hG4bK-3d89b93aFrom: "Mediatrix@"
<sip:1946@192.168.1.75:5060>;tag=1be50f305ef95759o4;ref=110To:
<sip:648204822@192.168.1.75:5060>;tag=c1f2604c716fd1eCall-ID: 24a3813c-
5308acbd@192.168.1.10CSeq: 102 BYEMax-Forwards: 70User-Agent: Linksys/
SPA9000-5.1.6(a)Allow-Events: talk, hold, conference, x-spa-ctiContent-
Length: 0

[4]<<192.168.1.75:5060(358)

[4]<<192.168.1.75:5060(358)

SIP/2.0 200 OKCall-ID: 24a3813c-5308acbd@192.168.1.10CSeq: 102 BYEFrom:
"Mediatrix@"
<sip:1946@192.168.1.75:5060>;tag=1be50f305ef95759o4;ref=110To:
<sip:648204822@192.168.1.75:5060>;tag=c1f2604c716fd1eVia: SIP/2.0/UDP
192.168.1.10:5062;branch=z9hG4bK-3d89b93aContent-Length: 0Supported:
replacesUser-Agent: MxSipApp/v1.1.3.19 MxSF/v3.2.1.2
```

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