ESI SIP Trunking

Installation Guide

0450-1227
Rev. B

esi

We Make It Easy To Communicate

Contents

Introduction

SIP (Session Initiation Protocol) **trunks** are offered by Internet telephony service providers (ITSPs) to connect business communications systems to the public telephone network (PSTN, for *public switched telephone network*), using VoIP (voice over Internet Protocol).

System requirements

SIP trunking is supported on the following ESI Communications Servers when equipped with one or more ESI **SIP Trunking Cards** and running these minimum system software versions:

System	Minimum software version
ESI-1000	18.5.10.0
ESI-600	16.5.10.0
ESI-200	14.5.10.0

System	Minimum software version
ESI-100	12.5.10.0
ESI-50	11.5.10.0
ESI-50L	(Not supported)

Hardware specifications

Per system	ESI-1000		ESI-600		ESI-200		ESI-100		ESI-50	
	4-digit	3-digit	4-digit	3-digit	4-digit	3-digit	4-digit	3-digit	4-digit	3-digit
Maximum SIP Trunking Cards	10	3	6	3	3	3	1	1	1	1
Total SIP trunks	240	72	144	72	72	72	24	24	24	24
Maximum lines available	240	84	168	84	84	84	42	42	35	35

> **Note:** An ESI Communications Server can support multiple SIP Trunking Cards, **but** each installed SIP Trunking Card will subtract from the maximum number of CO lines (*i.e.*, trunks) the system can support.

Codec support

As of system software version xx.5.10.0, only the G.711 codec is supported. For G.711 support, SIP calls use 106 Kbps of bandwidth across a T1. In order to support 24 channels, you would need two T1s if using G.711 or another Internet connection with enough available bandwidth to support 24 concurrent call sessions.

Feature summary

DID for SIP trunks

Inbound SIP calls will always be 10-digit direct inward dialing (DID) numbers. ESI Communications Servers now support 10 digits for DID translations, in Function 224, for routing of all circuits that support DID numbers.

ITSPs supported

As of this revision of this document, ESI SIP trunking supports the following ITSP:

• Broadvox (*www.broadvox.com*)

  – Broadvox ND

  – Broadvox NS

NAT traversal

ESI Communications Servers support carriers regardless of whether they host network address translation (NAT). To handle a carrier that **doesn't** support hosted NAT, it will be necessary to configure the firewall to allow communications between SIP trunks and the ESI Communications Server.

Installation

Perform the following steps to ensure proper SIP trunking operation on the ESI Communications Server:

1. Unbox the ESI SIP Trunking Card and install it in the desired card slot on the ESI Communications Server. (If installing multiple SIP Trunking Cards on the system, repeat this step as many times as needed.)

2. Contact the ESI Technical Support Center at **800 491-3609** or ***techsupp@esi-estech.com*** to have the SIP license activated.

> **Notes:** To have **each** site's license activated, you'll need the site name and sales order number.
>
> The license is for a site so, if you're installing multiple SIP Trunking Cards at a site, one license is sufficient.
>
> Each SIP Trunking Card installation will automatically activate the number of licenses assigned to that card. You will need to contact the ESI Technical Support Center **only** if you're upgrading a SIP8 card to either 16 or 24 channels.

3. In **Function 84**, perform the related programming for the installed **SIP Trunking Card**. (See "Function 84: SIP Trunking Card programming," page 5, for instructions.)

4. In **Function 2142**, perform the related programming to create a **SIP account**. (See "Function 2142: SIP account programming," page 7, for instructions.)

5. In **Function 2141**, perform the related programming for the **SIP trunk ports**. (See "Function 2141: SIP trunk programming," page 11, for instructions.)

6. In **Function 2145**, perform the related programming for the **SIP pilot table**. (See "Function 2145: SIP pilot table programming," page 14, for instructions.)

7. In **Function 224**, perform the related programming for **DID and DNIS/ANI tables**. (See "Function 224: DID and DNIS/ANI translation table programming," page 15, for instructions.)

8. In **Function 225**, perform the related programming for the **pilot number translation table**. (See "Function 225: Pilot number translation table programming," page 16, for instructions.)

Programming

> **Note:** These instructions assume familiarity with the programming of an ESI Communications Server. If necessary, consult the *ESI Communications Servers Programming Manual* (ESI # 0450-1050).

Overview

SIP trunking and feature parameters are assigned using the following programming functions, **in this order**:

- Function 84: SIP Trunking Card programming
- Function 2142: SIP account programming
- Function 2141: SIP trunk programming
- Function 2145: SIP pilot table programming
- Function 224: DID and DNIS/ANI translation table programming
- Function 225: Pilot number translation table programming

Function 84: SIP Trunking Card programming

Use this function **only** if any parameters must be changed for an **individual** SIP Trunking Card ("SIP" in the displays).

> **Note:** For each IP address in the following steps, enter the address in a dotted-decimal format consisting of four numbers separated by dots. Valid numbers are *0* through *255*. Use **#** to enter the dots between numbers.

1. Use the scroll keys to **select an active SIP Trunking Card to program**. Press **#** to continue.
   In this example, the SIP Trunking Card is in port card slot 11 *(PC11)*, which utilizes trunk ports 241–264.

   ```
   SELECT SIP
   PC11 SIP 241-264
   ```

2. The display now shows the **SIP Trunking Card password**, which can be 2–8 characters in length. Enter your desired password and press **#** to continue.

   ```
   SUPPORT PASSWORD
   73829164
   ```

3. Use the scroll keys to **select the type of SIP Trunking Card** that will be used. Press **#** to continue.

   ```
   SIP CARD TYPE
   SIGNALING
   ```

> **Note:** When the ESI Communications Server is behind a firewall, select *ENABLED* for the NAT traversal option (step 7). Program the SIP Trunking Card's IP address (step 4) and the RTP media IP address (step 9) using private IP addresses that are valid on the LAN side of the firewall. Program the SIP public IP address (step 8) using the public IP address of the WAN side of the firewall.
>
> When the ESI Communications Server is **not** behind a firewall, select *DISABLED* for the NAT traversal option (step 7). Program the SIP Trunking Card's IP address (step 4) and the RTP media IP address (step 9) using public IP addresses that are accessible by the ITSP. **With the SIP NAT option (step 7) disabled, the SIP public IP address (step 8) is not needed and this step will not appear.**

4. Enter (or view) the **SIP Trunking Card's IP address**. Press **#** to continue.
   Range: *0.0.0.0* through *255.255.255.255*. **Default:** *0.0.0.0*.

   ```
   PC11 SIP IP ADDR
   0.0.0.0
   ```

5. Enter (or view) the **SIP Trunking Card's subnet mask**. Press **#** to continue.
   Range: *0.0.0.0* through *255.255.255.252*. **Default:** *0.0.0.0*.

   ```
   PC11 SIP IP MASK
   0.0.0.0
   ```

6. Enter (or view) the **SIP Trunking Card's gateway address**. Press **#** to continue.
   Range: *0.0.0.0* through *255.255.255.255*. **Default:** *0.0.0.0*.

   ```
   PC11 SIP GATEWAY
   0.0.0.0
   ```

(Continued)

7. Use the scroll keys to enable or disable **NAT traversal**. Press **#** to continue.

```
PC11 SIP NAT
DISABLED
```

8. Enter (or view) the **public IP address** to use when NAT is enabled. Press **#** to continue.
Range: *0.0.0.0* through *255.255.255.255*. **Default:** *0.0.0.0*.

```
PC11 SIP PUB IP
0.0.0.0
```

9. Enter the **media IP address** to use for RTP[1] traffic to and from the ITSP. Press **#** to continue.
Range: *0.0.0.0* through *255.255.255.255*. **Default:** *0.0.0.0*.

```
PC11 RTP MEDIA IP
0.0.0.0
```

10. Enter the IP address for the **primary DNS[2] server**. Press **#** to continue.
Range: *0.0.0.0* through *255.255.255.255*. **Default:** *0.0.0.0*.

```
PC11 PRI DNS IP
0.0.0.0
```

11. Enter the IP address for the **secondary DNS server**. Press **#** to continue.
Range: *0.0.0.0* through *255.255.255.255*. **Default:** *0.0.0.0*.

```
PC11 SEC DNS IP
0.0.0.0
```

12. Either press **1** to save your changes to the SIP Trunking Card or press **2** to exit without restarting the SIP Trunking Card.

```
RESTART SIP ITSP
 1=YES   2=NO
```

> **Note:** Changes made to SIP Trunking Card programming require that the card be restarted in order for the changes to be uploaded to the card. Press **1** to restart the SIP ITSP now or press **2** to exit and return to the main menu.

> **Warning:** Restarting the SIP ITSP will cause any calls in progress to be disconnected.

---

[1] Real-time Transport Protocol. For more information about RTP, see RFC 3550 *(www.ietf.org/rfc/rfc3550.txt)*.
[2] Domain Name System.

Function 2142: SIP account programming

The procedure in Function 2142 varies after step 2, depending on your selection in that step. Therefore, we have separated the first two (common) steps from the subsequent steps, as you see here.

Common steps 1–2: Selecting SIP provider and ITSP

1. Enter the name for the SIP provider number (or, to select a previously entered SIP provider, use the scroll keys, and press **#** to confirm.
For this example, we'll be using "SIP Provider 1," named *Broadvox*, with a SIP Trunking Card installed in slot number 11.
Range: Up to 10 characters in length.

```
SIP PROVIDER #1
BROADVOX        >
```

2. The ITSP is currently programmed as shown on the display. To select a new value, select a new value, and press **#** to confirm (or, to confirm the existing value, just press **#**).
Choices: *Broadvox-ND*, *Broadvox-NS*. **Default**: *Broadvox-ND*.

```
ITSP SELECTION
BROADVOX-ND    >
```

Subsequent steps for "Broadvox-ND" . . .

If you selected *Broadvox-ND* in step 2 (see "Common steps 1–2: Selecting SIP provider and ITSP," *above*), the **subsequent** steps are as follows:

3. Use the scroll keys to select which SIP Trunking Card will be associated with the selected ITSP, and press **#** to confirm.
Choices: Any available SIP Trunking Card. **Default**: First installed SIP Trunking Card.

```
ITSP ASSOC
PC11 SIP XX-XX >
```

4. Enter the **primary billing numbe**r assigned by the ITSP, and press **#** to confirm.
If you need detailed information about entering characters, press **HELP**. (See Appendix A, page 19.)
Range: Up to 10 characters. **Default**: [Blank].

```
PRIMARY NUMBER
```

5. Enter the **optional account name** assigned by the ITSP, and press **#** to confirm.
If you need detailed information about entering characters, press **HELP**. (See Appendix A, page 19.)
Range: Up to 32 characters. **Default**: [Blank].

```
ACCOUNT NAME
```

(Continued)

6. Enter the **account password** assigned by the ITSP, and press **#** to confirm.
If you need detailed information about entering characters, press **HELP**. (See Appendix A, page 19.)
Range: Up to 32 characters. **Default**: [Blank].

```
ACCOUNT PASSWORD
```

7. Enter the **port number** that will be used for inbound SIP traffic to the SIP Trunking Card, and press **#** to confirm.
Range: *00000–65535*. **Default**: *5060*.

```
LOCAL SIP PORT
5060
```

8. Enter the **URI of the ITSP**, and press **#** to confirm.
Range: Up to 64 characters. **Default**: [Blank]. (See Appendix A, page 19.)

```
ITSP URI
```

9. Enter the **starting** port number of the RTP port range to be used, and press **#** to confirm.
Range: *00000–65535*. **Default**: *10000*.

```
RTP PORT RANGE
BEGIN: 10000
```

10. Enter the **ending** port number of the RTP port range to be used, and press **#** to confirm
Range: *00000–65535*. **Default**: *11000*.

```
RTP PORT RANGE
END: 11000
```

11. Use the scroll keys to select a codec, and press **#** to confirm.
Choices: *G.711, G.729.a.* **Default**: *G.711*.

```
SIP CHAN CODEC
G.711
```

12. Either press **1** to save your changes to the SIP Trunking Card or press **2** to exit without restarting the SIP Trunking Card.

```
RESTART SIP ITSP
 1=YES    2=NO
```

> **Note:** Changes made to SIP Trunking Card programming require that the card be restarted in order for the changes to be uploaded to the card. Press **1** to restart the SIP ITSP now or press **2** to exit and return to the main menu.

> **Warning:** Restarting the SIP ITSP will cause any calls in progress to be disconnected.

Subsequent steps for "Broadvox-NS" . . .

If you selected *Broadvox-NS* in step 2 (see "Common steps 1–2: Selecting SIP provider and ITSP," page 7), the **subsequent** steps are as follows:

3. Use the scroll keys to select which SIP Trunking Card will be associated with the selected ITSP, and press **#** to confirm.
Choices: Any available SIP Trunking Card. **Default**: First installed SIP Trunking Card.

```
ITSP ASSOC
PC11 SIP XX-XX >
```

4. Enter the **primary billing numbe**r assigned by the ITSP, and press **#** to confirm.
If you need detailed information about entering characters, press **HELP**. (See Appendix A, page 19.)
Range: Up to 10 characters. **Default**: [Blank].

```
PRIMARY NUMBER
```

5. Enter the **optional account name** assigned by the ITSP, and press **#** to confirm.
If you need detailed information about entering characters, press **HELP**. (See Appendix A, page 19.)
Range: Up to 32 characters. **Default**: [Blank].

```
ACCOUNT NAME
```

6. Enter the **port number** that will be used for inbound SIP traffic to the SIP Trunking Card, and press **#** to confirm.
Range: *00000–65535*. **Default**: *5060*.

```
LOCAL SIP PORT
5060
```

7. Enter the **URI of the ITSP**, and press **#** to confirm.
Range: Up to 64 characters. **Default**: [Blank]. (See Appendix A, page 19.)

```
ITSP URI
```

8. Enter the **starting** port number of the RTP port range to be used, and press **#** to confirm.
Range: *00000–65535*. **Default**: *10000*.

```
RTP PORT RANGE
BEGIN: 10000
```

9. Enter the **ending** port number of the RTP port range to be used, and press **#** to confirm
Range: *00000–65535*. **Default**: *11000*.

```
RTP PORT RANGE
END: 11000
```

(Continued)

10. Use the scroll keys to select a codec, and press **#** to confirm.
Choices: *G.711, G.729.a*. **Default**: *G.711*.

```
SIP CHAN CODEC
G.711
```

11. Either press **1** to save your changes to the SIP Trunking Card or press **2** to exit without restarting the SIP Trunking Card.

```
RESTART SIP ITSP
 1=YES    2=NO
```

> **Note:** Changes made to SIP Trunking Card programming require that the card be restarted in order for the changes to be uploaded to the card. Press **1** to restart the SIP ITSP now or press **2** to exit and return to the main menu.

> **Warning:** Restarting the SIP ITSP will cause any calls in progress to be disconnected.

Function 2141: SIP trunk programming

1. Select the line keys that are to have the same programming during **day** mode, and press **#** to confirm.
   If multiple SIP Trunking Cards have been installed, use the scroll keys to select the desired SIP Trunking Card. For this example, we'll be using a SIP Trunking Card in port card slot 11 ("PC 11").

   ```
   PC11 SIP XX-XX D
   CO: CIRCUIT
   ```

   > **Note:** You will be able to select only the number of CO lines for which you have licenses.

2. Use the scroll keys to select the **SIP provider** to be used for the selected trunks, and press **#** to confirm
   For this example, we'll be using SIP provider 1.
   Range: *1–10* (as previously defined in Function 2142; for instructions on programming that function, see page 7).

   ```
   SEL PROVIDER #
   1-BROADVOX
   ```

3. Enter the **line name** (if you want to use a different one than what appears), and press **#** to confirm.
   If you need detailed information about entering characters, press **HELP**.
   Range: Up to 10 characters. **Default**: [Blank].

   ```
   CO LINE
   NAME:
   ```

   > **Note:** Entering a line name delays presentation of inbound Caller ID data until the call is answered.

4. If tenant service is enabled in Function 169 (for more information on this Function, refer to the *ESI Communications Servers Programming Manual*, ESI # 0450-1050), use the scroll keys to select the **tenant number**, and press **#** to confirm. (If tenant service isn't enabled, press **#** to move to the next step.)
   Range: [Any valid tenant number]. **Default**: *1*.

   ```
   CO LINE
   TENANT:  1
   ```

5. This line is programmed for the line groups as displayed. Enter the new **group number** (if there is one), and press **#** to confirm.
   Range: *9, 8, 71–76* (if *71–76* were assigned as CO lines, in Function 164; for more information on this Function, refer to the *ESI Communications Servers Programming Manual*, ESI # 0450-1050). **Default**: *9*].

   ```
      DAY CO LINE
   CO GROUP:  9
   ```

6. Use the scroll keys to select a **distinctive ring tone** for the new line, and press **#** to confirm.
   Range: Normal and ring tones 1–6. **Default**: Normal.

   ```
   CO RING TONE
   TONE NORMAL
   ```

(Continued)

7. This line is programmed to be answered as displayed. To select a specific station, mailbox, or branch ID which should answer this SIP trunk, press a scroll key and enter the new value, and press **#** to confirm. (If the current value is acceptable, just press **#** to confirm.)
   Default: ID 1.

```
RING 1
ID: 1
```

8. Select the line keys that are to have the same programming during **night** mode, and press **#** to confirm.
   If multiple SIP Trunking Cards have been installed, use the scroll keys to select the desired SIP Trunking Card.
   For this example, we'll be using a SIP Trunking Card in port card slot 11 ("PC 11").

```
PC11 SIP XX-XX N
CO: CIRCUIT
```

9. Use the scroll keys to select the **SIP provider** to be used for the selected trunks, and press **#** to confirm
   Again in this example, we'll be using SIP provider 1.
   Range: *1–10* (as previously defined in Function 2142; for instructions on programming that function, see page 7).

   > **Note:** During night mode programming, this is a read-only field.

```
SEL PROVIDER #
1-BROADVOX
```

10. Enter the **line name** (if you want to use a different one than what appears), and press **#** to confirm.
    If you need detailed information about entering characters, press **HELP**.
    Range: Up to 10 characters. **Default**: [Blank].

```
CO LINE
NAME:
```

11. If tenant service is enabled in Function 169 (for more information on this Function, refer to the *ESI Communications Servers Programming Manual*, ESI # 0450-1050), use the scroll keys to select the **tenant number**, and press **#** to confirm. (If tenant service isn't enabled, press **#** to move to the next step.)
    Range: [Any valid tenant number]. **Default**: *1*.

```
CO LINE
TENANT: 1
```

12. Use the scroll keys to select a **distinctive ring tone** for the new line, and press **#** to confirm.
    Range: Normal and ring tones 1–6. **Default**: Normal.

```
CO RING TONE
TONE NORMAL
```

(Continued)

13. This line is programmed to be answered as displayed. To select a specific station, mailbox, or branch ID which should answer this SIP trunk, press a scroll key and enter the new value, and press **#** to confirm. (If the current value is acceptable, just press **#** to confirm.)
Default: ID 1.

```
RING 1
ID: 1
```

14. Either press **1** to save your changes to the SIP Trunking Card or press **2** to exit without restarting the SIP Trunking Card.

```
RESTART SIP ITSP
 1=YES    2=NO
```

> **Note:** Changes made to SIP Trunking Card programming require that the card be restarted in order for the changes to be uploaded to the card. Press **1** to restart the SIP ITSP now or press **2** to exit and return to the main menu.

> **Warning:** Restarting the SIP ITSP will cause any calls in progress to be disconnected.

Function 2145: SIP pilot table programming

With DID enabled, call processing will check first the pilot table and then the DID table (programmed in Functions 225 and 224, respectively), and route the call accordingly. Any number not found in the pilot table or DID table will be routed to the DID exception routing. With DID disabled, all calls will follow the routing as programmed in SIP trunk programming (Function 2141, beginning on page 11).

Default: Disabled.

Function 224: DID and DNIS/ANI translation table programming

1. **Either** enter the **DNIS or DID number or** use the scroll keys to select a previously programmed number or exception, and press **#** to confirm. For this example, we'll use *972 555-2145* as the DID number and will route to extension 245.

> **Notes**: Pressing the right scroll key will start with a blank number field for entry and continue through all previously programmed numbers, ending with the exception entry (continuing to scroll will start over the blank entry). Pressing the left scroll key will start with the exception entry, continue in reverse order with previously programmed numbers and end with the blank number entry (continuing to scroll will start over with the exception entry).
>
> Pressing **HOLD** will delete an entry.

```
DID/DNIS NUMBER
#:  9725552145
```

2. Enter the desired **name** (if no change is desired, enter nothing), and press **#** to confirm.
   Range: Up to 10 characters. Default: *[Blank.]*

> **Note:** If a name is entered for a DID and there is an inbound CO call, the DID name (instead of the Caller ID number) will be displayed until the call is answered.

```
DID#:  9725552145
NAME:
```

3. The system now shows **day** routing on the display.
   To select a branch ID, extension, or mailbox, use the scroll keys (or, if no change is desired, enter nothing), and press **#** to confirm.
   Choices: Any valid ID branch, extension, or mailbox. **Default**: *[Blank]*.

```
DID#:  9725552145
DAY EXT:  245
```

4. The system now shows **night** routing on the display.
   To select a branch ID, extension, or mailbox, use the scroll keys (or, if no change is desired, enter nothing), and press **#** to confirm.
   Choices: Any valid ID branch, extension, or mailbox. **Default**: *[Blank]*.

```
DID#:  9725552145
NGT EXT:  245
```

Function 225: Pilot number translation table

> **Note:** This feature is accessible only if Function 2135 or Function 2145 is enabled.[1]

Every call on a SIP span is sent with the called number in the setup message. This means you can determine whether to use a number as a pilot number or as a DID number. Pilot numbers are a means of routing a company's primary published phone number differently from a DID. Each ESI Communications Server has a different capacity for pilot entries, each with its own answer ring assignment.[2] This is important, because the dynamic channel allocation prevents you from routing based on channel.

Pilot numbers can be routed to an ID branch, department, extension or mailbox. This varies from DIDs in that, when a DID number is routed to an extension, it is considered answered and follows the busy/no-answer routing of the extension; but, with pilot numbers, the call will be routed to the extensions for live answer for the designated number of rings, then can be routed to an ID branch in the event of busy/no-answer. If the PRI span is used for outgoing calls, pilot table entry 1 is used for outgoing Caller ID, as well. Because of this, pilot number entry 1 should always be programmed with 10 digits.

> **Note:** An extension's Caller ID number setting (programmed in Function 31[1]) overrides the pilot number's Caller ID information set here.

Max Channels value

Max Channels is the maximum number of channels that can be used simultaneously for incoming calls to a particular pilot number. Incoming calls to that pilot number are rejected (CO plays busy signal) when the Max Channels value is exceeded for the pilot number. Max Channels is applicable only to incoming calls.

> **Notes:** This setting works **only** if the digits in the pilot number entry **exactly match** the digits that the PRI or SIP carrier sends in its messaging.
>
> The sum of maximum channels of any PRI pilot number must not exceed 23.

To program the pilot number translation table:

1. Use the scroll keys to select the entry number.

2. Enter the pilot number, two to 10 digits in length.[3]

3. Enter the name for the pilot number.

4. Enter the maximum channels allowed for this pilot number.
   Range: 1–24 (SIP) or 1–23 (PRI). **Default**: 24 (SIP) or 23 (PRI).

5. Enter the distinctive CO ring tone for this pilot number.
   Range: 1–6 or normal (disabled). **Default:** Normal (disabled).

(Continued)

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[1] See the *ESI Communications Servers Programming Manual* (ESI # 0450-1050).

[2] See "Translation tables" under "System capacities"" in the *ESI Communications Servers Programming Manual* (ESI # 0450-1050).

[3] If the carrier sends a pilot number with fewer than 10 digits, the system will use only the rightmost digits of the called number.

6. Set answer ring assignments for both day and night modes.

> **Note:** For programming information, see step 7 in Function 2131.

7. Repeat steps 1–6 for the remaining pilot numbers.

Example:

Entry no.	Pilot no.	Name	Max chs.	Ring tone	Day				Night			
					Ring 1	Ring 3	Ring 5	Ring 9	Ring 1	Ring 3	Ring 5	Ring 9
1	2145551452	ABC Co.	16		X100	X100, X101	X101	ID1	MB100	MB100	MB100	MB100
2	1245	Private line	1		X106	X106	MB106		MB106	MB106	MB106	
3												
4												
5												

Tenant service and pilot numbers

If tenant service is enabled, the first pilot number (*i.e.*, entry 1) is used for tenant 1's outgoing Caller ID, and so on through entry 8 for tenant 8's outgoing Caller ID. When a station (or the station's voice mailbox) originates an outgoing call via a PRI/SIP channel, the pilot number of the station's tenant will be used for that call's outgoing Caller ID. Also: any auto attendant outdial branches that originate from branch ID 1 will use Caller ID for tenant 1, and so on through branch ID 8 for tenant 8.

Special characters

As a security measure, some ITSPs require SIP account passwords to contain **all** of the following:

(a.) Lower-case letters.

(b.) Upper-case letters.

(c.) Numbers.

> **Examples**: | *mypassword* | = | Not OK |
> | --- | --- | --- |
> | *myPassword* | = | Not OK |
> | *myPassword123* | = | OK |

Here is a table of ESI Feature Phone keystrokes to enter characters:

Pressing this key . . .	Will produce the following characters (in this order) . . .
1	1 Q Z q z
2	2 A B C a b c
3	3 D E F d e f
4	4 G H I g h i
5	5 J K L j k l
6	6 M N O m n o
7	7 P R S p r s
8	8 T U V t u v
9	9 W X Y w x y
0	0 . - _ : ; / @ (*i.e.*, zero, period, dash, underscore, colon, semicolon, slash, and at-symbol)

In each case, pressing **#** will confirm the entry in the display and move the cursor to the next character.

Troubleshooting • Port forwarding

Troubleshooting chart

Issue	Solution
When using Broadvox version ND trunks, if you dial **9**, you get the response, "That line group is not valid."	If the site is using version ND trunks, registration isn't taking place correctly. Ensure port forwarding in the router is correct. Also, ensure that the correct DNS IP addresses are programmed in Function 84.
An incoming call results in one-way audio, in which the caller is the only person who can hear audio.	Contact Broadvox and ensure that Broadvox has automatic NAT detection enabled on its (Broadvox's) side.
There is no audio in either direction.	Ensure that the correct RTP ports have been forwarded to the SIP Trunking Card.
During the placing of an outbound call, the display returns to idle as soon as the number is dialed.	Verify that the ITSP URI has been entered correctly (a misspelled URI will result in failed calls).
You cannot dial a seven-digit local number.	A SIP trunking call requires 10 digits for call completion.
You are unable to connect to the system using a modem over a SIP trunk.	Due to limitations from SIP callers, modem connections won't be successful and you'll need to use the NSP to connect remotely to the system.

Port forwarding

To ensure that there is proper signaling between the ITSP and the SIP Trunking Card, the following port forwarding must be performed in order for the SIP trunks to function correctly:

• Port 5060 (UDP) must be forwarded to the primary IP address assigned to port 1 of the SIP Trunking Card.

• Ports 10000–11000[1] (UDP) must be forwarded to the secondary (media) IP address assigned to port 2 of the SIP Trunking Card.

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[1] Port numbers are dependent on the defined range in Function 2142.

Appendix A

Add to inbound filter list of router

Allow the media IPs below and all UDP ports up to 65535	
209.249.3.58	209.249.3.60

Function 2142 — Primary number and account name

Function 84 — Public IP address

Function 2142 — Account password

Trunk number	Turn-up ticket	Trunk type	BTN	Source IP	Password
1000458	320133	GoLocal	2144735188	68.165.86.131	

*If turn-up ticket has a password and the source IP shows "dynamic," then trunk type is **ND**.*

Function 2142 — ITSP URI

*If turn-up ticket has a source IP only, then trunk type is **NS**.*

Trunk number	DNS A record	DNS SRV record	IP address 1
1000458	dfwnx01ga1.pa.broadvox.net	dfwnx01ga1.psrv.broadvox.net	209.249.3.59

Functions 224 and/or 225 — DID number programming

Trunk number	DIDs
1000458	2144735188
	2142272130
	2142272131
	2142272132
	2142272133
	2142272134
	2142272135
	2142272136
	2142272137
	2142272138
	2142272139

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