

# Managing SIP Trunks

A SIP trunk is an IP based network connection between Nyquist and an Internet Telephony Service Provider (ITSP), which is also known as a VoIP telephone service provider. It allows you to use VoIP telephony beyond the facility's firewall without the need for a Public Switched Telephone Network (PSTN), providing a configuration that is easier and less expensive to operate and maintain. SIP trunks can carry VoIP calls, Enhanced 911, and other real-time communications services.

A SIP Tie-Trunk (or a SIP Tie-Line) functions similarly to a SIP trunk, but serves as an IP based inter-connection between Nyquist and a local (premises-based) or hosted IP-PBX. For information about using SIP Tie-Trunks with Nyquist, refer to [SIP Trunk Custom Configuration Settings](#).

Before you can set SIP parameters in Nyquist, you must first set up the system's DISA function by creating a station with the **Type** of **DISA Line**. (See [Adding a Station](#).)

If you will require support from Bogen during the implementation process, you must install Teamviewer on the Nyquist system server and enable Remote access capabilities in the LAN to allow a remote connection to the server. In addition, Wireshark must be installed in the Nyquist system server to allow packet inspection during SIP implementation and troubleshooting. It is recommended that the technician performing the install be skilled with SIP implementation and testing. The Firewall setting must allow Port 5060 and Ports 10000 – 20000 for the RTP traffic. Any other type of SIP connectivity could be available via an individual case basis process.

*See also:*

- [How Nyquist Handles Incoming SIP Calls](#)
- [How Nyquist Handles Outgoing SIP Calls](#)
- [Viewing SIP Trunks](#)
- [Adding SIP Trunk Configuration Parameters](#)
- [Viewing Outside Line Status](#)

## How Nyquist Handles Incoming SIP Calls

*Note:* If a SIP trunk is configured but disabled in the Nyquist System Parameters, then incoming calls from the SIP trunk are ignored.

When a SIP trunk is enabled and Nyquist receives an incoming call on that SIP trunk, Nyquist routes the call based on the **Access** parameter set in the Nyquist SIP trunk's parameters. The **Access** parameter is set when editing or adding SIP trunk configuration (see [Editing a SIP Trunk Configuration](#) and [Adding SIP Trunk Configuration Parameters](#)) and must match the ITSP switch configuration.

Based on the **Access** parameter setting, Nyquist does one of the following:

- Routes the call to a DISA function/station
- Routes the call to a Security DISA function/station
- Routes the call to a defined Day Admin or Night Admin

The following **Access** settings route the incoming SIP trunk call to the DISA functionality:

- DISA – Bi-directional – No password
- DISA – Bi-directional – Password
- DISA – Incoming only – No password
- DISA – Incoming only – Password

When the Nyquist system receives an incoming call, the caller will hear a dial tone. The caller can then dial any Nyquist extension or any Dual Tone Multi-Frequency (DTMF) code that is associated with a Nyquist feature. For example, after hearing the dial tone, the caller can dial #0911 to launch an Emergency All-Call page. (See *"Nyquist Dial Codes and DTMF Codes"* on page 311 for a list of DTMF codes.)

When incoming calls are routed to the DISA function, the SIP trunk **Extension** setting is used to map the SIP trunk to a station of type **DISA Line**; the associated station's CoS configuration is used to determine what Nyquist functions the incoming DISA user is allowed to initiate. (See [Adding CoS Parameters for a Station](#).) The **Extension** parameter is also used as the caller ID when a DISA user dials a station extension.

The following **Access** settings route the incoming SIP trunk call to Security DISA functionality:

- Security DISA – Bi-directional – Listen only
- Security DISA – Incoming – Listen only

The caller must enter a valid extension to be monitored and can only listen to calls or station locations.

The following **Access** settings route the incoming SIP trunk call to the defined Day or Night Admin:

- Unrestricted
- Incoming Only

If the Day or Night Admin does not answer the incoming call, the call is routed, or rolled over, to the Admin Group. (See [Using Admin Groups](#).)

If the call is not answered during night hours and the Night Ring option is enabled, the call is routed using the Night Ring functionality. (See [Setting Night Call Options](#).)

*See also:*

- [Managing SIP Trunks](#)
- [How Nyquist Handles Outgoing SIP Calls](#)
- [SIP Trunk Custom Configuration Settings](#)
- [Adding SIP Trunk Configuration Parameters](#)

## How Nyquist Handles Outgoing SIP Calls

Nyquist users can initiate outside calls by dialing a number that starts with the outside call prefix (98). Outside calls are only sent to SIP Trunks that have one of the following **Access** settings:

- DISA – Bi-directional – No Password
- DISA – Bi-directional – Password
- Security DISA – Bi-directional – Listen only
- Unrestricted
- Outgoing Only

When outside calls are placed, the outbound caller ID is set to the Direct Inward Dial (DID) defined for the SIP Trunk. (See [Editing a SIP Trunk Configuration](#) or [Adding SIP Trunk Configuration Parameters](#).)

When **Access** is set to **911 Only**, only outgoing 911 calls may be routed on the associated SIP trunk. The **DID** parameter is not used for outgoing 911 calls; it is expected that the SIP trunk provider has mapped the Billing Telephone Number (BTN) to E911 service.

All outgoing calls are routed through the SIP trunk provider using the following SIP trunk configuration parameters:

- Host IP Address
- Username
- Password

If the SIP trunk provider expects all called numbers to be prefixed with a specific code, the **Dial Prefix** setting can be used to satisfy the requirement.

If the SIP trunk provider expects all called 7-digit numbers to be prefixed with a specific local area code, the **Local Area Code** setting can be used to satisfy the requirement. (See [Adding SIP Trunk Configuration Parameters](#).)

*See also:*

- [Managing SIP Trunks](#)
- [How Nyquist Handles Incoming SIP Calls](#)

## **Adding SIP Trunk Configuration Parameters**

Before adding a SIP trunk, the following parameters must be configured or known:

- Extension for the Nyquist DISA line station associated with the specified SIP trunk; the station will have **DISA Line** selected as **Type** (see [Viewing Station Configuration Settings](#)).
- Dial prefix and format used to connect to an outside line
- Local area code
- Username and password for the DISA line

- DID phone number associated with the DISA line
- Codecs allowed
- Admin Group set up for the SIP extension (see [Using Admin Groups](#))

To add a SIP trunk:

- Step 1 On the navigation bar, select **SIP Trunks**.
- Step 2 On the SIP Trunks page, select the
- Step 3 **Add** icon.
- Step 4 On the Add SIP Trunk page, complete the parameters. (See [Adding SIP Trunk Configuration Parameters](#).)
- Step 5 Select **Save**.

**Table 2-8, Add SIP Trunk Page Parameters**

<b>Name</b>	Enter the name for the SIP trunk. The name cannot contain spaces or a slash (/) and cannot exceed 16 characters.
<b>Extension</b>	Select the DISA station extension that is associated with the SIP trunk.
<b>Dial Prefix</b>	Enter the dial prefix required by the ITSP to complete an outbound call over the SIP trunk. For example, :9, 9+1, etc.
<b>Local Area Code</b>	Enter the local area code if the ITSP requires 7-digit telephone numbers be prefixed with the area code. Otherwise, this field is left blank.
<b>Enabled</b>	Specify if the SIP trunk is enabled. The SIP trunk should be configured before it is enabled to avoid erratic system behavior.
<b>Access</b>	Select the outside access permissions for the SIP trunk. Options are: <ul style="list-style-type: none"> <li>• 911 Only</li> <li>• DISA – Bi-directional – No password</li> <li>• DISA – Bi-directional – Password</li> <li>• DISA – Incoming only – No password</li> </ul>

- DISA – Incoming only – Password
- Incoming Only
- No Access
- Outgoing Only
- Security DISA – Bi-directional – Listen only
- Security DISA – Incoming – Listen only
- Unrestricted

<b>Username</b>	Enter the user name that is required to access the SIP trunk.
<b>Password</b>	Enter the password that is required to access the SIP trunk. The password is provided by the SIP trunk provider or hosted VoIP provider.
<b>Password Confirmed</b>	Re-enter the SIP trunk password.
<b>Host</b>	Enter the host name or IP address for the ITSP.
<b>DID</b>	Enter the 10-digit DID telephone number that the SIP trunk or hosted VoIP provider has assigned the SIP trunk, using the format NPANXXxxxx (that is, no spaces or dashes).
<b>Day Admin</b>	Select the Admin Station to call during daytime hours.
<b>Night Admin</b>	Select the Admin Station to call during nighttime hours.
<b>Admin Group</b>	Select the Admin Group to call if the day or night Admin Stations do not answer the call.
<b>Call Recording</b>	Provide if incoming and outgoing calls are to be recorded.
<b>Allow</b>	Provide a list of media codecs that are allowed. PSTN standard codecs are G.711 or ULAW.
<b>Description</b>	Provide user provided description for the SIP trunk.

## Custom Settings

Provide custom setting configurations that are provided by Technical Support. (See [SIP Trunk Custom Configuration Settings](#).)

*See also:*

- [Managing SIP Trunks](#)
- [Understanding System Parameters](#)

## SIP Trunk Custom Configuration Settings

The **Custom Settings** parameter can be used if additional parameters are needed during SIP trunk configuration to satisfy SIP trunk provider requirements. Bogen Technical Support will provide any necessary custom settings. In most cases, **Custom Settings** are not required.

**Custom Settings** can be entered when adding or editing SIP Trunk parameters (see [Adding SIP Trunk Configuration Parameters](#) or [Editing a SIP Trunk Configuration](#)) using the following format:

:TABLE-NAME:<variable>=<value>:TABLE-NAME:

In this format, TABLE-NAME is one of the following:

- ENDPOINT
- AOR
- AUTH
- SIPTRUNK

Variable is a valid variable from one of the pjsip tables (ps\_endpoints, ps\_aors, ps\_auths) or sip\_trunk table.

Value is a valid value for a specified variable.

Contact Technical Support for information regarding Custom Settings for your specific IP-PBX type and configuration.

*See also:*

- [Viewing SIP Trunks](#)
- [Understanding System Parameters](#)

## Viewing SIP Trunks

To view available SIP trunks for your facility:

On the navigation bar, select **SIP Trunks**.

The following table describes the details that can be viewed for each SIP trunk:

**Table 2-9, SIP Trunks Page Parameters**

<b>Name</b>	Provides the name for the SIP trunk. The name cannot contain spaces or a slash (/).
<b>Description</b>	Provides user provided description for the SIP trunk.
<b>Enabled</b>	Specifies if the SIP trunk is enabled.
<b>Status</b>	Provides registration status of the SIP trunk. If the credentials are used by the SIP end point, the <b>Status</b> might be <b>Register</b> . If the SIP end point does not use credentials, the <b>Status</b> might be <b>Rejected</b> but this will not impede a successful SIP connection.
<b>Access</b>	Identifies the outside access permissions for the SIP trunk. Options are: <ul style="list-style-type: none"><li>• 911 Only</li><li>• DISA – Bi-directional – No password</li><li>• DISA – Bi-directional – Password</li><li>• DISA – Incoming only – No password</li><li>• DISA – Incoming only – Password</li><li>• Incoming Only</li><li>• No Access</li><li>• Outgoing Only</li><li>• Security DISA – Bi-directional – Listen only</li><li>• Security DISA – Incoming – Listen only</li><li>• Unrestricted</li></ul>



<b>Extension</b>	Identifies the station extension that is associated with the outside line (DISA station mapping and caller ID for incoming DISA).
<b>DID</b>	Identifies the DID telephone number associated with the outside line. This number is assigned by the ITSP provider.
<b>Username</b>	Identifies the user name that is required to access the SIP trunk.
<b>Host</b>	Identifies the host name or IP address for ITSP.
<b>Dial Prefix</b>	Provides the prefix required by the ITSP to complete an outbound call over the SIP trunk. For example, :9, 9+1, etc.
<b>Local Area Code</b>	Provides the local area code if the ITSP requires the 7-digit telephone number be prefixed with the area code. Otherwise, this field is left blank.
<b>Type</b>	Identifies this system trunk's type as a SIP trunk.
<b>Day Admin</b>	Identifies the Admin Station to call during daytime hours.
<b>Night Admin</b>	Identifies the Admin Station to call during nighttime hours.
<b>Admin Group</b>	Identifies the Admin Group to call if the day or night Admin Stations do not answer the call.
<b>Codecs Allowed</b>	Provides a list of media codecs that are allowed. Codecs are separated by a semi-colon.
<b>Call Recording</b>	Indicates if incoming and outgoing calls are being recorded.
<b>Custom Settings</b>	Identifies custom setting configurations that are provided by Technical Support. For more information, refer to <a href="#">SIP Trunk Custom Configuration Settings</a> .

*See also:*

- [Managing SIP Trunks](#)
- [Understanding System Parameters](#)

## Editing a SIP Trunk Configuration

You cannot edit the name given to a SIP trunk, but you can edit other fields, such as the Dial Prefix and Local Area Code.

*To edit a SIP Trunk's configuration:*

- Step 1 On the navigation bar, select **SIP Trunks**.
- Step 2 On the SIP Trunks page, select the **Edit** icon for the SIP trunk that you want to edit.
- Step 3 On the Edit SIP Trunk page, make the desired changes. (See [Edit SIP Trunks Page Parameters](#).)
- Step 4 Select **Save**.

**Table 2-10, Edit SIP Trunks Page Parameters**

<b>Name</b>	Displays the name for the SIP trunk. This parameter cannot be edited.
<b>Extension</b>	Enter the Nyquist DISA station extension that is associated with the outside line (DISA station mapping and caller ID for incoming DISA).
<b>Dial Prefix</b>	Enter the dial prefix required by the ITSP to complete an outbound call over the SIP trunk. For example, :9, 9+1, etc.
<b>Local Area Code</b>	Enter the local area code if the ITSP requires the 7-digit telephone number be prefixed with the area code. Otherwise, this field is left blank.
<b>Enabled</b>	Specify if the SIP trunk is enabled. SIP trunks should be fully configured before they are enabled to avoid erratic system behavior.
<b>Access</b>	Enter the outside access permissions for the SIP trunk. Options are:

- 911 Only
- DISA – Bi-directional – No password
- DISA – Bi-directional – Password
- DISA – Incoming only – No password
- DISA – Incoming only – Password
- Incoming Only
- No Access
- Outgoing Only
- Security DISA – Bi-directional – Listen only
- Security DISA – Incoming – Listen only
- Unrestricted

**Username**

Enter the user name that is required to access the SIP trunk.

**Change Password**

Enter the password that is required to access the SIP trunk. The password is provided by the SIP trunk provider or hosted VoIP provider.

**Confirm Password Change**

Re-enter the SIP trunk password.

**Host**

Enter the host name or IP address for the ITSP.

**DID**

Enter the DID telephone number associated with the outside line.

**Day Admin**

Select the Admin Station to call during daytime hours.

**Night Admin**

Select the Admin Station to call during nighttime hours.

**Admin Group**

Select the Admin Group to call if the day or night Admin Stations do not answer the call.

**Call Recording**

Select if incoming and outgoing calls are to be recorded.

Enter a list of media codecs that are allowed. PSTN

<b>Allow</b>	standard codecs are G.711 or ULAW.
<b>Description</b>	Edit user provided description for the SIP trunk.
<b>Custom Settings</b>	Edit custom setting configurations that are provided by Technical Support. For more information, refer to <a href="#">SIP Trunk Custom Configuration Settings</a> .

*See also:*

- [Managing SIP Trunks](#)
- [Understanding System Parameters](#)

## Deleting a SIP Trunk Configuration

If you are no longer paying for or using a SIP trunk service via an external provider, you may want to delete a previously added SIP trunk.

*To delete a SIP trunk:*

- Step 1 On the navigation bar, select **SIP Trunks**.
- Step 2 On the SIP Trunks page, select the
- Step 3 **Delete** icon next to the SIP trunk that you want to delete.
- Step 4 Select
- Step 5 **Delete**.

*See also:*

- [Managing SIP Trunks](#)
- [Understanding System Parameters](#)