NEC

DSX

Skype SIP Trunk Setup

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Revision History		
Issue	Date	Revisions
1.00	4/23/11	• Initial release.

Contents

Tables

1 Overview

The DSX is compatible with Skype Connect SIP Trunking. This setup guide summarizes the account information you will receive from Skype and provides step-by-step instructions on how to program that information into the DSX.

You can set up SIP trunks as standard loop start lines or DID lines. All the feature available with other types of loop start and DID lines are also available with Skype Connect SIP trunk lines. For details on the features available with loop start and DID lines, refer to the on-line DSX Software Manual at: http://www.necdsx.com/docs/dita/dsx_v3_complete/index.html

2 Skype Setup

To set up your Skype Connect service, contact Skype at http://www.skype.com. Once your Skype Connect service is set up you need the following information from your Skype Connect profile:

Table 1: Skype Account Information

Option	Description
Skype Connect Address	The Skype Connect Address is the SIP registration domain for your Skype Connect SIP trunks. When your SIP trunks initially register they use this domain.
SIP User	This is the username that Skype Connect uses to authenticate your registration.
Password	This is the password required with your SIP User.
Incoming Calls	These are the telephone numbers that users will dial to reach you over your Skype Connect SIP trunks.
! Important ! Skype Connect does not support emergency calls, fax, or Caller ID name.	

3 System Router Setup

3.1 NAT and SIP

In the router to which the DSX is connected, enable NAT and disable all special SIP management features (such as SIP ALG).

3.2 Port Forwarding

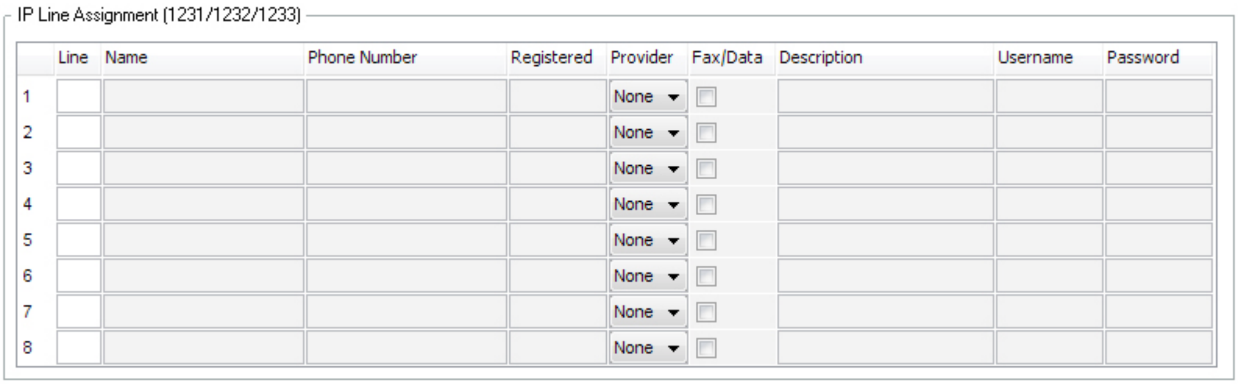
Forward UDP ports 5060 and 1024 through 1215 to the DSX system's IP address.

4 DSX Setup

4.1 Software Level

3.31.96 or higher.

4.2 Assign SIP Lines



1231-01: IP Lines [SYSTEM: PORTS: IP LINES: IP LINE ASSIGNMENT]
The first DSX SIP line is the first line beyond the last assigned line. Each SIP line uses an IP resource (licensed VoIP port) in DSX. The maximum number of VoIP licenses is 8 in DSX-40 and 16 in DSX-80/160.

Example with DSX-40:

- The first SIP line in a DSX-40 without a COIU expansion card is line 5. Available SIP line numbers are 5-12.
- With the COIU expansion card installed, the first SIP line is line 9. Available SIP line numbers are 9-16.
- DSX-40 supports a maximum of 8 VoIP licenses. When the total number of licenses are in use, only Peer-to-Peer calls are available.

Example with DSX-80/160:

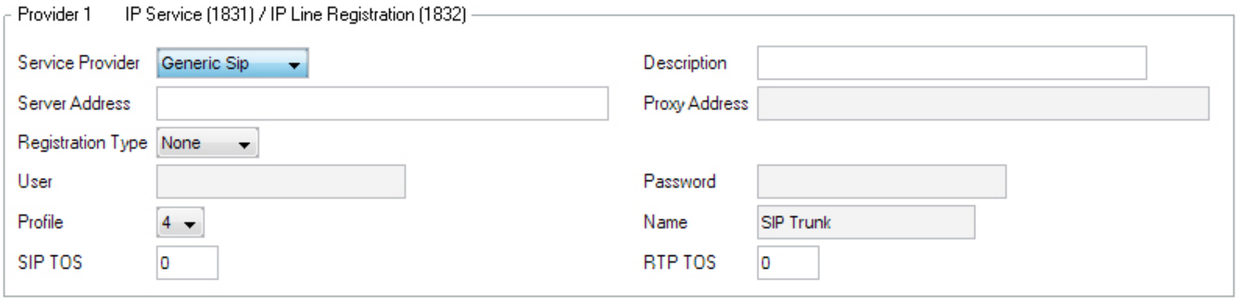
- The first SIP line in a DSX-80/160 with a single 8COIU card is line 9. Available SIP lines are 9-17.
- The last line number in DSX-80/160 is 64. In a system with an 8COIU and two T1/PRI cards, this will limit the available number of SIP lines.
- DSX-80/160 supports a maximum of 16 VoIP licenses. When the total number of licenses are in use, only Peer-to-Peer calls are available.

1232-01: IP Line Provider Number [SYSTEM: PORTS: IP LINES: IP LINE ASSIGNMENT]
Assign each SIP line in DSX to a provider number (1 or 2). DSX supports up to two providers simultaneously.

Example: If Skype is your only SIP trunk provider, assign all the SIP lines designated in 1231-01 to provider 1.

1232-02: IP Line Username [SYSTEM: PORTS: IP LINES: IP LINE ASSIGNMENT]
1232-03: IP Line Password [SYSTEM: PORTS: IP LINES: IP LINE ASSIGNMENT]
These options are not used with Skype Connect SIP trunks.

4.3 Select the SIP Trunk Provider Type



1831-01: Provider Number [System: VoIP: Providers: Provider (1 or 2)]
Select the provider number you want to set up. This selection corresponds to the provider number you assigned to the SIP lines in *1232-0*1.

Example: If Skype is your only SIP trunk provider, and you have chosen provider 1 in *1232-01*, select provider 1 for this option also.

1831-02: Provider Type [System: VoIP: Providers: Provider (1 or 2): Service Provider]
Dial 06 to choose *Skype* as the type for the provider selected in 1831-01.

1831-03: Provider Name [System: VoIP: Providers: Provider (1 or 2): Description]
The provider name is an optional 18-character entry that describes the provider. Enter any name you like – the entry does not affect provider setup or registration.

1831-04: Profile [System: VoIP: Providers: Provider (1 or 2): Profile]
Select the VoIP profile that the system will use when connecting to the SIP trunk provider. The entry you make here corresponds to the profiles set up in programs 1811-1815. ***Choose profile 4***.

4.4 Set Up the SIP Trunk Registration

1832-01: Provider IP Registration [System: VoIP: Providers: Provider (1 or 2): Server Address]
Select the provider number you want to set up (see 1831-01) and enter the Table 1 *Skype Connect Address* as it was provided to you by Skype. This option uses the text entry method.

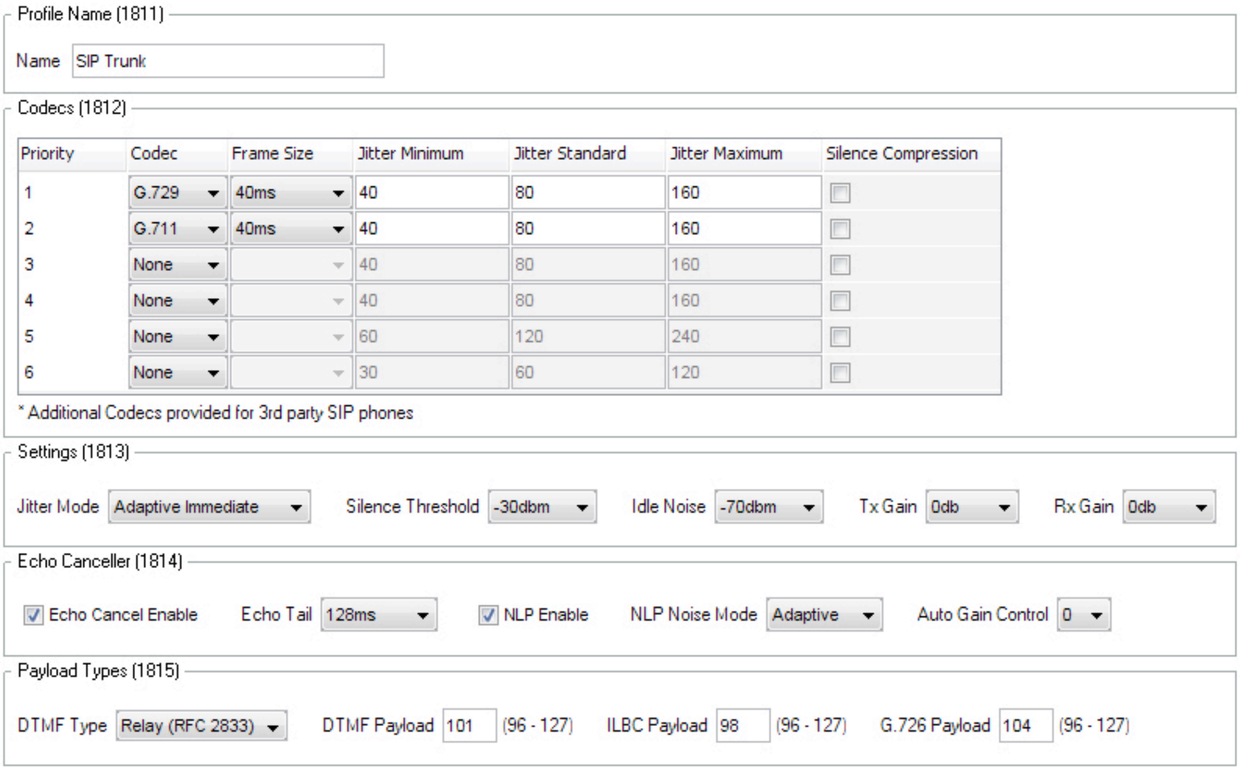
1832-02: Registration Type [System: VoIP: Providers: Provider (1 or 2): Registration Type]
Choose the IP line registration type. DSX IP lines can share the same registration (*01: Common*) or have a unique registration for each line (*02: Per Line*). For Skype, choose *01: Common*.

1832-03: Account Username [System: VoIP: Providers: Provider (1 or 2): User]
Enter the username for your Skype SIP trunk account. This option uses the text entry method and is the *SIP User* entry in Table 1.

1832-04: Account Password [System: VoIP: Providers: Provider (1 or 2): Password]
This option is the password associated with the Account Username assigned in 1832-03. The password can be up to 24 characters long. This is the *Password* entry in Table 1.

1832-05: Proxy [System: VoIP: Providers: Provider (1 or 2): Proxy Address]
1832-06: SIP Type or Service (ToS) [System: VoIP: Providers: Provider (1 or 2): SIP TOS]
1832-07: RTP Type of Service (ToS) [System: VoIP: Providers: Provider (1 or 2): RTP TOS]
These options are not used with Skype Connect SIP trunks.

4.5 Codec Setup



1812: Codecs [SYSTEM: VOIP: PROFILE 4: CODECS]
For Skype Connect SIP Trunks, for Profile 4 set:
- Codec 1 to *G.729, 40mS Frame Size*.
- Codec 2 to *G.711, 40mS Frame Size.*

Set all other priorities to *None*.

4.6 Assign DID Numbers to your SIP Lines

3101-07: Telephone Number [LINES: CONFIG: SETUP: TYPE: PHONE NUMBER]
For each of your SIP lines, enter a telephone number that corresponds to one of the *Trunk Numbers* in Table 1. Each line must have a unique entry. This entry is required for both Loop Start and DID lines. ***These are 11-digit entries. Always include the leading 1.***

4.7 Fax Setup

Skype Connect does not support fax.

4.8 Outgoing Caller ID

Skype delivers the Caller ID number to outside callers using the entry in *3101-07: Telephone Number* [LINES: CONFIG: SETUP: TYPE: PHONE NUMBER**.**