

NEC

DSX

Link2VoIP SIP Trunk Setup

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**NEC Corporation of America
6535 N. State Highway 161
Irving, TX 75039-2402**

Communications Technology Group

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1 Overview

The DSX is compatible with Link2VoIP SIP Trunking. This setup guide summarizes the account information you will receive from Link2VoIP 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 features available with other types of loop start and DID lines are also available with Link2VoIP 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 Link2VoIP Setup

To set up your SIP trunking account, contact Link2VoIP at <http://www.link2voip.com>. Once your account is set up you need the service information:

Table 1: Link2VoIP Account Information

Option	Description
Server Address	The Server Address is the SIP registration domain for your Link2VoIP SIP trunks. When your SIP trunks initially register they use this domain. Use: sip.us2.link2voip.com
SIP Username	This is the username that Link2VoIP uses to authenticate your registration.
SIP Password	This is the password required with your SIP Username.
DID Number(s)	These are the telephone numbers that users will dial to reach you over your Link2VoIP SIP trunks. In the Link2VoIP <i>Management Panel</i> , set up your DIDs to route to a unique SIP Uri for each DID number. This ensures correct routing within the DSX.

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

IP Line Assignment (1231/1232/1233)

Line	Name	Phone Number	Registered	Provider	Fax/Data	Description	Username	Password
1				None ▼	<input type="checkbox"/>			
2				None ▼	<input type="checkbox"/>			
3				None ▼	<input type="checkbox"/>			
4				None ▼	<input type="checkbox"/>			
5				None ▼	<input type="checkbox"/>			
6				None ▼	<input type="checkbox"/>			
7				None ▼	<input type="checkbox"/>			
8				None ▼	<input type="checkbox"/>			

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 Link2VoIP 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 Link2VoIP SIP trunks.

4.3 Select the SIP Trunk Provider Type

Provider 1 IP Service (1831) / IP Line Registration (1832)

Service Provider	Generic Sip	Description	
Server Address		Proxy Address	
Registration Type	None	User	
Profile	4	Password	
SIP TOS	0	Name	SIP Trunk
		RTP TOS	0

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 1232-01.

Example: If Link2VoIP 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 00 to choose *Generic SIP* 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 *Server Address* as it was provided to you by Link2VoIP. 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: *Single*) or have a unique registration for each line (02: *Per Line*). For Link2VoIP, choose 01: *Common*.

1832-03: Account Username [SYSTEM: VOIP: PROVIDERS: PROVIDER (1 OR 2): USER]

Enter the username for your Link2VoIP SIP trunk account. This option uses the text entry method and is the *SIP Username* entry in Table 1.

1832-04: Account Password [SYSTEM: VOIP: PROVIDERS: PROVIDER (1 OR 2): PASSWORD]

This option is the password associated with the SIP Username assigned in 1832-03. The password can be up to 24 characters long. This is the *SIP 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 Link2VoIP SIP trunks.

4.5 Codec Setup

Profile Name (1811)

Codecs (1812)

Priority	Codec	Frame Size	Jitter Minimum	Jitter Standard	Jitter Maximum	Silence Compression
1	G.729	40ms	40	80	160	<input type="checkbox"/>
2	G.711	40ms	40	80	160	<input type="checkbox"/>
3	None		40	80	160	<input type="checkbox"/>
4	None		40	80	160	<input type="checkbox"/>
5	None		60	120	240	<input type="checkbox"/>
6	None		30	60	120	<input type="checkbox"/>

* Additional Codecs provided for 3rd party SIP phones

Settings (1813)

Jitter Mode Silence Threshold Idle Noise Tx Gain Rx Gain

Echo Canceller (1814)

Echo Cancel Enable Echo Tail NLP Enable NLP Noise Mode Auto Gain Control

Payload Types (1815)

DTMF Type DTMF Payload (96 - 127) ILBC Payload (96 - 127) G.726 Payload (96 - 127)

1812: Codecs [SYSTEM: VOIP: PROFILE 4: CODECS]
 For Link2VoIP 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

Type (3101)

Type Name DTMF Dialing PBX Line
 Phone Number

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 DID Numbers in Table 1. Each line must have a unique entry. This entry is required for both Loop Start and DID lines.

4.7 Fax Setup

Note: Inbound fax only. Outbound faxes will not complete.

1812: Codecs [SYSTEM: VOIP: PROFILE X: CODECS]
 1815: DTMF [SYSTEM: VOIP: PROFILE X: PAYLOAD TYPES]
 SIP trunks to be used for fax must be set for G.711, 20mS Frame Size, with Inband DTMF. The fax is transmitted in-band DTMF only.

4.8 Outgoing Caller ID

Link2VoIP delivers Caller ID to outside callers using the following priority:

1. *2101-01: Outgoing ANI ID* [STATIONS: CONFIG: SETUP: TYPE: ANI ID]
2. *3101-07: Telephone Number* [LINES: CONFIG: SETUP: TYPE: PHONE NUMBER]
3. *1011-02: Telephone Number* [SYSTEM: CONFIG: SETUP: IDENTIFICATION: PHONE NUMBER]