

# NEC

# DSX

## **AccessLine 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 AccessLine SIP Trunking. This setup guide summarizes the account information you will receive from AccessLine 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 AccessLine SIP trunks. 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](http://www.necdsx.com/docs/dita/dsx_v3_complete/index.html)

## 2 AccessLine Setup

To set up your SIP Trunking account, contact AccessLine at <http://www.accessline.com>. Once your account is set up they will provide you with the following service information:

**Table 1: AccessLine Account Information**

Option	Description
AccessLine Domain	The AccessLine domain is the FQDN to which the DSX SIP lines will connect. This is normally: <i>usbc.accessline.com</i>
SIP Port	This is the SIP port AccessLine will use to connect to the DSX. It is normally: <i>6060</i>
SIP Trunk ID	The SIP Trunk ID is the username for your AccessLine SIP account.
Password	This is the password for your AccessLine SIP account.
DID(s)	These are the telephone numbers that users will dial to reach you over your AccessLine SIP trunks.

## 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 6060 and 1024 through 1215 to the DSX system's IP address, which is required for remote IP extensions. If you have no remote IP extensions, do not port forward.

DSX VoIP Resources

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.

4 DSX Setup

4.1 Software Level

3.42.04 or higher.

4.2 Basic SIP Setup

The screenshot shows a configuration window titled "SYSTEM VOIP" with a "Default" button in the top right corner. Below the title bar, there are tabs for "Setup", "Profile 1", "Profile 2", "Profile 3", "Profile 4", and "Providers". The "Setup" tab is selected. Underneath, there is a section labeled "SIP Ports (1801)". Within this section, the "SIP UDP Port" is set to "6060" (with a default of 5060), "RTP Range Start" is "1024" (with a default of 1024), and "RTP Range End" is "1215" (with a default of 1215).

*1801-01: SIP UDP Port* [SYSTEM: VOIP: SETUP]  
 Assign the SIP UDP port to 6060. (The default is 5060.)

4.3 CODEC Setup

**SYSTEM VOIP** Default

Setup Profile 1 Profile 2 Profile 3 Profile 4 Providers

Profile Name (1811) \_\_\_\_\_

Name

Codecs (1812)

Priority	Codec	Frame Size	Jitter Minimum	Jitter Standard	Jitter Maximum	Silence Compression
1	G.729	20ms	40	80	160	<input type="checkbox"/>
2	None		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"/>

**SYSTEM VOIP** Default

Setup Profile 1 Profile 2 Profile 3 Profile 4 Providers

Profile Name (1811) \_\_\_\_\_

Name

Codecs (1812)

Priority	Codec	Frame Size	Jitter Minimum	Jitter Standard	Jitter Maximum	Silence Compression
1	G.711	20ms	40	80	160	<input type="checkbox"/>
2	None		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"/>

1811/1812 [SYSTEM: VOIP: PROFILE 4: CODECS]

Assign the Priority 1 CODEC in Profile 4 as either [G.729@20ms](#) or [G.711@20ms](#). Priority settings 2-6 must be *None*. **Use one or the other – never both.**

The G.711 CODEC provides better voice quality but uses more bandwidth. G.729 uses less bandwidth but sacrifices some voice quality. Choose the CODEC that works best in your area.

4.4 AccessLine Account Setup

The screenshot shows the 'SYSTEM VOIP' configuration window. At the top right is a 'Default' button. Below the title bar are tabs for 'Setup', 'Profile 1', 'Profile 2', 'Profile 3', 'Profile 4', and 'Providers'. The 'Providers' tab is active, showing 'Provider 1' with the label 'IP Service (1831) / IP Line Registration (1832)'. The configuration fields are as follows:

- Service Provider: Generic SIP (dropdown)
- Server Address: usbc.accessline.com:6060
- Registration Type: Common (dropdown)
- User: Your SIP Trunk ID
- Password: Your Password
- Profile: 4 (dropdown)
- SIP TOS: 0
- Description: AccessLine
- Domain: (empty)
- Name: SIP Trunk
- RTP TOS: 0

1831/1832: Provider [SYSTEM: VOIP: PROVIDERS: PROVIDER 1]

For *Service Provider*, select *Generic SIP* (00 in telephone programming). DSX supports up to two providers simultaneously.

For *Description*, enter a descriptive 18-character name for your AccessLine account.

In *Server Address*, enter *usbc.accessline.com:6060*.

For *Registration Type*, select *Common* (01 in telephone programming).

In *User*, enter the account *SIP Trunk ID* provided to you by AccessLine.

For *Password*, enter the account *Password* provided to you by AccessLine.

The remaining fields on this page should be left at their default settings.

4.5 Assign Line Type and DID Numbers

The screenshot shows the 'LINE CONFIG' window. At the top right are 'Copy' and 'Default' buttons. Below the title bar are fields for 'Line', 'Ext', and 'Name'. The 'Options' tab is active, showing 'Type (3101)' configuration:

- Type: Loop Start (dropdown)
- Name: Optional Name
- Phone Number: 10-digit DID from AccessLine
- DTMF Dialing:
- PBX Line:

The first DSX SIP line is the first line beyond the last assigned line. In a base DSX-40, for example, your SIP lines would begin with line 5.

3101-07: Telephone Number [LINES: CONFIG: SETUP: TYPE: PHONE NUMBER]

For each of your SIP lines, enter one of the DID numbers provided by Access Line.

3101-01: Line Type [LINES: CONFIG: SETUP: TYPE: TYPE]

Your SIP lines can be either Loop Start or DID Immediate Start.

*Loop Start*

With Loop Start lines, the number entry in 3101-07 can either be unique for each line or the same. With unique entries, DSX operates as a “square key” where each line has its own line key. With identical entries, DSX provides a top down line rotary limited only by the number of concurrent circuits provided by your AccessLine account. With 6 lines set up this way, for example, line 6 rings first, followed by line 4, then line 4, etc.

*DID*

When set for DID, calls route into the DID table (3301/3302 [LINES: DID]) like any other DID line based on the number the caller dialed.

**4.6 Assign SIP Lines**

The screenshot shows the 'SYSTEM PORTS' configuration window with the 'SIP Lines' tab selected. It displays a table for 'IP Line Assignment (1231/1232/1233)' with the following data:

Line	Name	Phone Number	Registered	Provider	Fax/Data	Description	Username	Password
1	5	Optional Name 1	First 10-digit DID	Yes	1	Provider's Name		
2	6	Optional Name 2	Second 10-digit DID	Yes	1	Provider's Name		

*1231/1232/1233 [SYSTEM: PORTS: SIP LINES]*

The first DSX SIP line is the first line beyond the last assigned line. The example above shows two SIP lines enabled in a base DSX-40. Under *Line*, enter 5 and 6, select your provider from the *Provider* drop-down and the rest of the fields will fill in automatically. If the data you entered in the other programs is correct, after about 20 seconds *Registered* will show *Yes*. You can then start using your SIP lines to place and answer calls.

**4.7 Extension Level Caller ID**

The screenshot shows the 'STATION CONFIG' window for extension 301. Under the 'Type (2101)' section, the 'ANI ID' is set to 2039265400. Other visible settings include 'Type' as 'DSX 34 Button Super Display', 'Language' as 'English', and 'Door Chime' as 'None'.

*2101-05: Outgoing ANI ID [STATIONS: CONFIG: SETUP: TYPE: ANI]*

DSX and AccessLine SIP lines support Extension Level Caller ID. If Caller ID is enabled for the SIP line (*3121-01: Caller ID [LINES: CONFIG: SETUP: CALLER ID SETUP]*), DSX will deliver the DID number (*3101-07: Telephone Number [LINES: CONFIG: SETUP: TYPE: PHONE NUMBER]*) as the CID data. However, if an alternate number is entered in the extension's ANI option, that alternate number will be delivered instead.

In the above example, when extension 301 places a call over the SIP line, the system will deliver 2039265400 as the CID data, not the line's DID number.

**4.8 Fax Setup**

Fax services with AccessLine require a SIP line using CODEC type G.711 with inband DTMF. If you have AccessLine SIP lines using G.729, you must request a separate account from AccessLine that you can set up with G.711. The option *1232-04: Fax/Data Line [SYSTEM: PORTS: IP LINES: IP LINE ASSIGNMENT: FAX]* cannot work with AccessLine since a G.729 account will not switch to G.711.

Note that T.38 fax is not supported.



4.9 Remote IP Extensions and Security

VoIP (2106)

Options

Profile  Local (LAN)  LAN Multicast Peer-To-Peer:  On LAN  Over WAN

Security

Password  Restrictions:  Local Only  Keypset Only  Match Keypset MAC Address

2106 [STATIONS: CONFIG: SETUP: VOIP ]

To help ensure the integrity of their SIP services and minimize the likelihood of toll fraud, AccessLine requests that remote SIP endpoints registered with the DSX be DSX IP Keypsets with MAC address matching enabled. Remote endpoints that do not match these requirements will be unable to register with the DSX.