

NEC

DSX

ATC SIP Trunk Setup

April 22, 2011

Issue 1.01

©Copyright 2011

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

Communications Technology Group

Revision History		
Issue	Date	Revisions
1.00	4/11/11	<ul style="list-style-type: none">Initial release.
1.01	4/22/11	<ul style="list-style-type: none">Updated required software level, expanded RTP port forwarding range, and added System Administrator screen shots.

Contents

1 Overview..... 4

2 ATC Setup 4

3 System Router Setup..... 4

3.1 NAT and SIP..... 4

3.2 Port Forwarding 4

4 DSX Setup 4

4.1 Software Level 4

4.2 DSX WAN IP Address 5

4.3 Assign SIP Lines..... 5

4.4 Codec Setup..... 6

4.5 Select the SIP Trunk Provider Type 7

4.6 Set Up the SIP Trunk Registration..... 7

4.7 Assign DID Numbers to your SIP Lines..... 8

4.8 Fax Setup 8

Tables

Table 1: ATC Account Information 4

1 Overview

The DSX is compatible with ATC SIP Trunking. This setup guide summarizes the account information you will receive from ATC 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 ATC 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

2 ATC Setup

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

Table 1: ATC Account Information

Option	Description
ATC Server Address	The ATC Server Address is the IP address to which the DSX sip lines will connect.
Fixed WAN IP Address	ATC SIP Trunking terminates to a fixed WAN IP address at your site. Normally, this is the WAN IP address of the router to which the DSX is connected.
DID(s)	These are the telephone numbers that users will dial to reach you over your ATC 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 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 DSX WAN IP Address

Ethernet (1104)

IP Address Subnet Mask Router

DNS #1 DNS #2

System Admin Port (1024-85535)

Phone Manager Port (1-85535) MAC Address WAN Address

1104-09: WAN IP [SYSTEM: CONFIG: COMMUNICATION: ETHERNET: WAN ADDRESS]
 Enter the WAN IP address of the router to which the DSX system is connected.

4.3 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 ATC 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 ATC SIP trunks.

4.4 Codec Setup

Profile Name (1811)

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"/>

* 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 X: CODECS]

1815: *DTMF* [SYSTEM: VOIP: PROFILE X: PAYLOAD TYPES]

For the Profile you are using for ATC SIP trunks (normally 3 or 4), set Priority 1 for *G.711*, *20ms* Frame Size, with DTMF set to *Relay (RFC 2833)*. Set Priority 2-6 to *None*.

4.5 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 ATC 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 12 to choose ATC 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 a profile that corresponds to the entries you made in 4.4 *Codec Setup* above.

4.6 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 *ATC Server Address* entry as it was provided to you by ATC. 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*), have a unique registration for each line (02: *Per Line*), or require no SIP registration (00: *None*). For ATC, choose 00: *None*.

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

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

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 ATC SIP trunks.

4.7 Assign DID Numbers to your SIP Lines

Type (3101)

Type	Loop Start	Name	<input type="text"/>	<input checked="" type="checkbox"/> DTMF Dialing	<input type="checkbox"/> PBX Line
		Phone Number	<input type="text"/>		

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.

4.8 Fax Setup

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.