SIP Trunking using the Optimum Business SIP Trunk Adaptor and the NEC DSX-40 IP-PBX

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Overview

The purpose of this configuration guide is to describe the steps needed to configure the NEC DSX-40 PBX for proper operation with the Optimum Business SIP Trunk adaptor. Please note that this guide documents the basic configuration needed in the NEC DSX-40 PBX.

Prerequisites

Please follow the instructions in the Optimum Business SIP Trunk Set-up Guide. The Set-up Guide was left by the Optimum Business technician at installation. If you do not have the Set-up Guide, go to **optimumbusiness.com/SIP** to download a copy. Follow the instructions to configure the LAN side settings.

This knowledge base solution provides the configuration steps for PBX registration mode only. Static Mode is not supported.

The PBX used in the lab comprises of the following:

PBX Information

Manufacturer:	NEC
Model:	NEC DSX-40
System Version:	3.44
Does the PBX send SIP Registration messages (Yes/No)?	Yes
Vendor Contact	www.necam.com

Optimum SIP Truck Adaptor Information

Manufacturer:	Edgewater Network, Inc.
Model:	4552
Software Version:	11.6.19.0.1

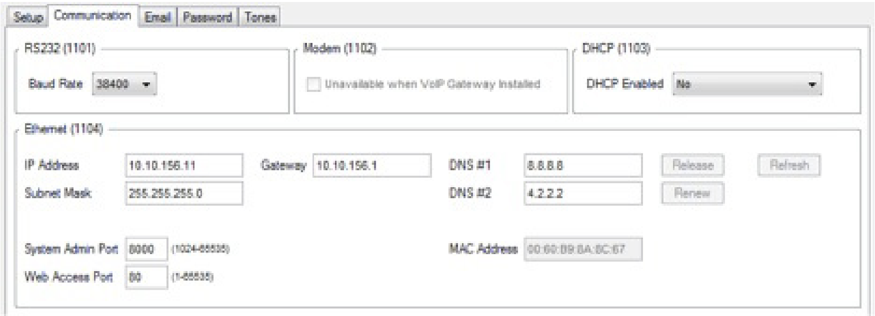
NEC DSX-40 Configuration

The steps below describe the basic configuration required to enable the PBX to use Optimum Business SIP Trunking for inbound and outbound calling. Please refer to the NEC DSX-40 documentation for other advanced PBX features.

The configuration described here assumes that the PBX is already configured and operational with station side phones using assigned extensions or DIDs. This configuration is based on the NEC DSX-40 version 3.44.

Network Settings

Device configuration requires the “NEC DSX System Administrator” software tool. Once in the system the configuration file can be accessed and edited under **Database ▸ Edit**. To change network settings once in the configuration file navigate to **System ▸ Config ▸ Communication** and enter the address of the PBX next to **IP Address** and the address of the Optimum Business SIP Trunk Adaptor next to **Gateway**. Here the PBX was assigned 10.10.156.11/24 and the Optimum Business SIP Trunk Adaptor was assigned 10.10.156.1/24. To keep these as Static addresses, **DHCP Enabled** above needs to be set to **No**.

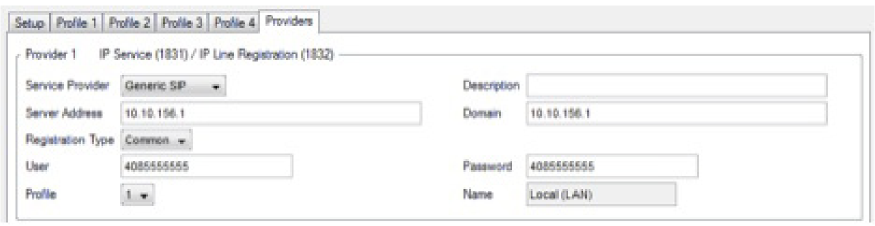


When done click the **Apply** icon from above.

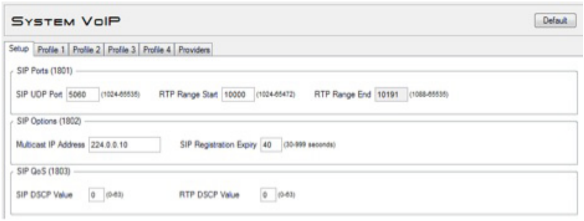
NOTE: Certain changes require disconnecting and resetting the system. The system will prompt this message when required.

SIP Programming

To configure SIP navigate to **System ▸ VoIP ▸ Providers** and set the **Service Provider** field to **Generic SIP**. Enter the IP address of the Optimum Business SIP Trunk Adaptor in both the **Server Address** and **Domain** fields. Set **Registration Type** to **Common**. Add the credentials next to **User** and **Password**. Use the same User ID and Password that you configured in the Optimum Business SIP Trunk Adaptor. In this example **1** was selected as the corresponding **Profile**. The **WAN Address (1104)** field should be **0.0.0.0**.

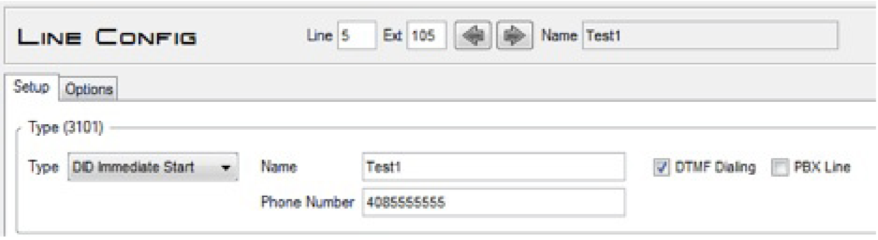


Now navigate to **System ▸ VoIP ▸ Setup**. Enter **5060** next to **SIP UDP Port** and **10000** in the **RTP Range Start** field.



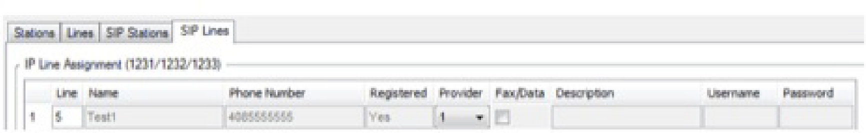
NOTE: The next section is dependent on how many lines are available on the system. In this example four lines were used and began with Line 5.

Navigate to **Lines ▸ Config** and each time the line from above should correspond to the SIP Line being configured. Next to **Type** select **DID Immediate Start**. Enter an appropriate name and next to **Phone Number** enter the Pilot DID. This should correspond to the first line which in this case was SIP Line 5.



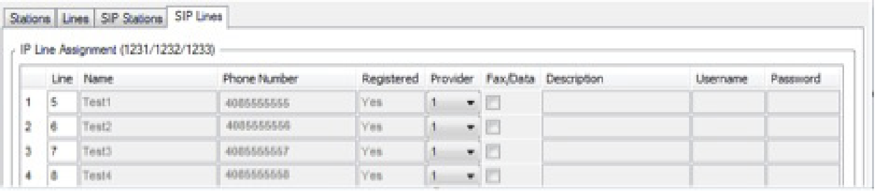
When done click **Apply** from above.

Navigate to **System ▸ Ports ▸ SIP Lines**. As shown **5** was added as the first line. If the registration is successful, **Yes** will appear under **Registered**.



	Line	Name	Phone Number	Registered	Provider	Fax/Data	Description	Username	Password
1	5	Test1	4085555555	Yes	1				

Add remaining DIDs similarity for remaining lines. The **Phone Number** field for these subsequent lines should individually include the remaining DID numbers.

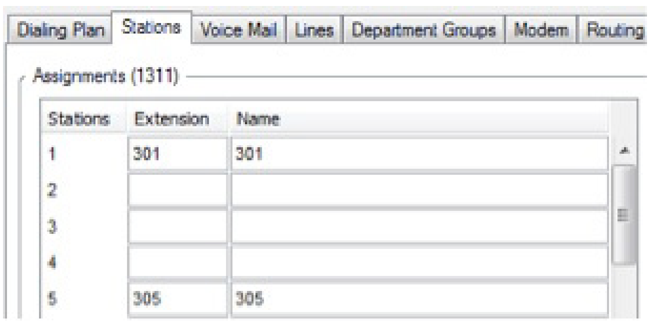


	Line	Name	Phone Number	Registered	Provider	Fax/Data	Description	Username	Password
1	5	Test1	4085555555	Yes	1				
2	6	Test2	4085555556	Yes	1				
3	7	Test3	4085555557	Yes	1				
4	8	Test4	4085555558	Yes	1				

NOTE: During Static mode on the PBX, two-way audio is lost. This is because the DSX device automatically enters the Optimum Business SIP Trunk Adaptor's WAN address under (1104) and therefore uses it as the source RTP address. Nothing can be done to stop the PBX from automatically entering the Optimum Business SIP Trunk Adaptor's Public address in the (1104) field during Static mode as this originates from the PBX itself. Although the SIP signaling will work, RTP will fail for Static mode and consequently, Static mode of operation is not supported and configuration for it is not included in this document.

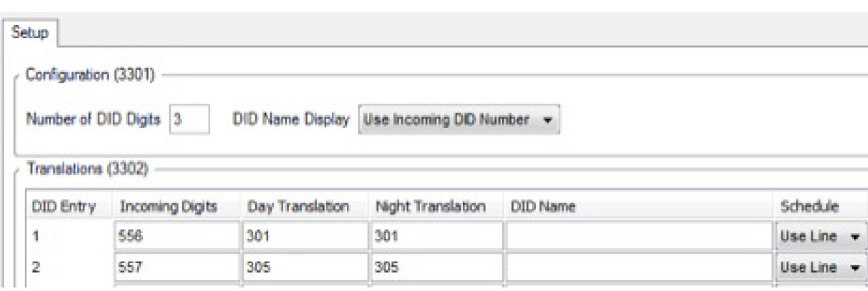
Extensions/DID

To configure extensions navigate to **System ▸ Numbering ▸ Stations** and assign extensions to valid station ports along with a name for each. In this case extension “301” was assigned to station port 1 and extension “305” was assigned to station port 5.



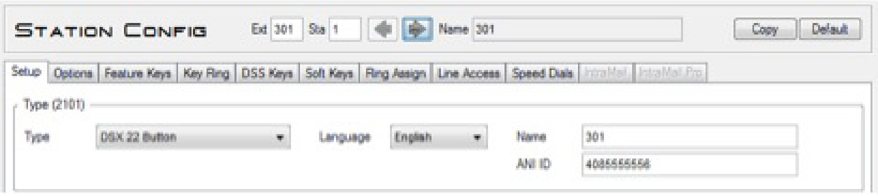
When done click the **Apply** from above.

Now navigate to **Lines ▸ DID** and here is where extensions are associated to DIDs for incoming calls. **DID Entry 1** was used for extension 301 which had DID 4085555556 and **DID Entry 2** was used for extension 305 and had DID 4085555557. The last 3 digits of each DID needs to be entered under **Incoming Digits**. Both **Day Translation** and **Night Translation** should contain the extension that the DID should ring.



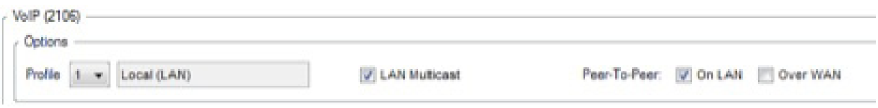
When done click the **Apply** from above.

Navigate to **Stations ▸ Config** and select the type of phone being used. Then enter the full DID for each extension next to **ANI ID**.

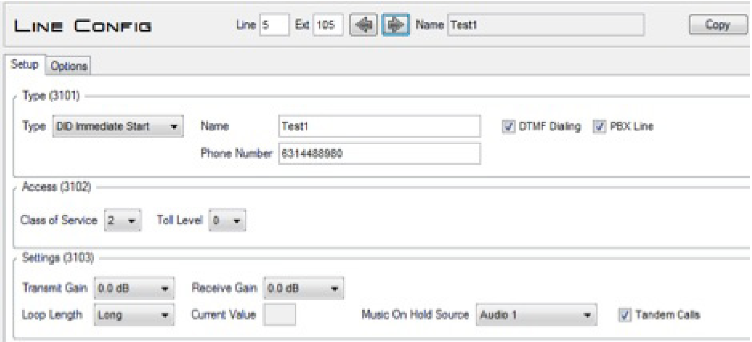


NOTE: The right and left arrow buttons from above may be used to direct to the appropriate extensions.

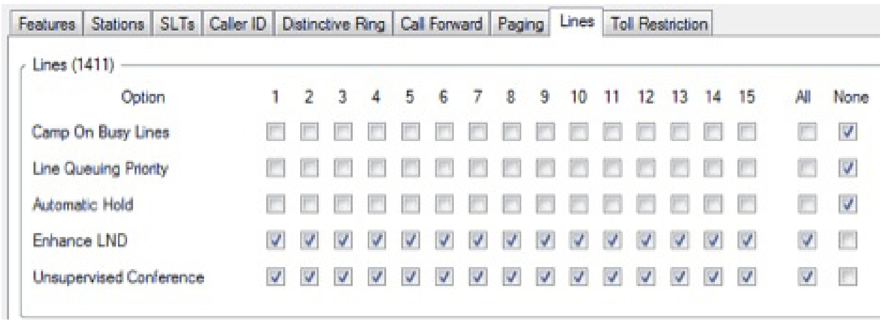
The VoIP profile under **Stations ▸ Config ▸ VoIP (2106)** should also match the profile currently registered to the Optimum Business SIP Trunk Adaptor, in this example **Profile 1**.



To enable external transfer calls navigate to **Lines ▸ Config ▸ Setup** and for each line make sure **Tandem Calls** is checked under **Settings (3103)**.

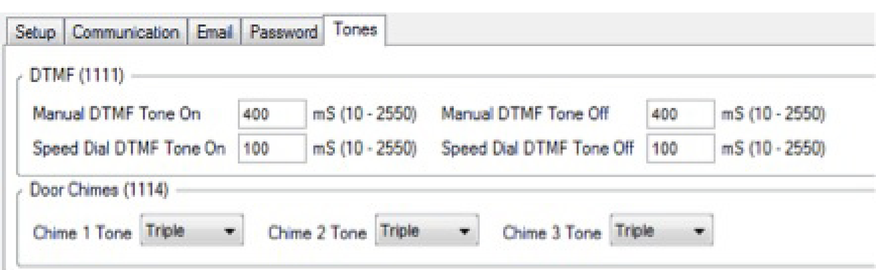


To enable conference calls navigate to **System** ▸ **Class of Service** ▸ **Lines** and check All next to **Unsupervised Conference**.

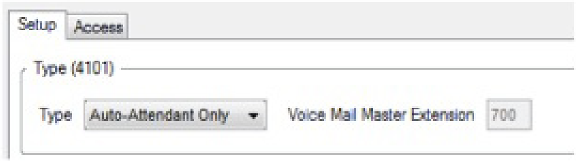


To modify the DTMF type navigate to **System** ▸ **VoIP** and click the Profile that is in use. Under **Payload Types (1815)** the **DTMF Type** may be changed.

NOTE: Due to the Cablevision DTMF network requirements, the DTMF tone duration generated by the phones and/or PBX may need to be increased to 400ms-600ms. To modify DTMF tone duration navigate to **System** ▸ **Config** ▸ **Tones** and under **DTMF(1111)** is where they may be changed.



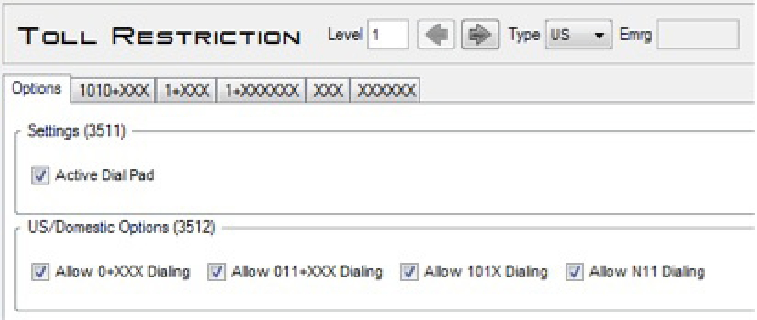
To enable Auto-Attendant navigate to **System ▸ Voice Mail** and next to **Type** select **Auto-Attendant Only**.



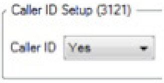
Thereafter the Auto-Attendant extension **700** should be under both **Day Translation** and **Night Translation** in the **Translations (3302)** table with the last 3 digits of the Auto-Attendant's DID under **Incoming Digits**.

DID Entry	Incoming Digits	Day Translation	Night Translation	DID Name	Schedule
1	556	301	301	301	Use Line
2	557	305	305	305	Use Line
3	558	700	700	700	Use Line

To restrict certain outbound calls navigate to **Lines ▸ Toll Restriction**. Here is where extensions can be restricted from dialing particular numbers.



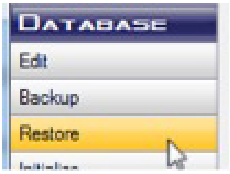
To enable Caller ID for Incoming calls navigate to **Lines ▸ Config ▸ Setup Tab ▸** and select **Yes** next to **Caller ID** for each Inbound Trunk.



Backup/Restore

To backup the configuration file navigate to **DATABASE ▸ Backup**.

To restore the configuration file navigate to **DATABASE ▸ Restore**.



NOTE: This must be performed from the initial page upon connection and cannot be done during Edit mode.