SL1100 SIP Trunk Help Guide

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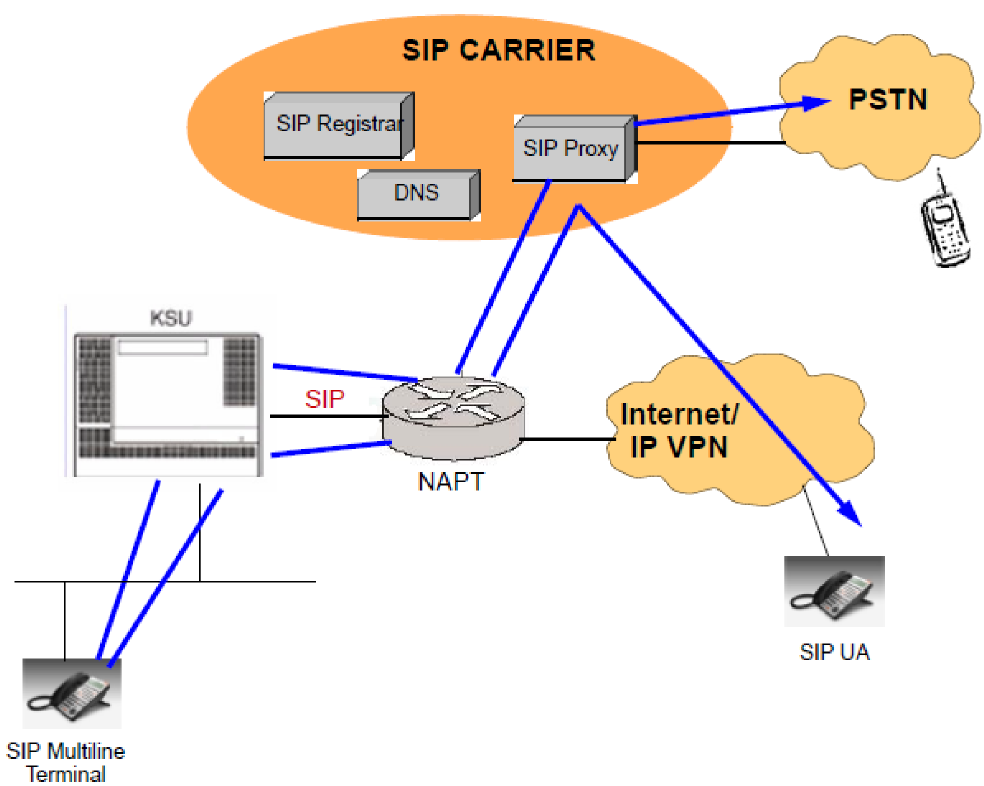
Document Description

This help document is to assist in the understanding of SIP Trunks and how they relate to the SL1100. This document covers the minimum configuration and programming required for SIP Trunk operation. For complete details for the SIP Trunk Feature please refer to the SL1100 Features & Specifications manual as well as the SL1100 Networking Guide, or the specific SL1100 SIP Trunk Guide for the provider you are using.

Network Diagram



Common IP Network using NEC SL1100 SIP Trunk

SIP Trunk Basics

The SL1100 SIP trunks utilize the **S**ession **I**nitiation **P**rotocol which allows the SL1100 system when programmed to connect directly to an ITSP (**I**nternet **T**elephony **S**ervice **P**rovider) via a data network for the purpose of originating and terminating Trunk calls.

The most significant advantages of SIP Trunks is that it allows the ability to combine voice and data into one single data stream, offering significant cost-savings and eliminating the need for local PSTN Lines.

SL1100 SIP trunk support three modes: Non-Registration, Registration mode and Registration mode with authentication:

- **Non-Registration Mode**- This is where the trunk simply sends the digits dialed, in an invite message (call setup message), to an IP-Address given to you by the SIP trunk provider. This is also the method used for SIP Tie lines between two phone systems.

- **Registration Mode**- This mode requires the SIP trunk to register with the SIP provider's Registration Server before being allowed to place calls over the provider's network. The system will register with the carrier when first connected and then at programmable intervals. It is a simple registration message notifying the SIP carrier of the SL1100 location/contact information.

- **Registration with Authentication Mode**- Authentication is an additional step to the registration mode. The SL1100 system will attempt to register to the SIP providers Registration Server and the carrier will respond with a request for authentication which is simply a password. The SL1100 forwards on the password and the carrier gives the ok allowing the SL1100 access to start making calls on the trunks. As with Registration Mode the authentication is also performed upon initial service connection and at programmable intervals thereafter.

Note: For more information on Registration Mode for SIP Trunks see the feature "IP Trunk – (SIP) Session Initiation Protocol – Registration Mode"

For a list of vendors that have successfully completed interoperability certification go to http://www.necntac.com and refer to Technical Documentation.

By default the SL1100 comes with **(4)** SIP trunk licenses. If you need more than (4) SIP trunks, then (1) SIP Trunk license, per SIP trunk, must be purchased. The license for SIP trunks is listed below:

- SL-IP-SIPTRUNK-1-LIC (License Code 5001)

SIP trunks require the VoIPDB to be installed in the SL1100.

The VoIPDB has up to 32 DSP resources on the unit (16 at default); each one can convert a speech channel from IP to TDM and vice versa.

Note – If an IP Phone is talking on a SIP trunk two DSP resources will be consumed. One resource is used for the IP Phone and the other resource is used for the SIP trunk.

To expand the VoIPDB resources from 16 to 32 you must purchase **(1)** of the following licenses:

- SL-IP-CHANNEL-16 LIC (License Code 0042)

Conditions

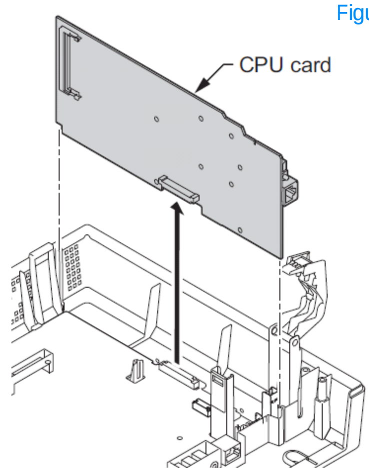
- The SL1100 supports a maximum of 32 SIP Trunks.
- Calling party name is not provided for outgoing calls on the SIP trunks.
- E.164 support is provided with CPU version 2.0 and above.
- The SL1100 software (V5.0 or higher) enables up to two SIP trunk carriers to be utilized.
- SIP Multi Profiles must be configured with unique SIP Port numbers per profile. i.g. Profile 1 could use the default SIP port 5060 and Profile 2 could be configured to use 5062. (V5.0 or higher)
- SIP Multi Profile carrier configurations must be reachable through the same IP gateway. i.g. the default gateway in PRGM 10-12-03 must be able to route traffic to the carrier configured in Profile 1 and also be able to route traffic to the System Interconnection configured in Profile 2. (V5.0 or higher)
- SIP Multi Profile carrier configurations must be reachable through the same DNS server settings. (V5.0 or higher)
- With Multi Profile each profile may use Registered SIP Trunks, or one profile using Registered and one Non-Registered but you may not have two Non-Registered SIP trunk profiles.

VoIPDB Connection

The VoIPDB is installed on the CPU card. To install the VoIPDB follow the steps below:

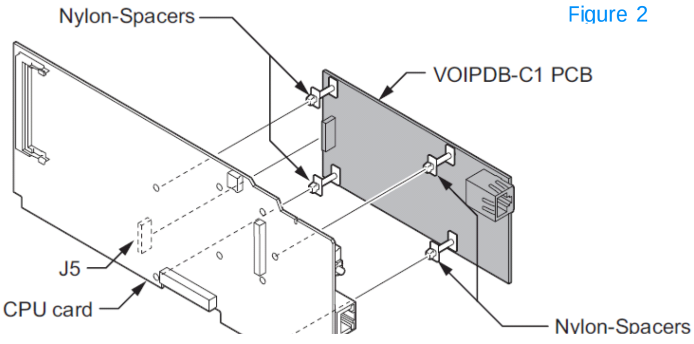
Step 1: Power down the system, remove the cover, and then remove the CPU as shown in Figure 1.

Figure 1



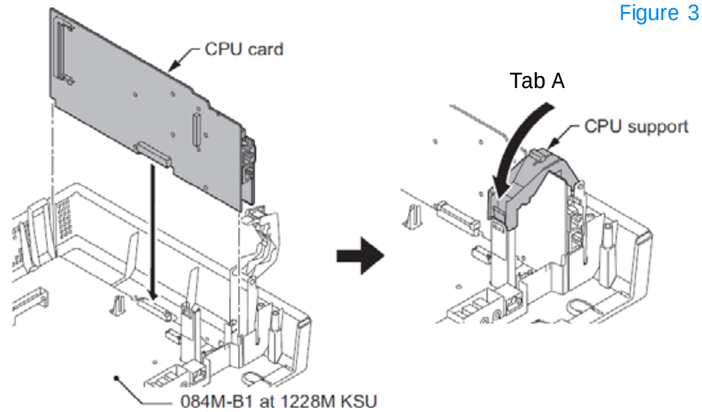
Step 2: Install the VoIPDB-C1 to the J5 connector on the CPU card as shown in Figure 2.

Figure 2



Step 3: Reinstall the CPU card into the 084M-B1 motherboard, and close the CPU support making sure **Tab A** locks into place as shown in Figure 3.

Figure 3



Netw

The voice bl
all of which are controlled by the network and/or the ISP and **not**
through a fully managed data network with QOS implemented.

For a network to be suitable for IP Phones it must pass the follow

- One way RTP must not exceed 150ms.
- Packet loss must not exceed 1%.
- Data switches must be manageable.
- No half duplex equipment installed anywhere within t
- Routers must provide QOS.
- Adequate bandwidth must be available for the estimated traffic (see Bandwidth Consumption section).

Bandwidth Consumption

Below is a table that shows the average bandwidth per call over Ethernet.

Codec	Packet Size	Bandwidth Used	Codec	Packet Size	Bandwidth Used
G.711	10ms	110.4kbps	G.729	10ms	54.4kbps
G.711	20ms	87.2kbps	G.729	20ms	31.2kbps
G.711	30ms	79.5kbps	G.729	30ms	23.5kbps
G.711	40ms	75.6kbps	G.729	40ms	19.6kbps
G.722	10ms	110.4kbps	G.729	50ms	17.3kbps
G.722	20ms	87.2kbps	G.729	60ms	15.7kbps
G.722	30ms	79.5kbps	G.723	30ms	20.8kbps
G.722	40ms	75.6kbps	G.723	60ms	13.2kbps
G.726	10ms	78.4kbps	iLBC	20ms	36.5kbps
G.726	20ms	55.2kbps	iLBC	30ms	28.8kbps
G.726	30ms	47.5kbps	iLBC	40ms	24.9kbps
G.726	40ms	43.6kbps			

Example:

Customer has a DSL connection which provides **5Mb** downstream and **512kbps** upstream.

> *Note – Most DSL connections provide more bandwidth in one direction than the other. Your typical DSL user (Home use) does not upload much data, usually they are downloading only.*

If the customer is using the default setting of G.711@ 30ms then only **6** calls at a time can be made. Any calls over **6** that are made **WILL** start causing choppy speech and even calls to drop.

> *Note – In this scenario possibly 4 or 5 calls will cause bad speech or dropped calls. 4 calls will use 318kbps alone and that leaves only 194kbps left on the circuit for any other data.*

Below is the reason for only **6** calls:

- Upstream bandwidth allows for only 512kbps. It does not matter that the downstream allows up to 5Mb the site can only transmit 512K total for Voice and Data.
- G.711 with a 30ms packet size uses 79.5kbps
- 6 calls * 79.5kbps = 477kbps
- If you were to place 7 calls it would be:
    - 7 calls * 79.5kbps = 556.6kbps (which is over the 512k limit)

If the customer would like to make more than 6 calls without purchasing more bandwidth they can change the system codec.

> *Note – Changing the system codec to **G.729** will compresses the call, which affects the quality of the speech. A compressed call will not sound as clear as an uncompressed call.*

For example if you change the system to use **G.729** with a 30ms packet size each call will only use **23.5kbps**.

This will allow up to **21 calls** to be made and fall within the 512kbps policy. Below is the reason for this:

- Policy allows for only 512kbps
- G.729 with a 30ms packet size uses 23.5kbps
- 21 calls * 23.5kbps = 493.5kbps
- If you were to place 22 calls it would be:
    - 22 calls * 23.5kbps = 517kbps (which is over the 512k limit)

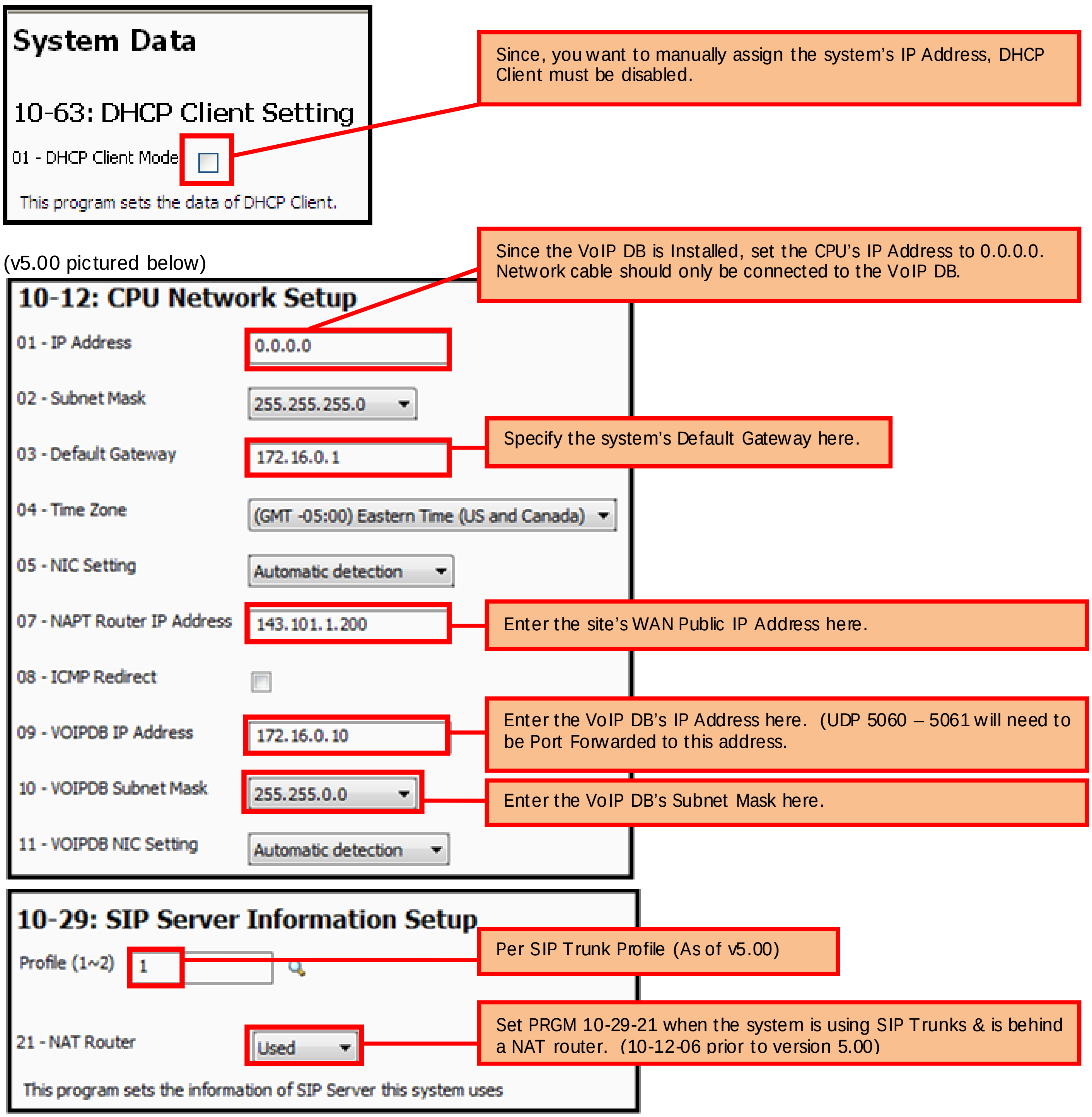
SL1100 Network Setup

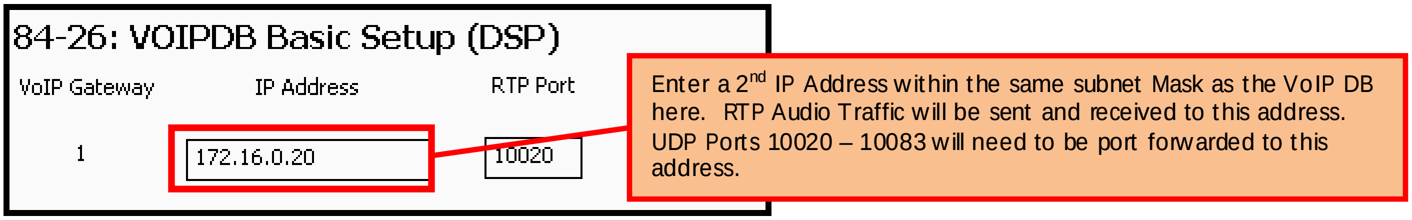
Start

	Description	Input Data
PRG 10-63-01	Disable the DHCP client. When using SIP trunks you always want the same IP Address to be assigned to the SL1100 therefore the DHCP Client function must be disabled.	**0 = Disabled** 1 = Enabled Default = 1 (Enabled)
PRG 10-12-01	Assign IP Address 0.0.0.0 and now all connections to the SL1100 are made through the VoIPDB and not the CPU network connection.	Assign IP Address 0.0.0.0 Default = 192.168.0.10
PRG 10-12-03	Assign the IP Address of the default Gateway for the SL1100.	Assign a valid IP Address within the same network as the IP Address assigned in PRG 10-12-09. Default = 0.0.0.0
PRG 10-12-07	If the SL1100 is connecting to the ITSP using NAT translations enter the WAN IP Address of the router that the SL1100 resides behind.	Assign a valid WAN IP Address. Default = 0.0.0.0
PRG 10-12-09	Assign the IP Address of the VoIPDB.	Assign a valid IP Address. Default = 172.16.0.10
PRG 10-12-10	Assign the Subnet Mask for the VoIPDB.	Assign a valid subnet mask. Default = 255.255.0.0
PRG 10-29-21 (*PRG 10-12-06 Prior to v5.0*)	If the SL1100 is connecting to the ITSP using NAT translations then this setting must be enabled. If NAT is not used in the connections to the ITSP leave this disabled.	0 = Disabled 1 = Enabled Default = 0 (Disabled Profile 1 & 2)
PRG 84-26-01	Assign the IP Address for the VoIPDB DSP's which are used to convert VoIP to TDM and vice versa.	Assign a valid IP Address within the same network as the IP Address assigned in PRG 10-12-09. Default = 172.16.0.20
System Reset	If changes were made to any of these programs a system reset is required before the changes will be applied.	

End

Programming Example:





Reboot system after the above program changes have been made and uploaded to the system.

NAT & Port Forwarding

Network **A**ddress **T**ranslations (NAT) translates between the private IP addresses in the customers local LAN and the public IP addresses on the internet. The use of NAT with SIP Trunks allows the customer to keep the SL1100 on the local LAN (for security reasons) and still have access to the ITSP through the Internet.

Note – When connecting to an ITSP through the Internet it is recommended to use NAT and to not assign Public IP Addresses to the SL1100.

In the router/firewall that the SL1100 resides behind port forwarding is required. The ports that must be forwarded to the SL1100 are as follows:

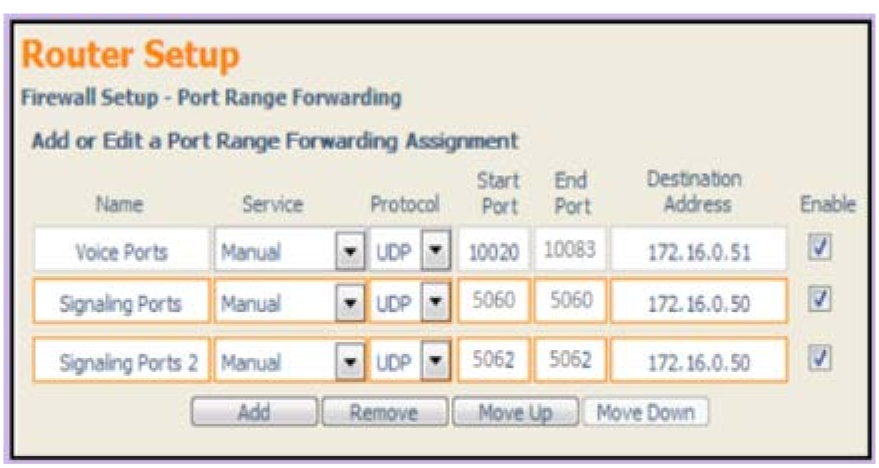
UDP Port 5060 **MUST** be forwarded to the IP Address assigned in PRG 10-12-09.

UDP Ports 10020 ~ 10083 **MUST** be forwarded to the IP Address assigned in PRG 84-26-01.

SL1100 systems version 5.00 and higher optionally support up to two different SIP Trunk Profiles. Both profiles must use different SIP Listen ports. By default PRG 84-14-06 uses UDP 5060 for Profile 1 and UDP 5062 for Profile 2.

If using Profile 2 you will also need to have a port forward rule for UDP 5062 that is port forwarded to the IP Address assigned in PRG 10-12-09.

An example of port forwarding in a router is displayed below. In this example the IP address assigned in program 10-12-09 is 172.16.0.50. The IP address assigned in program 84-26-01 is 172.16.0.51. UDP 5060 is for traffic for SIP Trunk profile 1, UDP 5062 is for traffic for SIP Trunk Profile 2:



SIP Trunk Information needed before programming

Before proceeding on with the SIP Trunk setup you must have the following information from the ITSP or the Network Administrator:

- Will the SIP Trunk Provider (ITSP) be accepting your outbound calls via the Public IP Address they are coming from or by a user ID, username & password they have issued you. (This will determine if you will be use Non-Registration vs. Registration mode)
- Are the SIP trunks registering via an IP Address (e.g. 10.10.10.10) or a Domain Name (e.g. thesipprovider.net)?
    - If you are registering via an IP Address see the section labeled "SIP Trunk Registration via IP Address".
    - If you are registering via a Domain Name see the section labeled "SIP trunk Registration via Domain Name".

  Note – You will only program one of the above sections, do not program both.
- If the SIP trunks are registering via a Domain Name what is the DNS IP Address?
- Is authentication required and if so what is the Username and Password?
- Is NAT going to be used to connect the SL1100 to the ITSP?
    - If so, what is the WAN Address of the router that the SL1100 resides behind? Refer to the feature "IP Trunk – (SIP) Session Initiation Protocol – NAT".
- Is there a separate Proxy Address or is everything sent to the Registration Address?
- How many SIP Trunks have been purchased?
- What telephone numbers are associated with this SIP Trunk Service? (Main Billing Number & DID numbers)

Multi Profile programming considerations (v5.00 and higher)

For SIP Multi Profile programming areas you will now require an index selection as to whether Profile 1 or Profile 2 is to be configured. (V5.0 or higher)

Program	Program Name	Note
10-28	SIP System Information Setup	Index added. Select Profile 1 or Profile 2.
10-29	SIP Server Information Setup	Index added. Select Profile 1 or Profile 2.
10-36	SIP Trunk Registration Information	Index added. Select Profile 1 or Profile 2.
15-16	SIP Register ID Setup for Extension	Index added. Select Profile 1 or Profile 2.
21-19	IP Trunk (SIP) Calling Party Number Setup for Extensions	Index added. Select Profile 1 or Profile 2.
84-13	SIP Trunk CODEC Information Setup	Index added. Select Profile 1 or Profile 2.
84-14	SIP Trunk Information Basic Setup	Index added. Select Profile 1 or Profile 2.
84-31	VoIPDB Echo Canceller Setup	Index added. Select Profile 1 or Profile 2.
84-33	FAX over IP Setup	Index added. Select Profile 1 or Profile 2.
84-34	VoIPDB DTMF Setup	Index added. Select Profile 1 or Profile 2.
84-38	VoIPDB Network Side Echo Canceller	Index added. Select Profile 1 or Profile 2.
84-39	SIP Trunk Message Customization	Index added. Select Profile 1 or Profile 2.

Note: Each SIP Trunk is assigned to their Profile in PRGM 14-18-05

SIP Trunk Assignment – (Any Mode)

Start

Program	Description	Input Data
PRG 10-40-01	Enable IP trunk availability.	0 = Disabled (SIP Trunks not available) **1 = Enabled** Default = 0 (Disabled)
PRG 10-40-02	Assign the number of SIP trunks that are to be used. Verify that the SL1100 contains one SIP Trunk license per SIP trunk assigned here.	0 = No SIP Trunks assigned 1~32 = 1~32 SIP Trunks Assigned Default = 0 (No SIP Trunks installed)
PRG 14-05-01	Assign each of the SIP trunks to a valid Trunk group. Do NOT assign valid SIP trunks to trunk group 0. After assigning the trunk to a trunk group assign the priority of the trunk within the group. The trunk with the lowest priority will be selected first. If two trunks have the same priority then the lowest numbered trunk will be selected first.	**Group** 0 = No trunk group assigned. 1~ 25 = Trunk group 1~25 **Priority** 1~84 = Priority 1~84, the lower priority trunks are selected first. Default = Refer to the programming manual.
PRG 14-18-05	SIP Profile (SIP Trunk) (V5.0 Added)	1=Profile 1 2=Profile 2 Default = 1 Profile 1

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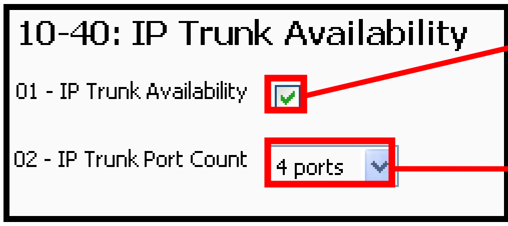
Continued		Assign Key ***01** (trunk Key) to an unused button. After assigning the trunk key, enter the trunk number to be assigned on the phone.
PRG 15-07-01	Assign the Trunk Key or a Loop Key to an unused button on the telephone. When assigning a Trunk key you must also select which trunk number you want to appear on the button. When assigning a Loop key you must also select whether it will be used for Outgoing Only, Incoming Only, or Both.	Assign Key ***05** (Loop Key) to an unused button. After assigning the key to the button select one of the following: 0 = Incoming Only 1 = Outgoing Only 2 = Both Default = Refer to the programming manual
PRG 22-02-01	Assign all SIP trunks to DID or TIE Line. It is recommended that all 8 day/night modes be set as either DID or TIE Line. *Note: Any trunk type can be assigned here however when using SIP Trunks to an ITSP the most common configuration is to assign the trunks as DID and perform DID Conversion on the digits received.* *When connecting the SL1100 to another phone system using SIP Trunks the most common configuration is to assign the SIP Trunks as Tie Line.*	0 = Normal 1 = VRS 2 = DISA **3 = DID** 4 = DIL **5 = Tie Line** 6 = Delayed VRS 7 = ANI/DNIS 8 = DID(DDI) Mode Switching Default = 0 (Normal)
End		

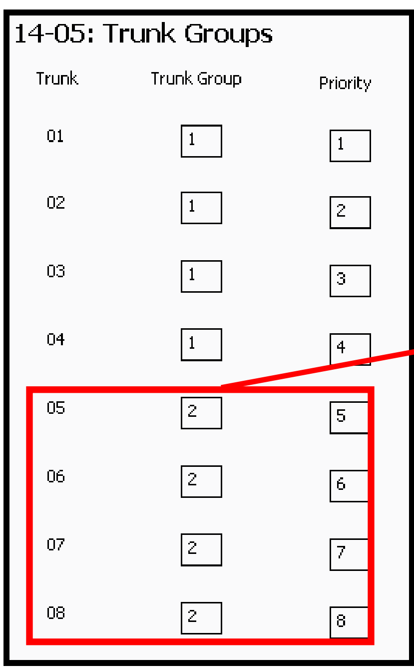
Programming Example



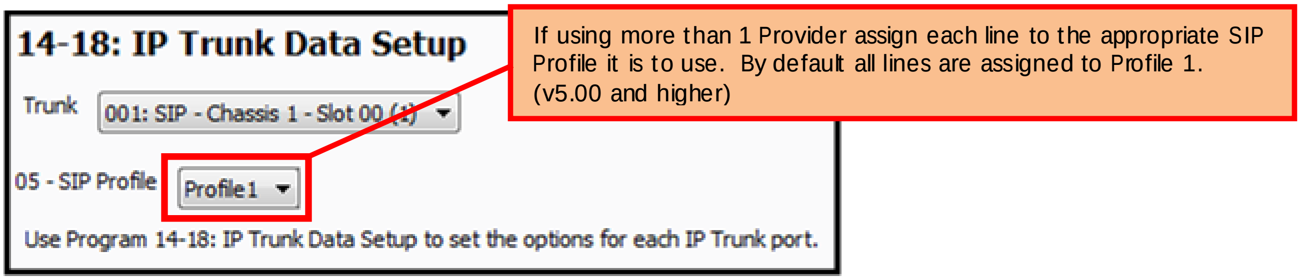
Check this box to enable SIP Trunk Availability.

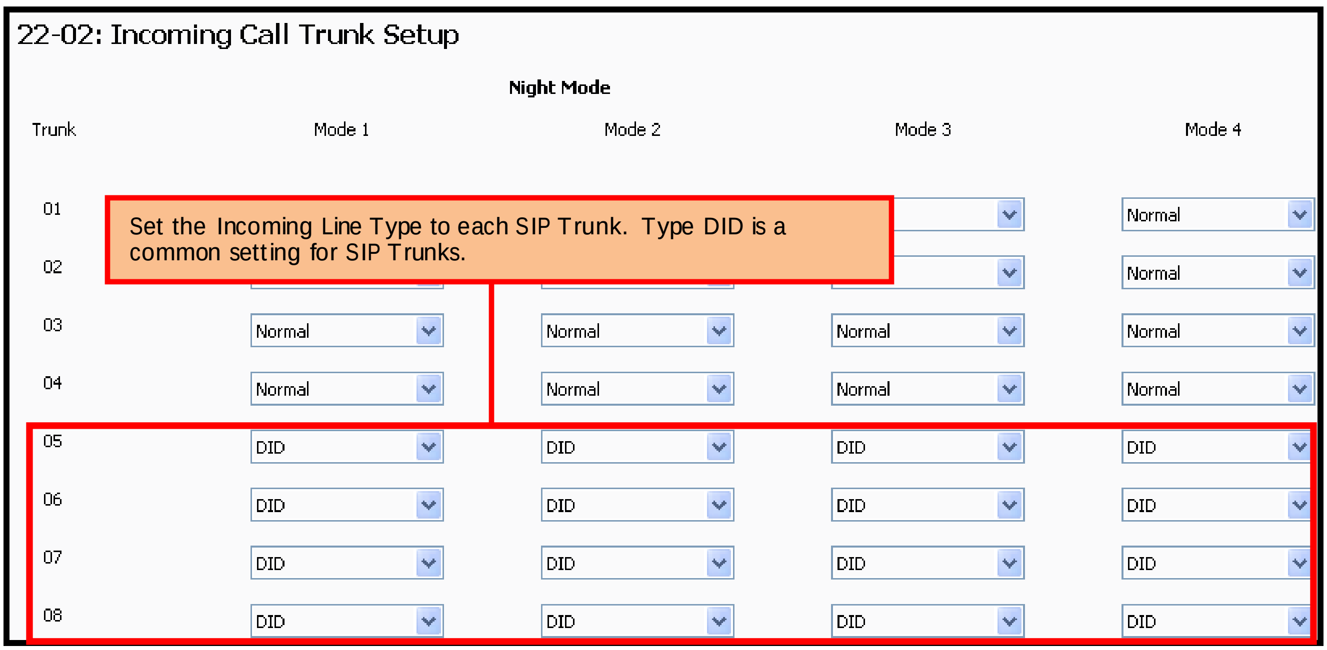
Set the number of SIP Trunks to be used.

After 10-40 is set, card view will show the Trunk Numbers assigned to SIP Trunks by the VoIP DB. As seen here.



Set the SIP Trunks to a different Trunk Group than the other lines


14-18: IP Trunk Data Setup
Trunk
001: SIP - Chassis 1 - Slot 00 (1)
05 - SIP Profile
Profile1
Use Program 14-18: IP Trunk Data Setup to set the options for each IP Trunk port.
If using more than 1 Provider assign each line to the appropriate SIP Profile it is to use. By default all lines are assigned to Profile 1. (v5.00 and higher)


22-02: Incoming Call Trunk Setup
Night Mode
Trunk
Mode 1
Mode 2
Mode 3
Mode 4
01
02
03
04
05
06
07
08
Normal
DID
Set the Incoming Line Type to each SIP Trunk. Type DID is a common setting for SIP Trunks.

Non Registration Mode - SIP Trunk Connection Assignment

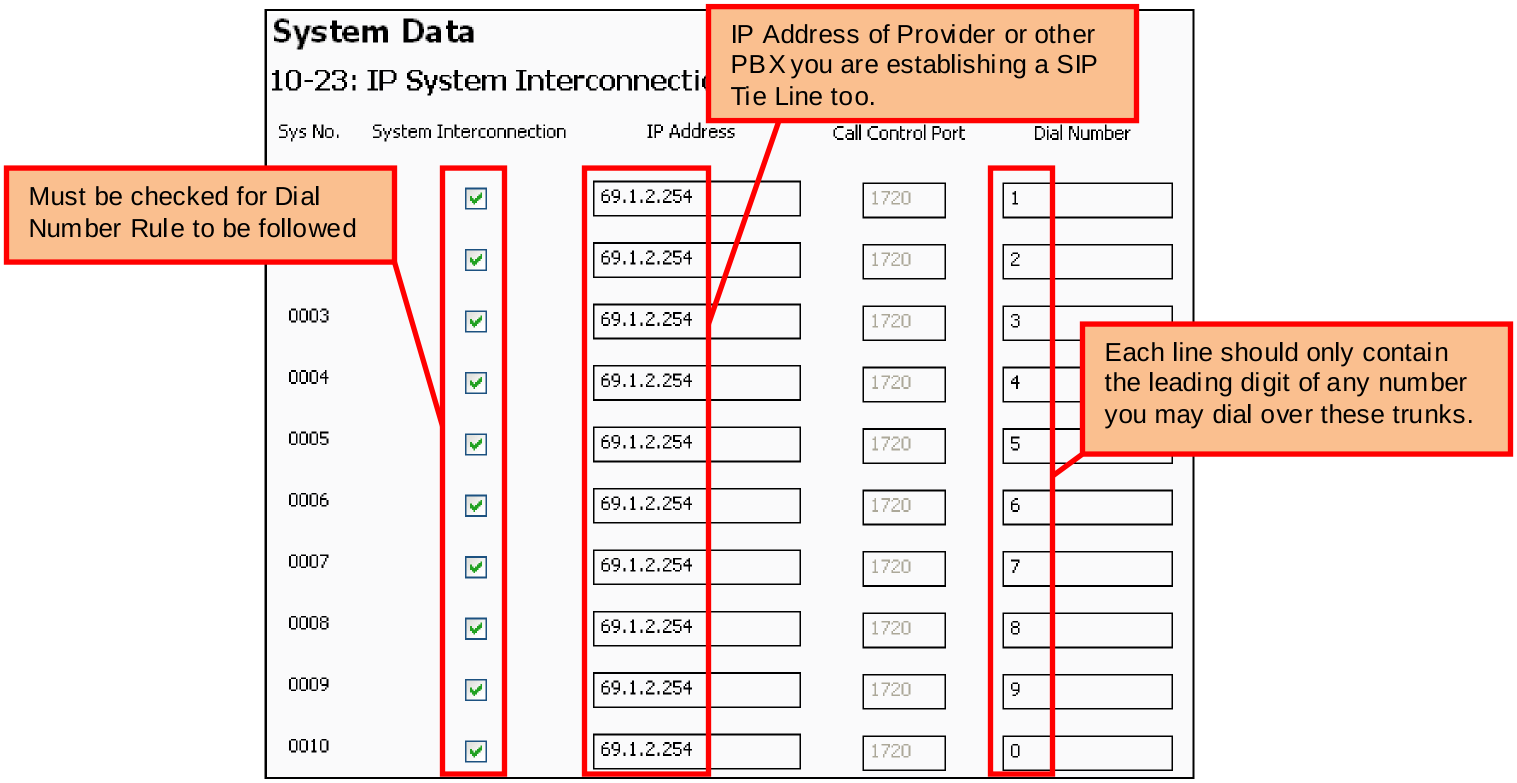
Start

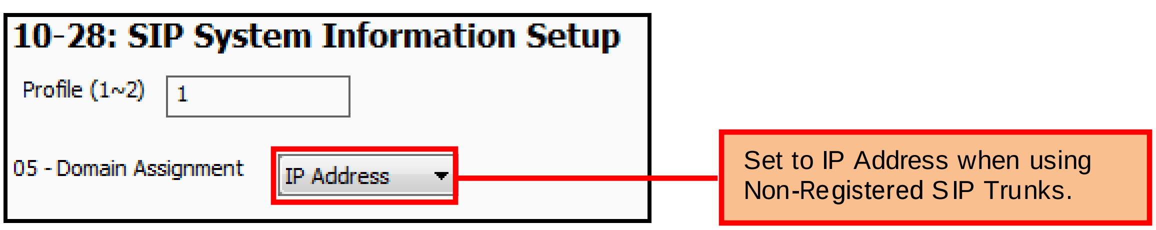
Program	Description	Input Data
PRG 10-23-01	This program determines if this dialing rule (1~1000) is Enabled/Disabled. When enabled, the dialing rule is followed and when disabled the dialing rule is ignored.	0 = Disable (Do not use dialing rule) 1 = Enabled (Use dialing rule) Default = 0 (Disable)
PRG 10-23-02	Assign the IP Address to send the SIP messages to. If you are connecting to an ITSP then assign the IP Address of the ITSP. If you are connecting to another telephone system using TIE Lines then assign the IP Address of the other system.	Assign a valid IP Address. Default = 0.0.0.0
PRG 10-23-04	Assign the leading digit/digits of the number routed out the SIP trunks. For example if you are doing SIP TIE Lines to another system and this other system has extensions in the 4XX range then assign the digit 4 in this program. The SL1100 will analyze the digits dialed and in this case you dialed a number starting with a 4 so the call will be routed to the IP Address in program 10-23-02.	Enter a number up to 12 digits in length. Default = Not Assigned
PRG 10-28-05	This program is used to decide if the SL1100 will connect to the ITSP using an IP Address or a Domain name. For Non Registration Trunks this MUST be set to IP Address.	**0 = IP Address** 1 = Domain Name Default = 0 (IP Address)
PRG 10-36-02	Use this program to input the User ID in the SIP Invite Setup message. A call cannot be completed across the span if there is no outbound CID info. (V5.0 Changed) (*Formally PRG 10-28-04 prior to v5.00*)	Default = Not set

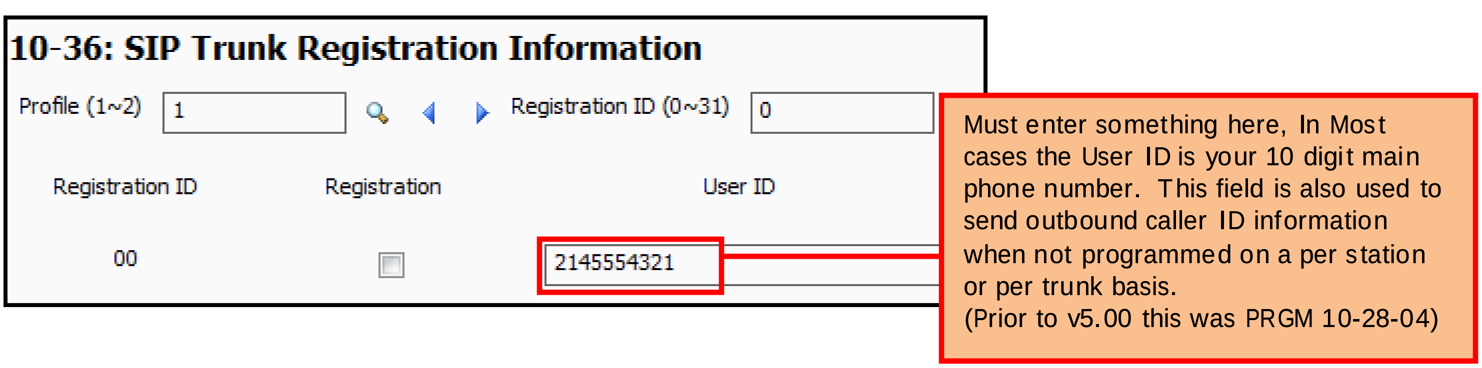
End

Programming example (v5.00 and higher)

The example below shows a typical configuration of program 10-23 when Non-Registration Mode is used:

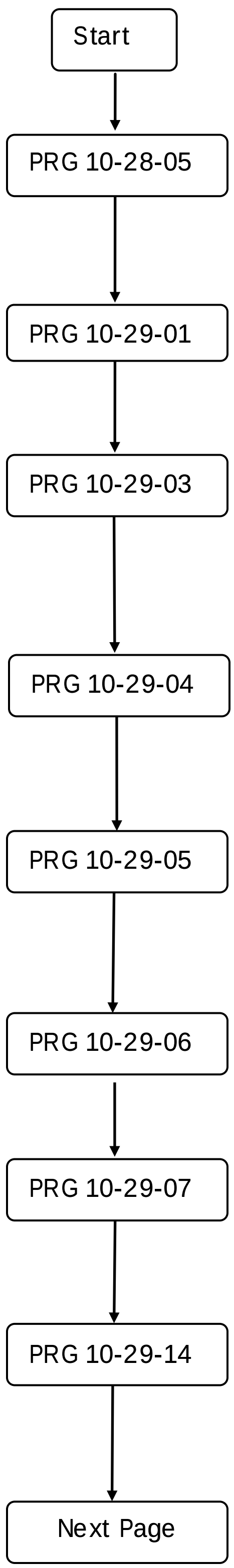






SIP Trunk Registration Mode

SIP Trunk Registration via IP Address (v5.00. ***The following Programs are set on a per Profile basis*****)**



Description	Input Data
This program is used to decide if the SL1100 will connect to the ITSP using an IP Address or a Domain name. For IP Registration this MUST be set to IP Address.	**0 = IP Address** 1 = Domain Name Default = 0 (IP Address)
Enable this setting during SIP registration Mode to allow for SIP Proxy settings.	0 = Disabled **1 = Enabled** Default = 0 (Disabled)
Assign the IP Address of the SIP Proxy Server provided by the ITSP. If no Proxy IP Address has been provided assign the same IP Address you will use for Registration.	Assign a valid IP Address. Default = 0.0.0.0
Assign the SIP Proxy port. This is usually port 5060 however verify this with the ITSP.	Assign any valid unused port number from 1 ~ 65535. Default = **5060**
This program is used to Enable/Disable the ability to Register to the ITSP. In Registration Mode (IP or Domain Name) this MUST be enabled.	0 = Disabled **1 = Enabled** Default = 0 (Disabled)
Assign the IP Address of the SIP Registration Server provided by the ITSP.	Assign a valid IP Address. Default = 0.0.0.0
Assign the SIP Registration port. This is usually port 5060 however verify this with the ITSP.	Assign any valid unused port number from 1 ~ 65535. Default = **5060**
Assign the SIP Carrier Mode. Each certified vendor may use a different carrier type. Visit the NTAC website (http://www.necntac.com) to verify the proper setting per vendor.	Valid Settings are 0 ~ 26 0 = None 1~26 = Carrier Type A ~ Carrier Type Z Default = 0 (None)

	Description	Input Data
Continued		
PRG 10-29-15	Registration Expiry (Expire) Time This sets the expiration time when the SIP trunk registers to the Sip server. When half the time set here passes, the registration update is automatically done.	120 ~ 65535 seconds (SIP Profile:1-2) (V5.0 Added) Default = 3600
PRG 10-29-20	Authentication Trial (V5.0 Added) This is how many times it will try to authenticate before timing out and not registering.	0 ~ 9 (SIP Profile:1-2) Default = 1
PRG 10-29-21	If the SL1100 is connecting to the ITSP using NAT translations then this setting must be enabled. If NAT is not used in the connections to the ITSP leave this disabled. (V5.0 Added)	0 = Disabled 1 = Enabled Default = 0 (Disabled Profile 1 & 2)
PRG 10-36-01	Registration This setting determines if the SIP trunk information is registered.	0 = Disable **1 = Enable** (SIP Profile:1-2) (V5.0 Added) Default = 0 Disable
PRG 10-36-02	User ID Use this program to input the User ID in the SIP Invite Setup message. A call cannot be completed across the span if there is no outbound CID info. (V5.0 changed) (*Formally PRG 10-28-04 prior to v5.00*)	Up to 32 Characters (alpha or numeric, spaces and/or special characters not allowed) (ex. : UserID@HostName.DomainName) (V5.0 Added) (SIP Profile:1-2) (V5.0 Added) Default = Not set
End		

SIP Trunk Registration via Domain Name (v5.00. ***The following Programs are set on a per Profile basis*****)**

Start

Program	Description	Input Data
PRG 10-28-01	Assign the Domain Name for the SIP-URL. For example if the FQDN (Fully Qualified Domain Name) address is mysipprovider.sipprovider.com you would assign sipprovider.com in this program. SIP message example: [10-28-04]@[10-28-02].[10-28-01]	Assign a valid Domain Name up to 64 characters in length. Default = Not Assigned
PRG 10-28-02	Assign the Host Name for the SIP-URL. For example if the FQDN (Fully Qualified Domain Name) address is mysipprovider.sipprovider.com you would assign mysipprovider in this program. SIP message example: [10-28-04]@[10-28-02].[10-28-01]	Assign a valid Host Name up to 48 characters in length. Default = Not Assigned
PRG 10-28-05	This program is used to decide if the SL1100 will connect to the ITSP using an IP Address or a Domain name. For Domain Name Registration this MUST be set to Domain Name.	0 = IP Address **1 = Domain Name** Default = 0 (IP Address)
PRG 10-29-01	Enable this setting during SIP registration Mode to allow for SIP Proxy settings.	0 = Disabled **1 = Enabled** Default = 0 (Disabled)
PRG 10-29-04	Assign the SIP Proxy port. This is usually port 5060 however verify this with the ITSP.	Assign any valid unused port number from 1 ~ 65535. Default = **5060**
PRG 10-29-05	This program is used to Enable/Disable the ability to Register to the ITSP. In Registration Mode (IP or Domain Name) this MUST be enabled.	0 = Disabled **1 = Manual** (Enabled) Default = 0 (Disabled)

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PRG 10-29-07	Assign the SIP Registration port. This is usually port 5060 however verify this with the ITSP.	Assign any valid unused port number from 1 ~ 65535. Default = **5060**
PRG 10-29-08	Enable DNS Mode so that the SL1100 will be able to translate the Domain Name to an IP Address.	0 = Disabled **1 = Enabled** Default = 0 (Disabled)
PRG 10-29-09	Assign the DNS Server IP Address which is provided by the ITSP or the network administrator.	Assign a valid IP Address. (*Uses Profile 1 Only*) (V5.0 Added) Default = 0.0.0.0
PRG 10-29-10	DNS Port Number If 10-29-08 is 1, this is effective. This sets the port number of the DNS server.	0 ~ 65535 (*Uses Profile 1 Only*) (V5.0 Added) Default = 53
PRG 10-29-11	Assign the SIP Registration Server Domain Name provided by the ITSP. For example: mysipserver.sipprovider.com	Assign a valid Domain Name up to 128 characters in length. (SIP Profile:1-2) (V5.0 Added) Default = Not Assigned
PRG 10-29-12	Assign the Domain Name of the SIP PROXY Server provided by the ITSP. For example if the SIP Proxy server address is proxy.sipprovider.com you would assign *sipprovider.com* in this program. If no SIP PROXY address is provided use the SIP Registration address as the proxy address.	Assign a valid Domain Name up to 64 characters in length. (SIP Profile:1-2) (V5.0 Added) Default = Not Assigned
PRG 10-29-13	Assign the Proxy Host Name of the SIP PROXY Server provided by the ITSP. For example if the SIP Proxy server address is proxy.sipprovider.com you would assign *proxy* in this program. If no SIP PROXY address is provided use the SIP Registration address as the proxy address.	Assign a valid Host Name up to 48 characters in length. (SIP Profile:1-2) (V5.0 Added) Default = Not Assigned

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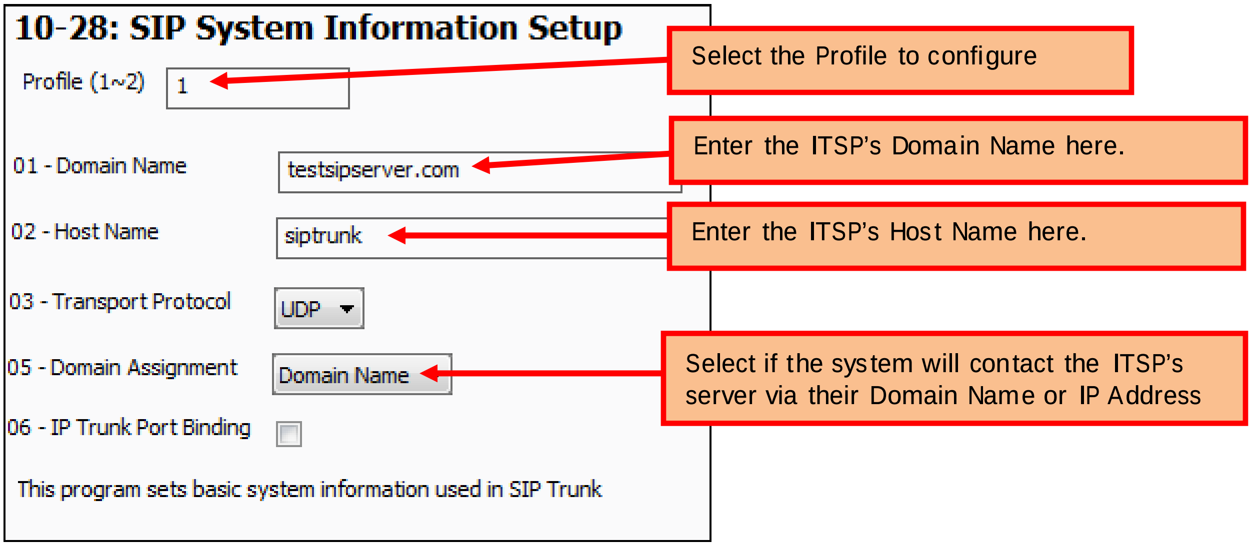
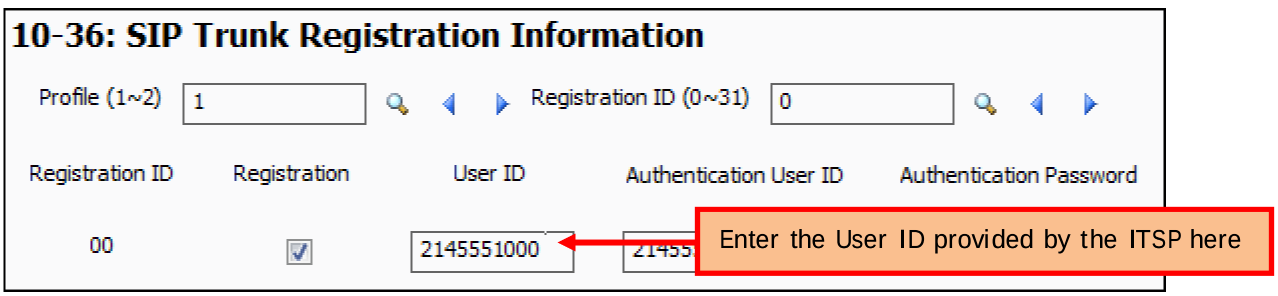
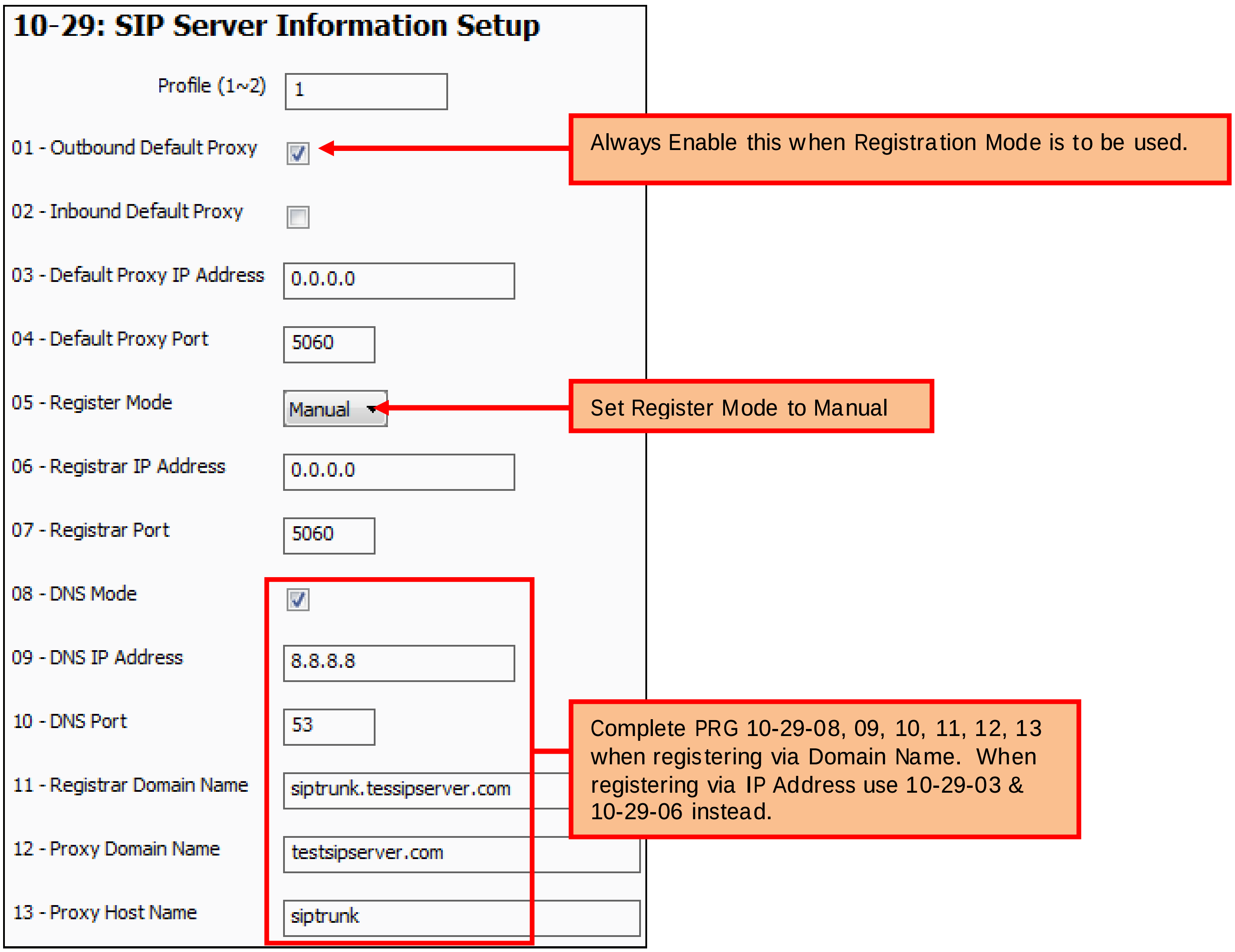
Program	Description	Input Data
PRG 10-29-14	Assign the SIP Carrier Mode. Each certified vendor may use a different carrier type. Visit the NTAC website (http://www.necntac.com) to verify the proper setting per vendor.	Valid Settings are 0 ~ 26 0 = None 1~26 = Carrier Type A ~ Carrier Type Z Default = 0 (None)
PRG 10-29-15	Registration Expiry (Expire) Time This sets the expiration time when the SIP trunk registers to the Sip server. When half the time set here passes, the registration update is automatically done.	120 ~ 65535 seconds (SIP Profile:1-2) (V5.0 Added) Default = 3600
PRG 10-29-17	DNS Source Port (Added v5.00) This sets the DNS Port Number when PRG10-29-08 is On.	0 ~ 65535 (***Uses Profile 1 Only***) Default = 53
PRG 10-29-20	Authentication Trial (V5.0 Added) This is how many times it will try to authenticate before timing out and not registering.	0 ~ 9 (SIP Profile:1-2) Default = 1
PRG 10-29-21	If the SL1100 is connecting to the ITSP using NAT translations then this setting must be enabled. If NAT is not used in the connections to the ITSP leave this disabled. (V5.0 Added)	0 = Disabled 1 = Enabled Default = 0 (Disabled Profile 1 & 2)
PRG 10-36-01	Registration This setting determines if the SIP trunk information is registered.	0 = Disable **1 = Enable** (SIP Profile:1-2) (V5.0 Added) Default = 0 Disable
PRG 10-36-02	User ID Use this program to input the User ID in the SIP Invite Setup message. A call cannot be completed across the span if there is no outbound CID info. (V5.0 changed) (*Formally PRG 10-28-04 prior to v5.00*)	Up to 32 Characters (alpha or numeric, spaces and/or special characters not allowed) (ex. : UserID@HostName.DomainName) (V5.0 Added) (SIP Profile:1-2) (V5.0 Added) Default = Not set

End

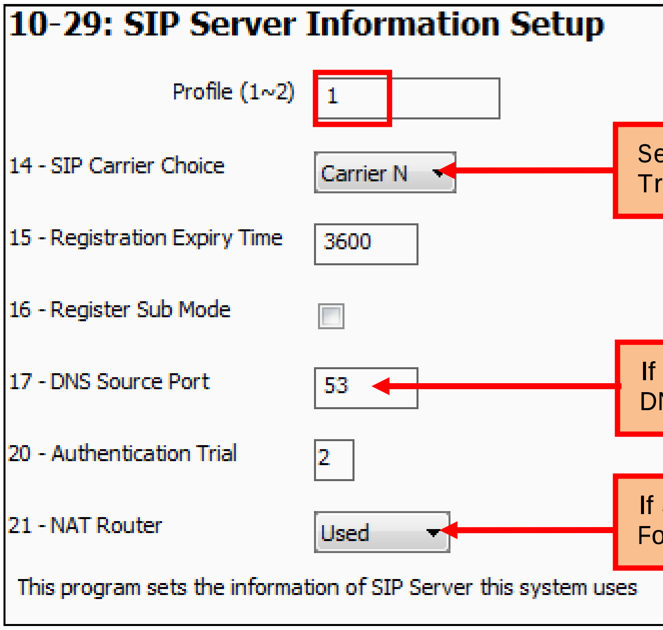
Programming Example (v5.00 Profile 1 using DNS):






10-29: SIP Server Information Setup
Profile (1~2)
1
01 - Outbound Default Proxy
02 - Inbound Default Proxy
03 - Default Proxy IP Address
0.0.0.0
04 - Default Proxy Port
5060
05 - Register Mode
Manual
06 - Registrar IP Address
0.0.0.0
07 - Registrar Port
5060
08 - DNS Mode
09 - DNS IP Address
8.8.8.8
10 - DNS Port
53
11 - Registrar Domain Name
siptrunk.tessipserver.com
12 - Proxy Domain Name
testsipserver.com
13 - Proxy Host Name
siptrunk
Always Enable this when Registration Mode is to be used.
Set Register Mode to Manual
Complete PRG 10-29-08, 09, 10, 11, 12, 13 when registering via Domain Name. When registering via IP Address use 10-29-03 & 10-29-06 instead.

Continued:



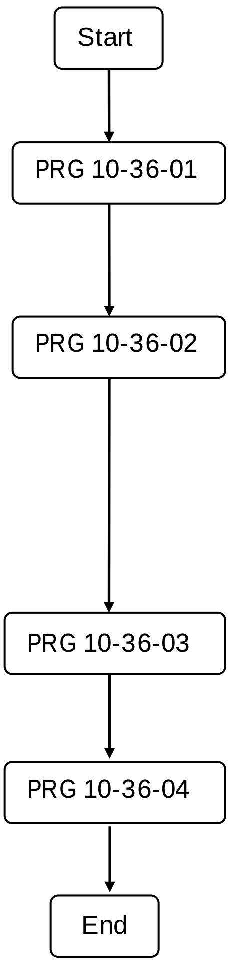
Set the Carrier Type here according the SL1100 SIP Trunk Guide for your ITSP

If the DNS server uses a different port other than DNS default port of 53 this will need to be changed.

If SL1100 is behind a NAT router with Port Forwarding in place set this to “Used”

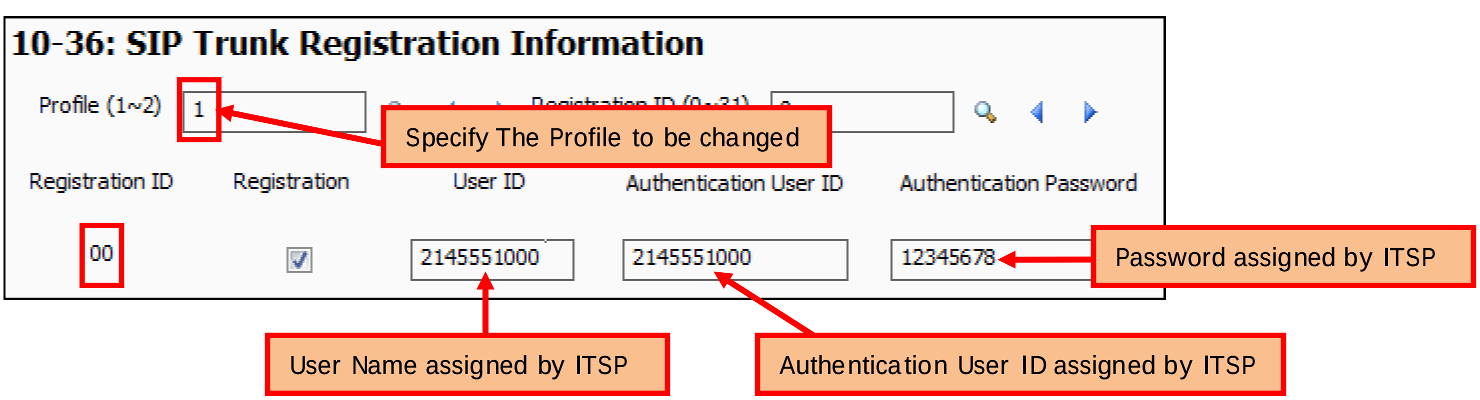
SIP Trunk Authentication:

As of version 5.00 Programs 10-30-02, 03, 04 have been deleted. The following is what is now required for Authentication for SL1100 versions 5.00 and higher:



Description	Input Data
Registration This setting determines if the SIP trunk information is registered.	0 = Disable 1 = Enable (SIP Profile:1-2) (V5.0 Added) Default = 0 Disable
User ID Use this program to input the User ID in the SIP Invite Setup message. A call cannot be completed across the span if there is no outbound CID info. (V5.0 changed) (*Formally PRG 10-28-04 prior to v5.00*)	Up to 32 Characters (alpha or numeric, spaces and/or special characters not allowed) (ex. : UserID@HostName.DomainName) (V5.0 Added) (SIP Profile:1-2) (V5.0 Added) Default = Not set
Authentication User ID This sets the user name of the SIP trunk. (V5.0 Changed)	Up to 64 Characters (SIP Profile:1-2) (V5.0 Added) Default = Not set
Authentication Password This sets the SIP trunk password. (V5.0 Changed)	Up to 32 Characters (SIP Profile:1-2) (V5.0 Added) Default = Not set

Programming Example:



SIP Trunk Sending Outbound Caller ID (CPN):

Start

	Description	Input Data
PRG 20-06-01	Assign stations to a class of service per Day/Night mode.	1~15 = Class of Service 1~15 Default = Extension 101 is in class 15 and all other extensions are in class 1.
PRG 20-08-13	Per class of service enable/disable ISDN Clip. ISDN Clip is used for ISDN and SIP trunks for the purpose of sending outbound caller ID.	0 = OFF (No CPN sent) 1 = ON (CPN is sent) Default = Classes 1~15 set to 0 (OFF, No CPN sent)
PRG 21-17-01	Per SIP trunk setup the outbound telephone number that is to be presented when a user makes a call on this line. Note: PRG 21-19-01 has a higher priority than this setting. If a user has a telephone number programmed in 21-19-01 then that number will be sent. If the user does not have a number programmed in 21-19-01 then the number specified here will be sent.	Assign a valid telephone number up to 16 digits in length. Default = Not Assigned
PRG 21-19-01	Per extension setup the outbound telephone number that is to be presented when a user makes an outbound call from this station.	Up to 16 Digits (1 ~ 0, *, #) (SIP Profile:1-2) (V5.0 Added) Default = Not Assigned

End

Programming Example

20-08: Class of Service Options (Outgoing Call Service)

13 - ISDN Clip

Check ISDN CLIP for each Class of Service

21-17: IP Trunk (SIP) Calling Party Number Setup for Trunks

Trunk	Calling Party Number
05	2142626539

Specify the number to be presented on a per Trunk Basis here. 21-19 for stations takes priority over 21-17. If no number is specified in the station in 21-19 this number will be used.

21-19: IP Trunk (SIP) Calling Party Number Setup for Extensions

Profile (1~2) 1

ICM Extension	Calling Party Number
101	2145554321

Specify the number to be presented on a per Extension, Per Profile basis here.

21-19: IP Trunk (SIP) Calling Party Number Setup for Extensions

Profile (1~2) 2

ICM Extension	Calling Party Number
101	4693219898

In this example note the number in profile 1 is different than the number in profile 2 for the same extension.