

SL2100

Networking Manual

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Introduction

Section 1 GENERAL OVERVIEW

This manual provides information for networking for the NEC SL2100 system.

Section 2 COMMON TERMS

The following terms and the associated abbreviations or alternate nomenclature may be found throughout this document.

Table 1-1 Common Terms and Associated Abbreviations

| Term | Abbreviation | Description |
|--|------------------------------------|--|
| IP7XX-CPU-C1 | CPU-C1 SL2100 CPU CPU | CPU-C1 with main software |
| IP7WW- VOIPDB- C1 | VOIPDB VOIP Daughter Board | Mount this card onto the CPU card (VoIPDB slot) at Main Chassis. Max. 128 channels by license control. |
| IP7WW-3COIDB-C1 | 3COIDB | The 3COIDB provides three analog trunks and is mounted on the 082U, 008U or 000U card. |
| IP7WW-082U-B1 | 082U card | 8-digital station + 2 Analog station interface card |
| IP7WW-12TXH-A1 TEL IP7WW-24TXH-A1 TEL | 4-wire Hybrid Multiline Telephone | 12TXH-A1 TEL 12-button hybrid telephone 24TXH-A1 TEL 24-button hybrid telephone |
| IP7WW- 12TXH- B1 TEL IP7WW- 24TXH- B1 TEL | 2-wire Digital Multiline Telephone | 12TXH-B1 TEL 12-button digital telephone 24TXH-B1 TEL 24-button digital telephone |
| IP7WW-8IPLD-C1 TEL | IP Multiline Telephone | 8IPLD-C1 TEL 8-button IP DESI-Less telephone |
| DTK-12D-1P TEL DTK-24D-1P TEL | Digital Multiline Telephone | DTK-12D-1P TEL Digital 12-button Display Telephone DTK-24D-1P TEL Digital 24-button Display Telephone |
| ITK-24CG-1P TEL | IP Multiline Telephone | IP 24-button Color Display Gigabit Ethernet |

MEMO

General Information

Section 1 VOICE OVER IP

Voice over IP (VoIP) is a technology that converts speech into data packets and transmits these packets over TCP/IP networks. The technology also facilitates compression and silence suppression to reduce network bandwidth demands.

As most organizations already have existing data networks in place, considerable cost savings can be achieved by utilizing spare bandwidth on these networks for voice traffic.

NEC SL2100 supports the use of IP Phones. These telephones provide the same functionality as a multiline telephone but use the data network rather than the traditional telecoms infrastructure. This can reduce costs and allow the use of NEC SL2100 telephones in locations that would not normally be supported by multiline telephones.

[Table 2-1 Table 2-1 VoIP Specifications](#) lists the specifications for various aspects of NEC SL2100 VoIP system.

Table 2-1 VoIP Specifications

| Category | Feature | Notes |
|--------------------------|---------------------------|------------------------|
| IP Address | DHCP Server ^{*1} | CPU-C1/VoIPDB |
| | DHCP Client ^{*1} | VoIPDB/IP Phone |
| QoS | 802.1p/1q | CPU-C1 |
| | L3 QoS (ToS) | Diffserv/IP Precedence |
| Maintenance | HTTP Server | CPU-C1/VoIPDB |
| VLAN | Tag and port-based VLAN | |
| VoCoder | G.711 μ -law/A-law | |
| | G.729a | |
| | G.722 | |
| | G.726 | |
| Jitter Buffer Size | Set by system programming | |
| RTP Length | Set by system programming | |
| Echo Canceller Tail Size | Set by system programming | |
| Level Adjustment | Set by system programming | |
| IP Phone | SIP Phone | SIP Phone |
| SIP Trunk | SIP Trunk | Maximum 128 Trunks |

^{*1}. DHCP Server and DHCP Client cannot be used at the same time. When the DHCP Server is enabled the DHCP Client function cannot be activated.



When the VoIPDB is installed, the CPU-C1 is no longer active, all IP connections go through the VoIPDB.

MEMO

IP Networking

Section 1 INTRODUCTION

IP Networking uses VoIP technology to connect two or more telephone systems together. This allows calls to be made between sites without using the public telephone network. This saves considerable money, and makes communication between sites much easier.



The voice quality of VoIP is dependent on variables such as available bandwidth, network latency and Quality of Service (QoS) initiatives, all of which are controlled by the network and Internet Service Providers. Because these variables are not in NEC control, it cannot guarantee the performance of the user's IP-based remote voice solution. Therefore, NEC recommends connecting VoIP equipment through a local area network using a Private IP address.

Section 2 IP TRUNKS

The SIP Trunks method of networking allows connection to SIP devices. This could be a PBX system or a third-party product. When using SIP, the feature set is limited and the advanced networking features cannot be used.

Refer to [SIP Trunking](#) for a detailed description of SIP trunking and for set up instructions.

To set up IP trunks:

1. Connect the system to the Data Network. (Refer to [General IP Configuration](#) for detailed instructions.)
2. Configure the IP trunks.
3. Configure the SIP information.
4. Configure the F-Route.

2.1 Configure IP Trunks

When installing a VoIP daughter board in the NEC SL2100 system, external line ports are allotted in accordance with the number of Licensed ports for the particular IP Application.

The NEC SL2100 system now has the required information about the remote destinations and the SIP configuration is complete. The only remaining task is to configure F-Route to route calls to remote destinations via the IP trunks. F-Route configuration is discussed in detail in the Automatic Route Selection (ARS) feature in the NEC SL2100 Programming Manual.

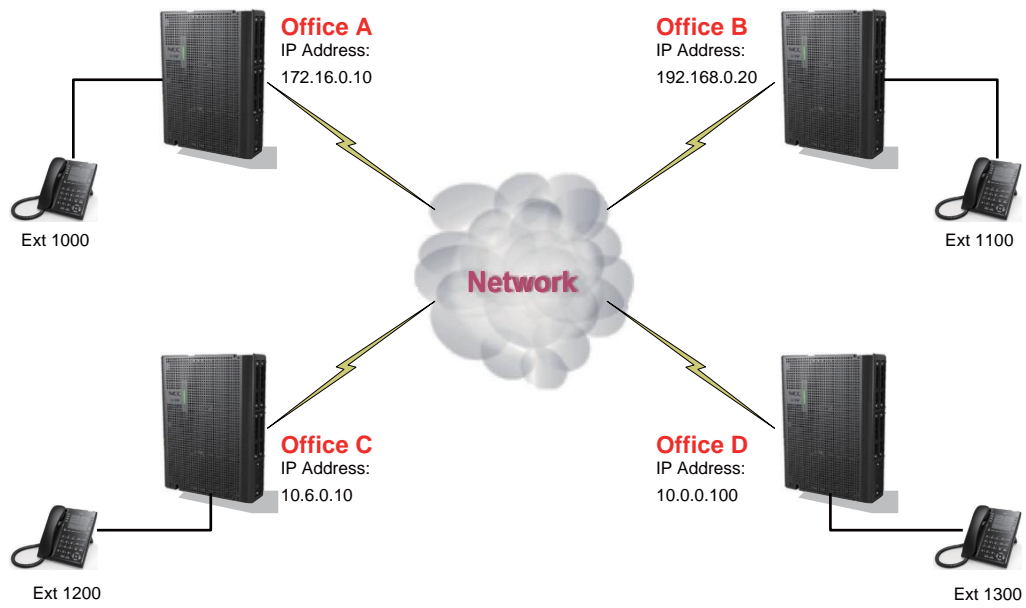


Figure 3-1 Example of SL2100 IP Network Configuration

General IP Configuration

Section 1 INTRODUCTION

The NEC SL2100 system uses IP for various applications, including:

- System Programming
- Voice Over IP

This section describes the procedure for connecting the NEC SL2100 system to an existing data network and configuring TCP/IP. This is the first step in implementing VoIP and other IP applications.

Section 2 NETWORK ADDRESSING OVERVIEW

Before connecting the system to a data network, it is necessary to obtain the relevant IP Addressing information. This information is supplied by the IT Manager or Network Administrator at the customer site.

2.1 IP Address

All equipment/devices used in the LAN setup must have an IP address assignment. An IP address assigns a unique address for each device. There are two types of IP addresses: Private and Global. A Private IP address is not accessible through the Internet; a Global IP address can be accessed through the Internet.

In most cases, a Private address is used, as LAN devices are not usually directly connected to the Internet. Private addresses are usually taken from the following ranges:

- Class A 10.0.0.0 ~ 10.255.255.255
- Class B 172.16.0.0 ~ 172.31.255.255
- Class C 192.168.0.0 ~ 192.168.255.255

A Public address is normally only used when a device is directly connected to the Internet. This is unlikely in the case of the equipment. If Public addressing is used, the numbers are normally allocated by an ISP.

2.2 Subnet Mask

As the IP address includes information to identify both the network and the final destination, the Subnet Mask sets apart the network and destination information. The default subnet masks are:

- Class A 255.0.0.0
- Class B 255.255.0.0
- Class C 255.255.255.0

The Subnet Mask is made up of four groups of numbers. When a group contains the number 255, the router ignores or masks that group of numbers in the IP address as it is defining the network location of the final destination.

For example, if the IP address is: 172.16.0.10 and the Subnet Mask used is Class B (255.255.0.0), the first two groups of numbers (172.16) are ignored once they reach the proper network location. The next two groups (0.10) are the final destination within the LAN to which the connection is to be made.



For sub-netted networks, the subnet mask may be different from the default subnet masks listed above.

2.3 DHCP

Dynamic Host Configuration Protocol (DHCP) assigns a dynamic IP address. Network control may be easier with DHCP as there is no need to assign and program individual IP addresses for the LAN equipment. To use a dynamic IP address, a DHCP server must be provided. The SL2100 can be configured to be the DHCP server for the customers network. Before the DHCP server in the SL2100 can be enabled, the DHCP client function must first be disabled.

When equipment, which is connected to the LAN (the DHCP client), is requesting an IP address, it searches the DHCP server.

When the request for an address is recognized, the DHCP server assigns an IP address, Subnet mask, and the IP address of the router, based on NEC SL2100 system programming.

Section 3 CONFIGURATION EXAMPLES

The following configuration examples illustrate a typical network configuration for an existing network that uses a static address and a typical configuration for a new network that uses a dynamic address.

3.1 Example Configuration 1 - Existing Network with Static Addressing

[Figure 4-1 Example Configuration 1 - Existing Network with Static IP Address](#) shows a typical network configuration that uses Static IP Addressing.

Each client device has a manually assigned IP address in the 192.168.1.0/24 network (i.e., 192.168.1.1 to 192.168.1.254 with a subnet mask of 255.255.255.0). They also have a default gateway address configured (192.168.1.254) this allows the device to route packets to destinations that exist outside of their own LAN.

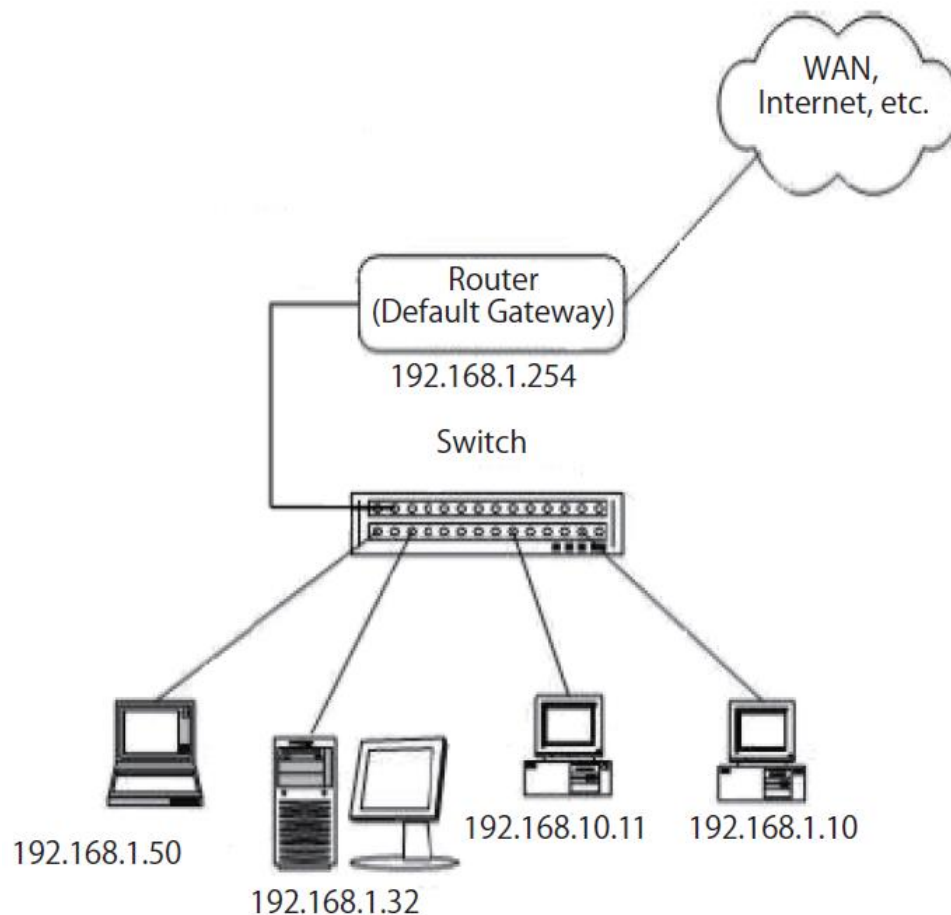



Figure 4-1 Example Configuration 1 - Existing Network with Static IP Address

Assume that a NEC SL2100 is added to the existing data network. The Network Administrator (or IT Manager) should provide the following:

- IP Address (for the CPU-C1)
- IP Addresses (for the VoIP daughter board)
- Subnet Mask
- Default Gateway
- A spare switch

First, program the NEC SL2100:

- 192.168.1.200
- 255.255.255.0
- PRG10-12-03: 192.168.1.254

 A system reset is required for the IP Address changes to take effect.

Now connect the CPU-C1/VoIPDB Ethernet Port to the switch port, using a standard Cat-5 patch cable. The NEC SL2100 is now configured on the network and should be accessible by other devices on the network. Refer to [Figure 4-2 Example Configuration 1 - Adding the NEC SL2100 Chassis to the Network](#).

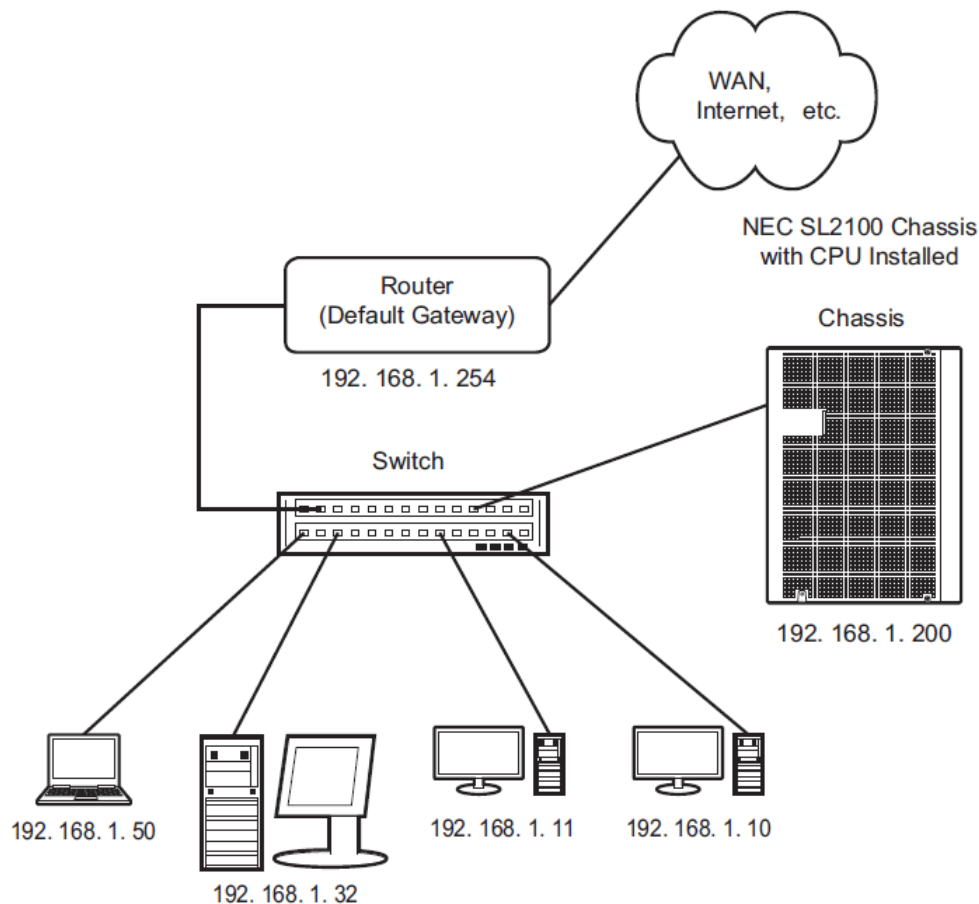


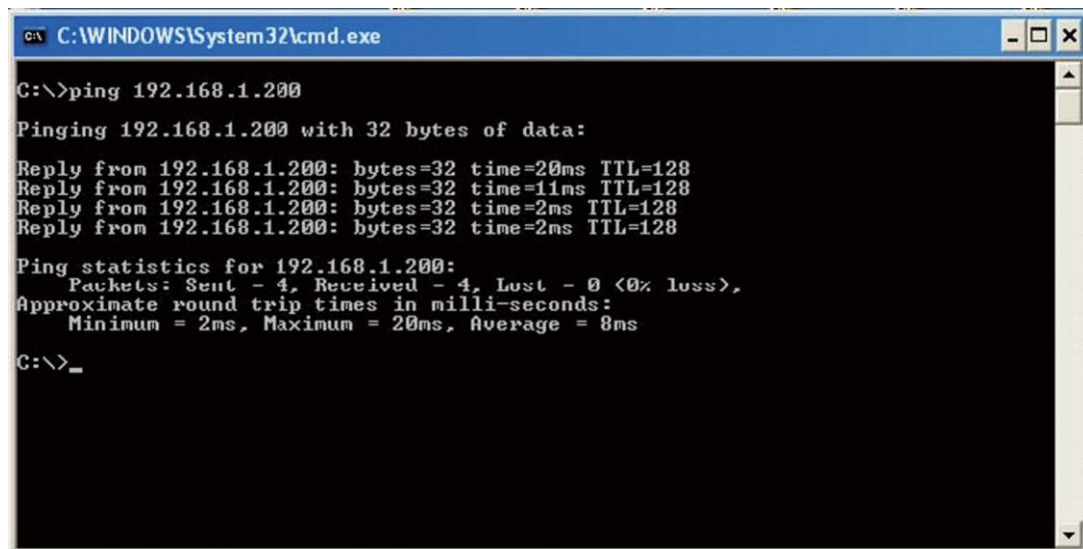
Figure 4-2 Example Configuration 1 - Adding the NEC SL2100 Chassis to the Network

Section 4 TESTING THE NEC SL2100 NETWORK CONNECTION

To test the NEC SL2100 network connection, it is possible to use the ICMP (Internet Control Message Protocol) Ping command. This basically sends a small amount of data from one device to another and then waits for a reply. This test confirms that the IP addressing and physical connection are good. To perform this test, from a Windows PC:

1. Click **Start**.
2. Click **Run...**
3. In the Open dialogue box, enter **command**.
4. Click **OK**. A Command prompt window opens.
5. Type **ping 192.168.1.200**.

[Figure 4-3 Testing the Network Connection](#) shows that the NEC SL2100 system has replied to the Ping request – this indicates that the NEC SL2100 system is correctly connected to the network.



```
C:\WINDOWS\System32\cmd.exe

C:\>ping 192.168.1.200

Pinging 192.168.1.200 with 32 bytes of data:

Reply from 192.168.1.200: bytes=32 time=20ms TTL=128
Reply from 192.168.1.200: bytes=32 time=11ms TTL=128
Reply from 192.168.1.200: bytes=32 time=2ms TTL=128
Reply from 192.168.1.200: bytes=32 time=2ms TTL=128

Ping statistics for 192.168.1.200:
    Packets: Sent = 4, Received = 4, Lost = 0 (0% loss),
    Approximate round trip times in milli-seconds:
        Minimum = 2ms, Maximum = 20ms, Average = 8ms

C:\>_
```

Figure 4-3 Testing the Network Connection

MEMO

PROGRAMMING

The remainder of this chapter provides instructions for programming individual program blocks.

Section 1 BEFORE YOU START PROGRAMMING

This chapter provides you with detailed information about the NEC SL2100 program blocks that may be required to connect the NEC SL2100 to a data network and to configure the VoIP function. The configuration and programming examples, found in the earlier chapters, can be a useful reference when programming the data.

Section 2 HOW TO USE THIS MANUAL

This section lists each program in numerical order. For example, Program 10-01 is at the beginning of the section and Program 92-01 is at the end. The information on each program is subdivided into the following headings :

Description describes what the program options control. The Default Settings for each program are also included. When you first install the system, it uses the Default Setting for all programs. Along with the Description are the **Conditions** which describe any limits or special considerations that may apply to the program.

The program access level is just above the Description heading. You can only use the program if your access level meets or exceeds the level the program requires. Refer to [How to Enter Programming Mode](#) for a list of the system access levels and passwords.

Feature Cross Reference provides you with a table of all the features affected by the program. You will want to keep the referenced features in mind when you change a program. Customizing a feature may have an effect on another feature that you did not intend.

Telephone Programming Instructions shows how to enter the program data into system memory. For example :

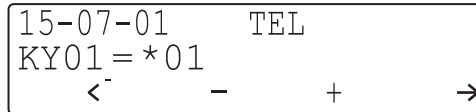
1. Enter the programming mode.
2. 15-07-01

```
1507-01 TEL200
KY01 = *01
```

Hybrid (4W) Multiline Terminal

```
15-07-01 TEL
KY01 = *01
< - + >
```

Digital (2W) Multiline Terminal



15-07-01 TEL
KY01 = *01
< - + >

IP Multiline Terminal

tells you to enter the programming mode, dial 150701 from the telephone dial pad. After you do, you will see the message “15-07-01 TEL” on the first line of the telephone display. This indicates the program number (15-07), item number (01), and that the options are being set for the extension. The second row of the display “KY01 = *01” indicates that Key 01 is being programmed with the entry of *01. The third row allows you to move the cursor to the left or right, depending on which arrow is pressed. To learn how to enter the programming mode, refer to [How to Enter Programming Mode](#).

Section 3 HOW TO ENTER PROGRAMMING MODE

To enter programming mode :

1. Go to any working display telephone.



In a newly installed system, use extension (port 1).

2. Do not lift the handset.
3. Press **Speaker**.
4. **##*.**

Password

5. Dial the system password + **Hold**.

Refer to the following table for the default system passwords. To change the passwords, use 90-02 : Programming Password Setup.

■ Program Mode

Hybrid (4W) Multiline Terminal

Program Mode
Base Service OP1 OP2

Digital (2W) Multiline Terminal, IP Multiline Terminal

| Password | User Name | Level | Programs at this Level |
|----------|-----------|--------|---|
| 12345678 | tech | 2 (IN) | Installation (IN) : IN, SA and SB levels defined by 90-26 are available. |
| 0000 | admin1 | 3 (SA) | System Administrator - Level 1 (SA) : 10-01, 10-02, 10-12, 10-13, 10-14, 10-15, 10-16, 10-17, 10-18, 10-23, 10-25, 10-28, 10-29, 10-45, 10-67, 12-02, 12-03, 12-04, 12-08, 15-01, 15-07, 15-09, 15-10, 15-11, 15-20, 20-16, 20-34, 20-57, 20-63, 20-64, 20-66, 20-70, 21-07, 21-14, 22-04, 22-11, 22-17, 25-08, 30-03, 30-0432-02, 45-02, 84-16, 84-22, 90-03, 90-04, 90-06, 90-07, 90-16, 90-19, 90-65, 93-01, 93-02, 93-03, 93-04, 93-05, 93-06 |
| 9999 | admin2 | 4 (SB) | System Administrator - Level 2 (SB) : 13-04, 13-05, 13-06, 13-11, 15-14, 21-20 |

Section 4 HOW TO EXIT PROGRAMMING MODE

To exit the programming mode :

When you are done programming, you must be out of a program option to exit (pressing the **Mute** key will exit the program option).

1. Press **Mute** key to exit the program options, if needed.

■ Program Mode

Hybrid (4W) Multiline Terminal

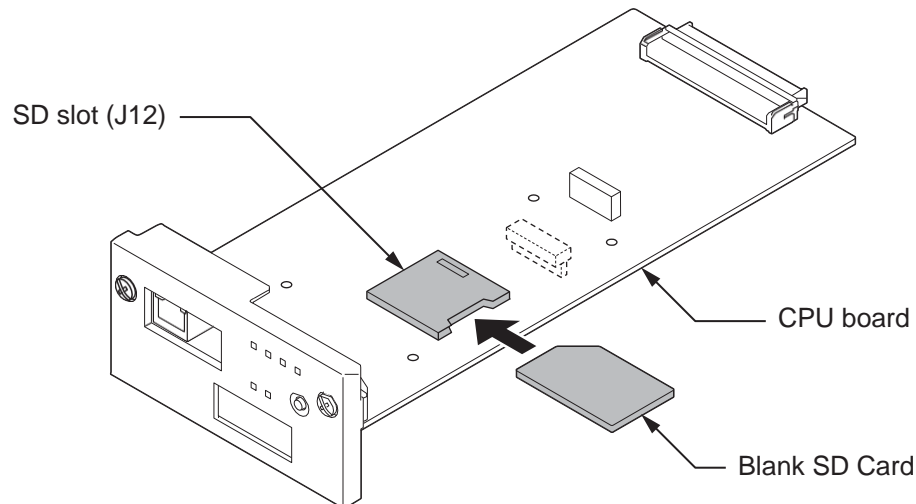
Program Mode
Base Service OP1 OP2

Digital (2W) Multiline Terminal, IP Multiline Terminal

2. Press **Speaker**. If changes were to the system programming, "Saving System Data" is displayed.
3. The display shows "Complete Data Save" when completed and exits the telephone to an idle mode.



To save a customer's database, a blank SD Card is required. Insert the SD Card into the CPU and, using Program 90-03, save the software to the SD Card. (Program 90-04 is used to reload the customer data if necessary.) Note that a SD Card can only hold one customer database. Each database to be saved requires a separate drive.



Section 5 USING KEYS TO MOVE AROUND IN THE PROGRAMS

Once you enter the programming mode, use the keys in the following chart to enter data, edit data and move around in the menus.

Table 5-1 Keys for Entering Data (Hybrid (4W) Multiline Terminal)

| When you want to ... | Telephone Programming |
|---------------------------------|-------------------------------|
| Enter Data into program | 0 ~ 9, *, #, Line Key (1 ~ 6) |
| Next Index | Cursor Key (Up) |
| Prior Index | Cursor Key (Down) |
| Select Data | Line Key (1 ~ 6) |
| All Clear | Flash |
| Apply | Hold Enter |
| Go Back to Prior Screen | Mute Clear / Back |
| Move Cursor Jump Up/Down | DND / CONF |
| Delete single character | Clear / Back |
| Next Page | - |
| Toggle between Number/Character | Help |
| While in a Entering Number | |
| Prior Page | - |
| Quit the programming | Speaker Exit |
| Move Cursor to Left | Cursor Key (Left) |
| Change Program Number | - |
| Change Index Number | |

| When you want to ... | Telephone Programming |
|-----------------------|-----------------------|
| Change Program Number | - |
| Change Index Number | |
| Move Cursor to Right | Cursor Key (Right) |

Table 5-2 Keys for Entering Data (Digital (2W) Multiline Terminal, IP Multiline Terminal)

| When you want to ... | Telephone Programming |
|---------------------------------|---------------------------------|
| Enter Data into program | 0 ~ 9, *, # Line Key (1 ~ 6) |
| Next Index | Cursor Key (Up) |
| Prior Index | Cursor Key (Down) |
| Select Data | Line Key (1 ~ 6) |
| All Clear | Flash |
| Apply | Hold Enter |
| Go Back to Prior Screen | Mute Clear / Back |
| Move Cursor Jump Up/Down | DND |
| Delete single character | Clear / Back |
| Next Page | Help |
| Toggle between Number/Character | |
| While in a Entering Number | |
| Prior Page | Transfer |
| Quit the programming | Speaker Exit |
| Move Cursor to Left | Cursor Key (Left) Soft Key1 |
| Change Program Number | Soft Key2 |
| Change Index Number | |
| Change Program Number | Soft Key3 |
| Change Index Number | |
| Move Cursor to Right | Cursor Key (Right) Soft Key4 |

Section 6 PROGRAMMING NAMES AND TEXT MESSAGES

Several programs (e.g., Program 20-16 : Selectable Display Messages) require you to enter text. Use the following chart when entering and editing text. When using the keypad digits, press the key once for the first character, twice for the second character, etc. For example, to enter a C, press the key **2** three times. Press the key six times to display the lower case letter. The name can be up to 12 digits long.

Table 5-3 Keys for Entering Names

| Use this keypad digit ... | When you want to ... |
|---------------------------|--|
| 1 | Enter characters : 1 @ [¥] ^ _ ` { } > < Á Â Ã Ä Å Æ Ç È É Ê Ë Ì Í Ö |
| 2 | Enter characters : A-C, a-c, 2. |
| 3 | Enter characters : D-F, d-f, 3. |

| Use this keypad digit ... | When you want to ... |
|---|--|
| 4 | Enter characters : G-I, g-i, 4. |
| 5 | Enter characters : J-L, j-l, 5. |
| 6 | Enter characters : M-O, m-o, 6. |
| 7 | Enter characters : P-S, p-s, 7. |
| 8 | Enter characters : T-V, t-v, 8. |
| 9 | Enter characters : W-Z, w-z, 9. |
| 0 | Enter characters : 0 ! " # \$ % & ' () ð ó ú á ä æ ö ü α ε θ Β |
| * | Enter characters : * + , - . / : ; < = > ? π Σ σ Ω ∞ € £ |
| # | # = Accepts an entry (only required if two letters on the same key are needed - ex : TOM). Pressing # again = Space. (In system programming mode, use the right arrow Softkey instead to accept and/or add a space.) |
| Clear/Back or DND/CONF (4W Hybrid Multiline Terminal) / DND (2W Digital/IP Multiline Terminal) | Clear the character entry one character at a time. |
| Flash | Clear all the entries from the point of the flashing cursor and to the right. |

Section 7 USING SOFTKEYS FOR PROGRAMMING

(This Feature is available for Digital (2W) Multiline Terminal and IP Multiline Terminal Only.)

Each Display telephone with Softkeys provides interactive Softkeys for intuitive feature access. The options for these keys will automatically change depending on where you are in the system programming. Simply press the Softkey located below the option you wish and the display will change accordingly.



Pressing the Cursor key Up or Down will scroll between the menus.



Section 8 WHAT THE SOFTKEY DISPLAY PROMPTS MEAN

(This Feature is available for Digital (2W) Multiline Terminal and IP Multiline Terminal Only.)

When using a display telephone in programming mode, various Softkey options are displayed. These keys will allow you to easily select, scan, or move through the programs.

Table 5-4 Softkey Display Prompts

| Softkey Display Prompts | |
|---------------------------------------|--|
| If you press this Soft-key ... | The system will ... |
| back | Go back one step in the program display. You can press Cursor Key (UP) or Cursor Key (Down) to scroll forward or backward through a list of programs. |
| ↑ | Scroll down through the available programs. |
| ↓ | Scroll up through the available programs. |
| select | Select the currently displayed program. |
| ← | Move the cursor to the left. |
| > | Move the cursor to the right. |
| - 1 | Move back through the available program options. |
| + 1 | Move forward through the available program options. |

Section 9 PROGRAMMING YOUR SYSTEM

This sections describes the programs used to connect the NEC SL2100 to a data network and to configure the VoIP functions.

Program 10 : System Configuration Setup

10-03 : ETU Setup

Level:

IN

Description

Use **Program 10-03 : ETU Setup** to setup and confirm the Basic Configuration data for each board. When changing a defined terminal type, first set the type to 0 and then plug the new device in to have the system automatically define it or you may have to reseal the board.



The items highlighted in gray are read only and cannot be changed.

Input Data

| | |
|----------|--------|
| Slot No. | 1 ~ 12 |
|----------|--------|

For HBI PKG Setup

| | |
|----------------------|-------|
| Physical Port Number | 1 ~ 8 |
|----------------------|-------|

| Item No. | Item | Input Data | Default |
|----------|-----------------------------|---|---------|
| 01 | Terminal Type | 0 = No Setting 1 = Multi-Line Telephone 2 = SLT 8 = Door Phone 10 = DSS Console 3 ~ 7, 9, 11, 12 = Not Used | 2 |
| 02 | Logical Port Number | [Terminal Type = 0] : 0 [Terminal Type = 1] : 1 ~ 128 [Terminal Type = 2] : 1 ~ 128 [Terminal Type = 8] : 1 ~ 2 [Terminal Type = 10] : 1 ~ 12 | 0 |
| 03 | Transmit Gain level for SLT | 1 ~ 57 (- 15.5 ~ + 12.5 dB) | 32 |
| 04 | Receive Gain level for SLT | 1 ~ 57 (- 15.5 ~ + 12.5 dB) | 32 |
| 05 | Select port type | 0 = Hybrid Port 1 = Door Phone | 0 |
| 06 | Multi-Line Telephone Line | 0 = None 1 = 12 Line 2 = 24 Line | 0 |

For ESI PKG Setup

| | |
|----------------------|-------|
| Physical Port Number | 1 ~ 8 |
|----------------------|-------|

| Item No. | Item | Input Data | Description | Default |
|----------|------------------------------|--|-------------|---------|
| 01 | Terminal Type | 0 = No Setting 1 = Multiline Telephone 10 = DSS Console | | 0 |
| 02 | Logical Port Number | 0 = No Setting 1 = Multiline Telephone (1 ~ 128) 10 = DSS Console (1 ~ 12) | | 0 |
| 08 | Multiline Telephone Type | 0 = DT3** 1 = DT4** 3 = SL1100 2W 4 = SL2100 2W | | 0 |
| 09 | Item Side option information | 0 = No Option 4 = EHS | | 0 |
| 10 | Bottom option information | 0 = None 4 = WHA 5 = BCA | | 0 |

For SLI PKG Setup

| | |
|----------------------|-------|
| Physical Port Number | 1 ~ 8 |
|----------------------|-------|

| Item No. | Item | Input Data | Description | Default |
|----------|-------------------------------|-----------------------------|--|-----------|
| 01 | Logical Port Number | 0 ~ 128 | | 0 |
| 03 | Transmit Gain Level (S-Level) | 1 ~ 57 (- 15.5 ~ + 12.5 dB) | | 32 (0 dB) |
| 04 | Receive Gain Level (R-Level) | 1 ~ 57 (- 15.5 ~ + 12.5 dB) | | 32 (0 dB) |
| 05 | Select port type | 0 = SLT 1 = Door Phone | HW switch can be change SLT or Door Phone. | 0 |

For COI PKG Setup



| | |
|----------------------|-------|
| Physical Port Number | 1 ~ 3 |
|----------------------|-------|

| Item No. | Item | Input Data | Default |
|----------|---------------------|------------|---------|
| 01 | Logical Port Number | 0 ~ 128 | 0 |

For BRI PKG Setup



| | |
|------------------|-------|
| ISDN Line Number | 1 ~ 2 |
|------------------|-------|

| Item No. | Item | Input Data | Description | Default |
|----------|----------------|---|-------------|---------|
| 01 | ISDN Line Mode | 0 = No Setting 1 = T-Point 2 = S-Point 3 ~ 5 = Not Used 6 = S-Point (Leased Line) | | 1 |

| Item No. | Item | Input Data | Description | Default |
|----------|-------------------------------------|---|--|---------|
| 02 | Logical Port Number | [0 : No Setting] = 0 [1 : T-Point] = 1 ~ 128 [2 : S-Point] = 1 ~ 128 [3 ~ 5] = Not Used [6 : S-Point (Leased Line)] = 1 ~ 128 |  The starting port number of a BRI line is displayed. Two logic ports are automatically assigned to a BRI line. | 0 |
| 03 | Connection Type | 0 = Point-to-Multipoint 1 = Point-to-Point | | 0 |
| 04 | Layer 3 Timer Type | 1 ~ 5 |  Each timer value of Layer 3 is set up for every type using Program 81-06 (T-Bus). | 1 |
| 05 | Calling Party Number | 0 = No 1 = Yes | Based on this setting, the system includes a Presentation Allowed (1) or Presentation Restricted (0) in the Setup message to allow or deny the Calling Party Number. Program 15-01-04 must also be set to 1 if this option is enabled. | 1 |
| 06 | Connection Bus Mode | 0 = Extended passive bus 1 = Short passive bus | | 0 |
| 07 | S-point DDI digits | 0 ~ 4 | | 0 |
| 08 | Dial Sending Mode | 0 = Enblock Sending 1 = Overlap Sending | ISDN Protocol definition | 1 |
| 09 | Dial Information Element | 0 = Keypad Facility 1 = Called Party Number | ISDN Protocol definition [Only when Dial Sending Mode (10-03-08) is set for 1 (Overlap Sending)] | 1 |
| 10 | Master/Slave System | 0 = Slave System 1 = Master System | If set to 0, system is synchronized to network clock. If set to 1, system is not synchronized to the network clock. | 0 |
| 14 | Service Protocol for S-point | 0 = Keypad facility 1 = Specified Protocol | | 0 |
| 15 | Call Busy Mode for S-point | 0 = Alerting 1 = Disconnect | | 0 |
| 17 | ISDN Line Ringback Tone | 0 = Disable 1 = Enable | System can provide ringback tone, if set to 1 : Enable. | 0 |
| 18 | Type of Number | 0 = Unknown 1 = International number 2 = National number 3 = Network specific number 4 = Subscriber number 5 = Speed Dial number | ISDN Protocol definition | 0 |
| 19 | Numbering Plan Identification | 0 = Unknown 1 = ISDN numbering plan 2 = Data numbering plan 3 = Telex numbering plan 4 = National standard numbering plan 5 = Private numbering plan | ISDN Protocol definition | 0 |
| 25 | Layer 1 Supervision | 0 = Disable 1 = Enable | | 0 |
| 26 | Screening Indicator (V1.5 added) | 0 = user-provided (not screened) 1 = user-provided (verified and passed) 2 = user-provided (verified and failed) | Calling Party Number information element Screening indicator (octet 3a) | 1 |

For PRI PKG Setup

| | |
|-------------|--------|
| SLOT Number | 1 ~ 12 |
|-------------|--------|

| Item No. | Item | Input Data | Description | Default |
|----------|--------------------------------|---|--|---------|
| 01 | ISDN Line Mode | 0 = No Setting 1 = T-Point 2 = S-Point 6 = S-Point (Leased Line) | | 1 |
| 02 | Logical Port Number | [0 : No Setting] = 0 [1 : T-Point] = 1 ~ 128 [2 : S-Point] = 1 ~ 128 [6 : S-Point (Leased Line)] = 1 ~ 128 |  The start port number of a PRI line is displayed. | 0 |
| 03 | CRC Multi-frame (CRC4) | 0 = off 1 = on | | 1 |
| 04 | Layer 3 Timer Type | 1 ~ 5 |  Each timer value of Layer 3 is set up for each type in Program 81-06 (T-Bus) | 1 |
| 05 | CLIP Information | 0 = No 1 = Yes | Based on this setting, the system includes a Presentation Allowed (1) or Presentation Restricted (0) in the Setup message to allow or deny the Calling Party Number. Program 15-01-04 must also be set to 1 if this option is enabled. | 1 |
| 06 | Length of Cable | 0 = Level 1 1 = Level 2 2 = Level 3 3 = Level 4 4 = Level 5 | | 2 |
| 07 | S-point DDI digits | 0 - 4 | | 0 |
| 08 | Dial Sending Mode | 0 = Enbloc Sending 1 = Overlap Sending | ISDN Protocol definition | 1 |
| 09 | Dial Information Element | 0 = Keypad Facility 1 = Called Party Number | ISDN Protocol definition Only when Dial Sending Mode (10-03-08) is set for 1 (Overlap Sending). | 1 |
| 10 | Master/Slave System | 0 = Slave System 1 = Master System | If set to 0, system is synchronized to network clock. If set to 1, system is not synchronized to the network clock. | 0 |
| 13 | Loss-Of-Signal Detection Limit | 0 = Level 0 (lowest sensitivity) 1 = Level 1 2 = Level 2 3 = Level 3 4 = Level 4 5 = Level 5 6 = Level 6 7 = Level 7 (highest sensitivity) | If the transmit/receive voltage is less than the setting in 10-03-13, the system considers this as Loss-Of-Signal and the PRI does not come up. Note that there are different values based on the setting in 10-03-12 for the PRI. | 2 |
| 14 | Service Protocol for S-point | 0 = Keypad facility 1 = Specified Protocol for Aspire system | | 0 |
| 15 | Call Busy Mode for S-point | 0 = Alerting 1 = Disconnect | | 0 |

| Item No. | Item | Input Data | Description | Default |
|----------|---|---|--|---------|
| 16 | Two B-Channel Transfer for PRI Service | 0 = off 1 = on | | 0 |
| 17 | ISDN Ringback Tone | 0 = Disable 1 = Enable | System can provide ring-back tone, by setting 10-03-17 is set to 1 : Enable. | 0 |
| 18 | Type of Number | 0 = Unknown 1 = International number 2 = National number 3 = Network Specific number 4 = Subscriber number 5 = Speed Dial number | ISDN Protocol definition. Select the number type for the ISDN circuit. | 0 |
| 19 | Numbering Plan Identification | 0 = Unknown 1 = ISDN numbering plan 2 = Data numbering plan 3 = Telex numbering plan 4 = National standard numbering plan 5 = Private numbering plan | ISDN Protocol definition. Select the Numbering Plan used for the ISDN circuit. | 0 |
| 20 | Network Exchanger Selection | 0 = Standard 1 = reserved 2 = reserved 3 = DMS (A211) 4 = 5ESS 5 = DMS (A233) 6 = 4ESS 7 = NI-2 | Select the ISDN protocol for the ISDN circuit | 0 |
| 21 | Number of Ports | 0 = Auto 1 = 4 Ports 2 = 8 Ports 3 = 12 Ports 4 = 16 Ports 5 = 20 Ports 6 = 24 Ports 7 = 28 Ports | | 0 |
| 26 | Screening Indicator (V1.5 added) | 0 = user-provided (not screened) 1 = user-provided (verified and passed) 2 = user-provided (verified and failed) | Calling Party Number information element Screening indicator (octet 3a) | 1 |

For VoIP PKG Setup

| | |
|-------------|---|
| SLOT Number | 0 |
|-------------|---|

| Item No. | Item | Input Data | Default |
|----------|--------------------------------|--|---------|
| 01 | VoIP Type | 4 = Built-in VoIP 5 = VoIPDB | |
| 02 | Number of Channel | 8 = Built-In VoIP 16 ~ 128 = VoIPDB | |
| 03 | Number of Voice Channel | 0 ~ 128 | 0 |

Conditions

- When changing a defined terminal type, first set the type to 0 and then plug the new device in to have the system automatically define it, or redefine the type manually.
- The system must have a board installed to view/change the options for that type of board.

Feature Cross Reference

None

Program 10 : System Configuration Setup

10-04 : Music On Hold Setup

Level:

IN

Description

Use **Program 10-04 : Music on Hold Setup** to set the Music on Hold (MOH) source.

Input Data

| Item No. | Item | Input Data | Description | Default |
|----------|---------------------------------------|--|---|-----------|
| 01 | Music on Hold Source Selection | 0 = Internal MOH (Tone is fixed by one music) 1 = External MOH (Tone set by PRG 10-60-01: Audio Port No.2) 2 = Service Tone (Tone set by PRG 80-01, tone 64) 3 = VRS (Tone set by PRG 10-04-02) | <ul style="list-style-type: none"> Internal Music Tune - The tune is fixed by one music tune. External Source - ACI input via 308U (J5)/082U (J4) audio connector (Program10-60-01). Silence - Callers on hold hear silence. | 0 |
| 02 | Music on Hold Tone Selection | [In case Item 1 is 3.] 1 ~ 100 = VRS Message Number | | 1 |
| 03 | Audio Gain Setup | 1 ~ 57 (- 15.5 ~ + 12.5 dB) | | 32 (0 dB) |

Conditions

None

Feature Cross Reference

- Background Music
- Music on Hold

Program 10 : System Configuration Setup

10-12 : CPU Network Setup

Level:

SA

Description


Use **Program 10-12 : CPU Network Setup** to setup the IP Address, Subnet-Mask, and Default Gateway addresses.

Caution!

If any IP Address or NIC settings are changed, the system must be reset for the changes to take affect.

Input Data

| Item No. | Item | Input Data | Description | Default |
|----------|-----------------|--|---|---------------|
| 01 | IP Address | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | Set for CPU. | 192.168.0.10 |
| 02 | Subnet Mask | 128.0.0.0 192.0.0.0 224.0.0.0 240.0.0.0 248.0.0.0 252.0.0.0 254.0.0.0 255.0.0.0 255.128.0.0 255.192.0.0 255.224.0.0 255.240.0.0 255.248.0.0 255.252.0.0 255.254.0.0 255.255.0.0 255.255.128.0 255.255.192.0 255.255.224.0 255.255.240.0 255.255.248.0 255.255.252.0 255.255.254.0 255.255.255.0 255.255.255.128 255.255.255.192 255.255.255.224 255.255.255.240 255.255.255.248 255.255.255.252 255.255.255.254 255.255.255.255 | The setting of Subnet Mask is invalid when all Host Addresses are 0. If the network section is: 0, 127, 128.0, 191.255, 192.0.0, 223.255.255 The setting of Subnet Mask is invalid. | 255.255.255.0 |
| 03 | Default Gateway | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | Assign the default gateway IP address. | 0.0.0.0 |

| Item No. | Item | Input Data | Description | Default |
|----------|------------------------|--|--|-------------|
| 05 | NIC Interface | 0 = Auto Detect 1 = 100Mbps, Full Duplex 2 = Reserved 3 = Reserved 4 = Reserved 5 = Reserved | NIC Auto Negotiate (CPU)  When this programming was changed, the LAN cable must be unplugged and then reconnected. | 0 |
| 07 | NAPT Router IP Address | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | Set the IP address on the WAN side of router. | 0.0.0.0 |
| 08 | ICMP Redirect | 0 = Enable 1 = Disable | When receiving ICMP redirect message, this determines if the IP Routing Table updates automatically or not. | 0 |
| 09 | VoIP IP Address | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | Assign the IP Address for the VoIPDB. | 172.16.0.10 |
| 10 | VoIP Subnet Mask | 128.0.0.0 192.0.0.0 224.0.0.0 240.0.0.0 248.0.0.0 252.0.0.0 254.0.0.0 255.0.0.0 255.128.0.0 255.192.0.0 255.224.0.0 255.240.0.0 255.248.0.0 255.252.0.0 255.254.0.0 255.255.0.0 255.255.128.0 255.255.192.0 255.255.224.0 255.255.240.0 255.255.248.0 255.255.252.0 255.255.254.0 255.255.255.0 255.255.255.128 255.255.255.192 255.255.255.224 255.255.255.240 255.255.255.248 255.255.255.252 255.255.255.254 255.255.255.255 | Assign the subnet mask for the VoIPDB. | 255.255.0.0 |
| 11 | NIC Setup | 0 = Auto Detect 1 = 100 Mbps, Full Duplex 2 = Not Used 3 = Not Used 4 = Not Used 5 = 1 Gbps, Full Duplex | Set for VoIPDB. | 0 |
| 13 | DNS Primary Address | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | In the future, use this system data when add a function with DNS. | 0.0.0.0 |

| Item No. | Item | Input Data | Description | Default |
|----------|------------------------------|--|--|---------|
| 14 | DNS Secondary Address | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | In the future, use this system data when add a function with DNS. | 0.0.0.0 |
| 15 | DNS Port | 1 ~ 65535 | In the future, use this system data when add a function with DNS. | 53 |
| 17 | VoIP NIC Port Setting | 0 = MDI 1 = MDI-X | | 0 |
| 18 | CPU MTU | 1000 ~ 1500 | The MTU value of CPU NIC. | 1450 |
| 19 | VoIP MTU | 1000 ~ 1500 | The MTU value of VoIP NIC. | 1450 |
| 20 | SSLv3 Setting | 0 = Disable 1 = Enable | Set whether to use the SSLv3 in SMTP. 0 (Disable) is encrypted with TLSv1. This data requires a reset to apply changed data to the system. | 0 |

Conditions

- The system must be reset for these changes to take affect.

Feature Cross Reference

None

Program 10 : System Configuration Setup

10-13 : In-DHCP Server Setup


Level:

SA

Description

Use **Program 10-13 : In-DHCP Server Setup** to setup the DHCP Server built into the CPU.

Input Data

| Item No. | Item | Input Data | Description | Default |
|----------|-------------------------|---------------------------|--|------------|
| 01 | DHCP Server Mode | 0 = Disable 1 = Enable | Enable or disable the use of the built-in DHCP Server. This program cannot be enabled if PRG10-63-01 is enabled. | 0 |
| 02 | Lease Time | Days 0 ~ 255 | Lease Time of the IP address to a client.  Pressing the Hold Key increments to the next setting data. | 0 day |
| | | Hour 0 ~ 23 | | 0 hour |
| | | Minutes 0 ~ 59 | | 30 minutes |
| 05 | Last DHCP Data | 0 = Disable 1 = Enable | If 10-13-01 is enabled, this setting determines if DHCP resource is enabled or disabled. | 1 |

Conditions

None

Feature Cross Reference

None

Program 10 : System Configuration Setup

10-14: Managed Network Setup

Level:

SA

Description

Use **Program 10-14 : Managed Network Setup** to set up the range of the IP address which the DHCP Server leases to a client.

Input Data

| Item No. | Item | Input Data | Description | Default | Related Program |
|----------|--|--|--|--------------|-----------------|
| 01 | The Range of the IP address to Lease. | Minimum : 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | When Maximum has not been entered, the maximum value equals the minimum value. | 172.16.0.100 | 10-13-04 |
| | | Maximum : 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | | 172.16.5.254 | |

Conditions

None

Feature Cross Reference

None

Program 10 : System Configuration Setup

10-15: Client Information Setup

Level:

SA

Description

Use **Program 10-15 : Client Information Setup** to set up the client information when the DHCP server needs to assign a fixed IP address to clients.

Input Data

| | |
|---------------|---------|
| Client Number | 1 ~ 128 |
|---------------|---------|

| Item No. | Item | Input Data | Description | Default |
|----------|-------------|--|---|-------------------|
| 01 | MAC Address | MAC : 00-00-00-00-00-00 ~ FF-FF-FF-FF-FF-FF | | 00-00-00-00-00-00 |
| | IP Address | 1.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | The IP address should be assigned out of the scope range set up in Program 10-14. | 0.0.0.0 |

Conditions

None

Feature Cross Reference

None

Program 10 : System Configuration Setup


10-16: Option Information Setup

Level:

SA

Description

Use **Program 10-16 : Option Information Setup** to set up the option given from the DHCP server to each client.

 The items highlighted in gray are read only and cannot be changed.

Input Data

| Item No. | Item | Input Data | Description | Default |
|----------|------------------------|---|---|-------------|
| 01 | Router | Code number 0 ~ 255 | Set the Router IP address. | 3 (Fixed) |
| | | IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | | 0.0.0.0 |
| 02 | DNS Server | Code number 0 ~ 255 | Set IP address of DNS Server. | 6 (Fixed) |
| | | IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | | 0.0.0.0 |
| 05 | MGC IP Address | Code number 0 ~ 255 | | 129 (Fixed) |
| | | IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | | 172.16.0.10 |
| 06 | Client Host Name | Code number 0 ~ 255 | Set the Client Host Name. | 12 (Fixed) |
| | | Maximum 64 character strings | | No Setting |
| 07 | DNS Domain Name | Code number 0 ~ 255 | Set the DNS Domain Name. | 15 (Fixed) |
| | | Maximum 20 character strings | | No Setting |
| 08 | Download Protocol | Code number 0 ~ 255 | Set Download Protocol used for AutoConfig (for 8IPLD-C1 Terminal). | 43 (Fixed) |
| | | Sub code number | | 163 (Fixed) |
| | | 1 = FTP 2 = HTTP | | 1 |
| 09 | Encryption Information | Code number 0 ~ 255 | Set an Encryption Information used for Auto-Config (for 8IPLD-C1 Terminal). | 43 (Fixed) |
| | | Sub code number | | 164 (Fixed) |
| | | Fixed 128 character strings | | No Setting |
| 10 | FTP Server Address | Code number 0 ~ 255 | Set a FTP Server Address used for AutoConfig. | 43 (Fixed) |
| | | Sub code number | | 141 |
| | | IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | | 0.0.0.0 |

| Item No. | Item | Input Data | Description | Default |
|----------|---------------------------------|---|--|-------------|
| 11 | Config File Name | Code number 0 ~ 255 | Set a File Name used for AutoConfig. | 43 (Fixed) |
| | | Sub code number | | 151 |
| | | Maximum 15 character strings | | No Setting |
| 12 | Vender Class ID | Code number 0 ~ 255 | | 60 (Fixed) |
| | | Maximum 256 character strings | | NEC DT700 |
| 13 | SNMP Server IP Address | Code number 0 ~ 255 | | 69 (Fixed) |
| | | IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | | 0.0.0.0 |
| 14 | POP3 Server IP Address | Code number 0 ~ 255 | | 70 (Fixed) |
| | | IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | | 0.0.0.0 |
| 16 | SIP Server (IP Address) | Code number 0 ~ 255 | | 120 (Fixed) |
| | | IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | | 172.16.0.10 |
| 17 | SIP Server (Domain Name) | Code number 0 ~ 255 | If there is setting in 10-16-16 this setting will be ignored | 120 (Fixed) |
| | | Maximum 20 character strings | | No Setting |
| 18 | FTP Server IP Address | Code number 0 ~ 255 | | 141 (Fixed) |
| | | IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | | 0.0.0.0 |
| 19 | Config File Name | Code number 0 ~ 255 | | 151 (Fixed) |
| | | Maximum 15 character strings | | No Setting |
| 20 | LDS Server 1 IP Address | Code number 0 ~ 255 | | 162 (Fixed) |
| | | IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | | 0.0.0.0 |
| 21 | LDS Server 2 IP Address | Code number 0 ~ 255 | | 162 (Fixed) |
| | | IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | | 0.0.0.0 |
| 22 | LDS Server 3 IP Address | Code number 0 ~ 255 | | 162 (Fixed) |
| | | IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | | 0.0.0.0 |
| 23 | LDS Server 4 IP Address | Code number 0 ~ 255 | | 162 (Fixed) |
| | | IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | | 0.0.0.0 |
| 24 | Next Server IP Address | IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 | | 0.0.0.0 |

| Item No. | Item | Input Data | Description | Default |
|----------|-------------------------|-------------------------|-------------|-------------|
| 27 | SIP Server Receive Port | Code number 0 ~ 255 | | 168 (Fixed) |
| | | Port: 1 ~ 65535 | | 5080 |
| 28 | Configuration File Name | Code number 0 ~ 255 | | 43 (Fixed) |
| | | Sub Code number 0 ~ 255 | | 152 |
| | | Up to 15 characters | | No Setting |
| 29 | Configuration File Name | Code number 0 ~ 255 | | 43 (Fixed) |
| | | Sub Code number 0 ~ 255 | | 153 |
| | | Up to 15 characters | | No Setting |

Conditions

None

Feature Cross Reference

None

Program 10 : System Configuration Setup

10-19 : VoIP DSP Resource Selection

Level:

IN

Description

Use **Program 10-19 : VoIP DSP Resource Selection** to define the criteria for each DSP resource on the VoIPDB.

Input Data

| | |
|---------------------|---------|
| Slot Number | 0 |
| DSP Resource Number | 1 ~ 128 |

| Item No. | Item | Input Data | Description | Default |
|----------|------------------------------------|---|--|----------------------|
| 01 | VoIP DSP Resource Selection | 0 = Common use for both IP extensions and trunks 1 = Use for IP extensions 2 = Use for SIP trunks 3 = Use for Networking 5 = Blocked 6 = Common without unicast paging 7 = Multicast paging 8 = Unicast paging | This program is used to assign the 128 resources of the VoIPDB. NEC recommends leaving the default settings as system default. | Resource 1 ~ 128 = 0 |

Conditions

None

Feature Cross Reference

None

Program 10 : System Configuration Setup

10-54 : License Configuration for Each Package

Level:

IN

Description

Use **Program 10-54 : License Configuration for Each Package** to set the license information for each board.

Input Data

| | |
|-------------|--------|
| Slot Number | 0 ~ 12 |
|-------------|--------|

| | |
|----------------------|--------|
| License Index Number | 1 ~ 32 |
|----------------------|--------|

| Item No. | Item | Input Data | Default |
|----------|------------------|-------------|------------|
| 01 | License Code | 0000 ~ 9999 | No Setting |
| 02 | License Quantity | 0 ~ 128 | 0 |

Conditions

None

Feature Cross Reference

None

Program 10 : System Configuration Setup

10-62 : NetBIOS Setting


Level:

IN

Description

Use **Program 10-62 : NetBIOS Setting** to set the data of NetBIOS.

Input Data

| Item No. | Item | Input Data | Description | Default |
|----------|--------------|-----------------------------|---|---------|
| 01 | NetBIOS Mode | 0 = Disabled 1 = Enabled | | 1 |
| 02 | NetBIOS Name | Maximum 15 characters |  Please avoid using Space between the words. Also when you create name please use all upper letters. | SL2100 |

Conditions

None

Feature Cross Reference

None

Program 10 : System Configuration Setup

10-63 : DHCP Client Setting

Level:

IN

Description

Use **Program 10-63 : DHCP Client Setting** to set the data of DHCP Client.

Input Data

| Item No. | Item | Input Data | Description | Default |
|----------|------------------|-----------------------------|--|---------|
| 01 | DHCP Client Mode | 0 = Disabled 1 = Enabled | If you are using IP phones/trunks it is recommended to not use the DHCP client function, a static IP address would be preferred. If you are going to still use the DHCP client function then the DHCP server should be setup so that the same IP address is always provided to the system. | 0 |

Conditions

None

Feature Cross Reference

None

Program 15 : Extension, Basic Setup

15-05: IP Telephone Terminal Basic Data Setup

Level:

IN

Description

Use **Program 15-05 : IP Telephone Terminal Basic Data Setup** to set up the basic settings for an IP telephone.

Input Data

| | |
|------------------|----------------------|
| Extension Number | Maximum eight digits |
|------------------|----------------------|

| Item No. | Item | Input Data | Description | Default | Related Program |
|----------|--------------------------------|---|--|-------------------|----------------------------|
| 01 | Terminal Type | 1 = H.323 2 = SIP 3 = None 4 = IP MLT 5 = WebRTC (V1.5 added) | | 3 | |
| 02 | IP Phone Fixed Port Assignment | MAC address 00-00-00-00-00-00 ~ FF-FF-FF-FF-FF-FF | MAC Address of registered SIP MLT phone is stored and/or can input the MAC address of an SIP MLT phone so when it comes online it is provided with the extension in which the MAC address matches. | 00-00-00-00-00-00 | 15-05-01 |
| 04 | Nickname | Up to 48 characters | Nickname section on Invite message. Example : Extension 100 has a Nickname set to PAUL. Extension 101 has command 15-05-17 set to Nickname. The inbound call to extension 101, from 100, shows PAUL. Nickname must be unique in the system. | No Setting | 15-05-17 |
| 07 | Using IP Address | 0.0.0.0 ~ 255.255.255.255 | | - | 15-05-01 |
| 09 | Call procedure port | 0 ~ 65535 | | - | 15-05-01 |
| 11 | IP MLT C/CTR Port | 0 ~ 65535 | | - | |
| 15 | CODEC Type | 1 = Type 1 2 = Type 2 3 = Type 3 4 = Type 4 5 = Type 5 | Assign the CODEC Type of the SIP MLT. | 1 | 84-24 84-11 15-05-01 |
| 16 | Authentication Password | Up to 24 characters | Assign the authentication password for SIP single line telephones. Web pro indicates password as * mark. | Random | 15-05-01 |

| Item No. | Item | Input Data | Description | Default | Related Program |
|----------|---|---|---|-------------|--|
| 18 | IP Duplication Allowed Group | 0 = Disable 1 = Enable | When this program is 1: Enable, the duplication of an IP address is allowed at the time of SIP/IP Multiline terminal registration. This PRG is valid, when PRG15-05-01 is SIP or IP Multiline Terminal. | 0 | 15-05-01 |
| 19 | Side Option Information | 0 = No Option 1 ~ 3 = Not used 4 = EHS | | 0 | |
| 20 | Bottom Option Information | 0 = No Option 1 ~ 2 = Not Used 3 = WHA 4 = BCA | Shows the type of adapter installed. | 0 | 10-03-10 |
| 26 | IP MLT Terminal Type | 0 = Not Set 1 = Not Used 2 = ITL-**-1D/ ITL-12BT-1D/ ITL-12PA-1D (without 8LKI (LCD) -L) 3 = UT880 4 = Not Used 5 = Softphone 6 = CTI 7 ~ 8 = Not Used 9 = IP4WW-24TIXH 10 ~ 12 = Not Used 13 = ITZ-**-D 14 = ITZ-**-CG 15 = ITZ-**-LDG/ITZ-**-LD 16 = ITY-6D 17 = ITY-8LDX 18 = IP7WW-8IPLD-C1 | | 0 | |
| 27 | Personal ID Index | 0 ~ 128 | When the SIP Multiline telephone is using manual/auto registration, assign each phone a unique personal index. Then go to command 84-22 to assign the user name and password. | 0 | 84-22 |
| 28 | Addition Information Setup | 0 = Do not inform 1 = Inform | Select whether to inform of additional information or not. | 0 | 15-01-01 15-02-13 15-02-15 15-02-34 |
| 29 | Terminal WAN-side IP Address | 0.0.0.0 ~ 255.255.255.255 | | 0.0.0.0 | |
| 30 | DTMF Play during Conversation at Receive Extension | 0 = Do Not Play 1 = Play | | 0 | |
| 31 | Alarm Tone during Conversation (RTP packet loss alarm) | 0 = Off 1 = On | | 1 | |
| 33 | LAN Side IP Address of Terminal | 0.0.0.0 ~ 255.255.255.255 | | 0.0.0.0. | |
| 35 | Encryption Mode On/Off | 0 = Off 1 = On | | 0 | |
| 36 | IP MLT Firmware Version | 00.00.00.00 ~ ff.ff.ff.ff | Indicate a current firmware Version. | 00.00.00.00 | |
| 38 | Paging Protocol Mode | 0 = Multicast 1 = Unicast 2 = Auto | Sets the protocol mode for the Paging function. | 0 | |
| 39 | CTI Override Mode | 0 = Disable 1 = Enable | | 0 | |

| Item No. | Item | Input Data | Description | Default | Related Program |
|----------|--|---|---|---------|-----------------|
| 40 | Calling name display info via trunk for standard SIP | 0 = Both name and number 1 = Name only 2 = Number only 3 = None | * Trunk name is the first priority and abbreviated (SPD) name is second priority | 0 | |
| 41 | Time Zone(hour) | 0 ~ 24 (- 12 ~ + 12 hour) | | 12 | |
| 42 | Time Zone(minute) | 0 ~ 120 (- 60 ~ + 60 minute) | | 60 | |
| 43 | Video Mode | 0 = Disable 1 = Enable | | 0 | |
| 44 | Using STD-SIP Display for CPN | 0 = Disable 1 = Enable | | 0 | |
| 45 | NAT plug & play | 0 = OFF 1 = ON | Effect this program only when PRG 10-46-14 (for IP Multiline Terminals) or PRG 10-33-05 (for Standard SIP Terminal) are set to NAT mode. Effect this program only when PRG 10-46-14 (for IP Multiline Terminals) is set to NAT mode. If you are setting the SIP Terminal using remote router by turning this setting ON you don't need to set the Port Forward on remote router side. Select sending RTP port number to remote Router, use from negotiation result (0) or received RTP packet (1) | 1 | 10-46-14 |
| 46 | Door Phone Number | 0 = Not door phone 1 ~ 6 = Door phone number | | 0 | |
| 47 | Registration Expire Timer for NAT | 0 = Disable 60 ~ 65535 (sec) | On a per station basis, this setting defines the SIP registration expiry timer. This setting applies to IP Multiline Terminals or Standard SIP Terminal connected via NAPT. If this value is set to 0, for a NAPT terminal, the value in PRG 84-23-01 (for IP Multiline Terminals) or PRG 10-33-01 (for Standard SIP Terminal) are applied. | 180 | 84-23-01 |
| 48 | Subscriber Expire Timer for NAT | 0 = Disable 60 ~ 65535 (sec) | On a per station basis, this setting defines the SIP subscribe expiry timer. This setting only applies to IP Multiline Terminals or Standard SIP Terminal connected via NAPT. If this value is set to 0, for a NAPT terminal, the value in PRG 84-23-02 (for IP Multiline Terminals) or PRG 10-33-01 (for Standard SIP Terminal) are applied. | 180 | 84-23-02 |
| 49 | Receiving SIP INFO | 0 = Disable 1 = Allowed any time 2 = Allowed while RTP is not available | Enables to receive a SIP INFO message from a standard SIP phone as a dial information. | 1 | |
| 50 | Peer to Peer Mode | 0 = Off 1 = On | | 1 | |
| 51 | Transport Protocol | 0 = UDP 1 = TCP 2 = TLS | This program is the transport protocol. (Read only) | 0 | |

Conditions

None

Feature Cross Reference

None

Program 84 : Hardware Setup for VoIPDB

84-09 : VLAN Setup

Level:

IN

Description

Use **Program 84-09 : VLAN Setup** to set up the VLAN data.

Input Data

| | | | |
|------------------|--|-------|--|
| Interface Number | | 1 ~ 2 | |
|------------------|--|-------|--|

| Item No. | Item | Input Data | Default |
|----------|----------|--------------------------------------|---------|
| 01 | VLAN | 0 = Disable (Off) 1 = Enable (On) | 0 |
| 02 | VLAN ID | 0 ~ 4094 | 0 |
| 03 | Priority | 0 ~ 7 | 0 |



Interface No.2: The packets send from LAN I/F on VoIPDB is set the VLAN Tag (VoIPDB 32ch/64ch/128ch).

Conditions

- System programming must be exited before these program options take affect.

Feature Cross Reference

None

Program 84 : Hardware Setup for VoIPDB

84-10 : ToS Setup

Level:

IN

Description

Use **Program 84-10 : ToS Setup** to set up the Type of Service data.

Input Data

| | |
|---------------|--|
| Protocol Type | 1 = Not Used 2 = Not Used 3 = Reserved 4 = H.323 5 = RTP/RTCP 6 = SIP 7 = Reserved 8 = SIP MLT 9 = SIP Trunk 10 = Reserved 11 = Reserved |
|---------------|--|

| Item No. | Item | Input Data | Description | Default |
|----------|--|--|---|---------|
| 01 | ToS Mode | 0 = Disable (Invalid) 1 = IP Precedence 2 = Diffserv | When Input Data is set to 1, Item No. 07 is invalid. When Data is set to 2, Item No. 02 ~ 05 are invalid. | 0 |
| 02 | Priority, IP Precedence | 0 ~ 7 0 = Low 7 = High | 1 = Router queuing priority | 0 |
| 03 | Low Delay | 0 ~ 1 0 = Normal Delay 1 = Low Delay | 1 = Optimize for low delay routing | 0 |
| 04 | Wideband (Throughout) | 0 ~ 1 0 = Normal Throughput 1 = High Throughput | 1 = Optimize for high bandwidth routing | 0 |
| 05 | High Reliability | 0 ~ 1 0 = Normal Reliability 1 = Low Reliability | 1 = Optimize for reliability routing | 0 |
| 07 | Priority (D.S.C.P. - Differentiated Services Code Point) | 0 ~ 63 | DSCP (Differentiated Services Code Point) | 0 |

Conditions

- The system must be reset for these program options to take affect.

Feature Cross Reference

None

Program 84 : Hardware Setup for VoIPDB

84-33 : FAX over IP Setup

Level:

IN

Description


This program sets the parameter of FAX over IP Function.

Input Data

| | |
|------|--|
| Type | 1 = H323 Trunk 2 = Networking 3 = SIP Trunk 4 = SIP Extension 5 = Reserved 6 = Reserved |
|------|--|

| | |
|---------|-------|
| Profile | 1 ~ 6 |
|---------|-------|

| Item No. | Item | Input Data | Description | Default |
|----------|---|---|---|---------|
| 01 | FAX Relay Mode | 0 = Disable 1 = Enable 2 = Each Port Mode | | 0 |
| 02 | T.38 Protocol mode | 0 = R/U 1 = U/R 2 = RTP 3 = UDPTL | | 1 |
| 04 | Maximum Jitter Buffer | 0 ~ 300 ms (for VoIPDB mode) 10 ~ 300 ms* (for Built-In VoIP mode) | | 160 |
| 05 | T.38 RTP Format Payload Number | 96 ~ 127 | | 100 |
| 06 | T.38 FAX Max Speed | 0 = V.27ter; 4800 bps 1 = V.29; 9600 bps 2 = V.17; 14400 bps | | 2 |
| 07 | T.38 Data Error Correction Mode | 0 = Redundancy 1 = FEC | | 0 |
| 08 | T.38 Error protection depth for Signaling | 0 ~ 2 | This item is only available when you use VoIPDB. It is unavailable when you use Built-in VoIP, which depends on the data of Item No. 09. | 0 |
| 09 | T.38 Error protection depth for Data | 0 ~ 2 | | 0 |
| 10 | T.38 TCF Method | 1 = VOIPDB 2 = G3FE | | 1 |
| 11 | T.38 ECM(Error Correction Mode) Mode | 0 = Disable 1 = Enable | | 1 |

| Item No. | Item | Input Data | Description | Default |
|----------|-----------------------|---|--|---------|
| 12 | FAX Codec | 1 = G.711 a-law 2 = G.711 u-law 3 = G726 | | 1 |
| 13 | Payload Size | 1 ~ 4 (10ms base) | | 2 |
| 14 | Jitter Buffer Mode | 1 = Static 2 = Self adjusting | | 1 |
| 15 | Minimum Jitter Buffer | 0 ~ 300 ms (for VoIPDB mode) | | 80 |
| 16 | Average Jitter Buffer | 10 ~ 300 ms* (for Built-In VoIP mode) | | 120 |
| 17 | Maximum Jitter Buffer | | | 160 |
| 18 | FAX RTP Payload Type | 96 ~ 127 (Not Sip Trunk) 0, 2, 8, 96 ~ 127 (SIP Trunk) |  <p>When the G,711 a-law is selected in Item 12, it is possible to set "8" in Item 18</p> <p>When the G,711 u-law is selected in Item 12, it is possible to set "0" in Item 18</p> <p>When the G,726 is selected in Item 12, it is possible to set "2" in Item 18</p> | 103 |
| 19 | FAX over IP Type | 0 = Type 1 1 = Type 2 | SIP Trunk : Type 1 : T.38 Type 2 : Path through | 0 |

Conditions

None

Feature Cross Reference

None

Program 90 : Maintenance Program

90-23 : Deleting Registration of IP Telephones

Level:

IN

Description



This program is available via telephone programming and not through PC Programming.

Use **Program 90-23 : Deleting Registration of IP Telephones** to delete the registered IP telephone from the system.

Input Data

| Extension Number | | Up to eight digits | | |
|------------------|---------------------|--------------------|---|---------|
| Item No. | Item | Input Data | Description | Default |
| 01 | Delete IP Telephone | 1→Hold | This assignment removes the station number association with the MAC address of the IP station. [Del.IP-Phone] : Dial 1 + press Hold (Press Hold only to cancel.) | - |

Conditions

None

Feature Cross Reference

None

Program 90 : Maintenance Program

90-34 : Firmware Information

Level:

IN

Description

Use **Program 90-34 : Firmware Information** to list the package type and firmware boards installed in the system.

Input Data

| | |
|-------------|---------|
| Slot Number | 00 ~ 12 |
|-------------|---------|

| Item No. | Item | Input Data | Default |
|----------|--------------------------|--|---------|
| 01 | Package Name | PKG Name (Up to 15 characters) | - |
| 02 | Firmware Version Number | 00.00 ~ 15.15 | - |
| 03 | VOIPDB Software Version | DEV/PR/REL - 00.00.00.00.00.00 DEV/PR/REL - FF.FF.FF.FF.FF.FF | - |
| 04 | DSP Project Number | 00000000 - FFFFFFFF | - |
| 05 | Vocoder Firmware Version | 00.00.00.00 - FF.FF.FF.FF | - |
| 06 | OCT1010ID Version | 00.00.00.00 - FF.FF.FF.FF | - |

Conditions

None

Feature Cross Reference

None

MEMO

Network Design Considerations

Section 1 INTRODUCTION

This chapter explains some issues that should be considered when planning a NEC SL2100 VoIP installation. This is a generalized explanation and therefore does not discuss vendor-specific issues and solutions. Typically, different solutions are implemented by different manufacturers.

Section 2 QoS

Quality of Service (QoS) is one of the most important factors for VoIP. This refers to the perceived quality of speech and the methods used to provide good quality speech transmission. Several factors that affect speech quality and several mechanisms can be used to ensure QoS.

This chapter also describes the problems that can occur and some possible solutions. Each network equipment manufacturer (NEC, 3Com, Cisco, etc.) has slightly different methods of implementing QoS and these are not discussed in this document. This chapter provides an overview to classify voice traffic on the NEC SL2100 so that the network equipment can impose QoS.

2.1 QoS Definitions

This section lists common definitions used with QoS for VoIP.

Latency (Delay):

If at any point the usage on the network exceeds the available bandwidth, the user experiences delay, also called latency. In more traditional uses of an IP data network, the applications can deal with this latency. If a person is waiting for a web page to download, they can accept a certain amount of wait time. This is not so for voice traffic. Voice is a real time application, which is sensitive to latency. If the end-to-end voice latency becomes too long (150ms, for example), the call quality is usually considered poor. It is also important to remember that packets can get lost. IP is a best effort networking protocol. This means the network tries to get the information there, but there is no guarantee.

Delay is the time required for a signal to traverse the network. In a telephony context, end-to-end delay is the time required for a signal generated at the talker's mouth to reach the listener's ear. Therefore, end-to-end delay is the sum of all the delays at the different network devices and across the network links through which voice traffic passes. Many factors contribute to end-to-end delay, which are covered next.

The buffering, queuing, and switching or routing delay of IP routers primarily determines IP network delay. Specifically, IP network delay is comprised of the following:

- **Packet Capture Delay**
Packet capture delay is the time required to receive the entire packet before processing and forwarding it through the router. This delay is determined by the packet length and transmission speed. Using short packets over high-speed networks can easily shorten the delay but potentially decrease network efficiency.

- **Switching/Routing Delay**

Switching/routing delay is the time the router takes to switch the packet. This time is needed to analyze the packet header, check the routing table, and route the packet to the output port. This delay depends on the architecture of the switches/routers and the size of the routing table.

- **Queuing Time**

Due to the statistical multiplexing nature of IP networks and to the asynchronous nature of packet arrivals, some queuing, thus delay, is required at the input and output ports of a packet switch. This delay is a function of the traffic load on a packet switch, the length of the packets and the statistical distribution over the ports. Designing very large router and link capacities can reduce but not completely eliminate this delay.

Jitter

Delay variation is the difference in delay exhibited by different packets that are part of the same traffic flow. High frequency delay variation is known as jitter. Jitter is caused primarily by differences in queue wait times for consecutive packets in a flow, and is the most significant issue for QoS. Certain traffic types, especially real-time traffic such as voice, are very intolerant of jitter. Differences in packet arrival times cause choppiness in the voice.

All transport systems exhibit some jitter. As long as jitter falls within defined tolerances, it does not impact service quality. Excessive jitter can be overcome by buffering, but this increases delay, which can cause other problems. With intelligent discard mechanisms, IP telephony/VoIP systems try to synchronize a communication flow by selective packet discard, in an effort to avoid the walkie-talkie phenomenon caused when two sides of a conversation have significant latency. NEC SL2100 incorporates a Jitter Buffer to avoid these problems.

Packet Loss

During a voice transmission, loss of multiple bits or packets of stream may cause an audible pop that can become annoying to the user. In a data transmission, loss of a single bit or multiple packets of information is almost never noticed by users. If packet drops become epidemic, the quality of all transmissions degrades. Packet loss rate must be less than five percent for minimum quality and less than one percent for toll quality.

2.2 Voice Quality Improvements

This section describes various techniques that can be used to improve the voice quality.

- **Increase available bandwidth:**

This can sometimes be the most basic solution and the easiest of the solutions. If running a System IP Phone using G.711 with a 30ms fill time over Ethernet, for only one call, 90Kbps of bandwidth is needed. If that same user only has a 64K line, they do not have a decent IP voice call. The user can increase the available bandwidth to slightly exceed the 90Kbps requirements and their voice quality dramatically increases. This solution might not be viable if no more bandwidth is available.

- **Use a different CODEC:**

The CODEC contains possible compression algorithms to be used on the voice. Let's take the example above again. The user only wants one voice line over a 64Kbps data connection. They also want to maintain their current fill time of 30ms. Change to a G.729. For one line, only 34Kbps is required for a call. This fits well within the 64Kbps of available bandwidth.

- **Increase the number of frames per packet:**

To continue with the example above, the user has moved to a G.729 CODEC. But now, the user wishes to add two more System IP Phones. Their current 64Kbps line can handle one call, because it is only 34Kbps. Two more System IP Phones would increase the total to 102Kbps so obviously there is not sufficient bandwidth.

The user can now increase the fill time to 50ms. This reduces the bandwidth per call to 19.8Kbps ($3 \times 19.8 = 59.4\text{Kbps}$). The savings in bandwidth comes from the fact that with a longer fill time, fewer packets are needed to send the voice. With fewer packets, less header information needs to be attached and transmitted.

- Change Layer 2 Protocols:

Ethernet is most commonly used for IP packets. Unfortunately, Ethernet has a fairly large overhead of 34 bytes. So every IP voice packet going over Ethernet has a 34-byte Ethernet header attached to it. As the number of packets add up, this header data can become significant. Frame Relay has a 7-byte header and Point-to-Point Protocol (PPP) has a 6-byte header. With this decrease in header length at layer 2, some significant savings in bandwidth use can be achieved.

The down side to this is that most networks may not have these services available, where Ethernet is very widely used. This is usually outside the control of the installer and therefore NEC strongly advises users to do more research on other layer 2 protocols before trying to implement them in their voice network.

- Implement Quality of Service (QoS):

Now, assume a derivative of the above example. The user needs only one voice line over their 64Kbps connection. They are using G.729 with a 30ms fill time. This requires 34Kbps of their available bandwidth. Also assume that this line is used at certain times of the day for data connectivity. This data connectivity is very light, only 20Kbps or so during most of the day, but does spike to 50Kbps during certain points of the day. This data is not time sensitive like the voice data, so if necessary it could be forced to wait.

Therefore, the user can implement a Quality of Service mechanism on the IP network. At its most basic form, this denotes certain IP packets as being more important than others. So they would tell this 64Kbps line that IP packets with voice deserve a higher priority than those without voice. This allows the network devices to give priority to the other data, so the quality of the call is not compromised.

2.3 Types of Classifications for Traffic for QoS

Classification uses information from a packet (or frame) to define the type of data and therefore how the data should be handled for QoS on the network. Using packet classification, you can partition network traffic into multiple priority levels or Types of Service (ToS). NEC SL2100 supports methods of marking a packet with its classification information in the Layer 2 or 3 headers.

- VLAN (802.1Q):

Virtual LANs work at Layer 2 of the OSI model and can be equated to a broadcast domain. More specifically, VLANs can be seen as a group of end stations, perhaps on multiple physical LAN segments that are not constrained by their physical location and therefore, communicate as if they were on a common LAN. Packets can be marked as important by using layer 2 classes of service (CoS) settings in the User Priority bits of the 802.1Q header. Refer to Program [84-09 : VLAN Setup](#) for information for VLAN configuration.

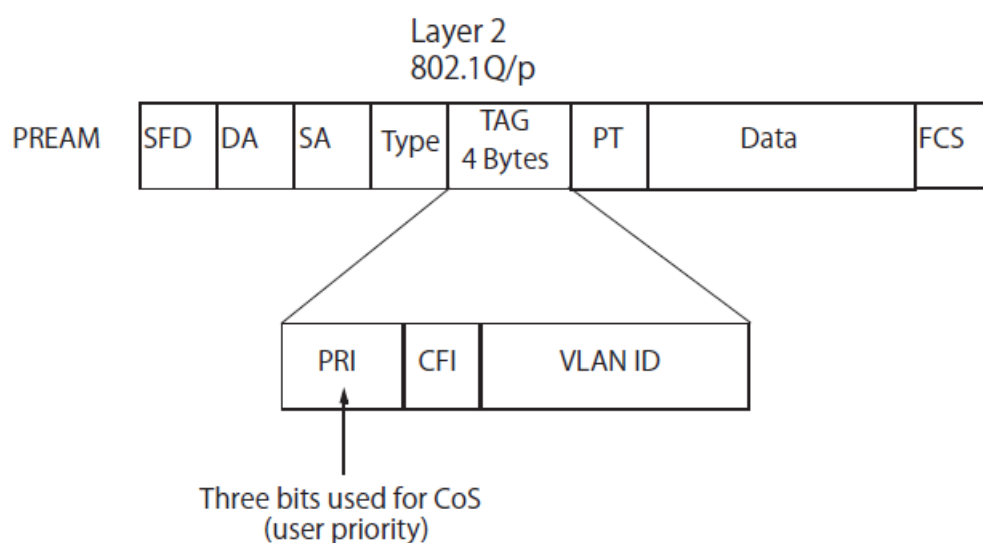


Figure 6-1 Layer 2 Diagram (802.1Q)

- **IP Precedence - Layer 3 QoS:**
Allows you to specify the class of service for a packet. You use the three precedence bits in the IPv4 header type of service (ToS) field for this purpose. Using the ToS bits, you can define up to six classes of service. Other devices configured throughout the network can then use these bits to determine how to treat the packet in regard to the type of service to grant it. These other QoS features can assign appropriate traffic-handling policies including congestion management and bandwidth allocation. By setting IP Precedence levels on incoming traffic and using them in combination with QoS queuing features, you can create differentiated service. (Refer to Program [84-10 : ToS Setup](#) for detailed programming information.)
- **Differentiated service (Diffserv) - Layer 3 QoS:**
Provides services differentiated on performance using weighted priority queuing. DiffServ requires that edge routers classify traffic flows into a member from a set of categories based on the TCP/IP header fields in what is called a micro flow. Because the Diffserv is present in every packet header, each node can provide differentiated services per-hop. Refer to Program [84-10 : ToS Setup](#) for detailed programming information.

Section 3 INTERNET BASED CONNECTIONS (XDSL, CABLE, ETC.)

Internet-based connections are becoming increasingly popular. This is mainly due to the speed and cost of xDSL and cable modem connections. For data applications, these types of connection are generally acceptable. For Voice over IP applications several issues should be taken into consideration.

Asymmetric Data Rates

On many Internet based connections, there are different data rates for upstream and downstream. For example 1Mbps down and 256Kbps up. This works well for Internet access, as generally you download files from the Internet to your PC and transmit less information in the other direction. For VoIP, speech uses the same amount of bandwidth in both directions, which means that the amount of simultaneous calls can not exceed the amount of “upstream” bandwidth available.

Contention

Most Internet based connections specify a contention ratio. This is typically 50:1 for home users or 20:1 for business users. This specifies the number of users subscribed to a single connection to the Internet Service Provider (ISP). This indicates how many users share the bandwidth with other users on the Internet, which means that the speeds that you are quoted are not necessarily accurate – you receive less than these figures.



It is unlikely that all subscribers are using a connection at the same time, so these figures are not quite as bad as they first seem.

Network Address Translation (NAT)

Usually, the equipment that your ISP provides (cable modem, ADSL router, etc.) uses Network Address Translation. This allows several devices to share one public IP address. The issues relating to the use of NAT are outlined in Firewalls and NAT below.

VPN

Due to the use of NAT, and non-routable IP addressing, it may be necessary to implement a VPN solution. This is outlined in VPN Tunneling below. (Refer to [Virtual Private Network \(VPN\) Tunneling](#).)

QoS

As discussed earlier, it is essential to have some form of Quality of Service implemented. With Internet based connections, we are not in control of the many routers, switches and other network hardware

that reside between our two VoIP endpoints. This means that we cannot specify any QoS parameter on these devices.

The only point where the QoS can be controlled is at the VPN or firewall. This allows VoIP traffic to be prioritized over any other data that is sent out to the Internet. This helps to maintain reasonable quality speech – but once the data has exited the local router/cable modem it is at the mercy of the Internet.

When implementing NEC SL2100 IP over Internet based connections it is very important that these factors are considered, and that the customer is made aware that neither the installer nor NEC are held responsible for any quality issues experienced.

Section 4 FIREWALLS AND NAT

The ways in which networks are designed to be secure (firewall, VPN services, proxy servers, etc.) and integration of NAT create problems for VoIP. This is due in part, to the endless number of different scenarios for non-real time protocols and their limited solutions.

4.1 Understanding the Infrastructure

The networks in place today look very different than the networks of yesterday. In the past, only computers and servers were connected to the network. The network was built to be as a best effort delivery mechanism, where delay and lost of information between devices was something we dealt with. Today, there is an over saturation of devices needing to gain access to the IP network. Desktop computers, fax machines, wireless PDAs, Servers, home appliances, video servers and now VoIP terminals all are fighting for bandwidth, precedence, and addresses on this converged network.

It is necessary to create some kind of Intranet environment (across the Internet), with fixed network characteristics, where VoIP solutions can tolerate some minor variations. IT personnel have been tasked with implementing different mechanisms in the network to support the new demands required on the converged network. Some solutions that have been implemented are:

- QoS devices to support precedence settings of voice packets.
- Elimination of hubs in place of switches to support 100Mbps full-duplex transmission.
- Firewall integration to protect the internal network from external attack.
- Network Address Translation (NAT) devices are widely deployed to support the addressing issues.
- Virtual Private Network (VPN) Servers were added to Enterprise networks to support the security and connectivity issues for remote users.

Some solutions, such as the hub replacement and integration of QoS, are done behind the scenes and should have no effect on the voice application. Other solutions such as NAT and Firewall cause major disturbance to VoIP.

4.2 Firewall Integration

Network security is always a concern when connecting the Local Area Network (LAN) to the Wide Area Network (WAN). There are many ways to integrate security in the network – the most popular are Firewalls and Proxy servers.

- **Firewalls**
Firewalls can be implemented in both hardware and software, or a combination of both. Firewalls are frequently used to prevent unauthorized Internet users from accessing private networks connected to the Internet, especially Intranets. All messages entering or leaving the Intranet pass through the firewall, which examines each message and blocks those that do not meet the specified security criteria.
- **Proxy Server**
Proxy server intercepts all messages entering and leaving the network. The proxy server effectively hides the true network address.

What should be noted is that no matter which security measure is implemented, the VoIP must have TCP/UDP ports open in the security wall (e.g., firewall/proxy) for the media and control streams to flow. If any point in the network prevents the ports from flowing from end-to-end, the VoIP application does not work.

The ports that need to be open on the firewall/proxy vary depending on the particular application being used. A list of these ports is shown below, however it should be noted that the preferred solution would be to allow all ports on the NEC SL2100 device to be open, or to place the NEC SL2100 outside of the firewall.

| Applications | Rx Port | NEC SL2100 Programming |
|--|--------------|------------------------|
| PC Programming | 8000 | 90-54-02 |
| Web Programming | 80 | 90-54-01 |
| DHCP Server | 67 | |
| SIP MLT Listening Port | 5080 5081 | 10-46-06 10-46-13 |
| SIP Trunk Listening Port | 5060 | 10-29-04 |
| SIP Single Line Stations | 5070 | 84-20-01 |
| Realtime Transport Protocol | 10020~10146 | 84-26-02 |
| Realtime Transport Control Protocol (RTCP) | 10021~10147 | 84-26-03 |

4.3 Virtual Private Network (VPN) Tunnelling

A Virtual Private Network is a private data network that maintains privacy through using a tunneling protocol and security procedures. Allowing for remote networks (including VoIP devices), which reside behind NATs and/or Firewalls to communicate freely with each other.

The idea of the VPN is to connect multiple networks together using public (i.e., Internet) based connections. This type of connection is ideal for those commuters, home workers, or small branch offices needing connectivity into the corporate backbone. It is possible to connect these remote networks together using private links (such as leased lines, ISDN, etc.) but this can be very expensive and there is now a high demand for low cost Internet connectivity.

Companies today are exploring the use of VPN for a variety of connectivity solutions, such as:

- Remote User to Corporate Site VPN
Allows employees to use their local ISP fastest connection such as cable modems, DSL, and ISDN. For traveling users, all they need to do is dial into their ISP local phone number.
- Site-to-site VPN
Allows companies to make use of the Internet for the branch-to-branch connections, cutting the cost of the expensive point to point leased line service.
- Extranet
Extranet describes one application using VPN technology. The concept allows a company and a vendor/supplier to access network resources at each site. For example, a customer may have access to a suppliers intranet for access to product information.

VPNs can be implemented in hardware or software. Single users, such as traveling sales personnel, may have a software based VPN client on their laptop computer. This connects back to the Head Office VPN server. For larger sites, the VPN is typically implemented using a hardware VPN – this is often incorporated in to a firewall solution.

The diagram below is example of how a VPN tunnel may be implemented. The red lines in the diagram show the tunnels that are created through the Internet. Each network can connect to the others as though they are connected with private connections (kilostream, etc.), without the issues relating to NAT.

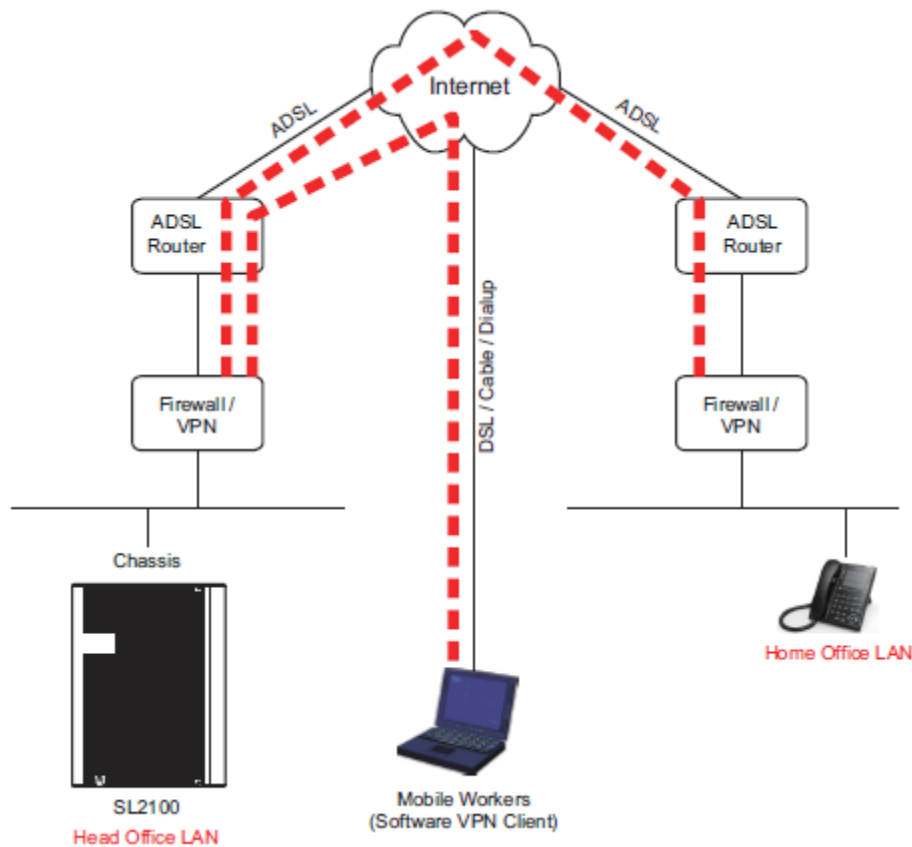



Figure 6-2 Virtual Private Network (VPN) Example

When IP address translation is applied to a VoIP packet, the application fails and the communication path is broken. VoIP packets contain the IP address information and the ports used as part of its payload. When NAT is applied, only the header parameter is changed, not the payload data that affects the process of data packets within the VoIP switch and terminal.

The common scenario for remote IP deployment is:

- Implementation of an IP Phone with a public IP address talking with an NEC SL2100 behind NAT. An example would be a telecommuter.
Implementation of an IP Phone behind a NAT, which connects to the Internet, terminates in a NEC SL2100 behind a different NAT.
When selecting VPN equipment it is important to consider Quality of Service. Generally, VPN hardware is connected to Internet connections which are unreliable and out of the control of the customer. However, it is possible to set prioritization on some VPN units for voice traffic. This does not solve the unreliability of the Internet, but helps to ensure that the data traffic to and from the LAN do not impair the quality of the voice traffic. (Refer to [QoS](#)).

 *NEC strongly recommends that any VPN hardware used for VoIP has the facility to prioritize voice traffic.*

Section 5 CODEC AND BANDWIDTH

This section describes CODEC and bandwidth and their application with the NEC SL2100 system.

5.1 CODECS

CODEC (COder/DECoder) uses the technology of encoding and decoding a signal. For VoIP, this specifically refers to the algorithm used to convert analog speech to digital data for transmission on an IP network.

The NEC SL2100 system supports three different CODECs:

- G.711
This is the ITU-T recommendation for coding of speech at 64kbps using PCM (pulse code modulation). This CODEC is often described as uncompressed as it uses the same sampling rate as Time-Division Multiplexing (TDM). G.711 has a MOS^{**1} score of 4.2 but uses a large bandwidth for transmission. This CODEC is not commonly used due to the bandwidth required, although it can be acceptable in LAN environment (i.e., IP Phones connected over a 100Mbps LAN).
- G.722
G.722 is an ITU standard CODEC that provides 7kHz wideband audio at data rates from 48 to 64kbps. This is useful in a fixed network Voice Over IP applications, where the required bandwidth is typically not prohibitive, and offers a significant improvement in speech quality over older narrowband CODECs such as G.711, without an excessive increase in implementation complexity.
- G.726
G.726 is an ITU-T ADPCM speech CODEC standard covering voice transmission at rates of 16, 24, 32, and 40kbit/s. It was introduced to supersede both G.721, which covered ADPCM at 32kbit/s, and G.723, which described ADPCM for 24 and 40kbit/s. G.726 also introduced a new 16kbit/s rate. The four bit rates associated with G.726 are often referred to by the bit size of a sample as 2-bits, 3-bits, 4-bits, and 5-bits respectively.
- G.729A
This ITU-T recommendation describes the algorithm for coding of speech signals at 8kbps using CS-ACELP (conjugate-structure algebraic code-excited linear prediction). This CODEC samples the analog signal at 8000Hz and uses a frame size of 10ms. This CODEC has a MOS score of 4.0. G.729 is the most commonly used CODEC for NEC SL2100 VoIP installations. This is due to the fact that it offers high compression (and therefore low bandwidth) while maintaining good speech quality.

Packet Size:

Each CODEC has a set frame length. This is the time that the frame encapsulates. For G.729 and G.711 the frame length is 10ms. It is possible to configure the packet size in the NEC SL2100 programming. To do this, we tell the NEC SL2100 how many frames to encapsulate into each packet for transmission.

For example, the G.729 has a frame length of 10ms - the packet size is set to 3 (in Program 84-11-01). This gives a 10ms x 3 = 30ms packet.

5.2 Bandwidth

The bandwidth required for VoIP calls depends on several factors, including:

- Number of simultaneous calls
- CODEC used
- Frame Size
- Data Networking Protocol used

The more frames encapsulated into each packet, the less bandwidth is required. This is because each packet transmitted has the same header size. Therefore, if numerous very small packets are sent then

^{**1}1. The Mean Opinion Score (MOS) provides a numerical measure of the quality of human speech at the destination end of the circuit. The scheme uses subjective tests (opinionated scores) that are mathematically averaged to obtain a quantitative indicator of the system performance.

bandwidth is also being used for a large amount of header information. If we add several frames to the packet, less packets are transmitted and therefore have less header information sent.

If we add many voice frames to each packet, less bandwidth is being used. However, this does have disadvantages. If there is a large packet size, and a particular voice packet is lost, this has a greater impact on the speech quality. If a small quantity of voice frames per packet is being used, the effect of losing a packet is reduced.

As a general rule: The more frames per packet, the less bandwidth is used, but the quality is also lower.

Examples:

Example 1: CODEC: G.729 Frame Size: 10ms Voice Frames per Packet: 2 Voice Sample Size: 20ms
(frame size x Voice Frames) Bandwidth Required: 24Kbps

Example 2: CODEC: G.729 Frame Size: 80ms Voice Frames per Packet: 8 Voice Sample Size: 80ms
(frame size x Voice Frames) Bandwidth Required: 12Kbps

Section 6 QUALITY OF SERVICE (QoS) IMPLEMENTATION

Section [Voice Quality Improvements](#) discusses some of the problems associated with voice quality. This section describes how QoS can be implemented on data networks to provide the “best case” for VoIP traffic.

Not all network hardware supports QoS and each manufacturer has their own methods of implementing QoS. The explanations below are as generic as possible. The installer/maintainer of the data network should be familiar with the QoS characteristics of their equipment and should be able to configure the equipment accordingly.

Quality of Service is commonly used to describe the actual implementation of prioritization on network hardware. This prioritization (at Layer 2 and Layer 3 of the OSI model) is described in [Figure 6-1 Layer 2 Diagram \(802.1Q\)](#).

6.1 Prioritization

When data is transmitted through a network, bottlenecks can occur causing the available bandwidth to be reduced or the data to increase. This impacts the packet delivery.

Consider data communication between the two computers shown in the diagram [Figure 6-1 Layer 2 Diagram \(802.1Q\)](#). The Hosts can transmit data at 100 Mbps. When a packet from Host A, destined for Host B, reaches the router, the available bandwidth is reduced to 256Kbps and the packet flow must be reduced. [Figure 6-3 Network Bottleneck Example](#) shows a diagram of this condition.

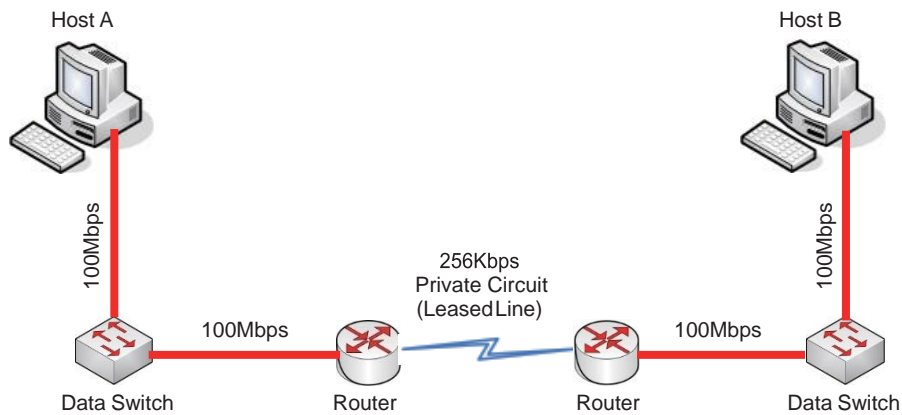


Figure 6-3 Network Bottleneck Example

For this example, each end of the network has only one host. Typically, many hosts are sending data over the narrow bandwidth. The routers buffer packets and transmit them over the WAN lines as efficiently as possible. When this occurs, certain packets are dropped by the router and some packets are delayed.

For most data applications this packet loss/delay is not critical. For example, a delay of one to five seconds to transmit an email is imperceptible. When VoIP is implemented, this loss/delay has a massive impact on the voice quality. The resulting gaps in speech, distortion and delay are unacceptable for voice traffic.

To avoid this problem, it is possible to prioritize the VoIP packets. The router examines all packets received, determines the priority level of the packet, and forwards it accordingly. The data^{**1} is assigned lower priority and the voice is transmitted before the data. This can have a negative impact on the data network if a lot of voice is transmitted.

Figure 6-4 Voice and Data Network Implementation shows how a voice and data network can be implemented.

***1. This description discusses voice and data. These terms are commonly used when describing QoS, although in the case of VoIP, the voice is actually converted to IP and transmitted as data. Therefore, everything transmitted on a Data Network is data, but logically we think of this as voice and data traffic.*

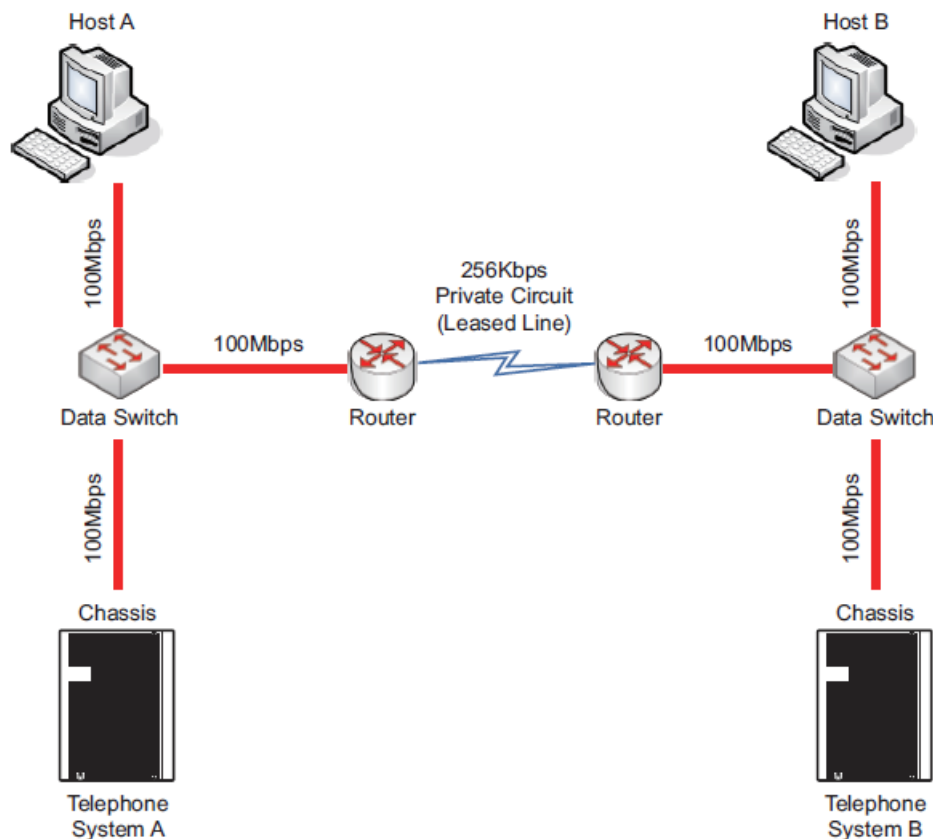


Figure 6-4 Voice and Data Network Implementation

After the router is configured for QoS, it examines incoming packets and allocates a priority to the packet. [Figure 6-5 Priority Queuing on Voice and Data Networks](#) shows the affect priority queuing has on voice and data networks. The packets arrive randomly. They are processed and output according to the QoS policy. The VoIP traffic is output first.

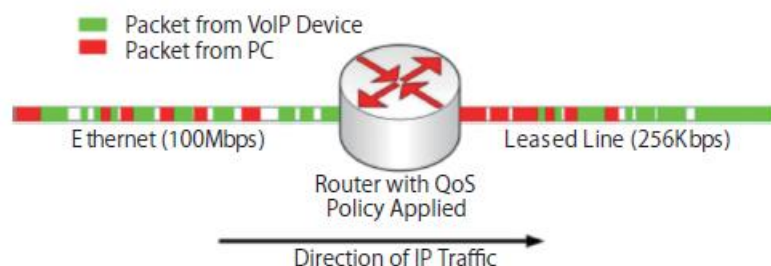


Figure 6-5 Priority Queuing on Voice and Data Networks

To enable this type of queuing it is necessary to:

1. Configure the VoIP equipment to mark its packets with a specific value so that the switches/routers can identify that it is voice – Called Marking.
2. Configure the network equipment to recognize the difference between the different Marked packets – Called Classification. (i.e., informs the router what a voice packet looks like.
3. Configure the network equipment to give priority to the packets that have been classified as voice – Called Priority Queuing.

6.2 Layer 2 QoS (802.1pq)

QoS is most commonly implemented at Layer 3 of the OSI model. This layer deals with IP addresses, and is usually handled by Routers. However, sometimes it is necessary to implement Layer 2 QoS – usually in large LAN environments with many IP phones.

Layer 2 devices work with Ethernet frames (encapsulated IP packets) rather than IP addresses. Layer 2 QoS uses the Priority field of the Ethernet frame. This field has three bits and can have eight possible values (000 to 111 in binary). Some switches can be configured to prioritize traffic based on these values. This field is available only if the Ethernet device is configured for VLAN (IEEE 802.1q) operation (VLAN is outside the scope of this document).

Protocol Structure - IEEE 802.1p: LAN Layer 2 QoS

Table 6-1 Protocol Structure for Layer 2 QoS shows the format of an Ethernet frame and the User Priority field that is used for Layer 2 QoS.

Table 6-1 Protocol Structure for Layer 2 QoS

| IEEE 802.1Q Tagged Frame for Ethernet | | | | | | | | |
|---------------------------------------|--------|---------|---------|---------|---------|-------------|---------------|---------|
| 7 Bytes | 1 Byte | 6 Bytes | 6 Bytes | 2 Bytes | 2 Bytes | 2 Bytes | 42~1496 Bytes | 4 Bytes |
| Preamble | SFD | DA | SA | TPID | TCI | Type Length | Data | CRC |

| Expanded View of TCI Field | | |
|----------------------------|-------|--|
| 3 Bits | 1 Bit | 12 Bits |
| User Priority | CFI | Bits of VLAN ID (VID) to identify possible VLANs |

The following define the fields used for the protocol structure:

Preamble (PRE) - The PRE is an alternating pattern of ones and zeros that tells receiving stations a frame is coming, and synchronizes frame-reception portions of receiving physical layers with the incoming bit stream.

Start-of-frame delimiter (SFD) - The SFD is an alternating pattern of ones and zeros, ending with two consecutive 1-bits indicating that the next bit is the left-most bit in the left-most byte of the destination address.

Destination Address (DA) - The DA field identifies which station(s) should receive the frame.

Source Addresses (SA) - The SA field identifies the sending station.

Tag Protocol Identifier (TPID) - The defined value of SL2100 in hex. When a frame has the EtherType equal to SL2100, this frame carries the tag IEEE 802.1Q / 802.1P.

Tag Control Information (TCI) - The field including user priority, Canonical format indicator and VLAN ID.

User Priority - Defines user priority, giving eight priority levels. IEEE 802.1P defines the operation for these three user priority bits.

CFI - Canonical Format Indicator is always set to zero for Ethernet switches. CFI is used for compatibility reason between Ethernet type network and Token Ring type network.

VID - VLAN ID is the identification of the VLAN, which is basically used by the standard 802.1Q. It allows the identification of 4096 VLANs.

Length/Type - This field indicates either the number of MAC-client data bytes that are contained in the data field of the frame, or the frame type ID if the frame is assembled using an optional format.

Data - Is a sequence of bytes of any value. The total frame minimum is 64 bytes.

Frame Check Sequence (FCS) - This sequence contains a 32-bit cyclic redundancy check (CRC) value, which is created by the sending MAC and is recalculated by the receiving MAC to check for damaged frames.

Example Ethernet Frame with Layer 2 QoS Enabled

The example below shows an Ethernet Frame containing one RTP (speech) packet. The Frame is VLAN tagged, has a VLAN ID of 99 and a VLAN Priority of 5. It is also possible to see that the Layer 3 QoS has not been set.

Ethernet Frame Example - Layer 2 QoS Enabled

```
Source Destination Protocol
172.16.0.101 172.16.0.21 RTP
Info
Payload type=ITU-T G.729, SSRC=701655963, Seq=28165, Time=21520
Frame 160 (78 bytes on wire, 78 bytes captured)
Arrival Time: Jan 18, 2005 13:55:44.842738000
Time delta from previous packet: 0.008241000 seconds
Time since reference or first frame: 2.910072000 seconds
Frame Number: 160
Packet Length: 78 bytes
Capture Length: 78 bytes
Ethernet II, Src: 00:60:b9:c6:6e:45, Dst: 00:60:b9:c1:ab:a3
Destination: 00:60:b9:c1:ab:a3 (Nitsuko_c1:ab:a3)
Source: 00:60:b9:c6:6e:45 (Nitsuko_c6:6e:45)
Type: 802.1Q Virtual LAN (0xSL2100)
802.1q Virtual LAN
101. .... = Priority: 5 (Layer 2 Priority = 5)
...0 .... = CFI: 0
.... 0000 0110 0011 = ID: 99
Type: IP (0x0800)
Internet Protocol, Src Addr: 172.16.0.101 (172.16.0.101), Dst Addr:
172.16.0.21 (172.16.0.21)
Version: 4
Header length: 20 bytes
Differentiated Services Field: 0x00 (DSCP 0x00: Default; ECN: 0x00)
0000 00.. = Differentiated Services Codepoint: Default
(0x00)
.... ..0. = ECN-Capable Transport (ECT): 0
.... ..0. = ECN-CE: 0
Total Length: 60
Identification: 0x0086 (134)
Flags: 0x00
0... = Reserved bit: Not set
.0.. = Don't fragment: Not set
..0. = More fragments: Not set
Fragment offset: 0
Time to live: 30
Protocol: UDP (0x11)
Header checksum: 0x4391 (correct)
Source: 172.16.0.101 (172.16.0.101)
Destination: 172.16.0.21 (172.16.0.21)
User Datagram Protocol, Src Port: 10022 (10022), Dst Port: 10020 (10020)
Source port: 10022 (10022)
Destination port: 10020 (10020)
Length: 40
Checksum: 0x0581 (correct)
Real-Time Transport Protocol
Stream setup by SDP (frame 1)
Setup frame: 1
Setup Method: SDP
10.. .... = Version: RFC 1889 Version (2)
..0. .... = Padding: False
...0 .... = Extension: False
.... 0000 = Contributing source identifiers count: 0
0... .... = Marker: False
..001 0010 = Payload type: ITU-T G.729 (18)
Sequence number: 28165
Timestamp: 21520
Synchronization Source identifier: 701655963
Payload: 76AC9D7AB6ACE2510B3A3338646DA738...
```

6.3 Layer 3 QoS

QoS is most commonly implemented at Layer 3. This allows the VoIP packets to be prioritized by routers, before they are forwarded to their next hop.

Layer 3 QoS uses the Type of Service (ToS) field of the IP packet. This is an 8-bit field in the header of the IP packet. The field can be used by Diffserv or IP Precedence. Although these are two different standards, the actual field in the IP packet is the same – Only the method of evaluating the bits differs.

QoS does not function only by using the ToS field (i.e., Marking the VoIP packets). It is an end-to-end process and requires configuration on all networking devices.

Packet Marking is the first step in this process and is often the only step that the NEC dealer performs.

Table 6-2 Layer 3 QoS Example

Protocol Structure - IP/Pv4 Header (Internet Protocol Version 4)

| 4 Bits | 4 Bits | 8 Bits | 16 Bits | |
|---------------------|--------|-----------------|-----------------|-----------------|
| Version | IHL | Type of Service | Total Length | |
| Identification | | | Flags | Fragment Offset |
| Time to Live | | Protocol | Header Checksum | |
| Source Address | | | | |
| Destination Address | | | | |
| Option + Padding | | | | |
| Data | | | | |

Type of Service Field (Diffserv)

| 6 Bits | 2 Bits |
|------------------------------------|-----------------------|
| Differentiated Services Code Point | ECN (Not QoS related) |

Type of Service Field (IP Precedence - Ref. RFC 1349)

| 3 Bits | 1 Bit | 1 Bit | 1 Bit | 1 Bit | 1 Bit |
|---------------------|-------|------------|-------------|-------|--------------------|
| IP Precedence Value | Delay | Throughput | Reliability | Cost | MBZ (must be zero) |

Listed below are the fields used in [Table 6-2 Layer 3 QoS Example](#).

Version – the version of IP currently used.

IP Header Length (IHL) – datagram header length. Points to the beginning of the data. The minimum value for a correct header is 5.

Type-of-Service – Indicates the quality of service desired by specifying how an upper-layer protocol would like a current datagram to be handled, and assigns datagrams various levels of importance. This field is used for the assignment of Precedence, Delay, Throughput and Reliability.

Total Length – Specifies the length, in bytes, of the entire IP packet, including the data and header. The maximum length specified by this field is 65,535 bytes. Typically, hosts are prepared to accept datagrams up to 576 bytes.

Identification – Contains an integer that identifies the current datagram. This field is assigned by sender to help receiver to assemble the datagram fragments.

Flags – Consists of a 3-bit field of which the two low-order (least-significant) bits control fragmentation. The low-order bit specifies whether the packet can be fragmented. The middle bit specifies whether

the packet is the last fragment in a series of fragmented packets. The third or high-order bit is not used.

Fragment Offset – This 13-bit field indicates the position of the fragment data relative to the beginning of the data in the original datagram, which allows the destination IP process to properly reconstruct the original datagram.

Time-to-Live – This is a counter that gradually decrements down to zero, at which point the datagram is discarded. This keeps packets from looping endlessly.

Protocol – Indicates which upper-layer protocol receives incoming packets after IP processing is complete.

Header Checksum – Helps ensure IP header integrity. Since some header fields change, e.g., Time To Live, this is recomputed and verified at each point that the Internet header is processed.

Source Address – Specifies the sending node.

Destination Address – Specifies the receiving node.

Options – Allows IP to support various options, such as security.

Data – Contains upper-layer information.

6.4 IP Precedence

IP Precedence is a QoS method that combines a priority value with different on/off parameters; Delay, Throughput, Reliability and Cost. The MBZ (Must be Zero) bit is not used.

Using the ToS bits, you can define up to eight classes of service. Other devices configured throughout the network can then use these bits to determine how to treat the packet in regard to the type of service to grant it. These other QoS features can assign appropriate traffic-handling policies including congestion management and bandwidth allocation. By setting IP Precedence levels on incoming traffic and using them in combination with QoS queuing features, you can create differentiated service.

Table 6-3 Type of Service Field (IP Precedence - i Ref. REC 1349)

| 3 Bits | 1 Bit | 1 Bit | 1 Bit | 1 Bit | 1 Bit |
|---------------------|-------|------------|-------------|-------|--------------------|
| IP Precedence Value | Delay | Throughput | Reliability | Cost | MBZ (must be zero) |

IP Precedence Value

| Value | Binary Value | Description |
|-------|--------------|----------------------|
| 0 | 000 | Routine |
| 1 | 001 | Priority |
| 2 | 010 | Immediate |
| 3 | 011 | Flash |
| 4 | 100 | Flash Override |
| 5 | 101 | CRITIC/ECP |
| 6 | 110 | Internetwork Control |
| 7 | 111 | Network Control |

Throughput

| Value | Description |
|-------|-------------------|
| 0 | Normal Throughput |
| 1 | High Throughput |

Reliability

| Value | Description |
|-------|--------------------|
| 0 | Normal Reliability |
| 1 | High Reliability |

Delay

| Value | Description |
|-------|--------------|
| 0 | Normal Delay |
| 1 | Low Delay |

Cost

| Value | Description |
|-------|-------------|
| 0 | Normal Cost |
| 1 | Low Cost |

6.5 Diffserv (Differentiated Service)

Differentiated Services (Diffserv) uses the ToS field in an IP header. Diffserv is now commonly used instead of IP Precedence (refer to [IP Precedence](#)) as it provides greater flexibility. This method uses six bits of the ToS field to determine the priority – which provides up to 64 possible values. The combination of binary digits is known as the Diffserv Codepoint (DSCP).

Table 6-4 Diffserv Parameters

| 6 bits | 2 bits |
|------------------------------------|-----------------------|
| Differentiated Services Code Point | ECN (Not QoS related) |

The example below shows an Ethernet Frame containing one RTP (speech) packet. The IP Packet has the ToS field set to 101000 (binary) which is the equivalent of Class Selector 5. The router(s) in this network should be programmed to prioritize based on CS5.

Ethernet Frame Example - Containing One RTP (Speech) Packet

```
Source Destination Protocol
172.16.0.21 172.16.0.101 RTP
Info
Payload type=ITU-T G.729, SSRC=732771006, Seq=30885, Time=20560
Frame 159 (65 bytes on wire, 65 bytes captured)
Arrival Time: Jan 18, 2005 13:55:44.834497000
Time delta from previous packet: 0.000445000 seconds
Time since reference or first frame: 2.901831000 seconds
Frame Number: 159
Packet Length: 65 bytes
Capture Length: 65 bytes
Ethernet II, Src: 00:60:b9:c1:ab:a3, Dst: 00:60:b9:c6:6e:45
Destination: 00:60:b9:c6:6e:45 (Nitsuko_c6:6e:45)
Source: 00:60:b9:c1:ab:a3 (Nitsuko_c1:ab:a3)
```



```

Type: IP (0x0800)
Internet Protocol, Src Addr: 172.16.0.21 (172.16.0.21), Dst Addr: 172.16.0.101 (172.16.0.101)
Version: 4
Header length: 20 bytes
Diff Services Field: 0xa0 (DSCP 0x28: Class Selector 5; ECN: 0x00)
1010 00.. = Diff Services Codepoint: Class Selector 5 (0x28)
.... ..0. = ECN-Capable Transport (ECT): 0
.... ...0 = ECN-CE: 0
Total Length: 44
Identification: 0x0069 (105)
Flags: 0x00
0... = Reserved bit: Not set
.0.. = Don't fragment: Not set
..0. = More fragments: Not set
Fragment offset: 0
Time to live: 30
Protocol: UDP (0x11)
Header checksum: 0x431e (correct)
Source: 172.16.0.21 (172.16.0.21)
Destination: 172.16.0.101 (172.16.0.101)
User Datagram Protocol, Src Port: 10020 (10020), Dst Port: 10022 (10022)
Source port: 10020 (10020)
Destination port: 10022 (10022)
Length: 24
Checksum: 0x5293 (correct)
Real-Time Transport Protocol
Stream setup by SDP (frame 112)
Setup frame: 112
Setup Method: SDP
10.. .... = Version: RFC 1889 Version (2)
..1. .... = Padding: True
...0 .... = Extension: False
.... 0000 = Contributing source identifiers count: 0
0... .... = Marker: False
.001 0010 = Payload type: ITU-T G.729 (18)
Sequence number: 30885
Timestamp: 20560
Synchronization Source identifier: 732771006
Payload: 3ED0
Padding data: 00
Padding count: 2

```

6.6 Comparison of IP Precedence and Diffserv Values

As stated earlier, IP Precedence and Diffserv use the same 8-bit ToS field in the IP header to mark packets. It is possible to have the same ToS value for either method which means that the two methods can work alongside each other.

For example, if the VoIP equipment supports IP Precedence and the router can prioritize only using the DSCP they can be set to the same value. Refer to [Table 6-5 IP Precedence and Diffserv Values Comparison](#) for the values.

Table 6-5 IP Precedence and Diffserv Values Comparison

| DSCP Decimal | DSCP Binary | IP Precedence | Description |
|--------------|-------------|---------------|------------------|
| 0 | 000000 | 0 | Class Selector 0 |
| 1 | 000001 | | |
| 2 | 000010 | | |
| 3 | 000011 | | |
| 4 | 000100 | | |
| 5 | 000101 | | |
| 6 | 000110 | | |
| 7 | 000111 | | |

| DSCP Decimal | DSCP Binary | IP Precedence | Description |
|--------------|-------------|---------------|---------------------------|
| 8 | 001000 | 1 | Class Selector 1 |
| 9 | 001001 | | |
| 10 | 001010 | | AF11 (Assured Forwarding) |
| 11 | 001011 | | |
| 12 | 001100 | | AF12 (Assured Forwarding) |
| 13 | 001101 | | |
| 14 | 001110 | | AF13 (Assured Forwarding) |
| 15 | 001111 | | |
| 16 | 010000 | 2 | Class Selector 2 |
| 17 | 010001 | | |
| 18 | 010010 | | AF21 (Assured Forwarding) |
| 19 | 010011 | | |
| 20 | 010100 | | AF22 (Assured Forwarding) |
| 21 | 010101 | | |
| 22 | 010110 | | AF23 (Assured Forwarding) |
| 23 | 010111 | | |
| 24 | 011000 | 3 | Class Selector 3 |
| 25 | 011001 | | |
| 26 | 011010 | | AF31 (Assured Forwarding) |
| 27 | 011011 | | |
| 28 | 011100 | | AF32 (Assured Forwarding) |
| 29 | 011101 | | |
| 30 | 011110 | | AF33 (Assured Forwarding) |
| 31 | 011111 | | |
| 32 | 100000 | 4 | Class Selector 4 |
| 33 | 100001 | | |
| 34 | 100010 | | AF41 (Assured Forwarding) |
| 35 | 100011 | | |
| 36 | 100100 | | AF42 (Assured Forwarding) |
| 37 | 100101 | | |
| 38 | 100110 | | AF43 (Assured Forwarding) |
| 39 | 100111 | | |
| 40 | 101000 | 5 | Class Selector 5 |
| 41 | 101001 | | |
| 42 | 101010 | | |
| 43 | 101011 | | |
| 44 | 101100 | | |
| 45 | 101101 | | |
| 46 | 101110 | | EF (Expedited Forwarding) |
| 47 | 101111 | | |
| 48 | 110000 | 6 | Class Selector 6 |
| 49 | 110001 | | |
| 50 | 110010 | | |

| DSCP Decimal | DSCP Binary | IP Precedence | Description |
|--------------|-------------|---------------|------------------|
| 51 | 110011 | | |
| 52 | 110100 | | |
| 53 | 110101 | | |
| 54 | 110110 | | |
| 55 | 110111 | | |
| 56 | 111000 | 7 | Class Selector 7 |
| 57 | 111001 | | |
| 58 | 111010 | | |
| 59 | 111011 | | |
| 60 | 111100 | | |
| 61 | 111101 | | |
| 62 | 111110 | | |
| 63 | 111111 | | |

6.7 Programming QoS in the NEC SL2100 System

6.7.1. Marking Voice Traffic - Program 84-10-XX

Before programming the NEC SL2100 system, discuss the requirements with the network engineering staff or the managed network provider. If the ToS markings that are used are not specifically configured into the network equipment, the voice traffic is handled by the default queue and is given lowest priority.

6.7.2. NEC SL2100 Voice Protocols

The NEC SL2100 system supports the following types of VoIP traffic.

| Number | Protocol Type | Description |
|--------|---------------|--|
| 1 | H.323 | Communication from VoIPDB to Network |
| 2 | RTP/RTCP | Voice (RTP) and Call Quality Data (RTCP) |
| 3 | SIP | Communication from VoIPDB to Standard SIP device. |
| 4 | SIP MLT | Communication from VoIPDB to SIP Multiline Terminal. |
| 5 | SIP Trunks | Communication from VoIPDB to Network |

6.7.3. Configuring Diffserv

Use Program 84-10-10 to select the logic for marking the ToS field. The choices are:

| Number | ToS Mode | Programs Enabled |
|--------|----------|-------------------------------|
| 0 | None | None – ToS bits are: 00000000 |

| Number | ToS Mode | Programs Enabled |
|--------|---------------|--|
| 1 | IP Precedence | 84-10-02 Priority – 0=Lowest ~ 7=Highest (ToS bits: 0~2) 84-10-03 Delay – 0=Normal, 1=Low (ToS Bit: 3) 84-10-04 Throughput – 0=Normal, 1=High (ToS Bit: 4) 84-10-05 Reliability – 0=Normal, 1=High (ToS Bit: 5) ToS Bit 7: Always 0 <i>Typically, only one of bits 3~6 is set to 1 and the other three bits are set to 0. For example, to maximize route reliability, set 84-10-05 to 1 and leave 84-10-03, 84-10-04 and 84-10-06 at 0 (default).</i> |
| 2 | Diffserv | 84-10-07 DSCP Value in Decimals: 0~63 (ToS bits: 0~5) ToS Bits 6 & 7 are not evaluated |

6.7.4. Configuration Examples for Classification and Queuing

Figure 6-6 Common Network with Cisco Router shows a typical network scenario and an example of a Cisco router configuration.



This document provides a general description of VoIP technology, but it does not discuss individual manufacturer solutions. This sample configuration is provided as a common scenario. It is a good example of how QoS can be implemented on a router.

NEC does not endorse or provide support on any third party equipment unless it is supplied by NEC.

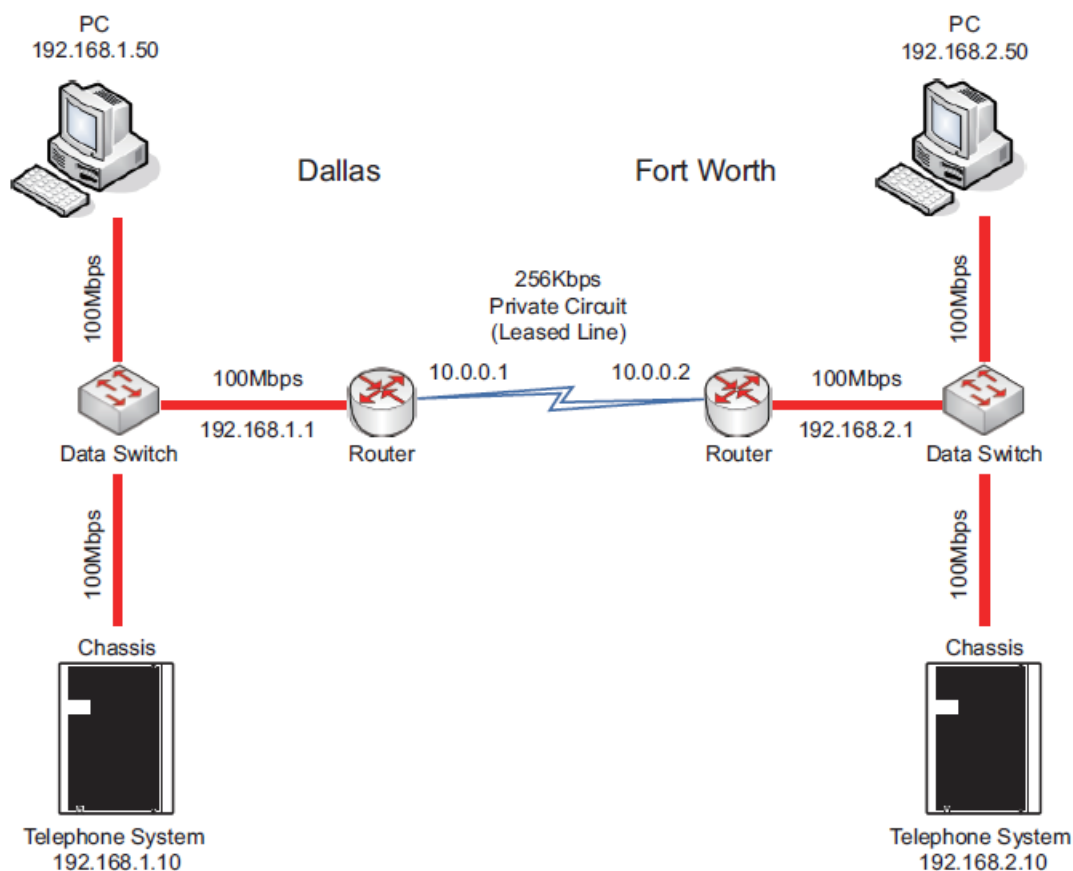


Figure 6-6 Common Network with Cisco Router

See Table 6-6 Cisco Router Configuration Example for configuration information about the Cisco 2621 router. A description of key commands follows.

Table 6-6 Cisco Router Configuration Example

```

Current Configuration : 2023 bytes
version 12.3
hostname Cisco2621
|
class-map match-any VoIPClass (1)
match ip dscp cs5 (2)
policy-map VoIPPolicy (3)
class VoIPClass(4)
priority 50(5)
class class-default(6)
fair-queue(7)
|
interface FastEthernet0/0
description Connects to Dallas LAN
ip address 192.168.1.1 255.255.255.0
|
interface Serial0/0
description Connects to Fort Worth via Kilostream
bandwidth 256(8)
ip address 10.0.0.1 255.255.0.0
service-policy output VoIPPolicy(9)
encapsulation ppp
|
ip route 0.0.0.0 0.0.0.0 10.0.0.2

```

Configuration Example Explanation:

1. Defines a Class Map called VoIPClass.
2. Matches any packets that have the ToS field set to IP Precedence 5 / DSCP 40 and assigns them to VoIPClass.
3. Defines a Policy Map called VoIPPolicy.
4. Creates a Class called VoIPClass and assigns this to the VoIPPolicy.
5. Allocates 50Kbps of bandwidth to the VoIPClass.
- 6 & 7. Determines that any data that does not match VoIPClass should be processed using the "fair-queue" method (i.e., No Prioritization).
8. Determines the amount of bandwidth available on the Serial interface – essential for the QoS calculations.
9. Applies the VoIP Policy to any packets that exit the serial interface. This means that data being received (input) does not use this policy.

Section 7 PORT DESIGNATIONS

Following table shows an IP Port related in the SL2100 System.

The assigning IP port number cannot be duplicated in the same port number listed below.

For the detailed programming information, refer to the programming manual.

Table 6-7 Port number list of SL2100 System

| Well-known port or Fixed port ("-" is fixed port (cannot change)) | IP Port Numbers | Description |
|--|-----------------|-------------|
| well-known port | 22 | ssh |
| well-known port | 53 | domain |
| well-known port | 67 | bootps |
| well-known port | 68 | bootpc |
| well-known port | 115 | sftp |
| well-known port | 123 | ntp |
| well-known port | 137 | netbios-dgm |

| Well-known port or Fixed port ("-" is fixed port (cannot change)) | IP Port Numbers | Description |
|--|-----------------|-----------------|
| well-known port | 161 | snmp |
| well-known port | 162 | snmptrap |
| - | 1718 | |
| - | 5555 | |
| - | 6666 | |
| - | 1900 | |
| - | 2567 | Lua application |
| - | 3000 | For InDECTPro |
| - | 7080 | For InDECTPro |
| - | 20020 ~ 20274 | Reserved |
| - | 20100 | |
| - | 20200 | |
| - | 20300 | |
| - | 20400 | |
| - | 27998 | For InDECTPro |
| - | 27999 | For InDECTPro |
| - | 33512 | Lua application |
| - | 33516 | Lua application |
| - | 33518 | For InDECTPro |
| - | 33519 | Lua application |
| - | 57000 | Reserved |
| - | 58000 | |
| - | 58001 | |
| - | 58002 | |
| - | 59000 ~ 59063 | |
| - | 60050 | Lua application |
| - | 60060 | Lua application |
| - | 60300 | Lua application |
| - | 63516 | Lua application |
| - | 63517 | Lua application |
| - | 63518 | For InDECTPro |
| - | 63519 | Lua application |
| Related IP Port programming | | |
| Program Number | Default | Description |
| 10-20-01 CTI Server | 0 | |
| 10-20-01 ACD-MIS | 0 | |
| 10-20-01 Networking System | 30000 | |
| 10-20-01 SMDR Output | 0 | |
| 10-20-01 DIM Output | 0 | |
| 10-20-01 UC Web Application | 0 | |
| 10-20-01 1st Party CTI | 0 | |
| 10-20-01 O&M Server | 8010 | |
| 10-20-01 Traffic Report Output | 0 | |

| Well-known port or Fixed port ("-" is fixed port (cannot change)) | IP Port Numbers | Description |
|--|--|--|
| 10-20-01 Room Data Output for Hotel Service | 0 | |
| 10-20-01 IP-DECT Directory Access | 0 | |
| 10-20-01 Presence | 0 | |
| 10-23-03 (Cannot be changing) | 1720 | |
| 10-29-04 | 5060 | SIP Trunk |
| 10-29-07 | 5060 | |
| 10-46-06 | 5080 | SIP MLT |
| 10-46-12 | 30000 | |
| 10-46-13 | 5081 | SIP MLT |
| 10-47-01 | 6080 | |
| 20-64-03 | 0 | |
| 20-64-06 | 0 | |
| 20-64-07 | 0 | When it is not 0, 512 port which continues from a set port is reserved. For example, when this port number is 50000, 50000 ~ 50511 ports are reserved. |
| 20-64-08 | 0 | When it is not 0, 460 port which continues from a set port is reserved. For example, when this port number is 10020, 10020 ~ 10479 ports are reserved. |
| 42-06-01 | 5129 | PMS Service |
| 84-02-33 | 1730 | |
| 84-02-36 | 20001 | |
| 84-14-06 | [profile 1]: 5060 [profile 2]: 5062 [profile 3]: 5090 [profile 4]: 5092 [profile 5]: 5094 [profile 6]: 5096 | |
| 84-20-01 | 5070 | SIP SLT |
| 84-26-02 | 10020 | RTP: 10020 ~ 10275 |
| 84-26-03 | 10021 | RTCP: 10021 ~ 10276 |
| 84-27-06 | 10100 | |
| 84-27-07 | 4000 | |
| 90-54-01 | 80 | WebPro (HTTP) |
| 90-54-02 | 8000 | PCPro |
| 90-54-03 | 443 | WebPro(HTTPS) |

MEMO

SIP Trunking

Section 1 VoIP

VoIP (Voice over Internet Protocol or Voice over IP) allows the delivery of voice information using the Internet protocol (sending data over the Internet using an IP address). This means that digital voice information can be sent in packets over the Internet rather than using the traditional public switch telephone network (CO lines). A major advantage of VoIP is that it avoids the tolls charged by ordinary telephone service.

Using VoIP equipment at a gateway (a network point that acts as an entrance to another network), the packetized voice transmissions from users in the company are received and routed to other parts of the company intranet (local area or wide area network) or they can be sent over the Internet using CO lines to another gateway.**1

Section 2 IP NETWORKING

IP Networking uses VoIP technology to connect two or more telephone systems together. This allows calls to be made between sites without using the public telephone network. This can save money and make communication between sites much easier.

The following Networking modes are available on the NEC SL2100 system:

- SIP TIE lines
- SIP Trunks (to a SIP Trunk Provider)

Section 3 SIP TRUNKING

3.1 Introduction

SIP (Session Initiation Protocol) is used for Voice over IP. It is defined by the IETF (Internet Engineering Task Force) in RFC2543 and RFC3261. SIP trunking is the term used for linking a PBX, such as the NEC SL2100 system, to the public telephone network by means of VoIP. This provides the ability for users to place and receive communications and services from any location and for networks to identify the users wherever they are located.

SIP analyzes requests from clients and retrieves responses from servers then sets call parameters at either end of the communication, handles call transfer and terminates.

The NEC SL2100 system implementation and programming for SIP are very similar. The call routing, call features and speech handling (RTP) are the same. Only the signaling protocol is different.

***1. The voice quality of VoIP depends on variables such as available bandwidth, network latency and Quality of Service (QoS) initiatives, all of which are controlled by the network and Internet Service Providers. Because these variables are not in NEC control, it cannot guarantee the performance of the user's IP-based remote voice solution. Therefore, NEC recommends connecting VoIP equipment through a local area network using a Private IP address.*

With the NEC SL2100 system, SIP trunks can receive incoming calls with Caller ID, place outgoing calls, and transfer SIP trunks to IP, SIP, analog and digital stations, and across a network.

Currently, however, SIP Centrex Transfer is not supported.

If a common carrier supports SIP, the NEC SL2100 can connect the SIP Carrier and outgoing calls to the PSTN (Public Switched Telephone Network) network and the common IP network using an NEC SL2100 SIP trunk.

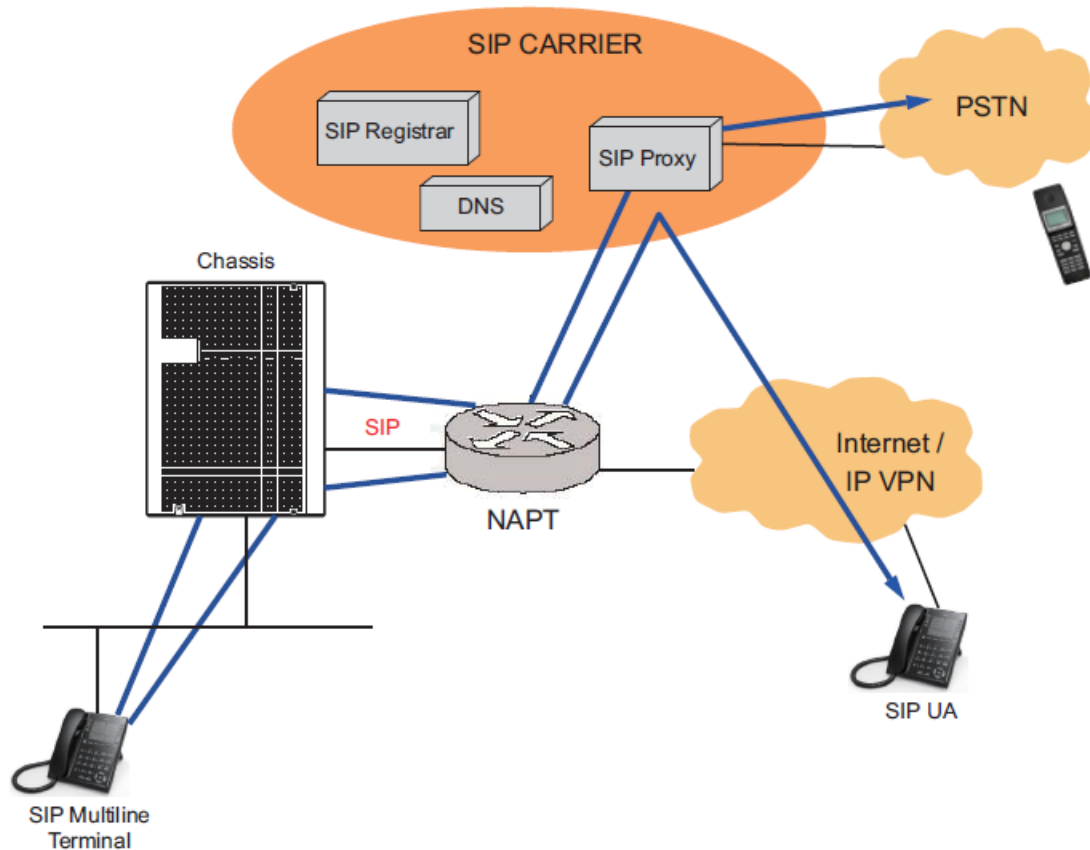


Figure 7-1 Common IP Network using NEC SL2100 SIP Trunk

3.1.1 SIP Trunking Requirements

The following are required when using the SIP trunk on the NEC SL2100 system:

- CPU-C1 software
- VOIPDB (VoIP Daughter Board) Programming Conditions



The VoIPDB is only a requirement if more than the built in (8) DSP resources on the CPU are required. Adding the VoIPDB provides an additional (8) DSP resources for a total of (16). If more than (16) DSP resources are required, additional IP Channel licenses are required.

The following conditions apply when programming the NEC SL2100 system for SIP Trunking:

- If entries are made in Program 10-29-xx for a SIP Server and the SIP Server is then removed or not used, the entries in Program 10-29-xx must be set back to their default settings. Even if **Program 10-29-01 : SIP Proxy Setup – Outbound Proxy** is set to 0, the NEC SL2100 system checks the settings in the remaining 10-29 programs.

- SIP Multi-profile support is added. The simultaneous use of a SIP trunk interconnection and a SIP trunk carrier connection or six SIP carriers at the same time is supported.

Limitations:

- Each profile requires a different number for every SIP trunk port set in PRG 84-14-06.
 - Six carriers must be connected using a single route. The system only supports one gateway.
 - Only one DNS server can be set for the SL2100 to connect to six different SIP Carriers.
- The NEC SL2100 system restricts an outgoing call under the following conditions:
 - SIP configuration failed
 - SIP registration failed
 - CPU-C1/VOIPDB daughter board link down
 - Lack of VOIPDB DSP resource
 - Lack of bandwidth

3.1.2 SIP Trunking Setup

Use the following steps to initially set up SIP Trunking for the NEC SL2100 system:

1. By default, the NEC SL2100 is assigned a static IP address and runs behind a NAT router.



When using an NEC SL2100 on a LAN behind an NAPT router, forward port 5060 to the IP address of the NEC SL2100 CPU-C1 (commonly called the SL2100 CPU) since the signaling is handled by the CPU-C1. Then, since the media stream (the speech) uses a large range of ports for the RTP packages, forward the ports (10020~10147) to the IP address of the VOIPDB. Or, use the DMZ option for the VOIPDB. This means that the VOIPDB is not actually behind the firewall. This is achieved by connecting the VOIPDB to a physical or virtual DMZ port. You can also achieve the same result by port forwarding 10020 to 10147.

2. Define the SIP Carrier account information (user name, password, domain name/IP address to the provider).
3. Define the trunk ports as SIP.
4. Set the Expire Time.

Section 4 SIP TRUNK OVERVIEW

This section is an overview of basic SIP trunk behavior, protocols, supported SIP trunking methods and options, supported CODEC as well as other supported functions of SIP Trunking.

4.1 General Information

The NEC SL2100 SIP Trunks behave like some SIP User Agents (UA). UA is the client application used with a particular network protocol.

The NEC SL2100 provides a maximum of 31 register IDs and can register these IDs with any SIP server.

A maximum number of 64 SIP trunks can be used with the NEC SL2100 system. The maximum number of simultaneous VoIP calls is 128 and is determined by the number of resources available on the VoIPDB.

The NEC SL2100 can connect any SIP server over a NAPT router by one static global IP address. The NEC SL2100 system supports a DNS resolution access and an IP address direct access for SIP servers and supports the sub-address feature with SIP trunk interconnection.

4.2 Supported Protocols

The following protocols are supported:

- SIP (RFC2543 bis04)
- SIP RFC3261 Supported [updated version of RFC2543]
- The SIP stack has been updated from RFC2543 Base to RFC3261 Base
- SDP
- RTP/RTCP
- UDP
- IPv4
- T.38 (draft-ietf-sipping-realtimefax-02.txt)

4.3 Supported SIP Methods

The following SIP methods are supported with the NEC SL2100 system:

- REGISTER
- INVITE
- BYE
- CANCEL
- ACK
- PRACK
- Response 1xx 2xx, 3xx, 4xx, 5xx, 6xx

The following features are available:

- **Support the 401 response for the Initial Invite**
If 401 message is sent for the Initial Invite, with previous software, the system cannot respond to the message correctly.
- **Support the 401/407 response for the Invite of Session Timer**
If 401/407 response is sent for the invite of Session Timer, the system can send the Invite message with Authentication header.
- **Support the 128 byte size of Nonce max value sent by the 401/407 message**
The Nonce maximum size sent by the 401/407 message is expanded to 128 byte. With previous software, 64 byte size can be received.

4.4 Supported SIP Trunking Options

The options listed below are supported with SIP Trunking.

- 100rel
- Session Timer
- Early Media

4.5 Supported CODEC

SIP Trunking can use the following CODECs.

- G.729. Low bandwidth requirement, and is used on most Wide Area Network links.
 - VAD
 - VIF size 20ms~30ms
- G.711 – High bandwidth requirement – usually used on Local Area Networks.
- G.722 – This CODEC is useful in fixed network, Voice over IP applications, where the required bandwidth is typically not prohibitive.
- G.726 – is an ITU-T ADPCM speech coded standard covering the transmission of voice at rates of 16, 24, 32, and 40Kbps.

Section 5 SUPPORTED SIP TRUNKING FUNCTIONS

The functions in this section are supported by NEC SL2100 SIP Trunking.

5.1 Address Resolution

If using any SIP Proxy Server, the NEC SL2100 always sends messages to an external SIP server.

If not using any SIP Proxy Server, the NEC SL2100 uses the internal address table (Program 10-23 : SIP System Interconnection Setup).

When a user creates an interconnection network with SIP trunks, **Program 10-29-14 : SIP Server Information Setup - SIP Carrier Choice** must be set to 0 (default).

5.2 Authentication Process

When using an external SIP Server provided by a carrier, an authentication process is usually needed.

NEC SL2100 SIP trunks support HTTP digest authentication process (MD5). This process is done on a Register process and Initial INVITE process.

5.3 Caller ID

Caller ID for SIP Trunks is set by **Program 21-17 : IP Trunk (SIP) Calling Party Number Setup for Trunk**.

Caller ID for SIP Extensions is **Program 21-19 : IP Trunk (SIP) Calling Party Number Setup for Extension**.

Programs follow program priority as follows: 21-19 > 21-17 > 10-36-02

With a trunk-to-trunk transfer and Trunk-to-Trunk Outgoing Caller ID Through Mode enabled (**Program 14-01-24**), the Caller ID/sub-address (received from the incoming trunk) is sent. If a SIP trunk is connected to any SIP carrier, the sub-address is not transferred.

Calling Party Name is not provided for outgoing calls on SIP trunks.

5.4 Carrier Support

If a common carrier supports SIP, the NEC SL2100 can connect:

- the SIP Carrier
- the outgoing call to the PSTN network
- the common IP network via an NEC SL2100 SIP trunk.

A conformity test with a carrier's SIP server is recommended.

5.5 Early Media

When the NEC SL2100 receives the response *18x w/SDP* and the CODEC negotiation is a success, the NEC SL2100 starts to send/receive RTP packets.

5.6 Fault Tolerance

When the NEC SL2100 uses an external SIP Server and if the registration process fails, the NEC SL2100 blocks outgoing calls with SIP trunks. All SIP trunk ports are placed into a busy status.

If the NEC SL2100 has trunk groups that include both SIP trunks and ISDN trunks, and all SIP trunks are busy, a user can make an outgoing call using an ISDN trunk as a bypass.

5.7 Network Address Port Translation (NAPT)

NEC SL2100 SIP trunk can pass through a NAPT router. The related system data is **Program 10-12-06 : CPU-C1 Network Setup – NAPT Router (On/Off)** and **Program 10-12-07 : CPU-C1 Network Setup – NAPT Router IP Address** (do not set with a dynamic router IP address).

Incorrect settings with these two programs can cause one-way audio problems.

The WAN global IP is set in the system data by the user or automatically using the NAT traversal feature (UPnP). The related system data is **10-37-01 : UPnP Setup – UPnP Mode (On/Off)** and **10-37-02 : UPnP Setup – UPnP Interval** (polling timer).

5.8 Quality of Service (QoS)

NEC SL2100 SIP trunks support Type of Service (ToS) (**Program 84-10 : ToS Setup**, protocol type = 9 (SIP Trunk)).

5.9 Registration

5.9.1 Registration Process

When the NEC SL2100 system registers its own IDs with an external SIP server, the following system data are sent:

| Register ID # | User ID | Authentication ID | Authentication Password |
|----------------|----------|-------------------|-------------------------|
| Register ID 0 | 10-36-02 | 10-36-03 | 10-36-04 |
| Register ID 1 | 10-36-02 | 10-36-03 | 10-36-04 |
| : | 10-36-02 | 10-36-03 | 10-36-04 |
| : | | | |
| Register ID 31 | 10-36-02 | 10-36-03 | 10-36-04 |

The NEC SL2100 sends the REGISTER Message when the system starts up, register timer expires, CPU LAN links and recover timer expires.






5.9.2 Registration Recover Process

The NEC SL2100 has a registration recovery process for registration failure. When a registration fails, the NEC SL2100 sets an internal recover timer. When the timer expires, the NEC SL2100 sends a REGISTER message per register ID again.

The recover timer is either five minutes or 30 minutes. Typically, five minutes is used.

Section 6 SIP TRUNK PROGRAMMING

6.1 SIP Trunk Basic Setup

| Program/ Item No. | Description/ Selection | Assigned Data | Comments |
|----------------------|--|---|--|
| 10-28-01 | SIP System Information Set-up – Domain Name | 64 characters maximum Default not assigned  <i>Select SIP Profile 1-6.</i> | Define the domain name. This information is generally provided by the SIP carrier. |
| 10-28-02 | SIP System Information Set-up – Host Name | 48 characters maximum Default not assigned  <i>Select SIP Profile 1-6.</i> | Define the Host name. This information is generally provided by the SIP carrier. |
| 10-28-03 | SIP System Information Set-up – Transport Protocol | 0 = UDP 1 = TCP 2 = TLS Default = 0  <i>Select SIP Profile 1-6.</i> | Define the Transport type. This option will always be set to UDP. |
| 10-28-05 | SIP System Information Set-up – Domain Assignment | 0 = IP Address 1 = Domain Name Default = 0  <i>Select SIP Profile 1-6.</i> | Define the Domain Assignment. This entry is determined by what information the SIP carrier provides. If the SIP carrier provides a server name: SIPconnect-sca@L0.cbeyond.net the domain would be: @L0.cbeyond.net and the host name would be: SIPconnect-sca . |
| 10-29-14 | SIP Server Information Setup – SIP Carrier Choice | 0 = Default 1 = Carrier A 2 = Carrier B 3 = Carrier C 4 = Carrier D 5 = Carrier E 6 = Carrier F 7 = Carrier G 8 = Carrier H 9 = Carrier I 10 = Carrier J 11 = Carrier K 12 = Carrier L 13 = Carrier M 14 = Carrier N 15 = Carrier O 16 = Carrier P 17 = Carrier Q 18 = Carrier R 19 = Carrier S 20 = Carrier T 21 = Carrier U 22 = Carrier V 23 = Carrier W 24 = Carrier X 25 = Carrier Y 26 = Carrier Z Default = 0  <i>Select SIP Profile 1-6.</i> | Define the SIP Carrier Choice. |
| 10-68-01 | IP Trunk Availability – Trunk Type | 0 = None 1 = SIP 2 = H.323 3 = Reserved | Assign the trunk type as (1) SIP. |

| Program/ Item No. | Description/ Selection | Assigned Data | Comments |
|----------------------|--|--|---|
| 10-68-02 | IP Trunk Availability – Start Port | 0~128 | Assign the Start Port for your SIP trunks. |
| 10-68-03 | IP Trunk Availability – Number of Port | 0~128 | Assign the number of SIP port trunks. |
| 14-18-05 | IP Trunk Data Setup – SIP Profile (SIP Only) | 1 = Profile 1 2 = Profile 2 3 = Profile 3 4 = Profile 4 5 = Profile 5 6 = Profile 6 Default = 1 | Define the SIP profile for each SIP trunk. |
| 22-02-01 | Incoming Call Trunk Setup | 0 = Normal 1 = VRS (second dial tone if no VRS installed) 2 = DISA 3 = DID 4 = DIL 5 = Tie line 6 = Delayed VRS 7 = ANI/DNIS 8 = DID Mode Switching Default = 0 | Define the SIP trunks as type 3 (DID). In addition to the SIP trunk programming, refer to the Direct Inward Dialing (DID) feature in the NEC SL2100 Features and Specifications Manuals and the NEC SL2100 Programming Manual for detailed information. |

6.2 IP DSP Resource




If any IP Address or NIC settings are changed, the system must be reset for the changes to take affect.



| Program/ Item No. | Description/ Selection | Assigned Data | Comments |
|----------------------|--|---|---|
| 10-03-01 | VoIPDB Configuration – Logical Port Number | 0 ~ 128 | The CPU-C1 automatically assigns each blade during installation. READ ONLY |
| 10-03-02 | VoIPDB Configuration – Trunk Type | 0 = H.323 1 = SIP Default = 1 | Assign each trunk type for each DSP Resource. |
| 10-19-01 | VoIP DSP Resource Selection –VoIP DSP Resource Selection | 0 = Commonly used for both IP Extensions and Trunks 1 = Use for IP Extensions 2 = Use for SIP Trunks 5 = Blocked 6 = Common without Unicast Paging 7 = Multicast Paging 8 = Unicast Paging Default: Resource 1 = 1 Resources 2~128 = 0 | Define the criteria for each DSP resource on the VoIP blade. |
| 14-05-01 | Trunk Group | Trunk Group = 0~50 Priority = 1 ~ 128 Default = Group1 for all trunk ports and priorities | Default priorities for trunks 1~128 is 1~128. Assign Trunks to Trunk Groups/ Outbound Priority |
| 22-02-01 | Incoming Call Trunk Setup | 0 = Normal 1 = VRS (second dial tone if no VRS installed) 2 = DISA 3 = DID 4 = DIL 5 = Tie line 6 = Delayed VRS 7 = ANI/DNIS 8 = DID Mode Switching Default = 0 | Set the feature type for the trunk you are programming. (Second dial tone for option 1 if no VRS is installed) |












| Program/Item No. | Description/Selection | Assigned Data | Comments |
|------------------|--|-----------------|--|
| 84-26-01 | VoIP Basic Setup – IP Address | XXX.XXX.XXX.XXX | Assign the VOIPDB Gateway IP Address. 172.16.0.20 |
| 84-26-02 | VoIP Basic Setup – RTP Port Number | 0 ~ 65534 | Default: 10020 |
| 84-26-03 | VoIP Basic Setup – RTCP Port Number (RTP Port Number + 1) | 1 ~ 65535 | Default: 10021 |











6.3 SIP Caller ID












| Program/Item No. | Description/Selection | Assigned Data | Comments |
|------------------|---|--|--|
| 14-01-24 | Basic Trunk Data Setup – Trunk-to-Trunk Outgoing Caller ID Through Mode | 0 = Disable 1 = Enable Default = 0 | Enable/Disable the Trunk-to-Trunk Outgoing Caller ID Through Mode. This option allows Caller ID from the original outside caller to be displayed when a trunk is forwarded off premise. This option can be used only with PRI and SIP trunks. |
| 21-17-01 | IP Trunk (SIP) Calling Party Number Setup for Trunk | 1 ~ 0, *, # (to equal up to 16 digits) Default not assigned | This program assigns the Caller Party Number for each IP trunk. The assigned number is sent to the central office when the caller places an outgoing call. If the Calling Party Number is assigned by both 21-17 and 21-18/21-19, the system uses the entry in 21-18/21-19. |
| 21-19-01 | IP Trunk (SIP) Calling Party Number Setup for Extension | 1 ~ 0, *, # (to equal up to 16 digits) Default not assigned  Select SIP Profile 1-6. | Assign the Calling Party Number for each extension. The assigned number is sent to the central office when the caller places an outgoing call. If the Calling Party Number is assigned by both Program 21-17 and 21-18/21-19, the system uses the data in Program 21-18/21-19. |



6.4 SIP CODEC Trunk

| Program/Item No. | Description/Selection | Assigned Data | Comments |
|------------------|---|---|--|
| 84-13-01 | SIP Trunk CODEC Information Basic Setup – G.711 Audio Frame Number | 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms Default is 2  Select SIP Profile 1-6. | Set the G.711 Audio Frame Number. |
| 84-13-02 | SIP Trunk CODEC Information Basic Setup – G.711 Voice Activity Detection Mode | 0 = Disable 1 = Enable Default is 0  Select SIP Profile 1-6. | Enable/Disable the G.711 VAD Detection Mode. |

| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|------------------|---|--|---|
| 84-13-03 | SIP Trunk CODEC Information Basic Setup – G.711 Type | 0 = A-law 1 = μ -law Default is 1  Select SIP Profile 1-6. | Define the G.711 type. |
| 84-13-04 | SIP Trunk CODEC Information Basic Setup - G.711 Jitter Buffer (min) | 0 ~ 255ms Default is 20  Select SIP Profile 1-6. | Set the minimum G.711 Jitter Buffer. |
| 84-13-05 | SIP Trunk CODEC Information Basic Setup – G.711 Jitter Buffer (average) | 0 ~ 255ms 40ms = Default is 40  Select SIP Profile 1-6. | Set the average G.711 Jitter Buffer. |
| 84-13-06 | SIP Trunk CODEC Information Basic Setup – G.711 Jitter Buffer (max) | 0 ~ 255ms Default is 80  Select SIP Profile 1-6. | Set the maximum G.711 Jitter Buffer. |
| 84-13-07 | SIP Trunk CODEC Information Basic Setup – G.729 Audio Frame Number | 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms 5 = 50ms 6 = 60ms Default is 2  Select SIP Profile 1-6. | Set the G.729 Audio Frame Number. |
| 84-13-08 | SIP Trunk CODEC Information Basic Setup – G.729 Voice Activity Detection Mode | 0 = Disable 1 = Enable Default is 0  Select SIP Profile 1-6. | Enable/Disable the G.729 VAD Detection Mode. |
| 84-13-09 | SIP Trunk CODEC Information Basic Setup –G.729 Jitter Buffer (min) | 0 ~ 300ms Default is 20  Select SIP Profile 1-6. | Set the minimum G.729 Jitter Buffer. |
| 84-13-10 | SIP Trunk CODEC Information Basic Setup – G.729 Jitter Buffer (average) | 0 ~ 300ms Default is 40  Select SIP Profile 1-6. | Set the average G.729 Jitter Buffer. |
| 84-13-11 | SIP Trunk CODEC Information Basic Setup – G.729 Jitter Buffer (max) | 0 ~ 300ms Default is 80  Select SIP Profile 1-6. | Set the maximum G.729 Jitter Buffer. |
| 84-13-17 | SIP Trunk CODEC Information Basic Setup – Jitter Buffer Mode | 1 = Static 3 = Self adjusting Default is 3  Select SIP Profile 1-6. | Set the Jitter Buffer Mode. |
| 84-13-18 | SIP Trunk CODEC Information Basic Setup – VAD Threshold | Entries 0 ~ 30 (-20dBm~10dBm) 1 = -19dB (-49dBm) : 20 = 0dB (-30dBm) : 29 = 9dB (-21dBm) 30 =10dB (-20dBm) Default is 20  Select SIP Profile 1-6. | Set the VAD (voice activity detection) threshold. |

| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|------------------|--|---|----------------------------|
| 84-13-28 | SIP Trunk CODEC Information Basic Setup – Audio Capability Priority | 0 = G.711_PT 1 = Not Used 2 = G.729_PT 3 = G.722 4 = G.726 5 = Not Used 6 = G.711 Only 7 = G.729 Only Default is 0  Select SIP Profile 1-6. | Define the CODEC Priority. |
| 84-13-33 | SIP Trunk CODEC Information Basic Setup – Number of G.722 Audio Frames | 1~4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms Default is 3  Select SIP Profile 1-6. | |
| 84-13-35 | SIP Trunk CODEC Information Basic Setup – G.722 Jitter Buffer (min) | 0~300ms Default is 30  Select SIP Profile 1-6. | |
| 84-13-36 | G.722 Jitter Buffer (average) | 0~300ms Default is 60  Select SIP Profile 1-6. | |
| 84-13-37 | G.722 Jitter Buffer (max) | 0~300ms Default is 120  Select SIP Profile 1-6. | |
| 84-13-38 | Number of G.726 Audio Frames | 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms Default is 3  Select SIP Profile 1-6. | |
| 84-13-39 | G.726 VAD Mode | 0 = Disable 1 = Enable Default is 0  Select SIP Profile 1-6. | |
| 84-13-40 | G.726 Jitter Buffer (min) | 0~300ms Default is 30  Select SIP Profile 1-6. | |
| 84-13-41 | G.726 Jitter Buffer (average) | 0~300ms Default is 60  Select SIP Profile 1-6. | |
| 84-13-42 | G.726 Jitter Buffer (max) | 0~300ms Default is 120  Select SIP Profile 1-6. | |

| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|------------------|---|---|-------------------------|
| 84-13-49 | RTP Filter | 0 = Disable 1 = IP 2 = IP + SSRC Default is 1  Select SIP Profile 1-6. | Set the FAX Relay Mode. |
| 84-33-01 | FAX Relay Mode | 0 = Disable 1 = Enable 2 = Each Port Mode Default is 0  Select SIP Profile 1-6. | |
| 84-33-02 | T.38 Protocol mode | 0 = R/U 1 = U/R 2 = RTP 3 = UDPTL Default is 1  Select SIP Profile 1-6. | |
| 84-33-05 | T.38 RTP Format Payload Number | 96 ~ 127 Default is 100  Select SIP Profile 1-6. | |
| 84-33-06 | T.38 FAX Max Speed | 0 = V.27ter: 4800 bps 1 = V.29: 9600 bps 2 = V.17: 14400 bps Default is 2  Select SIP Profile 1-6. | |
| 84-33-08 | T.38 Error protection depth for Signaling | 0 ~ 2 Default is 0  Select SIP Profile 1-6. | |
| 84-33-09 | T.38 Error protection depth for Data | 0 ~ 2 Default is 0  Select SIP Profile 1-6. | |
| 84-33-10 | T.38 TCF Method | 1 = VOIPDB 2 = G3FE Default is 1  Select SIP Profile 1-6. | |
| 84-13-65 | VAD Negotiation on SDP | 0 = Disable 1 = Enable Default is 0  Select SIP Profile 1-6. | |
| 84-13-66 | Voice Band Data (VBD) | 0 = Disable 1 = Enable Default is 0  Select SIP Profile 1-6. | |
| 84-13-67 | VBD Payload Type | 96 ~ 127 Default is 97  Select SIP Profile 1-6. | |

| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|------------------|------------------------|--|----------|
| 84-34-01 | DTMF Relay Mode | 0 = Disable 1 = RFC2833 2 = H.245 Default is 0  Select SIP Profile 1-6. | |
| 84-34-02 | DTMF Payload Number | 96 ~ 127 Default is 110  Select SIP Profile 1-6. | |


6.5 SIP NAPT Router Setup







| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|------------------|--|---|--|
| 10-12-07 | CPU Network Setup – NAPT Router IP Address | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.254.255.254 192.0.0.1 ~ 223.255.255.254 Default is 0.0.0.0 | Set the NAPT Router IP address. With SIP trunking, the IP address of the WAN side of the router must be entered. |
| 10-12-08 | CPU Network Setup – ICMP Redirect | 0 = Disable 1 = Enable Default is 0 | Enable/Disable ICMP (Internet Control Message Protocol) Redirect. |

6.6 SIP System Interconnection Setup



| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|------------------|---|---|--|
| 10-23-01 | SIP System Interconnection Setup – System Interconnection | 0 = Disable 1 = Enable Default is 0 | Enable/Disable system interconnection. |
| 10-23-02 | SIP System Interconnection Setup – IP Address | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 Default is 0.0.0.0 | Define the IP address of another system. |
| 10-23-04 | SIP System Interconnection Setup – Dial Number | Up to 12 digits (0 ~ 9) Default not assigned | Determine another system call control port number. |





6.7 SIP Protocol

| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|------------------|---|---|--|
| 84-14-06 | SIP Trunk Basic Information Setup – SIP Trunk Port Number | 1 ~ 65535 Default is 5060 = Profile 1 5062 = Profile 2 5090 = Profile 3 5092 = Profile 4 5094 = Profile 5 5096 = Profile 6  Select SIP Profile 1-6. | Set the SIP UA (User Authorized) Trunk port number (Receiving Transport for NEC SL2100 SIP). |







| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|------------------|--|--|--|
| 84-14-07 | SIP Trunk Basic Information Setup – Session Timer Value | 0 ~ 65535 seconds Default is 0  Select SIP Profile 1-6. | Set the Session Timer Value. 0 means session timer is Off |
| 84-14-08 | SIP Trunk Basic Information Setup – Minimum Session Timer Value | 0 ~ 65535 seconds Default is 1800  Select SIP Profile 1-6. | Set the Minimum Session Timer Value. |
| 84-14-09 | SIP Trunk Basic Information Setup – Called Party Information | 0 = Request URI 1 = To Header Default is 0  Select SIP Profile 1-6. | Set the Called Party Information. |
| 84-14-10 | SIP Trunk Basic Information Setup – URL Type | 0 = SIP-URL 1 = TEL-URL Default is 0  Select SIP Profile 1-6. | |
| 84-14-11 | SIP Trunk Basic Information Setup – URL/To HeaderSetting Information | 0 = Proxy Server Domain 1 = SIP UA Domain Default is 0  Select SIP Profile 1-6. | |
| 84-14-15 | SIP Trunk Basic Information Setup – 100rel Settings | 0 = Use Default Settings (100rel included) 1 = Use Opposite Settings (100rel not included) Default is 0  Select SIP Profile 1-6. | |


6.8 SIP Server Information Setup

| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|------------------|---|--|--|
| 10-29-01 | SIP Server Information Setup – Default Proxy (Outbound) | 0 = Off 1 = On Default is 0  Select SIP Profile 1-6. | Define the SIP Proxy setup, Default Proxy (Outbound). When SIP trunking is used, this must be on. If entries are made in Program 10-29-xx for a SIP Server and the SIP Server is then removed or not used, the entries in Program 10-29-xx must be set back to their default settings. Even if 10-29-01 is set to 0, the NEC SL2100 still checks the settings in the remaining 10-29 programs. |
| 10-29-02 | SIP Server Information Setup – Default Proxy (Inbound) | 0 = Off 1 = On Default is 0  Select SIP Profile 1-6. | Define the Default Proxy (inbound). |

| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|------------------|--|--|--|
| 10-29-03 | SIP Server Information Setup – Default Proxy IP Address | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.225.255.254 Default is 0.0.0.0  Select SIP Profile 1-6. | Enter the default Proxy IP Address if the SIP carrier is using an IP Address for the Proxy. In most cases, this is left at the default entry as the domain name is used. |
| 10-29-04 | SIP Server Information Setup – Default Proxy Port Number | 0 ~ 65535 Default is 5060  Select SIP Profile 1-6. | Define the Proxy Port Number. |
| 10-29-12 | SIP Server Information Setup – Proxy Domain Name | 64 characters maximum Default not assigned  Select SIP Profile 1-6. | Define the Proxy Domain Name (NEC SL2100 domain name). |
| 10-29-13 | SIP Server Information Setup – Proxy Host Name | 48 characters maximum Default not assigned  Select SIP Profile 1-6. | Define the Proxy Host name (NEC SL2100 proxy name). |

6.9 SIP Registrar Setup





| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|------------------|--|--|--|
| 10-29-05 | SIP Server Information Setup – Registrar Mode | 0 = None 1 = Manual Default is 0  Select SIP Profile 1-6. | Define the Registrar Mode. This should always be set to manual when using SIP trunking. |
| 10-29-06 | SIP Server Information Setup – Registrar IP Address | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.225.255.254 Default is 0.0.0.0  Select SIP Profile 1-6. | Define the Registrar IP Address. The carrier may provide an IP address. In most cases, a domain name will be used so this entry will be left at the default. |
| 10-29-07 | SIP Server Information Setup – Registrar Port Number | 0 ~ 65535 Default is 5060  Select SIP Profile 1-6. | Define the Registrar Port Numbers. |
| 10-29-11 | SIP Server Information Setup – Registrar Domain Name | 128 characters maximum Default not assigned  Select SIP Profile 1-6. | Define the Registrar Domain Name (normally provided by the SIP carrier). Example: SIPconnect-sca@L0.cbeyond.net |
| 10-29-15 | SIP Server Information Setup – Registration Expiry (Expire) Time | 120 ~ 65535 seconds Default is 3600  Select SIP Profile 1-6. | Define the Registration Expire time – the time allowed to register with the SIP carrier. This should stay at the default entry. |
| 10-29-20 | Authentication Trial | 0 ~ 9 0 = No Authentication Default is 1 (1 Authentication Attempt)  Select SIP Profile 1-6. | |

| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|------------------|------------------------|--|----------|
| 10-29-21 | NAT Router | 0 = Disabled 1 = Enabled Default is 0  Select SIP Profile 1-6. | |

6.10 SIP Server Status

| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|------------------|---------------------------------|--|--------------------------|
| 90-10-01 | System Alarm Setup – Alarm Type | Alarm 14 - CPU-LAN Link Error (IP Layer 1) Assign a Major or Minor alarm status for the LAN link. This program also assigns whether or not the alarm is displayed to a key telephone and whether or not the alarm information is reported to the predefined destination. Alarm 60 - SIP Registration Error Notification Assign a Major or Minor alarm status for the SIP Registration Error. This program also assigns whether or not the alarm is displayed to a key telephone and whether or not the alarm information is reported to the predefined destination. | Define alarms 14 and 60. |

6.11 SIP Trunk Registration Information Setup

| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|------------------|--|--|--|
| 10-36-01 | SIP Trunk Registration Information Setup – Registration | 0 = Disable 1 = Enable Default is 0  Select SIP Profile 1-6. | Enable/Disable the SIP trunk registration. |
| 10-36-02 | SIP Trunk Registration Information Setup – User ID | 32 characters maximum Default not assigned  Select SIP Profile 1-6. | Define the user ID. |
| 10-36-03 | SIP Trunk Registration Information Setup – Authentication User ID | 64 characters maximum Default not assigned  Select SIP Profile 1-6. | Define the authentication user ID. |
| 10-36-04 | SIP Trunk Registration Information Setup – Authentication Password | 32 characters maximum Default not assigned  Select SIP Profile 1-6. | Define the authentication password. |

6.12 SIP UPnP

| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|------------------|-------------------------|---|---|
| 10-37-01 | UPnP Setup – UPnP Mode | 0 = Disable 1 = Enable Default is 0 | Determine whether UPnP (Universal Plug and Play) task starts. If UPnP task starts, it obtains a NAPT router WAN IP Address by using NAT traversal and saves it in Program 10-12-07 automatically. |
| 10-37-02 | UPnP Setup – Retry Time | 0~3600 1~59 cannot be used. Default is 60 | UPnP (Universal Plug and Play) task tries to obtain the WAN IP Address of the NAPT router at the interval defined in this option. |

Section 7 SIP TRUNK E.164 SUPPORT

7.1 Description

With SIP Trunk E.164 Support enabled, the PBX is able to support SIP configurations where the number presentation within the SIP messages is formatted using the E.164 international numbering scheme. Specifically the system is able to handle the + digit when required as the International Access Code.

For example, a normal international SIP call can be dialed and displayed as follows:

Number dialed = 00441202223344

Request-URI: Invite sip: 00441202223344@172.16.18.100 SIP/2.0

With SIP Trunk E.164 Support enabled, the SIP call can be displayed once dialed as:

Request-URI: Invite sip:+441202223344@172.16.18.100 SIP/2.0

This display is a requirement of certain SIP ITSPs (Internet Telephony Service Providers) and may require that PBX handle these calls and modify any SIP messages to the correct format accordingly.

This feature uses the following SIP header fields:

To

From

P-Asserted Identity

P-Preferred Identity

7.1.1 Condition

- E.164 support is applied on the SIP trunk interface.
- E.164 is supported for all carrier choices (Program 10-29-14).
- Netlink multi-carrier support uses E.164 support across all carrier configurations at the secondary nodes.

7.1.2 Default Settings

Disabled

7.2 System Availability

7.2.1 Terminals

All Multiline Terminals

7.2.2 Trunks

IP SIP

7.2.3 Required Component(s)

IP7()-CPU-C1

IP7WW-VoIPDB-C1

SL2100 IP CHANNEL-16 LIC






SL2100 IP TRUNK-01 LIC

7.3 Programming

| Program Number | Program Name/Description | Input Data | Assigned Data | Level | | |
|----------------|---|---|---------------|-------|---|---|
| | | | | 1 | 2 | 3 |
| 10-02-01 | Location Setup – Country Code Enter the country code. | Dial (maximum of four digits) 0 ~ 9, *, # | 1 | ✓ | | |
| 10-02-02 | Location Setup – International Access Codes Enter the international access code. | Dial (maximum of four digits) 0 ~ 9, *, # | No Setting | ✓ | | |
| 44-01-02 | System Options for ARS/F-Route – Dial Tone Simulation Dial digits to be analyzed by the system. Numbering plan for the dial needs to be configured as F-Route in Program 11-01. | Dial (maximum of one digit) 0 ~ 9, *, # cannot be used Profile (1 ~ 6) | No Setting | ✓ | | |
| 44-01-03 | System Options for ARS/F-Route – Tone Type Set simulated DT to type which can change the tone used in Program 44-01-02 and Program 44-02-04. | 0 = Internal Dial Tone 1 = External Dial Tone | 0 | ✓ | | |
| 84-14-13 | SIP Trunk Basic Information Setup – Incoming/ Outgoing SIP Trunk for E.164 When this data is set to 1, then for any outbound SIP calls a + is added as a prefix to the Request-URI, To and From header fields of the SIP message. When it is set to 2 then if the dialed international access code matches the value in Program 10-02-02 this value is removed from the number dialed and the + added as a prefix to the Request-URI, To and From header fields of the SIP Message. | 0 = Off 1 = Mode 1 2 = Mode 2 3 = Mode 3 Profile (1 ~ 6) | 0 | ✓ | | |

7.4 Operation

7.4.1 To Transfer a Call into a Conference:

1. While on a call, press **Hold** key.
2. Dial the Transfer to Conference service code (**PRG 11-12-58**, default = **884**).
 -  *If the telephone does not have the proper COS, a warning tone is sent. After the user hangs up, the system automatically places a Callback to the extension.*
 -  *The display shows the line Number, Number/Name and Extension Name/Number.*
3. Dial the extension number or press a **DSS** key of a telephone in a Conference call.
 -  *If an error tone is heard, Barge-In is disable for the extension and the call cannot go through. Retrieve the call by pressing the flashing line Key or hang up and the call recalls the extension.*
 -  *When the call is transferred into the Conference, an intrusion tone is heard by all parties in the conference, depending on the entries in PRG 20-13-17 and PRG 80-01, and all display Multiline Terminals show the joined party.*
 -  *To cancel the transfer, press the flashing line Key to retrieve the call.*
4. Hang up.

Section 8 SIP TRUNK E.164 CLIP ENHANCEMENT

8.1 Description

With SIP Trunk E.164 Support enabled, the PBX is able to support SIP configurations where the number presentation within the SIP messages is formatted using the E.164 international numbering scheme. Specifically the system is able to handle the + digit when required as the International Access Code.

For example, a normal international SIP call can be dialed and displayed as follows:

Number dialed = 00441202223344

Request-URI: Invite sip: 00441202223344@172.16.18.100 SIP/2.0

With SIP Trunk E.164 Support enabled, the SIP call can be displayed once dialed as:

Request-URI: Invite sip:+441202223344@172.16.18.100 SIP/2.0

This display is a requirement of certain SIP ITSPs (Internet Telephony Service Providers) and may require that PBX handle these calls and modify any SIP messages to the correct format accordingly.

This feature uses the following SIP header fields:

To

From

P-Asserted Identity

P-Preferred Identity

8.1.1 Condition

- E.164 support is applied on the SIP trunk interface.

- E.164 is supported for all carrier choices (Program 10-29-14).
- Netlink multi-carrier support uses E.164 support across all carrier configurations at the secondary nodes.

8.1.2 Default Settings

Disabled

8.2 System Availability

8.2.1 Terminals

All Multiline Terminals

8.2.2 Trunks

IP SIP

8.2.3 Required Component(s)

IP7()-CPU-C1

IP7WW-VoIPDB-C1

SL2100 IP CHANNEL-16 LIC

SL2100 IP TRUNK-01 LIC

8.3 Programming

| Program Number | Program Name/Description | Input Data | Assigned Data | Level | | |
|----------------|---|--|---------------|-------|---|---|
| | | | | 1 | 2 | 3 |
| 10-02-01 | Location Setup – Country Code Enter the country code. | Dial (maximum of four digits) 0 ~ 9, *, # | 1 | ✓ | | |
| 10-02-02 | Location Setup – International Access Codes Enter the international access code. | Dial (maximum of four digits) 0 ~ 9, *, # | No Setting | ✓ | | |
| 10-02-03 | Location Setup – Other Area Access Code Enter the other area access code. | Dial (maximum of four digits) 0 ~ 9, *, # | 9 | | | |
| 84-14-13 | SIP Trunk Basic Information Setup – Incoming/ Outgoing SIP Trunk for E.164 When this data is set to 1, then for any outbound SIP calls a + is added as a prefix to the Request-URI, To and From header fields of the SIP message. When it is set to 2 then if the dialed international access code matches the value in Program 10-02-02 this value is removed from the number dialed and the + added as a prefix to the Request-URI, To and From header fields of the SIP Message. | 0 = Off 1 = Mode 1 2 = Mode 2 3 = Mode 3 Profile (1 ~ 6) | 0 | ✓ | | |

| Program Number | Program Name/Description | Input Data | Assigned Data | Level | | |
|----------------|--|---|---------------|-------|---|---|
| | | | | 1 | 2 | 3 |
| 84-14-16 | <p>SIP Trunk Basic Information Setup – SIP Trunk SIP-URI E.164 Incoming Mode</p> <p>When this data is set to 1, then for any inbound SIP calls that include a + and a country code not defined in Program 10-02-01, delete the + and add the International Access Code in Program 10-02-02. If the country code is a match then delete both the + and country code but do not add the International Access code.</p> <p>When it is set to 2, then for any inbound SIP calls that include a + and a country code not defined in Program 10-02-01, delete the + and add the International Access Code in Program 10-02-02. If the country code is a match then delete both the + and country code and add the Caller ID Edit Code from Program 10-02-03.</p> | <p>0 = Disable</p> <p>1 = Mode 1</p> <p>2 = Mode 2</p> <p>Profile (1 ~ 6)</p> | 0 | ✓ | | |

8.4 Operation

8.4.1 Delete the + only from an incoming SIP INVITE using E.164 numbering scheme:

Table 7-1 Delete + from Incoming SIP INVITE

| Program 84-14-16 | Program 84-14-13 | Description |
|------------------|-----------------------|---|
| 0: Off | 0: Off Or 1: On | When a + is presented as the international access code in a SIP INVITE for incoming calls then delete the + only. |

<Example Output>

Incoming call from: +4902131795770

Displayed in terminal incoming caller history as:

| | |
|-------|---------------|
| 01: | 4902131795770 |
| * 3-5 | 11:17 |
| ↑ | ↓ Store DEL |

Original

8.4.2 Delete and replace the + and matched country code from an incoming SIP INVITE using E.164 numbering scheme:

Table 7-2 Delete + and Country Code from Incoming SIP INVITE

| Program 84-14-16 | Program 84-14-13 | Description |
|------------------|------------------|---|
| 1: Mode 1 | 1: On | <p>With a SIP INVITE for incoming calls. When a + is presented as the international access code along with a country code that DOES NOT match the value in Program 10-02-01, then delete the + and add the international access code value in Program 10-02-02 only.</p> <p>- Or -</p> <p>With a SIP INVITE for incoming calls. When a + is presented as the international access code along with a country code that DOES match the value in Program 10-02-01, then delete the + and country code but DO NOT add the international access code value.</p> |

<Example Output>

Incoming call from: +4902131795770

Program 10-02-02 = 00

Displayed in terminal incoming caller history as:

| | |
|---------------|---------------|
| 01: | 4902131795770 |
| * 3-5 11:17 | |
| ↑ ↓ Store DEL | |
| Original | |

| | |
|----------------------|-----------------|
| 01: | 004902131795770 |
| * 3-5 11:17 | |
| ↑ ↓ Store DEL | |
| Program 10-02-01 = 0 | |

| | |
|-----------------------|-------------|
| 01: | 02131795770 |
| * 3-5 11:17 | |
| ↑ ↓ Store DEL | |
| Program 10-02-01 = 49 | |

Table 7-3 Delete + and Country Code from Incoming SIP INVITE

| Program 84-14-16 | Program 84-14-13 | Description |
|------------------|------------------|---|
| 2: Mode 2 | 1: On | <p>With a SIP INVITE for incoming calls. When a + is presented as the international access code along with a country code that DOES NOT match the value in Program 10-02-01, then delete the + and add the international access code value in Program 10-02-02 only.</p> <p>- Or -</p> <p>With a SIP INVITE for incoming calls. When a + is presented as the international access code along with a country code that DOES match the value in Program 10-02-01, then delete the + and country code but DO NOT add the international access code value.</p> |

<Example Output>

Incoming call from: +4902131795770

Program 10-02-02 = 00

Program 10-02-03 = 9

Displayed in terminal incoming caller history as:

| | |
|---------------|---------------|
| 01: | 4902131795770 |
| * 3-5 11:17 | |
| ↑ ↓ Store DEL | |
| Original | |

| | |
|----------------------|-----------------|
| 01: | 004902131795770 |
| * 3-5 11:17 | |
| ↑ ↓ Store DEL | |
| Program 10-02-01 = 0 | |

| | |
|-----------------------|--------------|
| 01: | 902131795770 |
| * 3-5 11:17 | |
| ↑ ↓ Store DEL | |
| Program 10-02-01 = 49 | |

8.4.3 Making an outgoing call from history of incoming calls:

1. From an idle multiline terminal.
2. Press soft key List.

3. Press soft key CID.
4. Press Speaker.

Section 9 SIP TRUNK KEEP ALIVE USING OPTION MESSAGE

Description

The SL2100 provides support for “SIP Trunk Keep Alive” using OPTION message for all six SIP Profiles, applicable to both IP System Interconnection and SIP Carrier mode.

9.1 Condition

- OPTION Keep Alive works for all SIP Carrier Type (PRG 10-29-14).
- Program 10-29-19 must be enabled for making OPTION Keep Alive to work for SIP Carrier mode.
- Program 10-23-05 must be enabled for making OPTION keep alive to work for IP system Interconnection mode.
- SL2100 sends the OPTION message at the interval of the value set in program 84-14-18.
- OPTION Keep Alive can be sent to the SIP carrier or IP system Interconnection of Net Link Secondary System.
- OPTION Keep Alive Call Restriction and Alarm of Net Link Secondary System are not supported.
- If SL2100 does not receive the 200-OK response from the SIP server then SL2100 would retry sending of OPTION message for 32 seconds.

9.1.2 Default Settings

None

9.2 System Availability

9.2.1 Terminals

None

9.2.2 Trunks

SIP Trunks

9.2.3 Required Component(s)

IP7WW-VoIPDB-C1

SL2100 IP CHANNEL-16 LIC

SL2100 IP TRUNK-01 LIC

9.3 Programming

| Program Number | Program Name/Description | Input Data | Default |
|----------------|--|--|---------|
| 10-23-05 | SIP System Interconnection Setup – Keep Alive Mode for SIP Enable/Disable OPTION Keep Alive Mode for IP system interconnection. | 0 = Disable 1 = Option Keep Alive Mode | 0 |
| 10-23-06 | SIP System Interconnection Setup – SIP Profile Assign the Interconnection to a SIP Profile. | 1 = Profile 1 2 = Profile 2 3 = Profile 3 4 = Profile 4 5 = Profile 5 6 = Profile 6 | 1 |
| 10-29-19 | SIP Server Information Setup – Keep Alive by OPTION Message Enable/Disable support of keep alive by OPTION Message for SIP carrier mode. | 0 = Disable 1 = Enable Profile (1 ~ 6) | 0 |
| 84-14-18 | SIP Trunk Basic Information Setup - Keep Alive by OPTION Interval Timer Define Keep Alive by OPTION Interval Timer. SL2100 sends the OPTION message at the interval of the value set in this program. | 60 ~ 3600 (seconds) Profile (1 ~ 6) | 180 |
| 84-14-19 | SIP Trunk Basic Information Setup - Keep Alive by OPTION Fail Limit Define Keep Alive by OPTION fail limit. | 1 ~ 5 Profile (1~6) | 1 |
| 84-14-20 | SIP Trunk Basic Information Setup - Option Keep Alive User ID Define Option Keep Alive User ID which is set in SIP URL of OPTION message. | Max. 32 String Profile (1 ~ 6) | ping |

9.4 Operation

None

DHCP Client

Section 1 DHCP CLIENT

DHCP Client will access an external DHCP server every time the LAN cable is connected to the CPU/VoIPDB, or when the system is powered up.

The system can receive the following information from the DHCP server: IP Address, Subnet Mask, and Default Gateway.

1.1 Conditions

- The DHCP Server should be configured to provide the system the same IP address every time. For example in the DHCP server extend the lease time to infinite or setup the server to provide the same IP address based on the systems MAC Address.
- DHCP client can set following programs automatically; however other IP related programs (such as PRG 84-26) have to set manually as required.
 - IP Addresses: PRG 10-12-01 (CPU)
 - Subnet Masks: PRG 10-12-02 (CPU)
 - Default Gateway: PRG 10-12-03
- DHCP Client (PRG 10-63) and existing DHCP Server feature (PRG 10-13) can not be used at the same time.

1.2 Default Settings

Disabled

1.3 System Availability

1.3.1 Required Component(s)

IP7[]-CPU-C1

IP7WW-VOIPDB-C1

1.4 Guide to Feature Programming

| Program Number | Program Name | Description/Comments | Assigned Data | Level | | |
|----------------|--|---|--|-------|---|---|
| | | | | 1 | 2 | 3 |
| 10-63-01 | DHCP Client Setting - DHCP Client Mode | Enable/disable the DHCP Client feature. | 0 = Disable 1 = Enable (default = 0) | ✓ | | |
| 10-12-01 | CPU Network Setup - IP Address | Assigns the IP address of the CPU card. | 192.168.0.10 | ✓ | | |

| Program Number | Program Name | Description/Comments | Assigned Data | Level | | |
|----------------|--|--|---|-------|---|---|
| | | | | 1 | 2 | 3 |
| 10-12-02 | CPU Network Setup - Subnet Mask | Assigns the Subnet Mask of the CPU card. | 255.255.255.0 | ✓ | | |
| 10-12-03 | CPU Network Setup – Default Gateway | Assign the default gateway IP address for the CPU. | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 (default = 0.0.0.0) | ✓ | | |
| 10-12-09 | CPU Network Setup – VoIP IP Address | Set for VoIPDB. | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 (default = 172.16.0.10) | ✓ | | |
| 10-12-10 | CPU Network Setup – Subnet Mask-VoIPDB | Define the Media Gateway Subnet Mask Address. | 128.0.0.0 192.0.0.0 224.0.0.0 240.0.0.0 248.0.0.0 252.0.0.0 254.0.0.0 255.0.0.0 255.128.0.0 255.192.0.0 255.224.0.0 255.240.0.0 255.248.0.0 255.252.0.0 255.254.0.0 255.255.0.0 255.255.128.0 255.255.192.0 255.255.224.0 255.255.240.0 255.255.248.0 255.255.252.0 255.255.254.0 255.255.255.0 255.255.255.128 255.255.255.192 255.255.255.224 255.255.255.240 255.255.255.248 255.255.255.252 255.255.255.254 255.255.255.255 (default = 255.255.0.0) | ✓ | | |

Section 2 DHCP SERVER CONFIGURATION EXAMPLE

The example below shows the necessary steps to add options to a Windows server so that the server will provide the IP Address of the SL2100 (PRG 10-12-09).

1. In the DHCP Server, right click on the actual server and select **Set Predefined Options**.

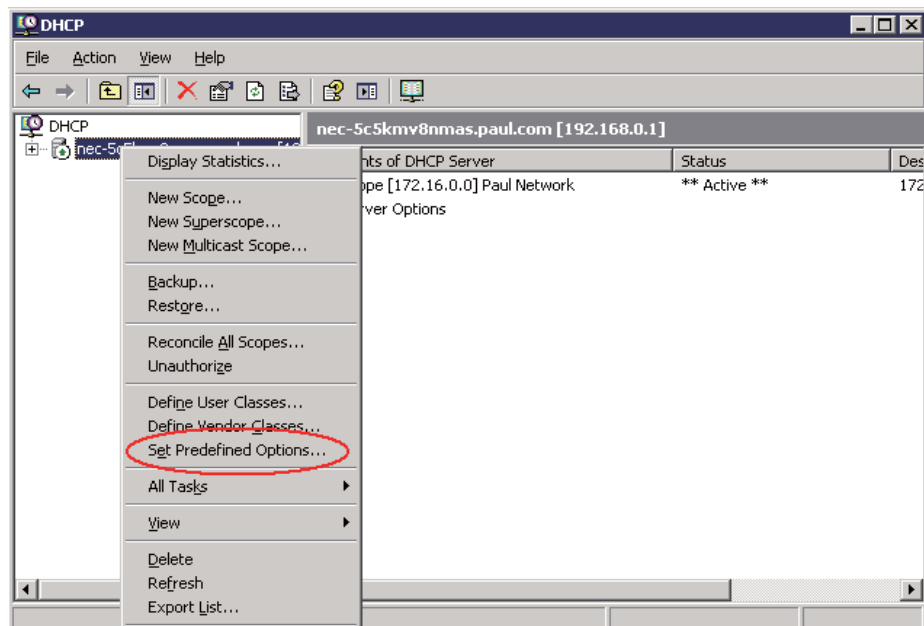
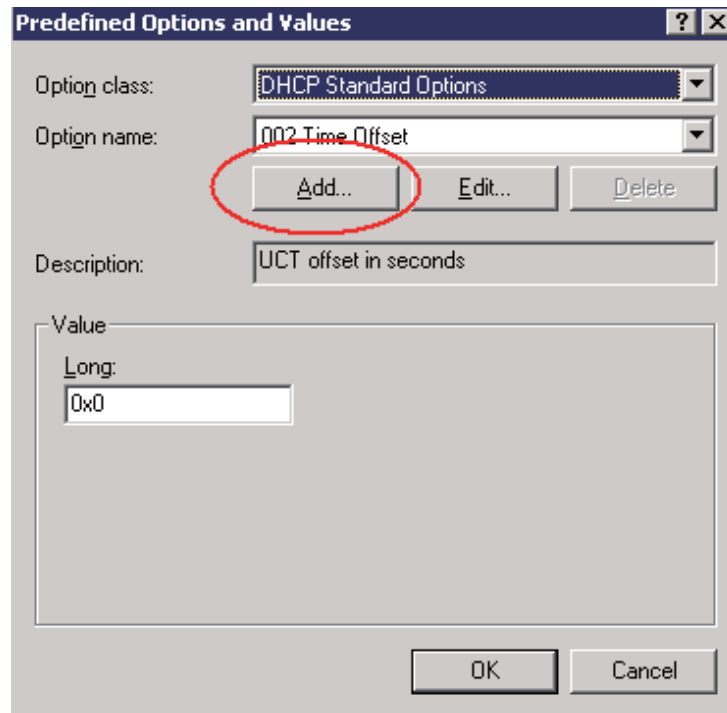


Figure 8-1 DHCP - Set Predefined Options

2. After clicking Set Predefined Options, a new window pops up. Select **Add** to create the new entry for the SL2100 system. Once the option type window is available, assign the following information:
 - Name = SIP Server
 - Data Type = Binary
 - Code = 120
 - Description = Any description you would like to enter



Predefined Options and Values

Option class: DHCP Standard Options

Option name: 002 Time Offset

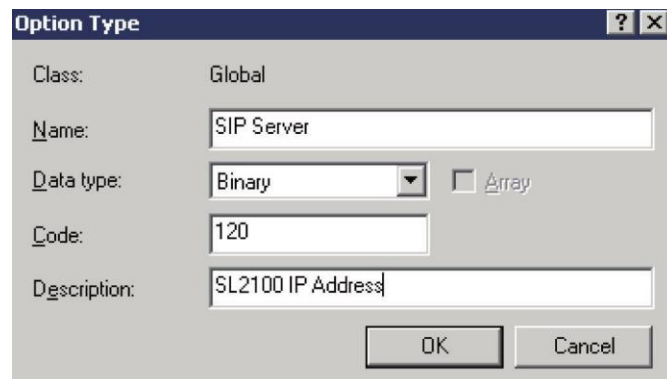
Add... Edit... Delete

Description: UCT offset in seconds

Value

Long: 0x0

OK Cancel



Option Type

Class: Global

Name: SIP Server

Data type: Binary ☐ Array

Code: 120

Description: SL2100 IP Address

OK Cancel

Figure 8-2 DHCP - Predefined Options and Values

3. In the DHCP server, select the scope of options for the DHCP scope that is being configured. Right click on the **Scope Options**, and select **Configure Options**.

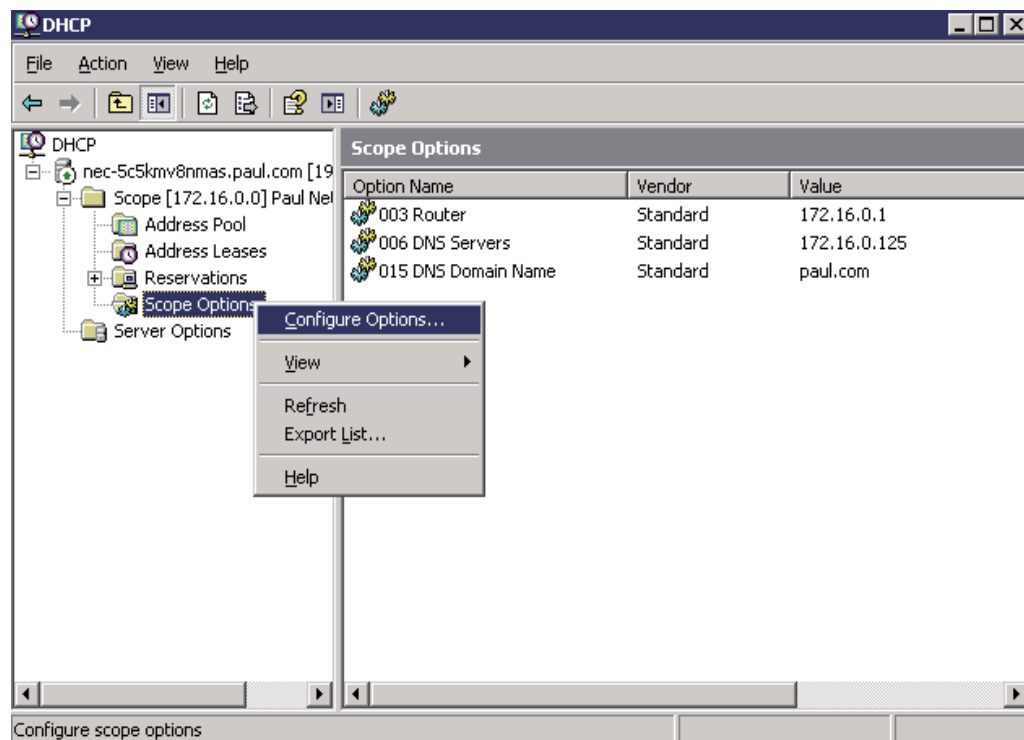


Figure 8-3 DHCP - Scope Options

4. In the Scope Options window, scroll down and place a check mark next to 120 SIP Server. Once the server is added, the data field needs to be changed. In the Data Entry default delete the

default value 00 and add the IP address of the SL2100 system as a Hex value preceded by a 01, for the first SIP server. Listed is an example of what data is to be entered:

- 01 = 1st SIP Server
- AC = Hex for 172
- 10 = Hex for 16
- 00 = Hex for 0
- 0A = Hex for 10

This tells the system that the first SIP server's IP address is: 172.16.0.10. Once assigned, click on **Apply** to update the DHCP server.

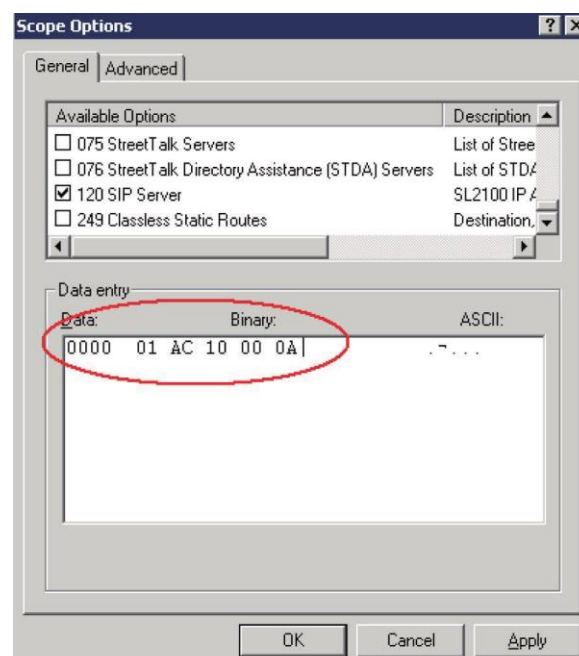
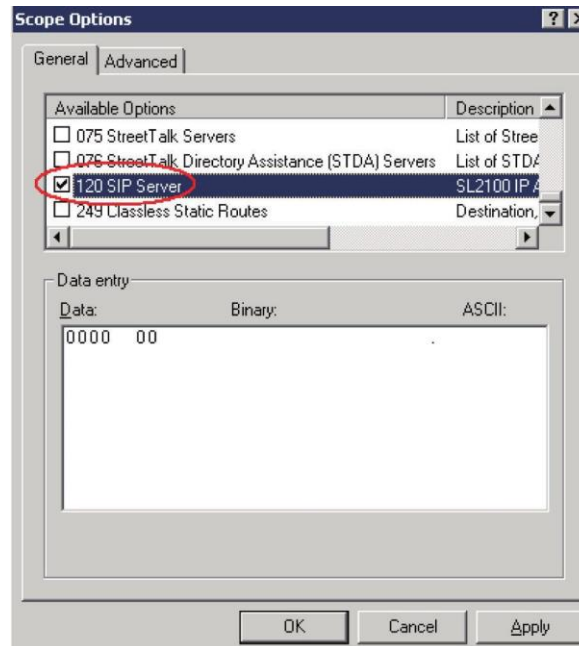


Figure 8-4 DHCP - Data Entry for 1st DHCP Server

IP Multiline Station (SIP)

Section 1 INTRODUCTION

The NEC SL2100 system supports IP extensions. These telephones have the same look and functionality as typical multiline telephones, but they are connected to the CPU via IP rather than by a hardwired connection to a station card.

The following terminal is supported.

- IP7WW-8IPLD-C1 TEL
- ITK-24CG-1P(BK/WH) TEL



The voice quality of VoIP is dependent on variables such as available bandwidth, network latency and Quality of Service (QoS) initiatives, all of which are controlled by the network and Internet Service Providers. Because these variables are not in NEC control, it cannot guarantee the performance of the user's IP-based remote voice solution. Therefore, NEC recommends connecting VoIP equipment through a local area network using a Private IP address.

Section 2 IP TO TDM CONVERSION

When an IP telephone calls a []TXH-A1/[]TXH-B1 multiline telephone, single line telephone or trunk, the speech must be converted from IP to TDM (Time Division Multiplexing) technology. The VoIPDB daughter board provides this function.

It is possible for 8IPLD/DT900 IP Phones to talk directly to other 8IPLD/DT900 IP Phones without using a VoIP DSP resource. For more information, refer to [Peer-to-Peer](#).

2.1 8IPLD/DT900 IP Multiline Telephones

The IP multiline telephone operates in the same way as a digital multiline telephone. It has all features and flexibility you expect from a digital multiline telephone. The difference is that the IP telephone has an RJ-45 for connection to an IP network, rather than an RJ-11 for connection to a digital extension port.



Figure 9-1 8IPLD Telephone

2.2 Conditions

When using 8IPLD/DT900 IP phones, it is not recommended to assign the following features to a large number of phones (16 or more):

- The same Trunk Line assignment (squared key system)
- The same Virtual Extension assignment
- Paging key with LED ON assignment
- The same location Park key
- The same BLF key assignment
- Day Night Mode Change key assignment
- The same VM Mail Box key assignment
- Trunk Group key
- Trunk Group All Line Busy Indication
- One call cannot ring more than 8 simultaneous IP extensions at the same time if the call originates from a ring group or a virtual.

2.3 LAN Connection

As illustrated in [Figure 9-2 Typical Network IP Connection](#), the IP telephone has two RJ-45 connections on the back side marked PC and LAN. This allows the IP telephone and a PC to share one cable run and switch/hub port.

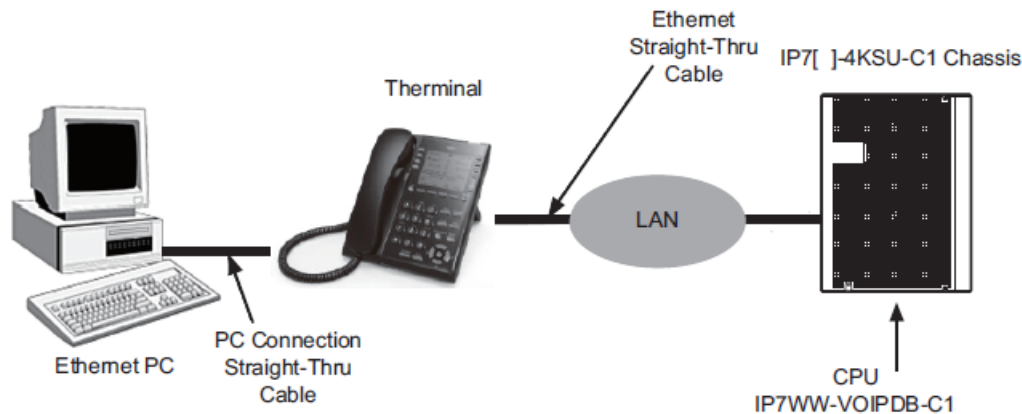


Figure 9-2 Typical Network IP Connection

If installing an IP telephone at a location that already has a PC connected to the data network, it is possible to use either of the following methods:

- Use a different cable and complete the following steps:
 - Leave the PC connected to the LAN.
 - Patch a switch port to a new cable run.
 - Connect a CAT 5 straight-through cable from the wall outlet to the LAN port on the IP telephone.
- Share the existing cable and complete the following steps:
 - Unplug the cable from the PC network card (NIC).
 - Connect that cable to the LAN port on the IP telephone.
 - Connect a new straight-through patch lead from the PC NIC to the PC port on the IP telephone.

Section 3 PROVIDING POWER

IP telephones require power to function. Power can be provided to the IP Terminal by the following methods

3.1 Power Over Ethernet (PoE)

A PoE switch provides power over the spare pairs. The switch can be used with any device (not just IP phones) and detects whether or not power is required. As all phones receive their power from one device, it is easy to protect the IP phones from loss of power (by connecting the PoE switch to a UPS).

3.1 Local Power

The ITK-24CG-1[] TEL has a connector on the back for external power. This is supplied by an AC adapter that outputs +27VDC requiring a separate power outlet per IP telephone. Loss of power in the building will prevent the telephones from functioning.



An external power supply (such as AC adapter) is not supported for IP7[]-8IPLD-C1 TEL



Only use the NEC supplied power supply.

Section 4 PEER-TO-PEER

An IP telephone can send and receive RTP packets to or from another IP telephone without using DSP resources on a VoIPDB. This operation supports only Intercom calls between the IP telephones.

If a 8IPLD/DT900 IP multiline telephone or trunk line is required, a DSP resource is needed and a VoIPDB must be installed. If a conference call is initiated while on a peer-to-peer call, the peer-to-peer connection is released and a new non peer-to-peer connection is created using the VoIPDB. If the third party drops out of the conversation, the call reverts to a peer-to-peer call. There is silence while this conversion is made by the system.

Although the peer-to-peer feature is supported for IP Station-to-IP Station calls, the NEC SL2100 Chassis must still have a registered VoIPDB installed in the system.


With Barge-In, a short silence may occur if the following occurs:

- Peer-To-Peer call receives a Barge-In without a Barge-In tone.
- Peer-To-Peer call receives a Barge-In with Monitor mode.
- Established Barge-In is disconnected.
- The Peer-to-Peer feature is a programmable feature that may be enabled or disabled by accessing Data Program 15-05-50 - Peer-to-Peer Mode.

Section 5 PROGRAMMING

The first step to connecting IP telephones to the NEC SL2100 system is to connect the NEC SL2100 system to the customer data network. Refer to [General IP Configuration](#). Next, program the VoIPDB and associated IP telephone settings. To complete the installation, program the IP telephone.

The programming commands required to complete this installation are located in [Programming](#). This section provides a brief description of the commonly used commands:

- **10-12-03 CPU Network Setup - Default Gateway**
If required, select the default gateway IP address to use when using a router (default: 0.0.0.0).
 - **10-12-09 CPU Network Setup - IP Address**
Select the IP address for the IP connection
 - **10-12-10 CPU Network Setup - Subnet Mask**
Assign the subnet mask for the VoIPDB
 - **15-05-50 Peer to Peer Mode**
Enable/Disable the Peer-to-Peer feature between IP Stations.
-  *Disabling this feature results in IP Station-to-IP Station calls using DSP Resources.*
- **15-05-15 IP Telephone Terminal Basic Data Setup - CODEC Type**
For each IP telephone, set which CODEC Type to use.
 - **84-26-01 VoIPDB Basic Setup - IP Address**
Enter the IP address for the VoIPDB.
 - **84-26-02 VoIPDB Basic Setup - RTP Port Number**
Enter the RTP port number range.

5.1 Music on Hold

Music on Hold is provided by the IP telephone. The settings in Program 10-04 : Music on Hold Setup are ignored except to determine whether or not music is to be provided. If 10-04-02 is set to 0, no Music on Hold is heard. If 10-04-02 is set to 1 or 2, music is provided by the IP telephone.

Section 6 CONFIGURATION EXAMPLES

The examples below show typical scenarios and basic programming required. These examples assume that the programming steps below are performed on a default system (i.e., no existing configuration).

6.1 Example Configuration 1 - Static IP Addressing, One LAN

This example shows IP Phone connected to a single LAN (no routers), with static IP Addresses.

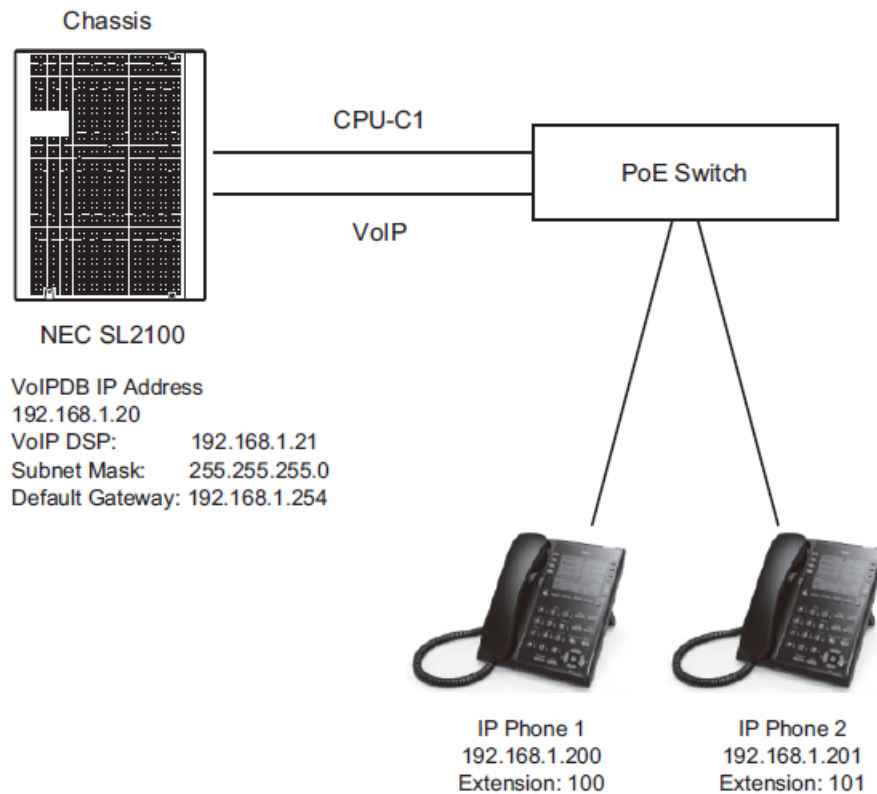


Figure 9-3 Example Configuration 1 - Static IP Addressing, One LAN

Programming - CPU:

| | |
|---|---------------|
| 10-12-01 : CPU Network Setup - IP Address (for CPU) | 0.0.0.0 |
| 10-12-10 : CPU Network Setup - Subnet Mask | 255.255.255.0 |
| 10-12-03 : CPU Network Setup - Default Gateway | 192.168.1.254 |
| 10-12-09 : CPUNetwork Setup - VoIPDB IP Address | 192.168.1.20 |

Programming - VoIP DSP Resource:

| | |
|---|--------------|
| 84-26-01 : VoIP Basic Setup - IP Address (Slot # - DSP) | 192.168.1.21 |
|---|--------------|

Programming - IP Phones:

| | |
|--------------------|---------------|
| DHCP Mode | Disabled |
| IP Address | 192.168.1.200 |
| Subnet Mask | 255.255.255.0 |
| Default Gateway | 192.168.1.254 |
| 1st Server Address | 192.168.1.20 |
| 1st Server Port | 5080 |

6.2 Example Configuration 2 - Dynamic IP Addressing, One LAN

These example shows System IP Phones connected to a single LAN (no routers) with dynamic IP Addresses. The DHCP server could be:

- Customer supplied (e.g., Windows server)
- inDHCP internal DHCP server

In this case, additional programming would be required. Refer to [General IP Configuration](#).



Built-in VoIP (by CPU-C1) also be supported for In-DHCP feature.

6.2.1 In-DHCP Ethernet port Only Configuration

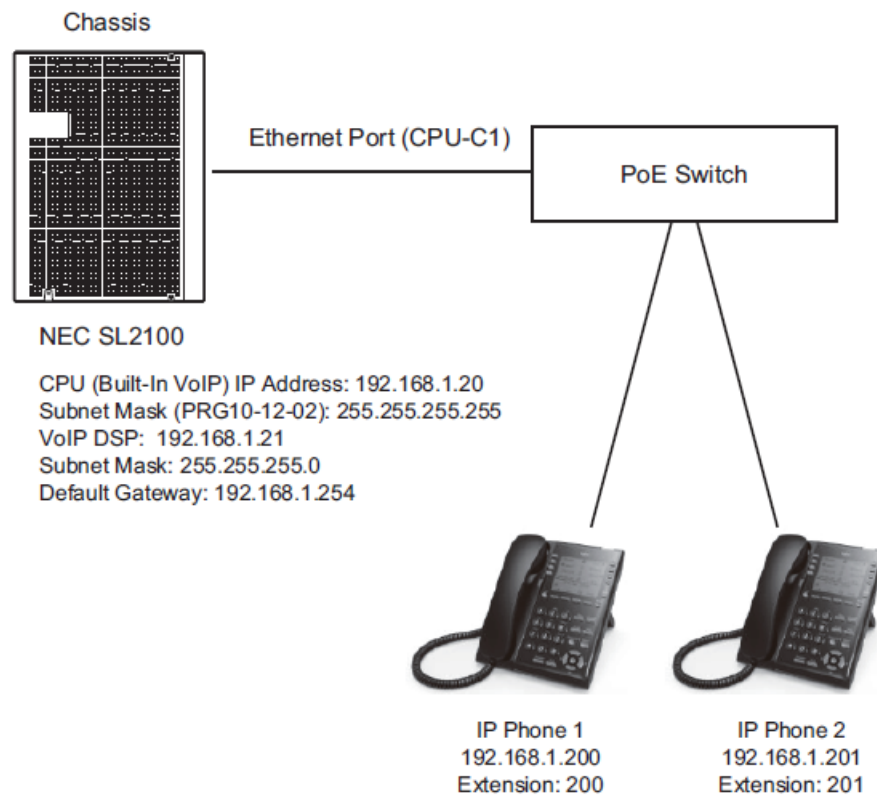


Figure 9-4 Example Configuration 2-1 - Dynamic IP Addressing, One LAN

Programming - CPU:

| | |
|-------------------------------------|-----------------|
| 10-12-01 : IP Address (for CPU) | 192.168.1.10 |
| 10-12-02 : Subnet Mask | 255.255.255.255 |
| 10-12-03 : Default Gateway | 192.168.1.254 |
| 10-12-09 : VoIP IP Address | 192.168.1.20 |
| 10-12-10 : VoIP IP Subnet Mask | 255.255.255.0 |
| 10-13-01 : DHCP Server Mode | Enable |
| 10-14-01 : Min | 192.168.1.200 |
| 10-14-02 : Max | 192.168.1.250 |
| 10-16-16 : SIP Server IP Address | 192.168.1.20 |
| 10-16-27 : SIP Server Receiver Port | 5080 |

Programming – VoIP DSP:

| | |
|--|--------------|
| 84-26-01 : VoIP Basic Setup - IP Address | 192.168.1.21 |
|--|--------------|

Programming - IP Phone 1:

| | |
|--------------------|---------------|
| DHCP Mode | Enabled |
| IP Address | 192.168.1.200 |
| Subnet Mask | 255.255.255.0 |
| Default Gateway | 192.168.1.254 |
| 1st Server Address | 192.168.1.20 |
| 1st Server Port | 5080 |

Programming - IP Phone 2:

| | |
|------------|---------------|
| DHCP Mode | Enable |
| IP Address | 192.168.1.201 |

| | |
|--------------------|---------------|
| Subnet Mask | 255.255.255.0 |
| Default Gateway | 192.168.1.254 |
| 1st Server Address | 192.168.1.20 |
| 1st Server Port | 5080 |

6.2.2 In-DHCP VoIPDB port Configuration

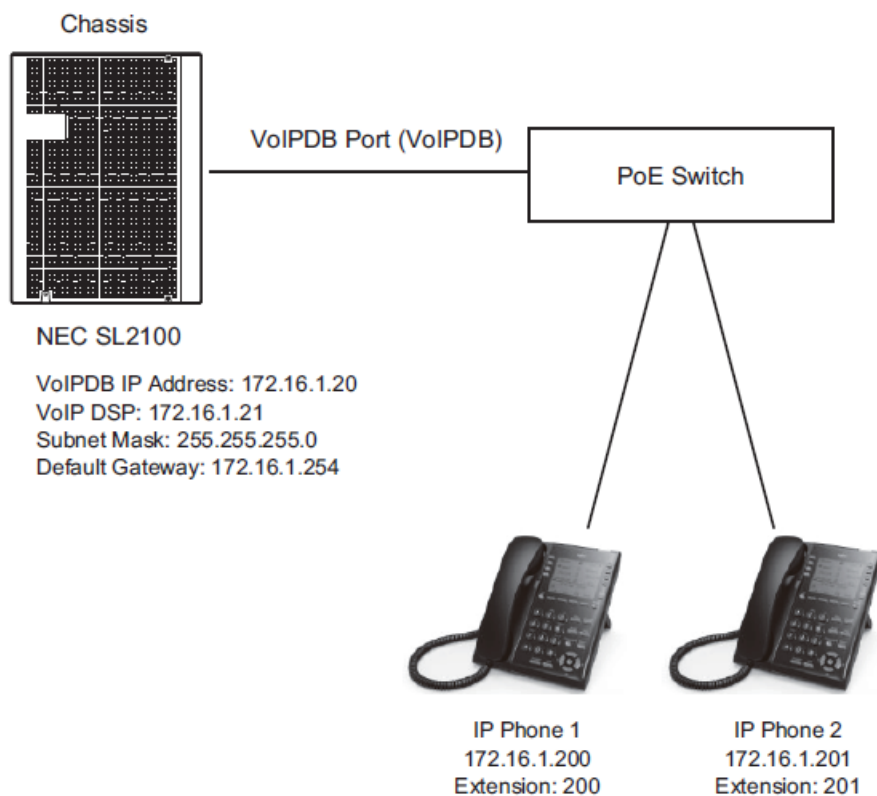


Figure 9-5 Example Configuration 2-2 - Dynamic IP Addressing, One LAN

Programming - CPU:

| | |
|--|---------------|
| 10-12-01 : IP Address (for CPU) | 0.0.0.0 |
| 10-12-02 : Subnet Mask | - |
| 10-12-03 : Default Gateway | 172.16.1.254 |
| 10-12-09 : VoIP IP Address | 172.16.1.20 |
| 10-12-10 : VoIP IP Subnet Mask | 255.255.255.0 |
| 10-13-01 : In-DHCP Server Setup - DHCP Server Mode | Enable |
| 10-14-01 : Min | 172.16.1.200 |
| 10-14-02 : Max | 172.16.1.250 |
| 10-16-16 : SIP Server IP Address | 172.16.1.10 |
| 10-16-27 : SIP Server Receiver Port | 5080 |

Programming – VoIP DSP:

| | |
|----------------------------------|-------------|
| 84-26-01 : VoIPDB DSP IP Address | 172.16.1.21 |
|----------------------------------|-------------|

Programming - IP Phone 1:

| | |
|-------------|---------------|
| DHCP Mode | Enabled |
| IP Address | 172.16.1.200 |
| Subnet Mask | 255.255.255.0 |

| | |
|----------------------------------|---------------|
| Default Gateway | 172.16.1.254 |
| 1st Server Address | 172.16.1.20 |
| 1st Server Port | 5080 |
| Programming - IP Phone 2: | |
| DHCP Mode | Enable |
| IP Address | 172.16.1.201 |
| Subnet Mask | 255.255.255.0 |
| Default Gateway | 172.16.1.254 |
| 1st Server Address | 172.16.1.20 |
| 1st Server Port | 5080 |

6.3 Example Configuration 3 - Static IP Addressing, Routed WAN

This example shows IP Phones connected to an NEC SL2100 system over a Wide Area Network (WAN), with static IP addressing. This is a typical scenario - a small branch office connecting to the main office.

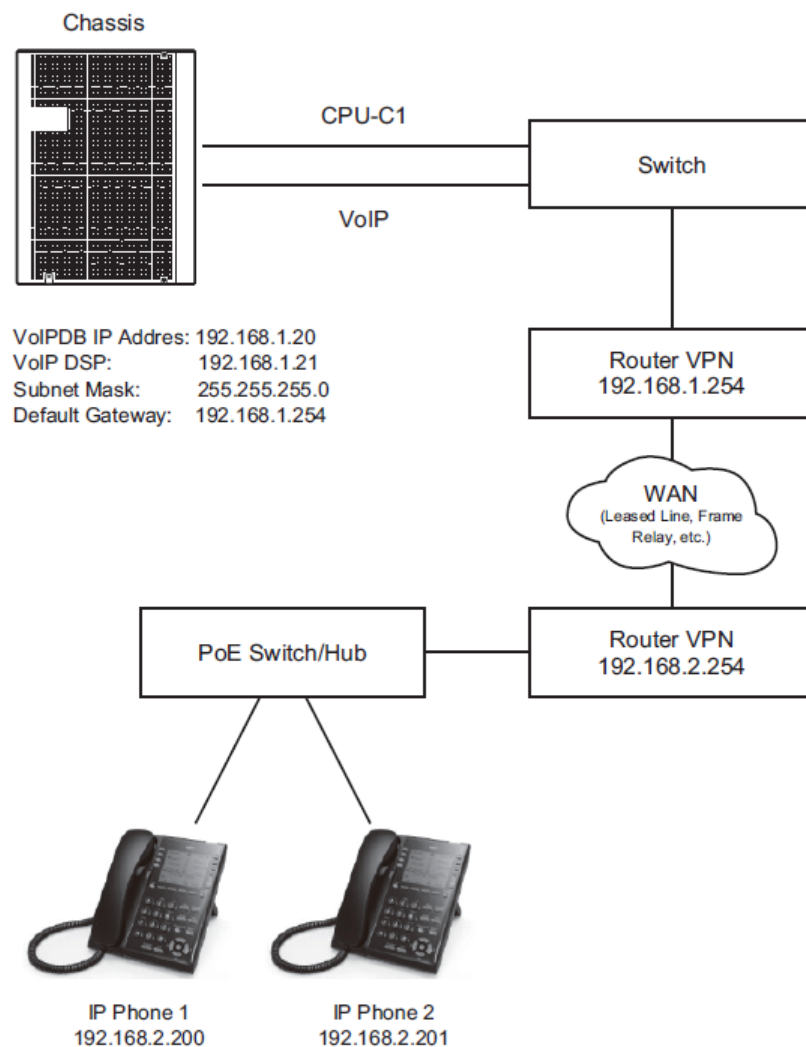


Figure 9-6 Example Configuration 3 Static IP Addressing, Routed WAN

Programming - CPU:

| | |
|---|---------------|
| 10-12-01 : CPU Network Setup - IP Address | 0.0.0.0 |
| 10-12-10 : CPU Network Setup - Subnet Mask | 255.255.255.0 |
| 10-12-03 : CPU Network Setup - Default Gateway | 198.168.1.254 |
| 10-12-09 : CPU Network Setup - VoIPDB IP Address | 192.168.1.20 |
| Programming - VoIPDB DSP: | |
| 84-26-01 : VoIPDB Basic Setup - IP Address (Slot # - DSP) | 192.168.1.21 |
| Programming - IP Phones: | |
| DHCP Mode: | Disabled |
| IP Address: | 192.168.2.200 |
| Subnet Mask: | 255.255.255.0 |
| 1st Server Address: | 198.168.1.20 |
| 1st Server Port: | 5080 |

Section 7 IP PHONE PROGRAMMING INTERFACE

This section describes how to access the programming interface for IP Phones. The following describes how to access the User Menu.

1. Using a 8IPLD/DT900 telephone, press **HOLD-Transfer-*#** buttons to enter program mode. The IP User Menu is displayed.
2. On the IP User Menu, enter the user name and password for the IP Phone. Settings are listed in [Table 9-1 IP Phone Programming Options User Menu](#).

Table 9-1 IP Phone Programming Options User Menu

| Programming Option | Default |
|--------------------|---------|
| UserName | ADMIN |
| Password | 6633222 |

Section 8 DHCP SERVER CONFIGURATION

It is possible to use either an external DHCP server (e.g., Windows Server) or the NEC SL2100 internal DHCP server. With IP Phones, either of these options requires the DHCP server to be configured to supply the IP terminal options.

If using the internal DHCP server, enable the DHCP server. Refer to [Example Configuration 2 - Dynamic IP Addressing, One LAN](#).

When using an external DHCP server, you must add a new Option Code to the DHCP scope for the VoIPDB IP address. The method for adding this service varies depending on the DHCP server used.

Section 9 CONFIGURING QUALITY OF SERVICE

NEC recommends configuring Quality of Service (QoS) to ensure that the speech quality is not affected by other data on the network. QoS is discussed further in [Network Design Considerations](#).

The NEC SL2100 system supports:

- IEEE802.1q (VLAN Tagging)
- IEEE802.1p (Priority)

With the VLAN tagging mode, the NEC SL2100 system can handle packets with or without a VLAN tag. If the VLAN ID of a packet is different from the registered one, that packet is dropped.

9.1 Layer 2 Priority Control

Layer 2 priority control can be enabled on an Ethernet switch, if it supports VLAN tagging. This allows layer 2 prioritization.

Using a switch that supports 802.1p allows:

- priority control
- reduction of unnecessary packets (e.g., broadcast packets) and provides circuitry by restricting a broadcast domain

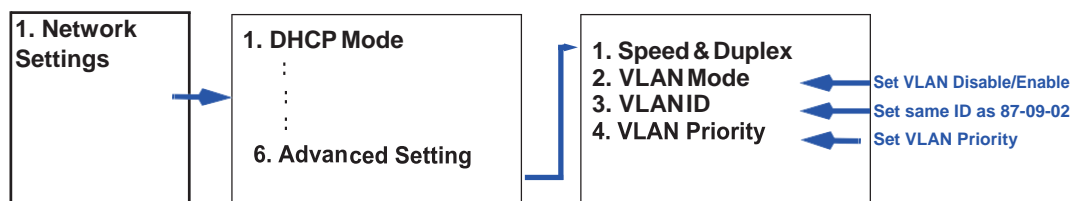
9.1.1 Programming Layer 2 Priority Control

To program Layer 2 Priority Control for IP Phones VLAN/QoS:

1. Press the **Menu** button on the IP Phone to enter the telephone program mode.



This enters the IP program mode to select the settings for the individual phone. The flashing item is the current selection.



2. Press **OK**, **EXIT**, **SAVE** softkeys to save the entries and return the telephone to idle.

9.2 Layer 3 (ToS) Priority Control

The router-supported ToS controls routing priority of packets by following the ToS field. You can give priority to the voice packet using the ToS field.

There are two types of ToS formats: DiffServ and IP Precedence. Before programming the router, make sure to check which type is supported by the router.

The NEC SL2100 system can set the ToS value for each protocol, and Voice Control. This setting allows flexible prioritization.

9.2.1 Programming Layer 3 Priority Control

To program Layer 3 VLAN/QoS use the following programs.

84-10-01 ToS Setup – ToS Mode

For the Protocol type (1= Not Used, 2= Not Used, 3= Reserved, 4= H.323, 5= RTP/RTCP, 6= SIP, 7= Reserved, 8= SIP MLT, 9= SIP Trunk, 10=Reserved, 11=Reserved), select the ToS mode (Disable, IP Precedence or Diffserv).

84-10-07 ToS Setup – Priority (D.S.C.P. - Differentiated Services Code Point)

For the Protocol type (1= Not Used, 2= Not Used, 3= Reserved, 4= H.323, 5= RTP/RTCP, 6= SIP, 7= Reserved, 8= SIP MLT, 9= SIP Trunk, 10=Reserved, 11=Reserved), if Program 84-10-01 is set to 2, select the Diffserv priority (0~63).

Section 10 IP TELEPHONE REGISTRATION AND DELETION

When an IP Phone connects to the NEC SL2100 system, it is assigned the first available port, starting from the value set in Program 11-02-01.

The ports are allocated in blocks of two.

For example:

- Insert a VoIPDB.
- Program 11-02-01 Extension Numbering.
- Configure a System IP Phone and connect to the LAN.

When connecting an IP Phone, the MAC address (ID) is automatically registered in Program 15-05-02. If the registration in Program 15-05-02 is made manually (before connecting the telephone) it uses the assigned port number when the telephone is connected. The MAC address is printed on the barcode label on the bottom of the telephone. It is a 12-digit alphanumeric number, ranging from 0~9 and A~F.

To delete a telephone registration:

Enter Program 90-23-01, and enter the extension number of the IP Phone. If connected to the SL2100 via Telephone Programming, enter a 1 to delete the IP phone and then press **Hold**. If connected to the SL2100 via Web Pro, place a check next to the extension and click **Apply**.

Section 11 SYSTEM IP PHONES AND ANALOG TRUNKS

Due to the nature of analog-to-digital conversion, considerable echo may be encountered when using Analog Trunks with IP Phones.

Due to all Analog trunks being different, padding of the Analog Trunks in PRG 81-07 and 14-01 may be necessary. Even after the pad changes are made, echo may still be present the first few seconds of the call while the echo cancellers are learning the characteristics of the circuit on this call.

It is recommended to use digital trunks when using IP phones for best performance.

Digital (ISDN, T-1, and SIP) trunks do not suffer from this problem.

When analog trunks are installed in an SL2100, PRG 90-68 (Side Tone Auto Setup) must be used. This program will automatically change the setting of 81-07 after multiple tests are performed on the analog line. During these tests the system will not be usable, and the system must be reset once the test are finished.

Anytime new analog trunks are added to the system, PRG 90-68 **MUST** be used.

Section 12 FIRMWARE UPGRADE PROCEDURE

A firmware for the IP Phones can be applied automatically or manually.



Automatically Upgrading Firmware is supported on IP7[J]-8IPLD-C1 TEL and DT800 series. DT900 series are not supported.

The upgrade requires using an FTP/TFTP server. This is a software package that runs on a PC. (These can be downloaded from the Internet, usually as freeware or shareware.)

12.1 Manually Upgrading Firmware

Manually upgrading the firmware uses an FTP/TFTP server, but requires the engineer visit each IP Phone individually. This may take longer, but is more controlled as the downloads can be staggered to avoid excessive bandwidth utilization.

To manually upgrade the firmware:

1. Install and configure an FTP/TFTP server.
2. Copy the firmware file **itlisipr.tgz** to the default FTP/TFTP directory.
3. To enter Programming Mode, press the **Hold, Transfer, *, #** button on the IP Phone.
4. To enter Maintenance Mode, press **3: Maintenance Settings**.
5. To access the Download menu, press **1**.
6. Enter the FTP/TFTP server IP address in Option 2 - Download Address.
7. To enter the protocol, press **#3** (Protocol FTP or TFTP).
8. To select download by file, press option **1** (Download File).
9. To boot the program, press **2** (Boot & Program).
10. Press the softkey.

The IP Phone downloads the firmware from the FTP/TFTP server and reboots when the download is complete.

12.2 Checking the Firmware Version

To check the IP Phone firmware version:

1. Press and hold the **HELP** button on the IP Phone.
2. Press **2** (System Information).
3. Press **5** (Keyset Information).
4. Press the **Next** softkey two times.

12.3 Upgrading Automatically

This procedure causes all IP Phones to attempt firmware upgrade the next time they connect to the CPU. This can make the upgrade procedure easier, as it is not necessary to visit every telephone to perform the upgrade.

This can cause problems if, for example, a PoE (Power over Internet) switch is used. When the PoE switch is powered up, all telephones connect to the FTP/TFTP server at the same time. This causes a large amount of data for the FTP/TFTP server to transfer over the data network.

To avoid this, connect the telephones to the PoE switch gradually, to allow time for each telephone to upgrade before connecting the next.

To enable automatic upgrade:

1. Install and configure an FTP/TFTP server.
2. Copy the firmware file **itlisipr.tgz** to the default FTP/TFTP directory.
3. Set the server mode to FTP/TFTP in Program 84-07-01 and specify the IP address of the FTP/TFTP server (this is the PC where the FTP/TFTP software is installed) in Program 84-07-02.
4. In Program 84-28 program the Firmware Directory path on the FTP/TFTP server and assign the file name (itlisipr.tgz).
5. In Program 90-42, assign the software and hardware versions. An example of the hardware assignment would be 09.01.03.06, this is Hardware Version 9.1.3.6.

Section 13 IP STATION (SIP MULTILINE TELEPHONE)

13.1 Description

The NEC SL2100 system supports IP phones using Session Initiation Protocol (SIP). These IP telephones have the same look and functionality as typical multiline telephones, but they are connected to an VoIPDB card in the CPU rather than hardwired to a Digital Station port.

Some of the benefits of an IP phone over a traditional TDM phone are described in the following list:

- Reduced telephone re-location costs. Unplug an IP telephone at one location and plug it into another VoIP ready jack at another location. The extension number can stay with the telephone, if programmed to do so.
- Multiple users can share the same IP telephone but keep their own personal extensions. With an IP phone you have the ability to log the phone out, then log back in with another extension number. All of your personal settings follow the login ID of your extension.
- The cabling infrastructure can be simplified. There is no longer a need for separate cabling for the phone system. Built into the IP phones is a 2 port 10/100 manageable data switch. The data connection for the PC is available on the back of the IP phone. This built in data switch also supports 802.1Q and 802.1P VLAN tagging abilities. The data and the voice can be tagged separately.
- Allows you to connect remote office workers with a telephone that has the look and same functionality as the telephones connected at the office.

SIP is used for VoIP as defined by the IETF (Internet Engineering Task Force) RFC3261. The NEC SIP MLT uses IP Multiline Station (SIP) (proprietary enhanced SIP protocol) to facilitate the multifunction, multiline telephone.

The SIP Multiline Telephone interfaces directly to the CPU that houses the VoIPDB daughter board. The VoIPDB provides the digital signal processors (DSPs) for IP stations and trunks.

- VoIPDB provides 128 DSPs

A DSP provides format conversion from circuit switched networks (TDM) to packet switched networks (IP). Each voice channel from the circuit switched network is compressed and packetized for transmission over the packet network. In the reverse direction, each packet is buffered for de-jittering, decompressed and sent to the circuit switched network. Each DSP converts a single speech channel from IP to TDM and vice versa.

The following are examples of DSP allocation.

- Calling from an IP phone to a TDM phone uses one DSP.
- Calling from an IP phone to another IP phone, registered to the same CPU, uses no DSPs.
- Calling from a TDM phone to another TDM phone uses no DSPs.
- Calling from a TDM phone and out an IP trunk uses one DSP.



An IP to IP call is peer-to-peer, and no DSP resource is used. When this feature is deactivated, an IP to IP call consumes two DSP resources.

13.2 IP Addressing

When a VoIPDB daughter board is installed in an SL2100 two IP addresses, in the same network, will be required. One IP Address will be for the VoIPDB and this address is assigned in Program 10-12-09. This IP Address is used for IP Phones/Trunks to register and pass signaling messages. The other IP Address is for the DSP's and this address is assigned in Program 84-26-01. This IP Address is used to pass voice traffic.

By default when an SL2100 with a VoIPDB is installed in a customer's network, the SL2100 will try and request IP Addressing information from the local DHCP server (DHCP Client Feature). The DHCP server can provide the SL2100 with the following information:

- IP Address for VoIPDB (PRG 10-12-09)
- Subnet Mask for VoIPDB (PRG 10-12-10)
- Default Gateway for VoIPDB (PRG 10-12-03)

The DHCP server will NOT provide the IP address for the DSP's (PRG 84-26-01). This IP Address will have to be manually configured.

The DHCP Client Feature can be disabled in Program 10-63-01. When this program is changed a reset of the SL2100 is required.

When the VoIPDB is installed, in the SL2100, the CPU NIC card is no longer available. All connections that previously were terminated to the CPU will now terminate to the VoIPDB. It is recommended when adding the VoIPDB to change the IP Address of the CPU NIC card (PRG 10-12-01) to be 0.0.0.0.

13.3 IP Phone Registration Modes

The SIP MLT supports three different registration modes.

- Plug and Play
- Automatic Login
- Manual Login

Plug and Play Registration

Plug and play registration mode allows for no authentication. As long as an IP terminal is configured with the proper IP addressing scheme, and plugged into the network, the phone comes on-line. In plug and play mode you may assign an extension number into the IP terminal or allow the system to automatically set an unused extension number for the station. When the system assigns unused extension numbers it starts searching for the first available port or starts at a preassigned port and works its way up from there.

Automatic Login

When set to automatic login the SIP user name and password must be entered in the configuration in the IP terminal. When the phone tries to register with the CPU it checks the user name and password against its database. If the user name and password match, the phone is allowed to complete registration. If the user name and password do not match, the phone cannot register with the CPU. The IP terminal displays an error message: *Unauthorized Auto Login*.

Manual Login

When set to manual login, no user name, password, or extension number is entered into the configuration of the telephone. The user is prompted to enter this user name and password into the IP phone. This information is cross referenced in the phone system to an associated extension number. If a match is found, the phone comes online. If there is no match, the phone cannot complete registration with the CPU. The IP terminal either returns to the login/password screen, or locks out the user and requires the administrator to unlock the IP terminal. Lockout on failed attempts is dependent on system programming. Manual mode is good for an environment where multiple users share the same IP phone at different times. As one user logs out the next user can login with their credentials and all of their associated programming follows.

In Manual mode a user can also logoff the IP phone to allow another user to login with their own login ID and password.

To logoff the IP Phone when the terminal is set to "Standard" softkey mode (PRG 15-02-60) use the following operation: Press the "Prog" softkey, press the "Down Arrow", press the "Down Arrow", press the "Down Arrow", and then press "Logoff". The IP Phone can also be logged off by resetting the IP terminal.

To logoff the IP Phone when the terminal softkey is set to "Advanced Mode 1/2" (PRG 15-02-60) the IP terminal must be reset.

Encryption

The SL2100 Supports AES 128-bit encryption between IP Multiline Terminals (8IPLD/DT900) and the VoIPDB.

Table 9-2 IP Multiline (8IPLD/DT900) Supported Encryption

| Source | Destination | SRTP | Comment |
|---------------------------------|-----------------------------|------|--|
| IP Multiline | STD SIP (P2P) | N | |
| IP Multiline | STD SIP (Non P2P) | S | The encryption will be between the IP terminal and the VoIPDB. Once the voice leaves the SL2100 to third party SIP terminal the voice cannot be encrypted. |
| IP Multiline | IP Multiline | S | |
| IP Multiline | PSTN | S | The encryption will be between the IP terminal and the VoIPDB. Once the voice leaves the SL2100 to the public network it cannot be encrypted. |
| IP Multiline | IP Network (SIP) | S | The encryption will be between the IP terminal and the VoIPDB. Once the voice leaves the SL2100 to the SIP network it cannot be encrypted. |
| IP Multiline Encryption On | IP Multiline Encryption Off | N' | |
| S = Supported N = Not Supported | | | |

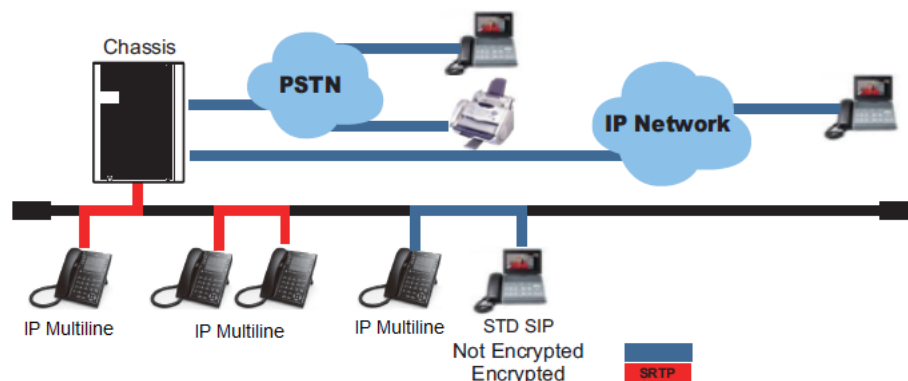


Figure 9-7 IP Multiline (8IPLD/DT900) Encryption

Conditions

- Encryption is not supported on 8IPLD/DT900 series phones that are connected via NAPT.
- If the encryption feature is enabled in terminal programming but not licensed, the terminal displays "Invalid server" and will not function.

13.4 General IP Configuration

The voice quality of VoIP depends on variables such as available bandwidth, network latency, and quality of service initiatives (QoS). These variables are controlled by the network and Internet service providers. Because these variables are not in NEC control, NEC cannot guarantee the performance of the users IP based voice solution. NEC recommends connecting the VoIP equipment through a local area network using private IP addresses.

To ensure a network meets the specific requirements for VoIP implementation, an IP ready check and a site survey must be completed at each site before VoIP implementation.

- One way delay must not exceed 100ms.
- Round trip delay must not exceed 200ms.
- Packet loss must not exceed 1%.
- Data switches must be manageable.
- Routers must provide QOS.
- There must be adequate bandwidth for estimated VoIP traffic. Refer to Section [Bandwidth](#).

Depending on how QOS policies are built in the network, assignments may be needed in both the CPU and IP terminal. The NEC SL2100 supports the flagging of packets at layer 2 (VLAN tagging 802.1Q/802.1P) and at layer 3 levels.

13.5 VLANs

A VLAN is used to logically break up the network and minimize broadcast domains. Without VLANs, the network must be physically segmented to break up broadcast domains. Each network segment is then connected through a routing device adding latency and cost. Latency is a delay in the transmission of data and is caused by routing packets from one LAN to another. In a VoIP environment latency must be kept to a minimum.

802.1Q allows a change in the Ethernet Type value in the Ethernet header tagging the Protocol ID 0x8100, identifying this frame as an 802.1Q frame. This inserts additional bytes into the frame that composes the VLAN ID (valid IDs = 1 ~ 4094).

802.1P allows you to prioritize the VLAN using a 3-bit priority field in the 802.1Q header. Valid VLAN priority assignments are 0 ~7. A tag of 0 is treated as normal data traffic giving no priority. Under normal circumstances the higher the tag numbers, the higher the priority. However this is left up to the network administrator as they could set the exact opposite where the lower tag numbers have a higher priority.

Tagging Voice and Data Packets

Built into the IP phones is a 2 port 10/100 manageable data switch allowing for a PC connection on the back of the IP phone. This built in data switch also supports 802.1Q and 802.1P VLAN tagging capabilities.

The following procedures describe two methods for tagging the voice packets and the data packets separately, using the PC, or using the phone keypad.

13.5.1 Logging In on the PC

1. Web browse to the IP address of phone.
2. To log in, enter default user name: **ADMIN**
3. Enter default password: **6633222**

4. Click **OK**.

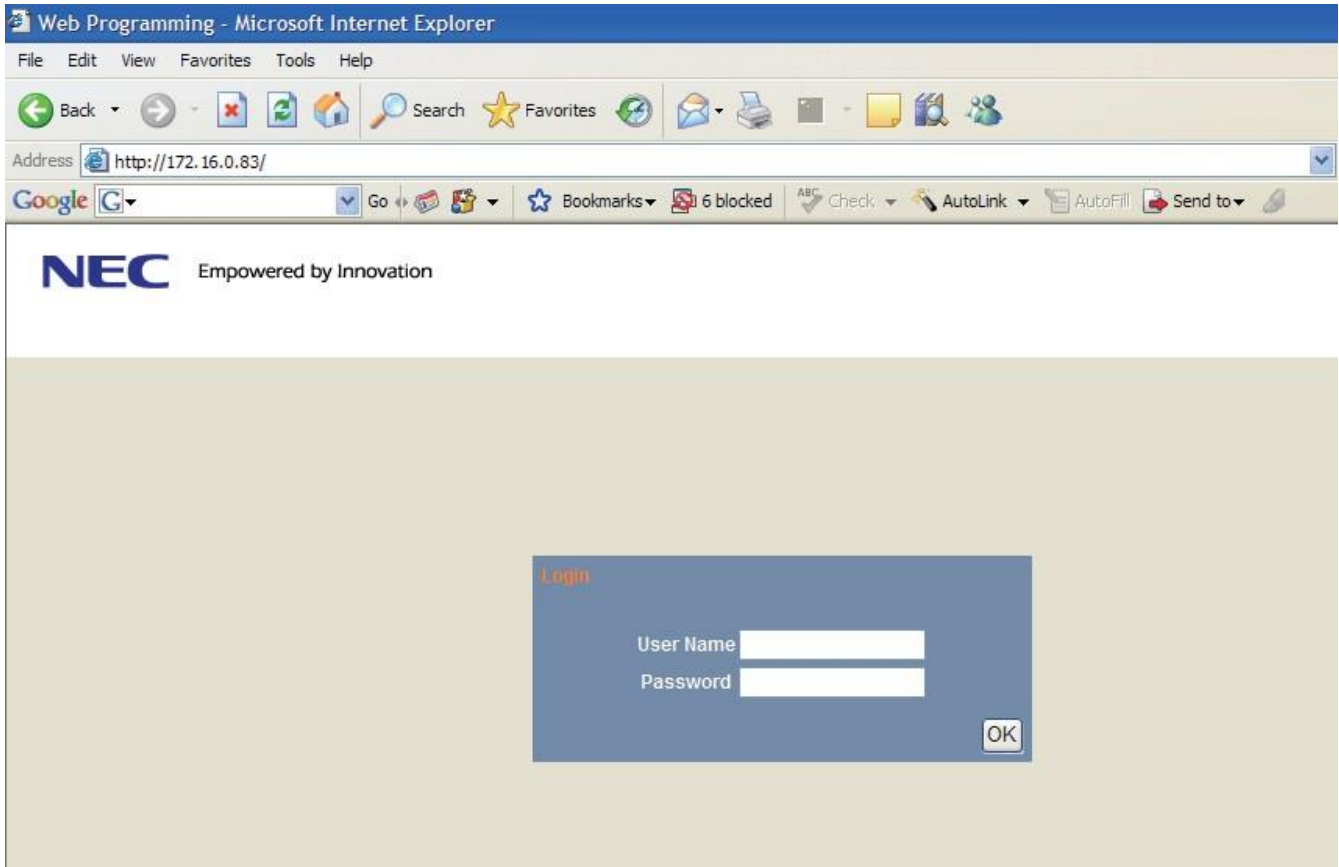


Figure 9-8 Log In to IP Phone

13.5.2 Tagging Voice Packets Using IP Phone

1. To apply a tag to the voice packets only, go to **Network Settings>Advanced Settings>LAN port settings**.

2. Access the following three menus to select options for LAN Port Settings:
 - VLAN Mode
 - VLAN ID
 - VLAN Priority

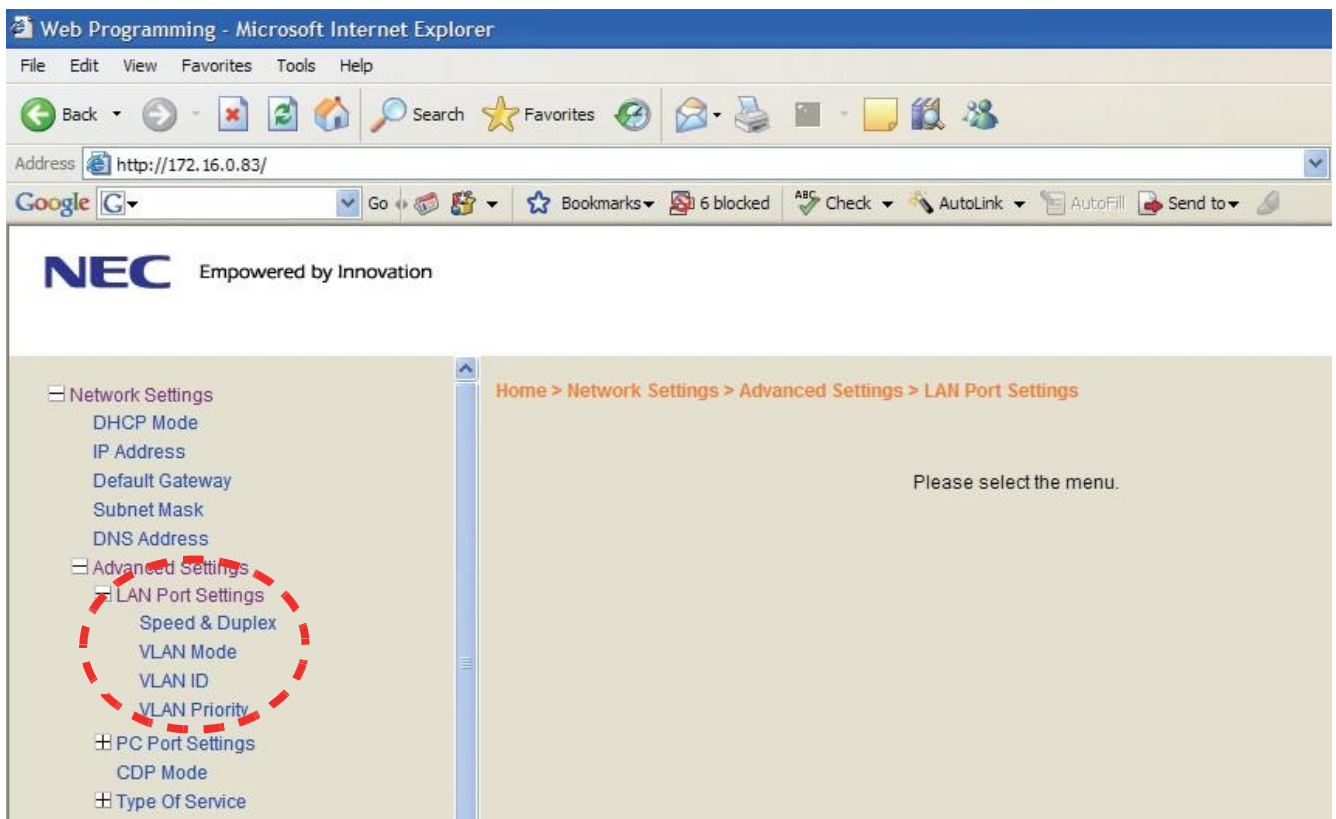


Figure 9-9 LAN Port Settings Window

3. Select the VLAN Mode to enable or disable this feature.
4. Select either Enable or Disable (default) and click **OK**.

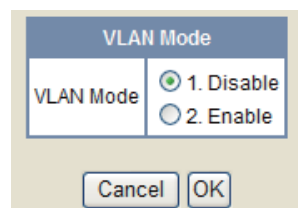


Figure 9-10 VLAN Mode

5. VLAN ID allows an entry of 1~4094 for the VLAN ID. VLAN Mode must be enabled for this entry to be valid.
Enter the VLAN ID and click **OK**.

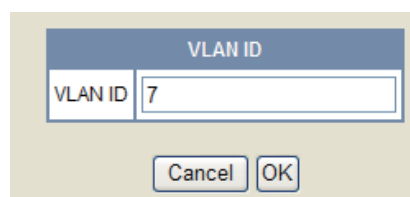


Figure 9-11 VLAN ID

6. VLAN Priority allows an entry of 0~7 for the VLAN Priority. VLAN mode must be enabled for this entry to be valid.
Enter the required priority, and click **OK**.

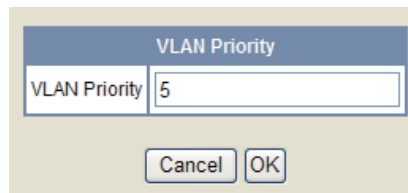


Figure 9-12 VLAN Priority

13.5.3 Tagging Data Packets Using IP Phone

1. While logged in to the IP address of the phone on the PC, go to **Network Settings>Advanced Settings>PC Port Settings**. Refer to Section [Logging In on the PC](#).
2. Access the following three menus to select options for PC Port Settings:
 - Port VLAN Mode
 - Port VLAN ID
 - Port VLAN Priority.

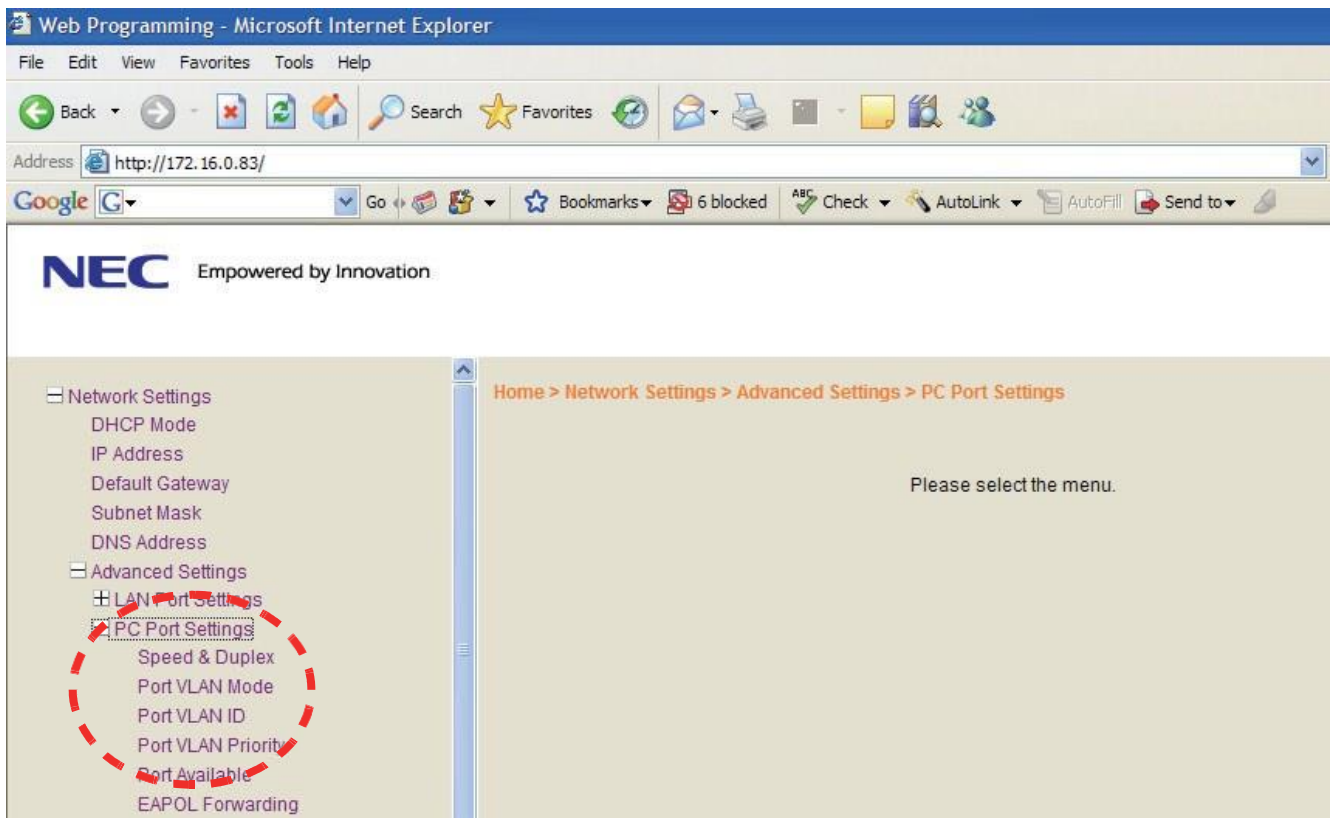


Figure 9-13 PC Port Settings Window

3. Select the VLAN Mode to enable or disable this feature.

4. Select either Enable or Disable (default) and click **OK**.

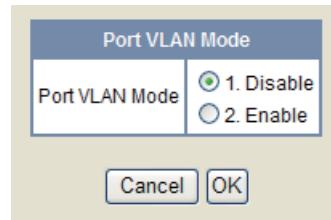



Figure 9-14 Port VLAN Mode

 *The remaining data packets settings for VLAN on the PC Port are the same as those for the voice packets.*

5. VLAN ID allows an entry of 1~4094 for the VLAN ID. VLAN Mode must be enabled for this entry to be valid.
Enter the VLAN ID, and click **OK**.

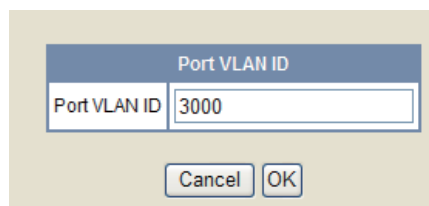


Figure 9-15 Port VLAN ID

6. VLAN Priority allows an entry of 0~7 for the VLAN Priority. VLAN mode must be enabled for this entry to be valid.
Enter the required priority, and click **OK**.

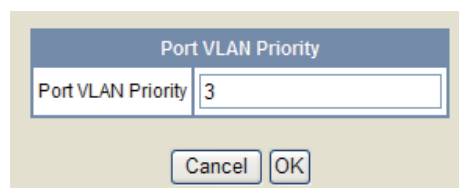


Figure 9-16 Port VLAN Priority

7. Click **Save**.

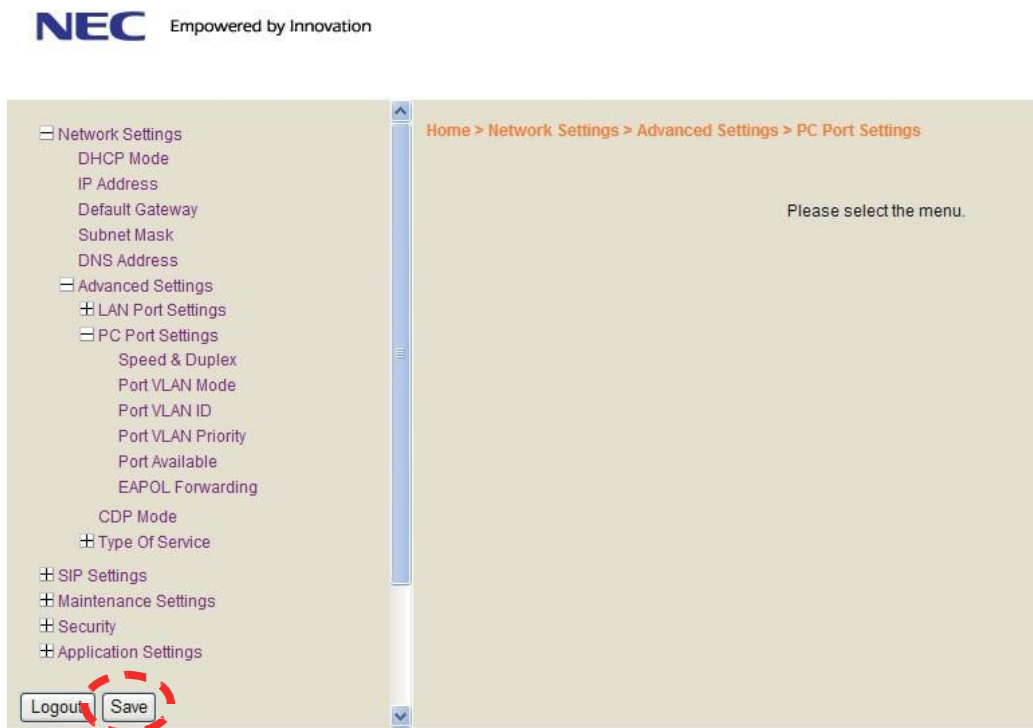


Figure 9-17 Save Network Settings

8. After saving settings, click **OK** to confirm. Telephone reboots and applies the VLAN settings.

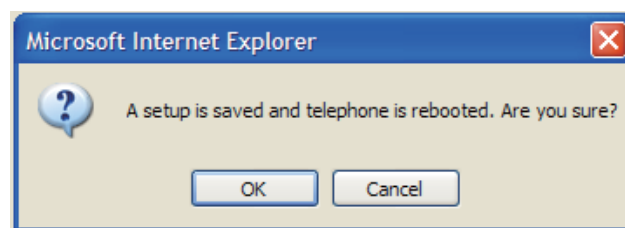


Figure 9-18 Save Confirmation Window

13.5.4 Entering VLAN Settings by Phone (Voice Packets Only)

1. Press **Hold, Transfer, *, #** to enter programming mode.
2. Enter the user name of **ADMIN** and password of **6633222**, then press the **OK** softkey.
3. Dial **1** for Network Settings.
4. Press **6** on the dial pad for Advanced Settings
5. Press **1** on the dial pad for LAN Port Settings (VLAN for the voice packets only)
6. Press **2** on the dial pad for VLAN Mode
7. Press **1** or **2** to disable/enable the VLAN for the voice packets. Press the **OK** soft key after the setting is changed.
8. Press **3** on the dial pad for VLAN ID
9. Enter a valid VLAN ID of 1~4094. Press the **OK** soft key after the setting is changed.
10. Press **4** on the dial pad for VLAN Priority
11. Enter the VLAN priority of 0~7. Press the **OK** soft key after the setting is changed.
12. If no more changes are made, press the **Exit** soft key three times. Then press the **Save** soft key, and the phone reboots.

13.5.5 Entering VLAN Settings for PC Port by Phone (Data Packets Only)

1. Press **Hold, Transfer, *, #** to enter programming mode.
2. Enter the user name of **ADMIN** and password of **6633222**, then press the **OK** softkey.
3. Press **1** on the dial pad for Network Settings.
4. Press **1** or **2** to disable/enable the VLAN for the data packets. Press the **OK** soft key after the setting is changed.
5. Press **3** on the dial pad for Port VLAN ID.
6. Enter a valid VLAN ID of 1~4094. Press the **OK** soft key after the setting is changed.
7. Press **4** on the dial pad for Port VLAN Priority.
8. Enter the VLAN priority of 0~7. Press the **OK** soft key after the setting is changed.
9. If no more changes are made, press the **Exit** soft key three times. Then press the **Save** soft key, and the phone reboots.

13.6 ToS Settings (Layer 3 QoS)

The marking of packets at layer 3 is done by marking the ToS byte in the IP header of the voice packet. The NEC SL2100 supports two methods for marking the ToS byte:

- IP precedence
- DSCP (Diffserv)

IP Precedence

IP Precedence uses the first 3 bits of the ToS field to give eight possible precedence values (0~7). Under normal circumstances the higher the number the higher the priority. However this is left to the network administrator for setup. The administrator may assign this in exactly the opposite manner with the lower values having a higher priority. Below are the eight common values for IP precedence.

- 000 is an IP precedence value of 0, sometimes referred to as routine or best effort.
- 001 is an IP precedence value of 1, sometimes referred to as priority.
- 010 is an IP precedence value of 2, sometimes referred to as immediate.
- 011 is an IP precedence value of 3, sometimes referred to as flash.
- 100 is an IP precedence value of 4, sometimes referred to as flash override.
- 101 is an IP precedence value of 5, sometimes called critical.
- 110 is an IP precedence value of 6, sometimes called internetwork control.
- 111 is an IP precedence value of 7, sometimes called network control.

Working in conjunction with IP precedence, the next 4 bits in the ToS field are designed to influence the delivery of data based on delay, throughput, reliability, and cost. However these fields are typically not used.

The following table shows the 8-bit ToS field and the associated IP precedence bits.

| IP Prece- dence | IP Prece- dence | IP Prece- dence | Delay | Through- put | Reliability | Cost | Not Used |
|----------------------------|----------------------------|----------------------------|-------|-----------------|-------------|------|----------|
| 1(on) here = value of 4 | 1(on) here = value of 2 | 1(on) here = value of 1 | | | | | |

Differential Services Code Point (DSCP)

DSCP stands for Differential Services Code Point (or Diffserv for short). It uses the first 6 bits of the ToS field therefore giving 64 possible values.

The following list shows the most common DSCP code points with their binary values and their associated names:

| DSCP Code Points | Binary Values | Names |
|------------------|---------------|-------------------------------|
| 000000 | 0 | Best Effort (BE) |
| 001000 | 8 | Class Selector 1 (CS1) |
| 001010 | 10 | Assured Forwarding 11 (AF11) |
| 001100 | 12 | Assured Forwarding 12 (AF12) |
| 001110 | 14 | Assured Forwarding 13 (AF13) |
| 010000 | 16 | Class Selector 2 (CS2) |
| 010010 | 18 | Assured Forwarding 21 (AF21) |
| 010100 | 20 | Assured Forwarding 22 (AF22) |
| 010110 | 22 | Assured Forwarding 23 (AF23) |
| 011000 | 24 | Class Selector 3 (CS3) |
| 011010 | 26 | Assured Forwarding 31 (AF31) |
| 011100 | 28 | Assured Forwarding 32 (AF32) |
| 011110 | 30 | Assured Forwarding 33 (AF33) |
| 100000 | 32 | Class Selector 4 (CS4) |
| 100001 | 34 | Assured Forwarding 41 (AF41) |
| 100100 | 36 | Assured Forwarding 42 (AF42) |
| 100110 | 38 | Assured Forwarding 43 (AF 43) |
| 101110 | 46 | Expedited Forwarding (EF) |
| 110000 | 48 | Class Selector 6 (CS6) |
| 111000 | 56 | Class Selector 7 (CS7) |

The following table shows the 8 bit TOS field and the associated Diffserv bits.

| Diffserv | Diffserv | Diffserv | Diffserv | Diffserv | Diffserv | Not Used | Not Used |
|-----------------------------|-----------------------------|-------------------------------|-------------------------------|-------------------------------|-------------------------------|----------|----------|
| 1(on) here = value of 32 | 1(on) here = value of 16 | 1(on) here = value of 8 | 1(on) here = value of 4 | 1(on) here = value of 2 | 1(on) here = value of 1 | | |

IP Precedence/Diffserv Values Submitted in Command 84-10

Assignments for the IP Precedence/Diffserv values in the system are submitted in command 84-10. This setting data affects only the packets sent by the VoIPDB card. This does not affect the packets sent from the IP terminals.

System Data
84-10: ToS Setup

| Protocol Type | ToS Mode | IP Precedence Priority | IP Precedence Delay | IP Precedence Throughput | IP Precedence Reliability | Diffserve |
|---------------|----------|------------------------|---------------------|--------------------------|---------------------------|-----------|
| Networking | Disabled | 0 | Normal | Normal | Normal | 0 |
| RTP/RTCP | Disabled | 0 | Normal | Normal | Normal | 0 |
| SIP | Disabled | 0 | Normal | Normal | Normal | 0 |
| DR700 | Disabled | 0 | Normal | Normal | Normal | 0 |
| SIP Trunk | Disabled | 0 | Normal | Normal | Normal | 0 |

Value should be in the range 0 to 7
This program sets the ToS Data.

Figure 9-19 84-10: ToS Setup

To set the IP Precedence/Diffserv bits for packets leaving the IP terminal there are the following two options:

- **System wide.** If all IP phones use the same ToS value, this can be assigned in commands 84-23-06 and 84-23-12. When an IP phone registers with the CPU, it looks for settings in these commands. If these are found, they override any previous individual settings.
- **Individual.** If different IP phones require different ToS assignments, due to the network configuration, these assignments must be set at each individual station

Command 84-23 requires a Hexadecimal representation of the 8 bit ToS field. For example, to assign the signaling packets an IP precedence value of 4 and the voice packets an IP precedence value of 5, it would be as follows. Refer to [Figure 9-20 SIP MLT Basic Setup](#).

- 80 in Hex is 10000000 - This represents the signaling packets leaving the IP phone
- A0 in Hex is 10100000 - This represents the voice packets leaving the IP phone

System Data
84-23 : DR700 Basic Setup

| | |
|--|-----------------------------------|
| 01 - Registration Expiry Time | <input type="text" value="180"/> |
| 02 - Subscribe Expiry Time | <input type="text" value="3600"/> |
| 03 - Session Expiry Time | <input type="text" value="180"/> |
| 04 - Minimum Session Expiry Time | <input type="text" value="180"/> |
| 05 - Invite Expiry Time | <input type="text" value="180"/> |
| 06 - Type of Service | <input type="text" value="80"/> |
| 07 - Registration Failure Display Time | <input type="text" value="0"/> |
| 08 - Digest Authorization Registration Expiry Time | <input type="text" value="0"/> |
| 10 - Password Retry Count | <input type="text" value="0"/> |
| 11 - Password Lock Time | <input type="text" value="0"/> |
| 12 - Inquiry Number | <input type="text"/> |
| 13 - Media ToS | <input type="text" value="A0"/> |
| 14 - Refer Expiry Time | <input type="text" value="60"/> |

Value should be in the range 0 to 65535

Figure 9-20 SIP MLT Basic Setup

The following table shows the common IP Precedence/Diffserv values and their hexadecimal equivalent.

Table 9-3 Common IP Precedence/Diffserv Values and Hexadecimal Equivalent

| IP Precedence Name | Hex Value |
|--------------------|-----------|
| IP Precedence 1 | 20 |
| IP Precedence 2 | 40 |
| IP Precedence 3 | 60 |
| IP Precedence 4 | 80 |
| IP Precedence 5 | A0 |
| IP Precedence 6 | C0 |
| IP Precedence 7 | E0 |
| DSCP Name | Hex Value |
| DSCP 10 | 28 |
| DSCP 12 | 30 |
| DSCP 14 | 38 |
| DSCP 16 | 40 |
| DSCP 18 | 48 |
| DSCP 20 | 50 |
| DSCP 22 | 58 |
| DSCP 24 | 60 |

| | |
|---------|----|
| DSCP 26 | 68 |
| DSCP 28 | 70 |
| DSCP 30 | 78 |
| DSCP 32 | 80 |
| DSCP 34 | 88 |
| DSCP 36 | 90 |
| DSCP 38 | 98 |
| DSCP 46 | B8 |
| DSCP 48 | C0 |
| DSCP 56 | E0 |

13.6.1 Enter Values on a Per Phone Basis

- By the web browser
- By configuration mode through the dial pad

To enter the values per phone, browse to the individual phone or enter the configuration mode through dial pad.

The following example describes assigning these fields via the web browser.

1. Log in on PC. Refer to Section [Logging In on the PC](#).

Go to Network Settings >Advanced Settings >Type Of Service.

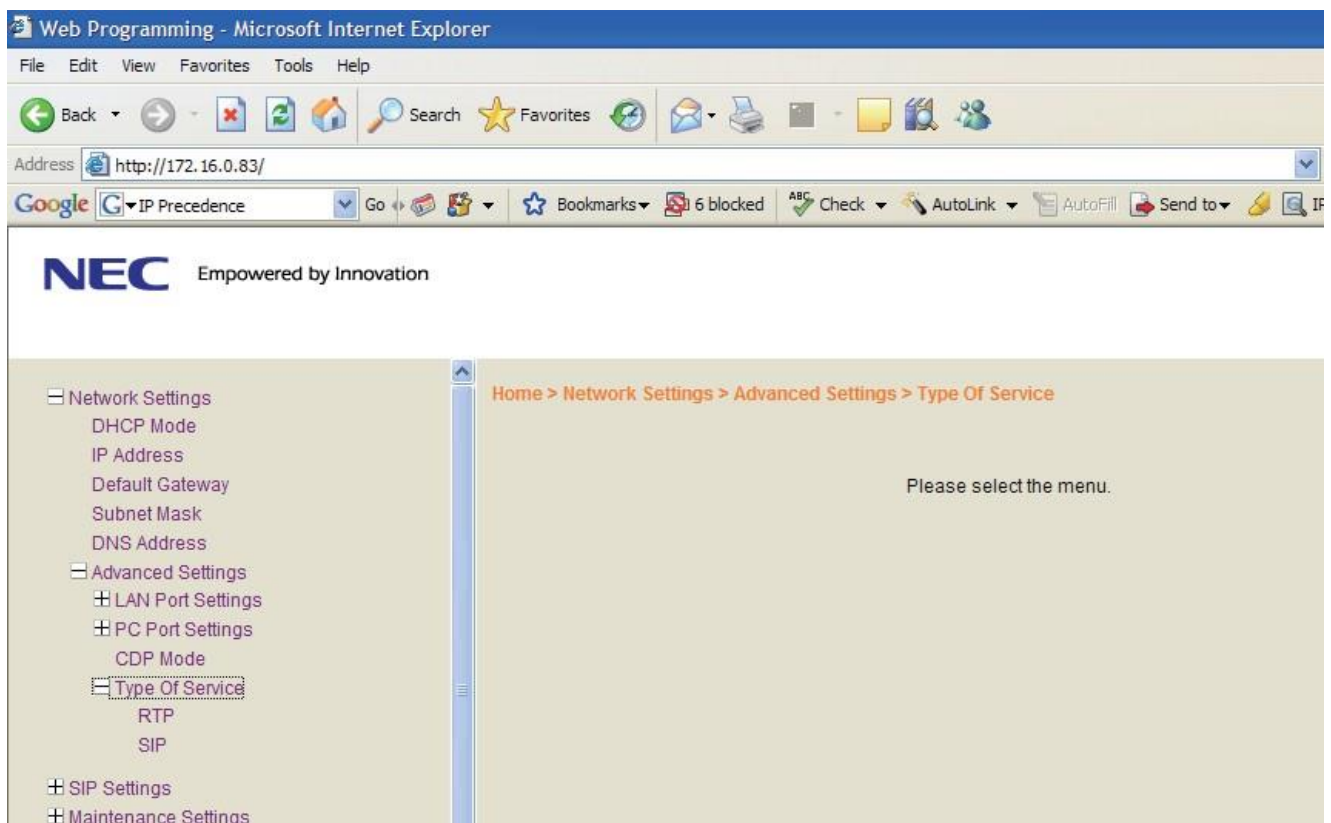


Figure 9-21 Type of Service Window

2. There are two choices: RTP and SIP. RTP = voice packets and SIP = signaling packets. Select each field and assign the appropriate value. Then select **OK**. These fields are also looking for a Hexadecimal value as with command 84-23. Refer to [Table 9-3 Common IP Precedence/Diffserv Values and Hexadecimal Equivalent](#). Access the following menus to select options:

- RTP - Voice Packets
- SIP - Signaling Packets

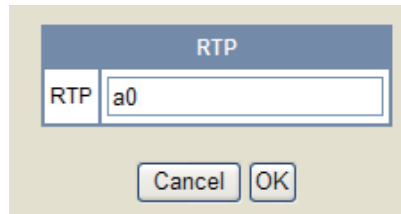


Figure 9-22 RTP - Voice Packets

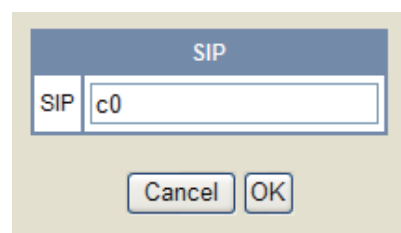


Figure 9-23 SIP - Signaling Packets

3. After selecting **Save**, the following message appears.
4. Select **OK** and the phone reboots. Once online, the phone tags all packets with the applied settings.

13.6.2 Assign Values on IP Terminal

The following is an example of assigning fields on the IP terminal.

1. Press **Hold, Transfer, *, #**.
2. Enter the user name of **ADMIN** and password of **6633222**, and press the **OK** softkey.
3. Press **1** on the dial pad for Network Settings.
4. Press **6** on the dial pad for Advanced Settings.
5. Press **4** on the dial pad for Type of Service.
6. There are two options (1) is RTP and (2) is SIP.
7. Press **1** on the dial pad for RTP (voice packets), enter the hexadecimal value, and then press the **OK** soft key.
8. Press **2** on the dial pad for SIP (Signaling packets), enter the hexadecimal value, and then press the **OK** soft key.
9. If no more changes are to be made, press the **Exit** soft key three times, and then press the **Save** soft key. The phone reboots.

13.7 Bandwidth

The bandwidth required for VoIP calls depends on the following factors.

- Layer 2 media
- CODEC
- Packet Size
- RTP Header Compression

- Voice Activity Detection (VAD)
- Number of simultaneous calls

Layer 2 media is concerned with moving data across the physical links in the network. A few of the most common layer 2 media types are Ethernet, PPP, and Frame Relay.

CODEC stands for Coder/Decoder and is the conversion of the TDM signal into an IP signal and vice versa. A CODEC can also compress/decompress the voice payload to save on bandwidth.

Packet Size is the amount of audio in each PDU (protocol data unit) measured in milliseconds. The larger the packet the less bandwidth used. This is because sending larger packets (more milliseconds of voice) requires, overall, less packets to be sent. The downside of this practice is if a packet is dropped/lost a larger piece of voice is missing from the conversation as the system waits the additional delay for the next packet arrival.

RTP Header Compression compacts the RTP header from 40 bytes in size to 2 ~ 4 Bytes in size. RTP header compression is used only on low speed links. Regularly on every voice packet there is an IP/UDP/RTP header that is 40 bytes in length. Compressing this header, down to 2 ~ 4 bytes, can save a considerable amount of bandwidth. The following is an example of a VoIP packet without RTP header compression and one of a packet with RTP header compression.

Notice that the overall packet size, when using RTP header compression, is considerably smaller.

- VoIP packet without RTP header compression

| | | | |
|-----------------------|-----------------------|------------------------|---------------|
| IP Header 20 bytes | UDP Header 8 Bytes | RTP Header 12 bytes | VOICE PAYLOAD |
|-----------------------|-----------------------|------------------------|---------------|

- VoIP packet with RTP header compression

| | |
|----------------------------------|---------------|
| Compressed Header 2 ~ 4 bytes | VOICE PAYLOAD |
|----------------------------------|---------------|

Voice Activity Detection (VAD) is suppression of silence packets from being sent across the network. In a VoIP network all conversations are packetized and sent, including silence. On an average a typical conversation contain anywhere from 35% ~ 45% silence. This can be interrupted as 35% ~ 45% transmission of VoIP packets, as having no audio, using valuable bandwidth. With the VAD option enabled, the transmitting of packets stops after a threshold is met determining silence. The receiving side then injects comfort noise into the call so it does not appear the call has dropped.

Bandwidth Calculations

The first step in calculating the bandwidth of a call is determining how many bytes the voice payload is going to use. The amount is directly affected by the CODEC and packet size. Below are the supported default CODEC speeds for SIP Multiline telephones.

- G.711 = 64000bps
- G.722 = 64000bps
- G.729 = 8000bps

Payload Calculation Voice

- $(\text{Packet size} * \text{CODEC bandwidth}) / 8 = \text{Voice Payload in Bytes}$
- Example of G.711 with a 20ms packet size
- $(.020 * 64000) / 8 = 160 \text{ Bytes}$
- Example of G.729 with a 30ms packet size
- $(.030 * 8000) / 8 = 30 \text{ Bytes}$

Now that you have the voice payload in bytes you can calculate the overall bandwidth including the layer 2 media. Below are some of the common layer 2 media types and their overhead.

- Ethernet = 18 Bytes
- 802.1Q/P Ethernet = up to 32 bytes
- PPP = 9 Bytes
- Frame Relay = 6 Bytes
- Multilink Protocol = 6 Bytes

Bandwidth Calculation

$([\text{Layer 2 overhead} + \text{IP/UDP/RTP header} + \text{Voice Payload}] / \text{Voice Payload}) * \text{Default CODEC speed} = \text{Total Bandwidth}$

Example of a G.711 call over Ethernet using a 20ms packet size and not using RTP header compression

$(.020 * 64000) / 8 = 160 \text{ Bytes for Voice Payload}$

$([18 + 40 + 160] / 160) * 64000 = 87200\text{bps}$

If VAD is not enabled each side of the conversation would be streaming 87.2kbps in one direction for a total of 174.4kbps.

The following chart shows the supported CODECS for IP phones with different packet sizes over PPP and Ethernet.

| CODEC | Packet Size | PPP | Ethernet |
|-------|-------------|------------|------------|
| G.711 | 10 | 103.2 kbps | 110.4 kbps |
| G.711 | 20 | 83.6 kbps | 87.2 kbps |
| G.711 | 30 | 77.1 kbps | 79.5 kbps |
| G.711 | 40 | 73.9 kbps | 75.6 kbps |
| | | | |
| G.722 | 10 | 103.2 kbps | 110.4 kbps |
| G.722 | 20 | 83.6 kbps | 87.2 kbps |
| G.722 | 30 | 77.1 kbps | 79.5 kbps |
| G.722 | 40 | 73.9 kbps | 75.6 kbps |
| | | | |
| G.729 | 10 | 47.2 kbps | 54.4 kbps |
| G.729 | 20 | 27.6 kbps | 31.2 kbps |
| G.729 | 30 | 21.1 kbps | 23.5 kbps |
| G.729 | 40 | 17.8 kbps | 19.6 kbps |
| G.729 | 50 | 15.9 kbps | 17.3 kbps |
| G.729 | 60 | 14.5 kbps | 15.7 kbps |
| | | | |

13.8 Some Network Considerations

When adding the SL2100 to a customers network there are many things to be aware of. Before implementation a detailed network diagram of the existing network must be obtained from the customer. This diagram may provide you with information about possible network conditions that can prevent or hinder the VoIP equipment from functioning correctly.

Firewalls

Another regular device in customer networks that can hinder VoIP performance is a firewall. Most corporate LANs connect to the public Internet through a firewall. A firewall is filtering software built into a router or a stand alone server unit. It is used to protect a LAN it from unauthorized access, providing the network with a level of security. Firewalls are used for many things, but in its simplest form, a firewall can be thought of as a one way gate. It allows outgoing packets from the local LAN to the Internet but blocks packets from the Internet routing into the local LAN, unless they are a response to query.

A firewall must be configured to allow specific traffic from the Internet to pass through onto the LAN. If an IP phone is deployed out over the Internet there is a very good chance it is passing through a firewall, either at the MAIN, the remote, or both locations.

The following diagram shows two IP phones on the corporate local LAN and one IP phone on a Remote network connected via the Internet. The two phones that are installed on the local LAN are functioning correctly. The IP phone at the remote site cannot register therefore it is not working.

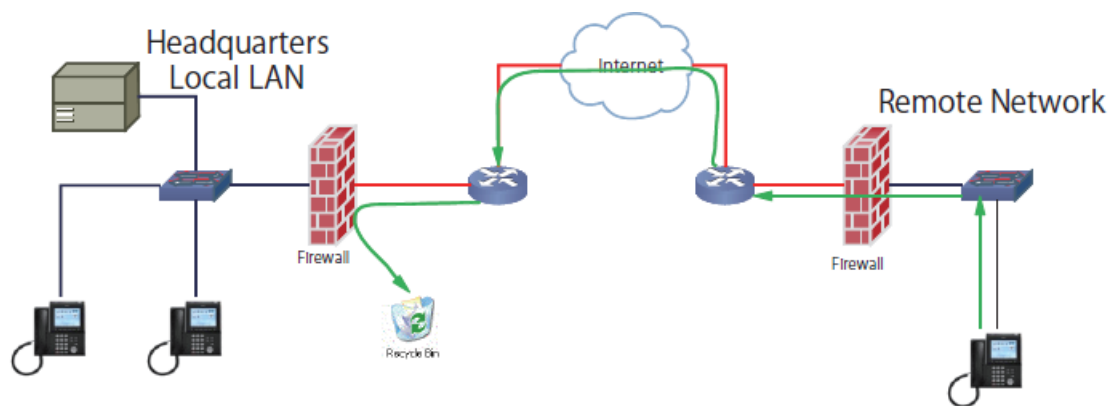


Figure 9-24 NEC SL2100 Network Example No. 1

The green arrow in the diagram above represents the data packets leaving the IP phone destined for the SL2100 on the Headquarters LAN. The firewall on the Headquarters network is not configured to recognize the UDP ports used by the NEC equipment thus blocking them and resulting in registration failure. To solve this issue the ports used by the NEC VoIP equipment must be opened in the firewall allowing the NEC traffic to pass through onto the SL2100.

The ports that are required to be opened on the headquarters locations are:

5080 and 5081 (UDP) for Signaling and 10020 ~ 10147 (UDP) for Voice.

The ports that need to be opened on the Remote network are **5060** (UDP) for signaling and ports **3462** and **3463** for voice (UDP).

VPN

Another common feature is the use of the Internet as the WAN between customer locations. When this is done VPNs are typically used between the locations. A VPN (Virtual Private Network) is a private data network that maintains privacy through the use of tunneling protocols and security features over the public Internet. This allows for remote networks (with private addresses), residing behind NAT routers and/or firewalls, to communicate freely with each other. When building the VPN tunnels, throughout the network, they must be assigned as a fully meshed network. This means that every network is allowed direct connection to each and every other network in the topology. Network equipment limitations may sometimes restrict this ability resulting in no voice path on VoIP calls between sites. When this happens Peer-to-Peer must be disabled in the SL2100. The downside to disabling Peer-to-Peer is using more DSPs and consumption of additional bandwidth at the MAIN location.

The following diagram shows three sites connected together via VPN. This network is not fully meshed due to the lack of a VPN tunnel between Sites B and C.

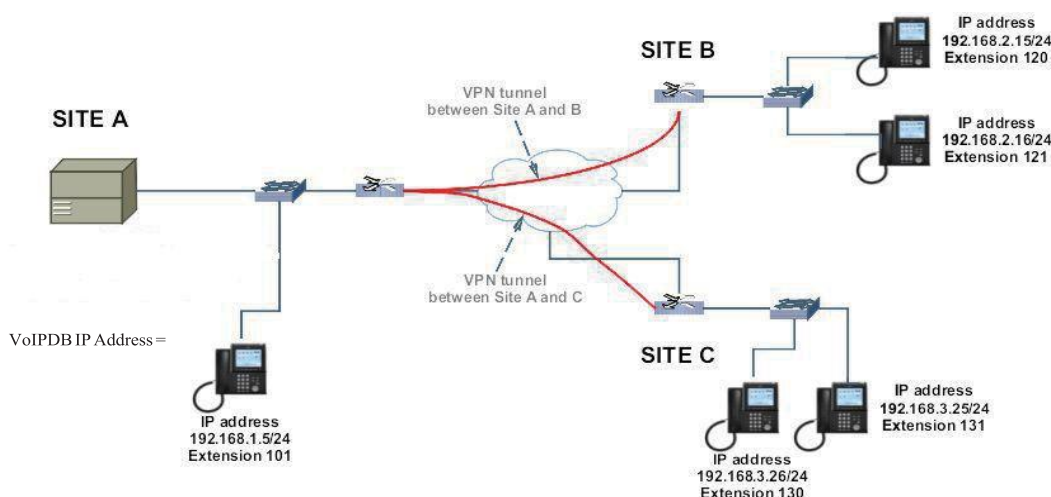


Figure 9-25 NEC SL2100 Network Example No. 2

With Peer-to-Peer enabled, the IP phones on site A can communicate with IP phones on sites B and C. IP Phones on sites B and C cannot communicate directly with each other though. The IP phone from site B can set up a call to the IP phone at site C, but there is no speech path. Here are the steps in the call scenario leading to the failed call.

- Extension 120 goes off hook and dials ext 130.
- An initial invite message is sent from 192.168.2.15 (ext 120) to 192.168.1.10 (VoIPDB).
- 192.168.1.10 (VoIPDB) forwards that message to 192.168.3.26 (ext 130).
- In the original setup message there is a field labeled SDP (Session Description Protocol). The SDP portion informs the IP phone where to send the media (voice) to. The SDP portion of this invite message contains the IP address of 192.168.2.15 (ext 120).
- 192.168.3.26 (ext 130) sends a 200 OK message to 192.168.1.10 (VoIPDB). In the 200 OK message is the SDP field reporting the IP address of 192.168.3.26 (ext 130).
- 192.168.1.10 (VoIPDB) forwards this message to 192.168.2.15 (ext 120).
- 192.168.2.15 (ext 120) sends an ACK message to 192.168.1.10 (VoIPDB).
- 192.168.1.10 (VoIPDB) forwards this message to 192.168.3.26 (130).
- At that point the two IP phones attempt to send voice packets directly to each other. As there is no VPN connection between these sites the call is set up with no voice path.

To correct this issue another VPN connection between sites B and C is required. If an additional VPN cannot be implemented, due to network limitations, the Peer-to-Peer feature can be disabled in the NEC SL2100. With Peer-to-Peer disabled, all packets (Signaling and Voice) route through the VoIPDB card. This also affects IP phones at the REMOTE locations calling other IP phones at the same location. Without Peer-to-Peer enabled the voice path must route to the MAIN location and then back to the REMOTE instead of directly between the two stations on the REMOTE network. This forces the use of additional bandwidth on the MAIN, and REMOTE locations. Peer-to-Peer is disabled in command 15-05-50 per IP Phone.



Figure 9-26 IP System Operation Setup

13.9 Guide to Feature Programming

| Program Number | Program Name | Description/Comments | Assigned Data | 1 | 2 | 3 |
|----------------|---|--|---|---|---|---|
| 10-12-03 | CPU Network Setup – Default Gateway | VoIPDB uses the Default Gateway that is assigned here. | 0.0.0.0~ 126.255.255.254 128.0.0.1~ 191.255.255.254 192.0.0.1~ 223.255.255.254 Default is 0.0.0.0 | X | | |
| 10-12-09 | CPU Network Setup – IP Address | Assign Layer 3 IP Address to the VoIPDB connected to CPU. | 0.0.0.0~ 126.255.255.254 128.0.0.1~ 191.255.255.254 192.0.0.1~ 223.255.255.254 Default is 172.16.0.10 | X | | |
| 10-12-10 | CPU Network Setup – Subnet Mask | Assign Subnet Mask to the VoIPDB connected to CPU. | Default is 255.255.0.0 | X | | |
| 84-26-01 | VoIPDB Basic Setup – IP Address | Assign an IP Address for the VoIPDB DSP's. | Default Values: Slot 1 = 172.16.0.20 | X | | |
| 84-26-02 | VoIPDB Basic Setup – RTP Port Number | | Range: 0 ~ 65534 Default Values: VoIP GW1 = 10020 | | | X |
| 84-26-03 | VoIPDB Basic Setup – RTP Port Number (RTP Port Number +1) | | Range: 1 ~ 65535 Default Values: VoIP GW1 = 10021 VoIP GW2 = 10053 VoIP GW3 = 10085 VoIP GW4 = 10117 VoIP GW5 = 10149 VoIP GW6 = 10181 VoIP GW7 = 10213 VoIP GW8 = 10245 | | | X |
| 10-46-01 | IP MLT Server Information Setup – Register Mode | Define which of the three Registration modes you wish the SIP MLTs to use. Plug and Play When the phone boots up it will report the ext assigned in the phone or choose the next available extension in the system. No password required. Auto If set to auto then the SIP user name and password must be entered into the actual IP phone. These settings have to match Programs 84-22/15-05-27 or the phone does not come on-line. Manual When the phone boots up it prompts you to enter a user ID and password before logging in. It checks this user ID/password against Programs 84-22/15-05-27. If there is not a match, the phone does not come on-line. | 0 = Plug and Play 1 = Automatic 2 = Manual Default is 0 | | X | |
| 10-46-04 | IP MLT Server Information Setup – Server Name | USER ID of the SIP URL if Program 10-46-05 is set to domain name. A SIP URL is made up of three parts. Domain name, host name, and server name. e.g. At default the server name is sipphd. The URL could look like the following: sipphd@voipu.nec.com | Up to 32 characters. Default is sipphd | | X | |

| Program Number | Program Name | Description/Comments | Assigned Data | 1 | 2 | 3 |
|----------------|--|--|---|---|---|---|
| 10-46-06 | IP MLT Server Information Setup – Register Port | Port the SIP messages are sent to on the VoIPU card. This same port number must be assigned in the SIP Multiline terminals. Changing this command also requires a CPU reset. | Range = 0 ~ 65535 Default is 5080 | | | X |
| 10-46-07 | IP MLT Server Information Setup – Encryption Mode | Enable or disable encryption mode. | 0 = Off 1 = On 1 = Default | | X | |
| 10-46-08 | IP MLT Server Information Setup – Encryption Type | Assign the encryption type. | 0 = Mode 1 Default is 0 READ ONLY | | X | |
| 10-46-09 | IP MLT Server Information Setup – One-Time Password | Password used when Program 10-46-07 is set to ON. Assign a character string of 10 characters or less. | Valid Characters (0-9, *, #) Default Not assigned | | X | |
| 10-46-10 | IP MLT Server Information Setup – Start Port | With Automatic logon the starting port number for automatic port allocation. | Range = (1 ~ 128) Default = 1 | | X | |
| 15-05-01 | IP Telephone Terminal Basic Data Setup – Terminal Type | Type of IP terminal registered with the specified extension number. | 1 = H.323 2 = SIP 3 = None 4 = SIP MLT Default is 3 READ ONLY | | | X |
| 15-05-02 | IP Telephone Terminal Basic Data Setup – Terminal MAC Address | Allow association of a MAC Address to an extension. When the IP phone sends a register message to the CPU the CPU responds back with the extension number associated to the MAC address. | 00.00.00.00.00.00~ FF.FF.FF.FF.FF.FF Default is 00.00.00.00.00.00 | | X | |
| 15-05-07 | IP Telephone Terminal Basic Data Setup – Using IP Address | IP address the IP Terminal is using for the specified extension number. | 0.0.0.0~ 255.255.255.255. Default is 0.0.0.0 READ ONLY | | | X |
| 15-05-15 | IP Telephone Terminal Basic Data Setup – CODEC Type | Assign CODEC type for IP Terminal. If SIP SLT, use Program 84-19. If SIP MLT, use Program 84-24. | 1 = Type 1 2 = Type 2 3 = Type 3 4 = Type 4 5 = Type 5 1 = Default is 1 | | X | |
| 15-05-20 | IP Telephone Terminal Basic Data Setup – Bottom Option Information | Read Only CM showing type of adapter installed on the ITH-style telephone. | 0 = No Option 1 = ADA 2 = BHA Default is 0 READ ONLY | | | X |
| 15-05-26 | IP Telephone Terminal Basic Data Setup – IP MLT Terminal Type | Assign type of SIP MLT terminal connected. | 0 = Not Set 1 = Not Used 2 = ITL Series-12/24 Button without 8LK Unit 3 = Not Used 4 = Not Used 5 = Softphone 6 = CTI 7 = Not Used 8 = Not Used 9 = IP4WW-24TIXH Default is 0 10 ~ 14 = Not Used 15 = ITZ-**LDG/ITZ-**LD 16 = ITY-6D 17 = ITY-8LDX 18 = IP7WW-8IPLD-C1 | | | X |

| Program Number | Program Name | Description/Comments | Assigned Data | 1 | 2 | 3 |
|----------------|--|---|--|---|---|---|
| 15-05-27 | IP Telephone Terminal Basic Data Setup –Personal ID Index | For SIP Multiline phone using Manual/ Auto registration. Assign each phone a unique personal index. When complete go to command 84-22 and set the user name and password. | 0 = Not Set 1-128 = Set Default is 0 | | X | |
| 15-05-28 | IP Telephone Terminal Basic Data Setup –Additional Information Setup | | 0 = Disable 1 = Enable Default is 0 | | X | |
| 15-05-29 | IP Telephone Terminal Basic Data Setup –Using Router | | 0.0.0.0 ~ 255.255.255.255 Default is 0.0.0.0 READ ONLY | | X | |
| 15-05-30 | IP Telephone Terminal Basic Data Setup –DTMF Play During Conversation at Receive Extension | | 0 = Disable 1 = Enable Default is 0 | | | X |
| 15-05-31 | IP Telephone Terminal Basic Data Setup –Alarm Tone During Conversation (RTP packet loss alarm) | | 0 = Disable 1 = Enable Default is 1 | | | X |
| 15-05-33 | IP Telephone Terminal Basic Data Setup – Using IP Address | | 0.0.0.0~ 255.255.255.255 Default is 0.0.0.0 READ ONLY | | | X |
| 15-05-35 | IP Telephone Terminal Basic Data Setup –Encryption Mode On/Off | | 0 = Off 1 = On Default is 0 READ ONLY | | | X |
| 15-05-36 | IP Telephone Terminal Basic Data Setup – IP MLT Firmware Version | | 00.00.00.00~ FF.FF.FF.FF Default is 00.00.00.00 READ ONLY | | | X |
| 15-05-38 | IP Telephone Terminal Basic Data Setup –Paging Protocol Mode | Sets the protocol mode for the Paging function. | 0 = Multicast 1 = Unicast 2 = Auto Default is 0 | | | X |
| 15-05-39 | IP Telephone Terminal Basic Data Setup – CTI Override Mode | | 0 = Disable 1 = Enable Default is 0 | | | X |
| 15-05-40 | IP Telephone Terminal Basic Data Setup –Calling Name Display Info via Trunk for Standard SIP | | 0 = Both Name and Number 1 = Name only 2 = Number only 3 = None Default is 0 | | | X |
| 15-05-41 | IP Telephone Terminal Basic Data Setup –Time Zone (Hour) | | 0~24 (-12 ~ +12 hour) Default is 12 | | | X |
| 15-05-43 | IP Telephone Terminal Basic Data Setup –Video Mode | | 0 = Disable 1 = Enable Default is 0 | | | X |
| 15-05-44 | IP Telephone Terminal Basic Data Setup – Using STD-SIP Display for CPN | | 0 = Disable 1 = Enable Default is 0 | | | X |

| Program Number | Program Name | Description/Comments | Assigned Data | 1 | 2 | 3 |
|----------------|--|---|---|---|---|---|
| 15-05-45 | IP Telephone Terminal Basic Data Setup – NAT Plug and Play | Effective when Program 10-46-14 is set to NAT Mode. Select sending RTP port number to remote router, use from negotiation result (0) or received RTP packet (1). | 0 = Off 1 = On Default is 0 | | | X |
| 15-05-47 | IP Telephone Terminal Basic Data Setup – Registration Expire Timer for NAT | On a per station basis, this setting defines the SIP registration expiry timer. If this value is set to 0, for a NAPT terminal, the value in Program 84-23-01 is applied. | 0 = Disable 60 ~ 65535 (sec) Default is 180 | | X | |
| 15-05-48 | IP Telephone Terminal Basic Data Setup – Subscribe Expire Timer for NAT | On a per station basis, this setting defines the SIP Subscribe expiry timer. If this value is set to 0, for a NAPT terminal, the value in Program 84-23-02 is applied. | 0 = Disable 60 ~ 65535 (sec) Default is 180 | | X | |
| 84-10-XX | ToS Setup | Assignments deal with setting of the Layer 3 IP Header ToS field as it leaves the VoIPDB unit. Specify the protocol to assign the ToS field for, and select the populated field to conform to either IP Precedence or Differentiated Services. When setting IP Precedence, assign Priority, Delay, Throughput, and Reliability in Programs 84-10-01/02/03/04/05. When setting DiffServ, only assign the DSCP in Program 84-10-07. | Protocol Type 1 = Not Used 2 = Not Used 3 = Reserved 4 = H.323 5 = RTP/RTCP 6 = SIP 7 = Reserved 8 = SIP MLT 9 = SIP Trunk 10 = Reserved 11 = Reserved | | X | |
| 84-22-01 | IP MLT Logon Information Setup – User ID | User ID for Manual or Auto registration (Program 10-46-01). | Assign up to 32 Alpha/Numeric Characters Default is No Setting | | X | |
| 84-22-02 | IP MLT Logon Information Setup – Password | Password for Manual or Auto registration (Program 10-46-01). | Assign up to 16 Alpha/Numeric Characters Default is No Setting | | X | |
| 84-22-03 | IP MLT Logon Information Setup – User ID Omission | When set to manual login mode, the user ID is omitted from the display during entry by the user. | 0 = Off 1 = On Default is 0 | | X | |
| 84-22-04 | IP MLT Logon Information Setup – Log Off | Allow the ability to log off from the IP terminal when using manual registration mode. | 0 = Off 1 = On Default is 1 | | X | |
| 84-22-05 | IP MLT Logon Information Setup – Nickname | Assign up to 32 Alpha/Numeric characters. | Assign up to 32 Alpha/Numeric Characters Default is No Setting | | X | |
| 84-23-01 | IP MLT Basic Information Setup – Registration Expire Timer | At half the value of this timer the IP terminal sends another registration message to the CPU. | Range: 60~65535 Sec. Default is 180 | | X | |
| 84-23-02 | IP MLT Basic Information Setup – Subscribe Expire Timer | At half the value of this timer the IP terminal sends another Subscribe message to the CPU. | Range: 60~65535 Sec. Default is 3600 | | X | |
| 84-23-03 | IP MLT Basic Information Setup – Session Expire Timer | At half the value of this timer the IP terminal sends a re-invite message. | Range: 0~65535 Sec. Default is 180 | | X | |
| 84-23-04 | IP MLT Basic Information Setup – Minimum Session Expire Timer | Minimum time the CPU accepts a session timer for a new call. | Range: 0~65535 Sec. Default is 180 | | X | |

| Program Number | Program Name | Description/Comments | Assigned Data | 1 | 2 | 3 |
|----------------|---|--|--|---|---|---|
| 84-23-05 | IP MLT Basic Information Setup – Invite Expire Timer | When INVITE message received from SIP MLT does not contain Expires header, the CPU uses this value for timeout of outgoing call. E.g. The SIP MLT hears RBT for duration of this timer and then is disconnected. | Range: 0~65535 Sec. Default is 180 | | X | |
| 84-23-06 | IP MLT Basic Information Setup – Signal Type of Service | Used for updating the IP terminals SIGNALING TOS values. | Range: 0x00 ~ 0xFF Default is 00 | | X | |
| 84-23-07 | IP MLT Basic Information Setup – Error Display Timer | The time that an IP terminal holds an error message in the display. Setting 0 holds the error message indefinitely. | Range: 0 ~ 65535 Sec. Default is 0 | | X | |
| 84-23-08 | IP MLT Basic Information Setup – Digest Authorization Registration Expire Timer | When Digest Authentication mode is ON, this value is available. After receiving Initial INVITE without authentication information, CPU will send 401 message to the SIP MLT, then waits for an INVITE message with the authentication message from SIP MLT within this timer. Additionally, after receiving Re-REGISTER message for Keep Alive purpose, the CPU sends a 401 message. | Range: 0 ~ 4294967295 Default is 0 | | | X |
| 84-23-10 | IP MLT Basic Information Setup – Number of Password Retries | The number of times an incorrect password can be entered when the security key is pressed. If set to (1), only one attempt is allowed. When number of password retries is met an error message displays on the phone: Incorrect security code password entered, press call key to contact an administrator | Range: 0 ~ 255 Default is 0 | | X | |
| 84-23-11 | IP MLT Basic Information Setup – Password Lock Time | Time to leave the terminal Locked Out after entering the wrong security code. | Range: 0 ~ 120 Default: 0 Default is 0 | | X | |
| 84-23-12 | IP MLT Basic Information Setup – Reference Number | Assign the network admin telephone number. When the user presses the Call key to contact the network administrator, this number is dialed. | Up to 32 Digits (0~9, *, #, P, R, @) Default is No Setting | | X | |
| 84-23-13 | IP MLT Basic Information Setup – Media Type of Service | Assign the IP terminals MEDIA TOS values. | Range: 0x00 ~ 0xFF (0~9, A~F) Default is 00 | | X | |
| 84-23-14 | IP MLT Basic Information Setup – Refer Expire Timer | The valid period of the REFER subscription. | Range: 0 ~ 65535 Sec. Default is 60 | | | X |
| 84-24-01 | IP MLT CODEC Basic Information Setup – Number of G.711 Audio Frames | Amount of audio in each RTP packet. | Range: 1~4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms Default is 2 | | X | |
| 84-24-02 | IP MLT CODEC Basic Information Setup - G.711 Silence Detection Mode | Select whether to compress silence with G.711. When there is silence, the RTP packet is not sent. When VAD is enabled the CPU will stop sending packets but the IP Phone will continue to transmit silence packets. | 0 = Disable 1 = Enable Default = 0 | | | X |

| Program Number | Program Name | Description/Comments | Assigned Data | 1 | 2 | 3 |
|----------------|---|---|--|---|---|---|
| 84-24-03 | IP MLT CODEC Basic Information Setup – G.711 Type | μ-law used in N.A. | 0 = A-law 1 = μ-law Default is 1 | | X | |
| 84-24-04 | IP MLT CODEC Basic Information Setup – G.711 Jitter Buffer Minimum | Minimum value of the dynamic jitter buffer. | Range: 0 ~ 255ms Default is 20 | | X | |
| 84-24-05 | IP MLT CODEC Basic Information Setup – G.711 Jitter Buffer Average | Average value of the dynamic jitter buffer. | Range: 0 ~ 255ms Default is 40 | | X | |
| 84-24-06 | IP MLT CODEC Basic Information Setup – G.711 Jitter Buffer Maximum | Maximum value of the dynamic jitter buffer. | Range: 0 ~ 255ms Default is 80 | | X | |
| 84-24-07 | IP MLT CODEC Basic Information Setup – Number of G.729 Audio Frames | Amount of audio in each RTP packet. | Range: 1~4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms Default is 2 | | X | |
| 84-24-08 | IP MLT CODEC Basic Information Setup – G.729 Voice Activity Detection | Select whether to compress silence with G.729. When there is silence, the RTP packet is not sent. When VAD is enabled the CPU will stop sending packets but the IP Phone will continue to transmit silence packets. | 0 = Disable 1 = Enable Default is 0 | | | X |
| 84-24-09 | IP MLT CODEC Basic Information Setup – G.729 Jitter Buffer Minimum | Minimum value of the dynamic jitter buffer. | 0~300ms Default is 20 | | X | |
| 84-24-10 | IP MLT CODEC Basic Information Setup – G.729 Jitter Buffer Average | Average value of the dynamic jitter buffer. | 0~300ms Default is 40 | | X | |
| 84-24-11 | IP MLT CODEC Basic Information Setup – G.729 Jitter Buffer Maximum | Maximum value of the dynamic jitter buffer. | 0~300ms Default is 80 | | X | |
| 84-24-17 | IP MLT CODEC Basic Information Setup – Jitter Buffer Mode | | 1 = Static 2 = Not used 3 = Adaptive immediate Default is 3 | | X | |
| 84-24-18 | IP MLT CODEC Basic Information Setup - Voice Activity Detection Threshold | | 0 ~ 30 Default is 20 | | X | |
| 84-24-28 | IP MLT CODEC Basic Information Setup – Audio Capability Priority | This assign the CODEC to be used. | Range: 0~3 0 = G.711_PT 1 = Not Used 2 = G.729_PT 3 = G.722_PT Default is 0 | | X | |
| 84-24-32 | IP MLT CODEC Basic Information Setup – G.722 Audio Frame Number | Amount of audio in each RTP packet. | Range: 1~4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms Default is 3 | | X | |

| Program Number | Program Name | Description/Comments | Assigned Data | 1 | 2 | 3 |
|----------------|--|---|------------------------------------|---|---|---|
| 84-24-34 | IP MLT CODEC Basic Information Setup – G.722 Jitter Buffer Minimum | Minimum value of the dynamic jitter buffer. | Range: 0 ~ 255ms Default is 30 | | X | |
| 84-24-35 | IP MLT CODEC Basic Information Setup – G.722 Jitter Buffer Average | Average value of the dynamic jitter buffer. | Range: 0 ~ 255ms Default is 60 | | X | |
| 84-24-36 | IP MLT CODEC Basic Information Setup – G.722 Jitter Buffer Maximum | Maximum value of the dynamic jitter buffer. | Range: 0 ~ 255ms Default is 120 | | X | |
| 84-28-01 | IP MLT Firmware Name Setup – Firmware Directory | Maximum 64 characters. | Default is No Setting | | X | |
| 84-28-02 | IP MLT Firmware Name Setup – Firmware File Name | Maximum 30 characters. | Default is No Setting | | X | |

13.10 SIP MLT Quick Startup Guide

The following guides describe the setup for a SIP MLT from a default state for these modes:

- Plug and Play
- Automatic Registration
- Manual Registration

13.10.1 Plug and Play

1. Program 10-12

Assign the VoIPDB registration/signaling IP address, subnet mask, and default gateway. If no customer provided default gateway is provided, leave Gateway IP address at 0.0.0.0.

System Data
10-12 : CPU Network Setup

01 - IP Address: 0.0.0.0

02 - Subnet Mask: 255.255.255.0

03 - Default Gateway: 172.16.0.1

04 - Time Zone: (GMT -05:00) Eastern Time (US and Canada)

05 - NIC Setting: Automatic detection

07 - NAPT Router IP Address: 0.0.0.0

08 - ICMP Redirect: ☐

09 - VOIPDB IP Address: 172.16.0.10

10 - VOIPDB Subnet Mask: 255.255.0.0


11 - VOIPDB NIC Setting: Automatic detection

Figure 9-27 System Data 10-12: CPU Network Setup

2. Program 84-26

Assign the IP addresses that the DSP is going to use. The IP address assigned must be in the same subnet as the address in Program 10-12-09.

After these commands are uploaded to the CPU, a system reset must be applied.



| VoIP Gateway | IP Address | RTP Port | RTCP Port |
|--------------|-------------|----------|-----------|
| 1 | 172.16.0.20 | 10020 | 10021 |

Figure 9-28 System Data 84-26: VoIPDB Basic Setup (DSP)

3. Program 11-02

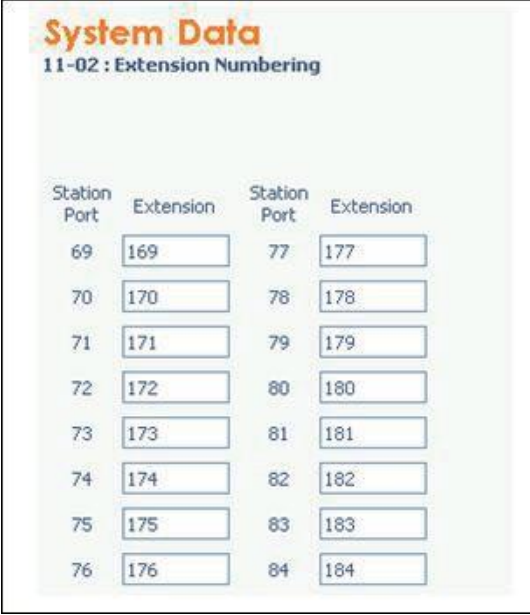
SIP MLT Stations are assigned to non-equipped hardware ports.

Physical Station ports are assigned automatically from lowest number ascending as cards are added to the system.

Because of this you should assign SIP MLT Stations starting with the higher number ports. By default all Station Ports are assigned numbers in the SL2100. These are easily changed in Program 11-02 to the required station number as long as the leading digit/digits are set in Program 11-01 as Extension.

Ports are dedicated to VoIP stations in groups of 2. E.g. In the image below if port 84 (Extension 283) is used for a SIP MLT Station that group of 2 ports (Ports 83 and 84) is now dedicated to VoIP use only.

After one port in a block of two is used by a VoIP station, the remaining port can be used only for another VoIP Extension.



| Station Port | Extension | Station Port | Extension |
|--------------|-----------|--------------|-----------|
| 69 | 169 | 77 | 177 |
| 70 | 170 | 78 | 178 |
| 71 | 171 | 79 | 179 |
| 72 | 172 | 80 | 180 |
| 73 | 173 | 81 | 181 |
| 74 | 174 | 82 | 182 |
| 75 | 175 | 83 | 183 |
| 76 | 176 | 84 | 184 |

Figure 9-29 System Data 11-02: Extension Numbering

- This step is optional. To enable Key data and other station feature programming (before IP Phone is brought online) the extensions must be identified as IP Phones. Once checked in the IPPhone

List in PC-Pro, the extensions are available for selection in Program 15 and other station related Programs.

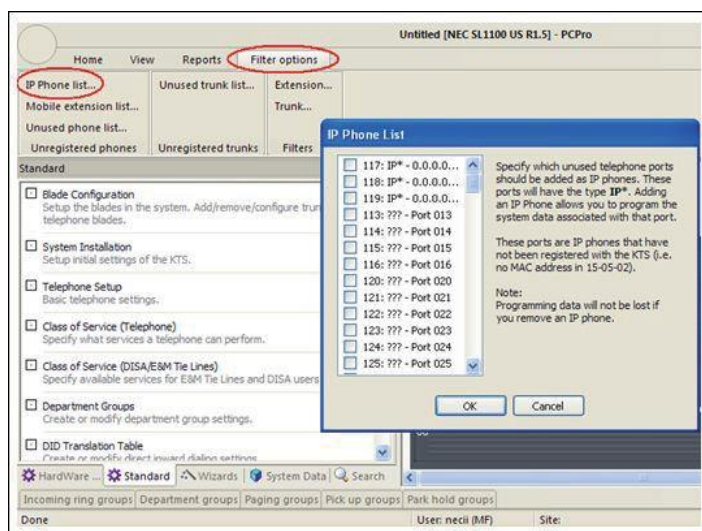


Figure 9-30 IP Phone List

- The SIP MLT Station requires assignments to be made in the phone itself. Enter the Program Mode in the station using the following steps.



- The station does not require an Ethernet connection to enter the program mode. Only power is required. Power can be provided by POE provided by a data switch. If the data switch is providing POE it must be using the **802.3AF** standard.

For Basic bench testing only the following assignments are required:

- At this point you are prompted with a User Name and Password. These are the defaults:
User Name: **ADMIN**
Password: **6633222**

- The user name should already be entered in the terminal.

- Press **Set** soft key to step down to the password field.
- After you enter the password, press **OK** soft key.

Network Settings

- DHCP Mode – DHCP Disable. Click **OK**.
- IP Address – Enter the IP Address for the station, and click **OK**.
- Default Gateway – Enter the Default Gateway Address, and click **OK**. If you are testing without a router/gateway, this must be left at the default 0.0.0.0
- Subnet Mask - Enter the Subnet Mask for the station, and click **OK**.

SIP Settings

- SIP User – Intercom Number
Enter the extension number for the IP station, and click **OK**.
- Server Address & URI – 1st Server Address
Enter the IP address assigned in command 10-12-09, and click **OK**.
- SIP Server Port - 1st Server Port
Enter port 5080, and click **OK**.
- Press the **EXIT** key until you are back at the Main menu.

- Press the **SAVE** key and the phone saves the configuration to memory, reboots itself and registers with the CPU.

13.10.2 Automatic Registration

1. Steps 1~3 are the same as for Plug and Play mode. Step 4 is not optional and **MUST** be assigned when using Automatic Registration.
2. Same as Plug and Play mode.
3. Same as Plug and Play mode.
4. To enable key data and other station feature programming before IP Phone is brought online, the extensions must be identified as IP Phones. Once checked in the IP Phone List in PC-Pro (see images below), the extensions are available for selection in Program 15 and other station related Programs.

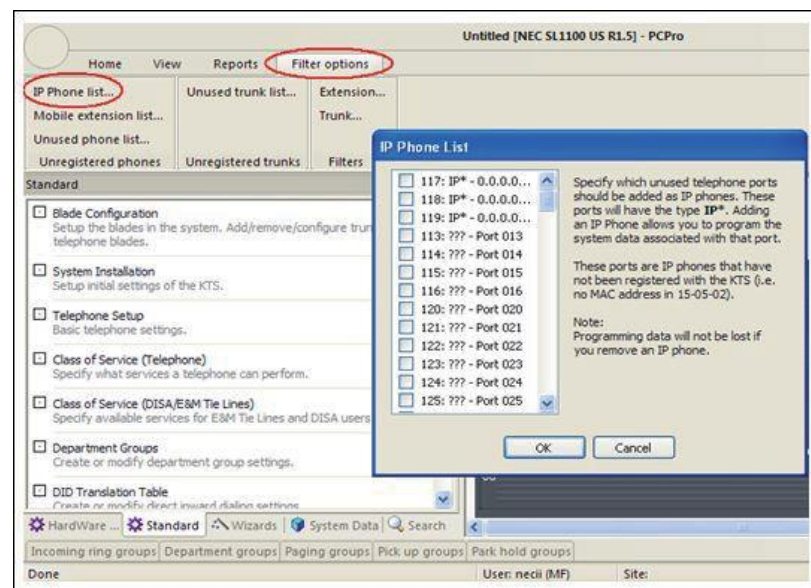


Figure 9-31 IP Phone List

5. Program 10-46
Change Program 10-46-01 to **Automatic**.



Figure 9-32 IP Multiline Terminal Server Information Setup

6. Program 15-05-27
Each IP phone requires a unique personal ID index. Valid settings are 1 ~ 512.

Figure 9-33 Automatic Registration Basic Setup

Figure 9-34 Automatic Registration Personal ID Index

7. Program 84-22-01
Assign the user ID and password to be associated with the Personal ID Index assigned in Step 6.

| Personal ID Index | User Id | Password |
|-------------------|---------|----------|
| 001 | 1234 | 4321 |
| 002 | | |
| 003 | | |
| 004 | | |
| 005 | | |
| 006 | | |
| 007 | | |
| 008 | | |

Figure 9-35 Automatic Registration User Name and Password Assignment

8. The SIP MLT Station requires assignments to be made in the phone itself. Enter the Program Mode in the station using the following steps.



- The station does not require an Ethernet connection to enter the program mode. Only power is required. Power can be provided by an AC adapter plugged into the phone or by POE provided by a data switch. If the data switch is providing POE it **MUST** be using the **802.3AF** standard.
For Basic bench testing, only the following assignments are required:

- At this point, you are prompted with a User Name and Password. These are the defaults:
User Name: **ADMIN**
Password: **6633222**

- The user name should already be entered in the terminal.

- Press **Set** soft key to step down to the password field.
- After you enter the password, press **OK** soft key.

Network Settings

- DHCP Mode - DHCP Disable. Click **OK**.
- IP Address - Enter the IP Address for the station, and click **OK**.
- Default Gateway - Enter the Default Gateway Address, and click **OK**. If you are testing without a router/gateway, this must be left at the default 0.0.0.0.
- Subnet Mask - Enter the Subnet Mask for the station, and click **OK**.

SIP Settings

- SIP User
 - **User ID** - Enter User ID assigned in command 84-22. Click **OK**.
 - **Password** - Enter the password assigned in command 84-22. Click **OK**.
 - **Incom Number** - Enter the extension number for the IP station. Click **OK**.
- Server Address & URI - 1st Server Address
Enter the IP address assigned in command 10-12-09, and click **OK**.
- SIP Server Port - 1st Server Port
Enter port 5080. Click **OK**.
- Press the **EXIT** key until you are back at the Main menu.
- Press the **SAVE** key, and the phone saves the configuration to memory, reboots itself and registers with the CPU.

13.10.3 Manual Registration

Steps 1–4 are the same as for Section [Automatic Registration](#).

1. Same as for Automatic Registration Mode.
2. Same as for Automatic Registration Mode.
3. Same as for Automatic Registration Mode.
4. Same as for Automatic Registration Mode.
5. Program10-46 – ChangeProgram10-46-01 to **Manual**.

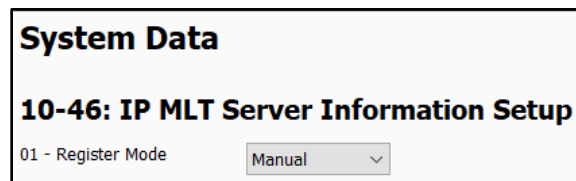



Figure 9-36 IP Multiline Terminal Server Information Setup


6. Same as for Automatic Registration Mode.
7. Same as for Automatic Registration Mode.

8. The SIP MLT Station requires assignments to be made in the phone itself. Enter the Program Mode in the station using the following steps.



 *The station does not require an Ethernet connection to enter the program mode. Only power is required. Power can be provided by an AC adapter plugged into the phone or by POE provided by a data switch. If the data switch is providing POE it must be using the **802.3AF** standard. For Basic bench testing only the following assignments are required.*

- At this point, you are prompted with a User Name and Password. These are the defaults:
User Name: **ADMIN**
Password: **6633222**

 *The user name should already be entered in the terminal.*

- Press **Set** soft key to step down to the password field.
- After you enter the password, press **OK** soft key.

Network Settings

- DHCP Mode - DHCP Disable. Click **OK**.
- IP Address - Enter the IP Address for the station, and click **OK**.
- Default Gateway - Enter the Default Gateway Address, and click **OK**. If you are testing without a router/gateway, this must be left at the default 0.0.0.0
- Subnet Mask - Enter the Subnet Mask for the station, and click **OK**.

SIP Settings

- Do not enter any information in the SIP user field. When the phone boots up, it requires a user name and password. These are preassigned in the system. When entered correctly, the phone is provided an extension number.
- Server Address & URI - 1st Server Address
Enter the IP address assigned in command 10-12-09, and click **OK**.
- SIP Server Port - 1st Server Port
Enter port 5080, and click **OK**.
- Press the **EXIT** key until you are back at the Main menu.
- Press the **SAVE** key, and the phone saves the configuration to memory, reboots itself and registers with the CPU.

13.11 IP Phone Relocation

The IP Phone Relocation feature gives users access to their IP telephone from any location by using the override login function. Users have the flexibility of logging into their IP Station in the office as well as remotely at the home office.

IP Phone Relocation Override

IP Phone Relocation is a feature for overriding the registration of an IP phone from various locations. To override the registration of an IP phone, you must have the login ID and Password of that IP phone.

Conditions

- Multiple IP Phones cannot use the same user ID and the same password at the same time.
- When a user is using multiple IP Phones at the same time, the user ID and password must be different for each phone.
- When a user is using SoftPhone (CTI mode) and controlling the IP Phone by this SoftPhone, the user ID and password should be different for the SoftPhone and IP Phone.
- An IP Phone (IP Phone and Soft phone) with DSS console cannot override another IP Phone.

- An IP Phone (IP Phone and Soft phone) with DSS console cannot be overridden from another IP Phone.
- The login ID and Password are programmed in Program 15-05-27 and Program 84-22.
- IP Phone Relocation can be used only in Manual Registration Mode.
- Two ports of the same terminal type (Program 15-05-26) cannot be assigned to the same Personal ID index (Program 15-05-27).
- When using Override with an active CTI connection, Program 15-05-39 must be enabled for the extensions that will be overridden. The overriding terminal must be of the same type and number of line keys as the terminal to be overridden. If the types of terminals and number of keys are different between overriding and overridden phones, the Telephony Service Providers (1st Party and 3rd Party) may not function properly.

Table 9-4 IP Phone Relocation

| Program/Item No. | Description/ Selection | Assigned Data | Comments |
|-------------------------|--|--|---|
| 10-46-01 | IP MLT Server Information Setup – Register Mode | 0 = Plug and Play 1 = Automatic 2 = Manual Default is 0 | Set up the information of the IP Multiline Server. This Program is a system-wide setting. |
| 15-05-27 | IP Telephone Terminal Basic Data Setup – Personal ID Index | 0~128 0 Default is 0 | Used when the SIP Multiline telephone is using manual/ auto registration. Assign each phone a unique personal index. Then go to command 84-22 to assign the user name and password. |
| 84-22-01 | IP MLT Logon Information Setup – User ID | Up to 32 characters Default not assigned | Input the User ID when using manual or auto registration (Program 10-46-01). |
| 84-22-02 | IP MLT Logon Information Setup – Password | Up to 16 characters Default not assigned | Input the Password when using manual or auto registration (Program 10-46-01). |
| 84-22-04 | IP MLT Logon Information Setup – Log Off | 0 = Off 1 = On Default is 1 | Input the Personal ID from terminal automatically when log on again. If set to 0, IP Phone Relocation fails. |

IP Single Line Telephone (SIP)

Section 1 INTRODUCTION

Session Initiation Protocol (SIP) Station feature provides Voice over Internet Protocol (VoIP) for IP stations. This feature is defined by the Internet Engineering Task Force (IETF) RFC3261.

SIP analyzes requests from clients and retrieves responses from servers, then sets call parameters at either end of the communication, handles call transfer and terminates. Typically, Voice over IP services are available from an SIP service provider.

With the VoIPDB up to 128 TDM talk paths are supported. This total may be shared among SIP stations or SIP trunks. Registered SIP stations and/or SIP trunks require a one-to-one relation with the VoIPDB DSP Resource. This is a required component of SIP implementation in the NEC SL2100. The NEC SL2100 VoIPDB contains a regular TCP/RTP/IP stack that can handle real-time media and supports industry standard SIP (RFC3261) communication on the WAN side.

For this feature, the VoIPDB is installed and assigned. The VoIPDB supports IP signaling for up to 128 (SIP Trunks and/or SIP Stations) and reduces the maximum capacity of system stations and/or Trunks in accordance with the number of registered SIP Stations.

The NEC SL2100 supports the following CODECS that are considered to provide toll-quality equivalent speech path.

The following voice compression methods are supported for the IP Station SIP feature:

- G.729. Low bandwidth requirement used on most Wide Area Network links.
- G.711. μ Law – High bandwidth requirement usually used on Local Area Networks.
- G.722 This CODEC is useful in fixed network, Voice over IP applications, where the required bandwidth is typically not prohibitive.
- G.726 is an ITU-T ADPCM speech-coded standard covering the transmission of voice at rates of 16-, 24-, 32-, and 40Kbps.

The SIP Station feature set supports the HOLD and TRF features based on RFC draft.

- Draft-IETF-sipping-service-examples-09.txt.
- Section 2.5 Draft-ietf-sipping-service-examples- (Transfer - Attended) 15.txt
- IETF RFC is defined as: Internet Engineering Task Force (RFC) Request for Comments.
- The SIP Station feature set supports the Message Waiting Indication (MWI) based on RFC3842.
- SIP INFO works independent from other DTMF methods such as RFC2833. This means SIP Terminals should send DTMF information by a single method, otherwise the system will receive both separately causing double digits.
- When PRG 15-05-49 is set to 2: Allowed while RTP is not available, SIP INFO will be received while RTP is not established. In-band method such as RFC2833 will be used once voice path is established.
- When PRG 15-05-49 is set to 1: Allowed any time, SIP INFO will be received whenever they arrive.

NAT Mode for SIP Phone

- NAT mode for SIP Phone which can not use SIP P2P mode and standard SIP video call feature uses P2P mode cannot establish in same system.
- When PRG 10-33-05 NAT mode for SIP phone is set to 1 - Enable, P2P mode for SIP Phone becomes always Off regardless PRG 15-05-50 setting.

- The system time can be provided using NTP time server updates to standard SIP terminals.
- When connecting multiple SIP Phones via NAT, PRG 15-05-18 has to be set to admit the registration of multiple SIP Phones which are using the same IP address. For example, if you had a STD SIP Terminal that had two lines registering with the same IP Address, you would need to flag PRG 15-05-18 for both Extension numbers.
- In the router/firewall that the SL2100 resides behind port forwarding is required. Port forwarding at the SIP Terminal end is not required as long as PRG 15-05-45 (Plug and Play) is enabled, which it is by default. The ports that must be forwarded to the SL2100 are as follows:
 - UDP Port 5070 **MUST** be forwarded to the IP Address assigned in PRG 10-12-09. UDP Ports 10020 ~ 10147 **MUST** be forwarded to the IP Address assigned in PRG 84-26-01.


Section 2 PROGRAMMING

2.1 Card Setup

| Program Number | Program Name | Description/Comments | Assigned Data | 1 | 2 | 3 |
|----------------|---|--------------------------------------|---|---|---|---|
| 10-03-01 | IPL Configuration - VoIP Type | VoIPDB Type | 0 ~ 3 = Not Used 4 = Built-in VoIP 5 = VoIPDB | X | | |
| 10-03-02 | IPL Configuration - Number of Channel | VoIPDB Channel | Built-in VoIP = 8 VoIPDB = 16 ~ 128 | X | | |
| 10-03-03 | IPL Configuration - Number of Voice Channel | Voice Channel on VoIPDB DSP Resource | 0 ~ 128 (Read Only) | X | | |

2.2 Configure VoIPDB Networking Information

The VoIPDB DSP's (PRG 84-26-01) should be connected to the same IP Subnet as the VoIPDB signaling address (PRG 10-12-09).

 If any IP Address or NIC setting is changed, the system must be reset for the changes to take affect.

| Program Number | Program Name | Description/Comments | Assigned Data | 1 | 2 | 3 |
|----------------|-------------------------------------|---------------------------------------|--|---|---|---|
| 10-12-03 | CPU Network Setup – Default Gateway | Assign the IP Address for the Router. | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 Default is 0.0.0.0 | X | | |
| 10-12-09 | CPU Network Setup – IP Address | Assign the IP Address of the VoIPDB. | 0.0.0.0 ~ 126.255.255.254 128.0.0.13~191.255.255.254 192.0.0.1~223.255.255.254 Default is 172.16.0.10 | X | | |
| 10-12-10 | CPU Network Setup – Subnet Mask | Assign the Subnet Mask of the VoIPDB. | 255.255.0.0 Default is 255.255.0.0 | X | | |

2.2.1 VoIPDB (DSP) Basic Setup

| Program Number | Program Name | Description/ Comments | Assigned Data | 1 | 2 | 3 |
|----------------|---|---|--------------------------------------|---|---|---|
| 84-26-01 | VoIP Basic Setup – IP Address | Assign the IP Address for the DSP resource. Note: The VoIPDB IP Addresses must be statically assigned. | Default: Slot 1 = 172.16.0.20 | X | | |
| 84-26-02 | VoIP Basic Setup – RTP Port Number | Assign the RTP Port Number. | 0~65534 Default for DSP1 is 10020 | | | X |
| 84-26-03 | VoIP Basic Setup – RTCP Port Number (RTP Port Number + 1) | Assign the RTCP Port Number. | 0~65534 Default for DSP1 is 10021 | | | X |

2.2.2 VoIP ToS Setup

| Program Number | Program Name | Description/ Comments | Assigned Data | 1 | 2 | 3 |
|----------------|----------------------|--|---|---|---|---|
| 84-10-01 | ToS Setup – ToS Mode | Use this field to define your SIP QoS marking for ToS or Diffserv. When Input Data is set to 1, Item No. 07 is invalid. When Data is set to 2, Item No. 02 ~ 06 are invalid. | 0 = Disabled (Invalid) 1 = IP Precedence 2 = Diffserv Default is 0 | | X | |

2.2.3 SIP Peer to Peer

| Program Number | Program Name | Description/ Comments | Assigned Data | 1 | 2 | 3 |
|----------------|--|--|--|---|---|---|
| 15-05-50 | IP Telephone Terminal Basic Data Setup – Peer to Peer Mode | Enable or disable Peer-to-Peer mode for SIP phones. When PRG 10-33-05 NAT mode is set to 1 (Enable), P2P mode for Sip phones always set to Off automatically regardless of this program setting. | 0 = No (Disable) 1 = Yes (Enable) (default is 0) | | X | |

2.2.4 IP Extension Numbering

| Program Number | Program Name | Description/ Comments | Assigned Data | 1 | 2 | 3 |
|----------------|---------------------|--|--|---|---|---|
| 11-01-01 | System Numbering | Refer to the SL2100 Programming Manual for all options and default settings. | 1 = 3 Digit; Intercom Default is 3 Digit; Intercom | X | | |
| 11-02-01 | Extension Numbering | Assign up to eight digits for extension numbering | Dial (Up to eight digits) Default Ports 1~84 = 101~184 | X | | |

2.2.5 SIP Extension CODEC Information

| Program Number | Program Name | Description/ Comments | Assigned Data | 1 | 2 | 3 |
|----------------|---|--|--|---|---|---|
| 84-19-01 | SIP Extension CODEC Information Basic Setup – Number of G.711 Audio Frames | Define the G.711 audio Frame Size | 1–4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms Default is 2 | | X | |
| 84-19-02 | SIP Extension CODEC Information Basic Setup – G.711 Voice Activity Detection Mode | Enable/Disable Voice Activity Detection for G.711. | 0 = Disable 1 = Enable Default is 0 | | | X |
| 84-19-03 | SIP Extension CODEC Information Basic Setup – G.711 Type | Define the G.711 Type – μ -law is recommended in USA. | 0 = A-law 1 = μ -law Default is 1 | | | X |
| 84-19-04 | SIP Extension CODEC Information Basic Setup – G.711 Jitter Buffer (min) | Define G.711 Jitter Buffer minimum accepted value. | 0–300ms Default is 20 | | | X |
| 84-19-05 | SIP Extension CODEC Information Basic Setup – G.711 Jitter Buffer (Average) | Define G.711 Jitter Buffer setting. | 0–300ms Default is 40 | | | X |
| 84-19-06 | SIP Extension CODEC Information Basic Setup – G.711 Jitter Buffer (max) | Define G.711 Jitter Buffer maximum accepted value. | 0–300ms Default is 80 | | | X |
| 84-19-07 | SIP Extension CODEC Information Basic Setup – Number of G.729 Audio Frames | Define the G.729 audio Frame Size. | 1–6 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms 5 = 50ms 6 = 60ms Default is 2 | | X | |
| 84-19-08 | SIP Extension CODEC Information Basic Setup – G.729 Voice Activity Detection Mode | Enable/Disable Voice Activity Detection for G.729. | 0 = Disable 1 = Enable Default is 0 | | | X |
| 84-19-09 | SIP Extension CODEC Information Basic Setup – G.729 Jitter Buffer (min) | Define G.711 Jitter Buffer minimum accepted value. | 0–300ms Default is 20 | | | X |
| 84-19-10 | SIP Extension CODEC Information Basic Setup – G.729 Jitter Buffer (average) | Define G.729 Jitter Buffer setting. | 0–300ms Default is 40 | | | X |
| 84-19-11 | SIP Extension CODEC Information Basic Setup – G.729 Jitter Buffer (max) | Define G.729 Jitter Buffer maximum accepted value. | 0–300ms Default is 80 | | | X |
| 84-19-17 | SIP Extension CODEC Information Basic Setup – Jitter Buffer Mode | Define the Jitter Buffer mode – supported Static or Immediate. | 1 = Static 2 = Not used 3 = Adaptive Immediately Default is 3 | | | X |
| 84-19-18 | SIP Extension CODEC Information Basic Setup – VAD Threshold | Define the VAD Threshold. | 0–30 Default is 20 | | | X |
| 84-19-28 | SIP Extension CODEC Information Basic Setup – Audio Capability Priority | Define Audio capability priority. | 0 = G.711_PT 1 = Not Used 2 = G.729_PT 3 = G.722 4 = G.726 5 = Not Used Default is 0 | | X | |

| Program Number | Program Name | Description/Comments | Assigned Data | 1 | 2 | 3 |
|----------------|---|--|---|---|---|---|
| 84-19-33 | SIP Extension CODEC Information Basic Setup – Number of G.722 Audio Frames | | 1~4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms Default is 3 | | X | |
| 84-19-35 | SIP Extension CODEC Information Basic Setup – G.722 Jitter Buffer (min) | | 0~255ms Default is 30 | | | X |
| 84-19-36 | SIP Extension CODEC Information Basic Setup – G.722 Jitter Buffer (Average) | | 0~255ms Default is 60 | | | X |
| 84-19-37 | SIP Extension CODEC Information Basic Setup – G.722 Jitter Buffer (max) | | 0~255ms Default is 120 | | | X |
| 84-19-38 | SIP Extension CODEC Information Basic Setup – Number of G.726 Audio Frames | | 1~4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms Default is 3 | | X | |
| 84-19-39 | SIP Extension CODEC Information Basic Setup – G.726 Voice Activity Detection Mode | | 0 = Disable 1 = Enable Default is 0 | | | X |
| 84-19-40 | SIP Extension CODEC Information Basic Setup – G.726 Jitter Buffer (min) | | 0~255ms Default is 30 | | | X |
| 84-19-41 | SIP Extension CODEC Information Basic Setup – G.726 Jitter Buffer (Average) | | 0~255ms Default is 60 | | | X |
| 84-19-42 | SIP Extension CODEC Information Basic Setup – G.726 Jitter Buffer (max) | | 0~255ms Default is 120 | | | X |
| 84-19-49 | SIP Extension CODEC Information Basic Setup – RTP Filter | | 0 = Disable 1 = Enable Default is 1 | | | X |
| 84-19-65 | SIP Extension CODEC Information Basic Setup – Video Quality Mode | This setup specifies the maximum capacity of video. Refer to following ※ about the definition in each mode. | 0 = mode1 1 = mode2 Default is 0 | | | |
| 84-19-66 | SIP Extension CODEC Information Basic Setup – Video codec | | 0 = H.264 Default is 0 | | | |
| 84-19-67 | SIP Extension CODEC Information Basic Setup – Jitter Buffer Mode for Video | This setup specifies the jitter buffer mode of the video stream. Jitter buffer size adjustment method decides depending on the mode setting. Please set to 2 (Self adjusting) usually. | 0 = Static 1 = Self Adjusting Default is 2 | | | |
| 84-19-68 | SIP Extension CODEC Information Basic Setup – Minimum Jitter Buffer for Video | This setup specifies the minimum value of jitter buffer for the video stream. This value must be smaller than the value of the maximum jitter buffer. | 0 ~ 1000 (ms) Default is 70 | | | |

| Program Number | Program Name | Description/ Comments | Assigned Data | 1 | 2 | 3 |
|----------------|---|---|---------------------------------|---|---|---|
| 84-19-69 | SIP Extension CODEC Information Basic Setup – Initial Jitter Buffer for Video | This setup specifies the initial value of jitter buffer for the video stream. This value must be smaller than the value of the maximum jitter buffer and be bigger than the value of the minimum jitter buffer. | 0 ~ 1000 (ms) Default is 140 | | | |
| 84-19-70 | SIP Extension CODEC Information Basic Setup – Maximum Jitter Buffer for Video | This setup specifies the maximum value of jitter buffer for the video stream. It is used only when "67-Jitter Buffer Mode for video" is 1:Fixed. This value must be bigger than the value of the minimum jitter buffer. | 0 ~ 1000 (ms) Default is 210 | | | |

2.2.6 SIP Extension Basic Information Setup

| Program Number | Program Name | Description/ Comments | Assigned Data | 1 | 2 | 3 |
|----------------|---|--|--|---|---|---|
| 84-20-01 | SIP Extension Basic Information Setup – Registrar/Proxy Port | Define SIP station Proxy Port. | 1~65535 Default is 5070 | | | X |
| 84-20-02 | SIP Extension Basic Information Setup – Session Timer Value | Define the periodic refresh time that allows both user agents and proxies to determine if the SIP session is still active. | 0~65535 Default is 180 | | | X |
| 84-20-03 | SIP Extension Basic Information Setup – Minimum Session Timer Value | Define to convey the minimum allowed value for the SIP session timer. | 0~65535 Default is 180 | | | X |
| 84-20-04 | SIP Extension Basic Information Setup – Called Party Info | Define the SIP Extension presented Caller ID information. | 0 = Request URI 1 = To Header Default is 0 | | | X |
| 84-20-05 | SIP Extension Basic Information Setup – Expire Value of Invite | Define the time out response value for SIP invite. | 0~256 seconds Default is 180 | | | X |
| 84-20-06 | Expire Value of Invite (send) | The expiration time is set for the Invite message. | 1~3600 seconds Default is 180 | | | X |

2.2.7 IP Phone Configuration

| Program Number | Program Name | Description/ Comments | Assigned Data | 1 | 2 | 3 |
|----------------|--|--|--|---|---|---|
| 15-05-01 | IP Telephone Terminal Basic Data Setup – Terminal Type | Review the type protocol support by the IP Phone. Viewing Only – No changes permitted. | 0 = NGT 1 = H.323 2 = SIP 3 = MEGACO 4 = SIP-MLT READ ONLY Default = 3 | | | X |

| Program Number | Program Name | Description/Comments | Assigned Data | 1 | 2 | 3 |
|----------------|--|--|--|---|---|---|
| 15-05-02 | IP Telephone Terminal Basic Data Setup – Terminal MAC Address | MAC Address of registered MLT SIP phone is stored and/or can input the MAC address of an MLT SIP phone so when it comes online it is provided with the extension in which the MAC address matches. | MAC address 00-00-00-00-00-00 to FF-FF-FF-FF-FF-FF Default is 00-00-00-00-00-00 | | | X |
| 15-05-07 | IP Telephone Terminal Basic Data Setup – Using IP Address | Informational Only registered IP Phones | 0.0.0.0~255.255.255.255 Default is 0.0.0.0 | | | X |
| 15-05-16 | IP Telephone Terminal Basic Data Setup – Password | Assign the authentication password for SIP single line telephones. This program is valid when PRG 15-05-01 Terminal Type is set to 2. (2 = SIP) | Up to 24 characters Default not assigned | | X | |
| 15-05-18 | IP Telephone Terminal Basic Data Setup – IP Duplication Allowed Group | If an adapter has one IP address coming into it but multiple extensions off of it, assign all extensions to a group so the CPU knows that the one IP address is assigned to multiple extensions. | 0 = Not Used 1 = Group 1 2 = Group 2 3 = Group 3 4 = Group 4 5 = Group 5 6 = Group 6 7 = Group 7 8 = Group 8 9 = Group 9 10 = Group 10 Default is 0 | | X | |
| 15-05-40 | IP Telephone Terminal Basic Data Setup – Calling Name Display | | 0 = Both Name and Number 1 = Name only 2 = Number only 3 = None Default = 0 | | | X |
| 15-05-41 | IP Telephone Terminal Basic Data Setup – Time Zone | | 0 ~ 24 (-12 ~ +12 hour) Default = 12 | | | X |
| 15-05-43 | IP Telephone Terminal Basic Data Setup – Video Mode | | 0 = Disable 1 = Enable Default = 0 | | | X |
| 15-05-44 | IP Telephone Terminal Basic Data Setup – Using STD-SIP Display for CPN | | 0 = Disable 1 = Enable Default = 0 | | | X |
| 15-05-49 | IP Telephone Basic Setup - Receiving SIP INFO | Determines whether SIP INFO messages are received by the system. | 0 = Disable 1 = Allowed any time 2 = Allowed while RTP is not available Default = 2 | | | |

2.2.8 NAT Mode for Standard SIP terminal

| Program Number | Program Name | Description/Comments | Assigned Data | 1 | 2 | 3 |
|----------------|--|--|---|---|---|---|
| 10-12-03 | CPU Network Setup – Default Gateway | Assign the IP Address for the Router. | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 Default is 0.0.0.0 | X | | |
| 10-12-07 | CPU Network Setup - NAPT Router IP Address (Default Gateway [WAN]) | Define the IP Address of the WAN side of the router. | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 Default is 0.0.0.0 | X | | |

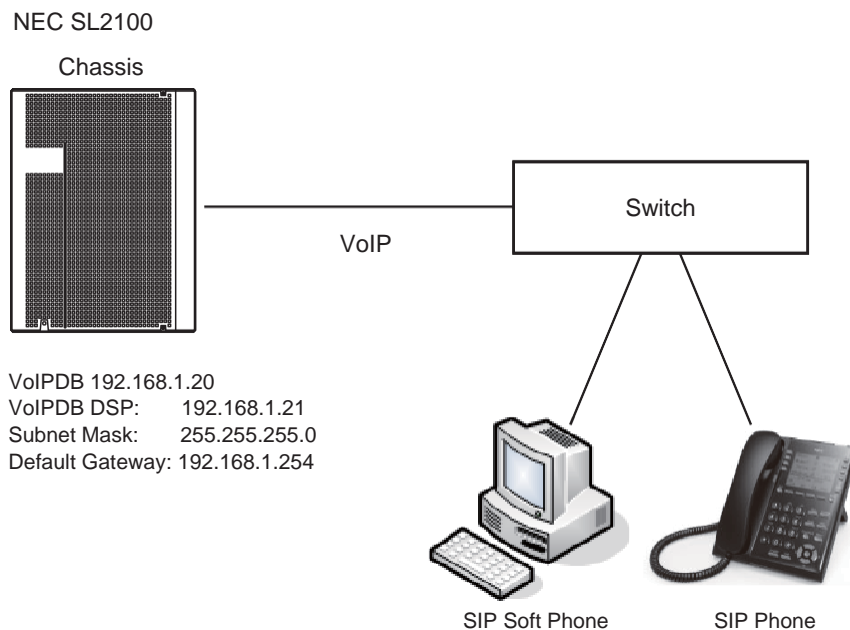
| Program Number | Program Name | Description/Comments | Assigned Data | 1 | 2 | 3 |
|----------------|---|--|--|---|---|---|
| 10-12-08 | CPU Network Setup - ICMP Redirect | When receiving ICMP redirect messages, this determines of the IP Routing Table updates automatically or not. | 0 = Enable 1 = Disable Default is 0 | | X | |
| 10-12-09 | CPU Network Setup – IP Address | Assign the IP Address of the VoIPDB. | 0.0.0.0 ~ 126.255.255.254 128.0.0.13~191.255.255.254 192.0.0.1~223.255.255.254 Default is 172.16.0.10 | X | | |
| 10-12-10 | CPU Network Setup – Subnet Mask | Define the Media Gateway Subnet Mask Address. | 128.0.0.0 192.0.0.0 224.0.0.0 240.0.0.0 248.0.0.0 252.0.0.0 254.0.0.0 255.0.0.0 255.128.0.0 255.192.0.0 255.224.0.0 255.240.0.0 255.248.0.0 255.252.0.0 255.254.0.0 255.255.0.0 255.255.128.0 255.255.192.0 255.255.224.0 255.255.240.0 255.255.248.0 255.255.252.0 255.255.254.0 255.255.255.0 255.255.255.128 255.255.255.192 255.255.255.224 255.255.255.240 255.255.255.248 255.255.255.252 255.255.255.254 255.255.255.255 default is 255.255.0.0 | X | | |
| 10-12-11 | CPU Network Setup - NIC Setup | Define the LAN interface speed and mode of the VoIP application supported. | 0 = Auto Detect 1 = 100Mbps, Full Duplex 2 = 10Mbps, Full Duplex 3 = 1Gbps, Full Duplex default is 0 | X | | |
| 15-05-50 | IP System Operation Setup – SIP Peer to Peer Mode | Enable or disable Peer-to-Peer mode for SIP phones. When PRG 10-33-05 NAT mode is set to 1 (Enable), P2P mode for SIP phones always set to Off automatically regardless of this program setting. | 0 = No (Disable) 1 = Yes (Enable) default is 0 | | X | |
| 10-33-02 | SIP Registrar/Proxy Information Basic Setup – Authentication Mode | When connecting STD SIP Terminal via NAT, this option must be enabled to prohibit illegal SIP phone registration. | 0 = No (Disable) 1 = Yes (Enable) default is 1 | X | | |
| 10-33-05 | SIP Registrar/Proxy Information Basic Setup NAT Mode | When system controls remote SIP phone via NAT router, set this program to 1 = Enable. | 0 = No (Disable) 1 = Yes (Enable) default is 0 | X | | |

| Program Number | Program Name | Description/Comments | Assigned Data | 1 | 2 | 3 |
|----------------|--|---|--|---|---|---|
| 10-37-01 | UPnP Setup – UPnP Mode | Default Gateway (WAN), PRG 10-12-07 can be set automatically when this program is set to 1 = Enable. | 0 = No (Disable) 1 = Yes (Enable) default is 0 | | X | |
| 10-58-01 | Network Address – Network Address | If there are other networks connected to this system that are not to be routed through NAT translations, then these networks must be identified in PRG 10-58. Use this PRG to assign the individual IP Address (e.g. 192.168.1.100) or assign the whole network address (e.g. 192.168.1.0). | Assign a valid IP Address or assign a valid Network Address. default is 0.0.0.0 | | X | |
| 10-58-02 | Network Address – Subnet Mask | If there are other networks connected to system that are not to be routed through NAT translations, then these networks must be identified in PRG 10-58. Use this PRG to assign the subnet mask for the IP Address entered in PRG 10-58-01. | Assign a valid subnet mask. default is 0.0.0.0 | | X | |
| 15-05-16 | IP Telephone Terminal Basic Data Setup – Password | Assign the authentication password for SIP single line telephones. This program is valid when PRG 15-05-01 Terminal Type is set to 2. (2 = SIP) | Up to 24 characters Default not assigned | X | | |
| 15-05-45 | IP Telephone Terminal Basic Data Setup- NAT Plug & Play | Per IP Single line telephone, enable NAT Plug and Play so that port forwarding is NOT required in the router the IP single line telephone resides behind. Effective only when PRG 10-46-14 is set to 1 = Enable. | 0 = No (Disable) 1 = Yes (Enable) default is 1 | X | | |
| 15-05-47 | IP Telephone Terminal Basic Data Setup – Registration Expire Timer for NAT | On a per station basis, this setting defines the SIP registration expiry timer. If this value is set to 0, for the terminal, the value in PRG 84-23-01 is applied. | 0 = User Timer in Program 84-23-01 60~65535(sec) default is 180 seconds | | X | |
| 15-05-48 | IP Telephone Terminal Basic Data Setup – Subscribe Expire Timer for NAT | On a per station basis, this setting defines the SIP subscribe expiry timer. If this value is set to 0, for the terminal, the value in PRG 84-23-02 is applied. | 0 = User Timer in Program 84-23-01 60~65535(sec) default is 180 seconds | | X | |
| 84-26-01 | VoIPDB Basic Setup – IP Address | Assign the IP Address for each DSP resource on the VoIPDB | xxx.xxx.xxx.xxx Default: Slot 1 = 172.16.0.20 | X | | |
| 84-26-02 | VoIPDB Basic Setup – RTP Port Number | Assign the RTP Port Number. | 0~65534 Default for DSP1 is 10020 | | | X |
| 84-26-03 | VoIPDB Basic Setup – RTCP Port Number | Assign the RTCP Port Number. | 0~65535 Default for DSP1 is 10020~10147 | | | X |

2.2.9 NTP Time Server

| Program Number | Program Name | Description/Comments | Assigned Data | 1 | 2 | 3 |
|----------------|-----------------|--|---|---|---|---|
| 10-65-01 | NTP Server | Set 1 when you want to start the NTP Server. | 0 = Disable (Stop) 1 = Enable (Start) default is 0 (Stop) | | | |
| 10-65-02 | NTP Server Port | Set the receiving port of the NTP Server. | 1 ~ 65535 default is 123 | | | |

2.3 SIP Phone Example



Section 3 IP Single Line Telephone (SIP) - TLS Connection Support

3.1 Description

With Ver 2.00 or higher, TLS protocol is supported on STD SIP terminals. The SIP message encryption acts only between "SL2100 and STD SIP terminals".

By default, the TLS Registrar/Proxy Port in Program 84-20-07 is set to 0. To register a STD SIP terminal using TLS it is necessary to assign the TLS Registrar/Proxy Port.

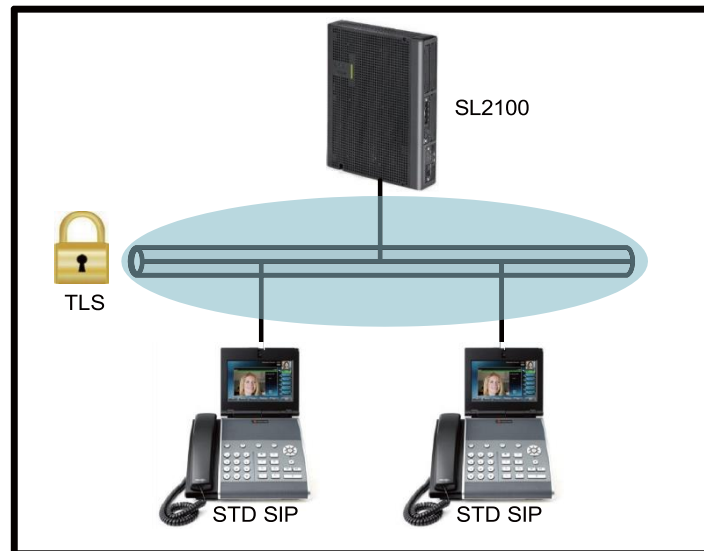


Figure 10-1 TLS Configuration

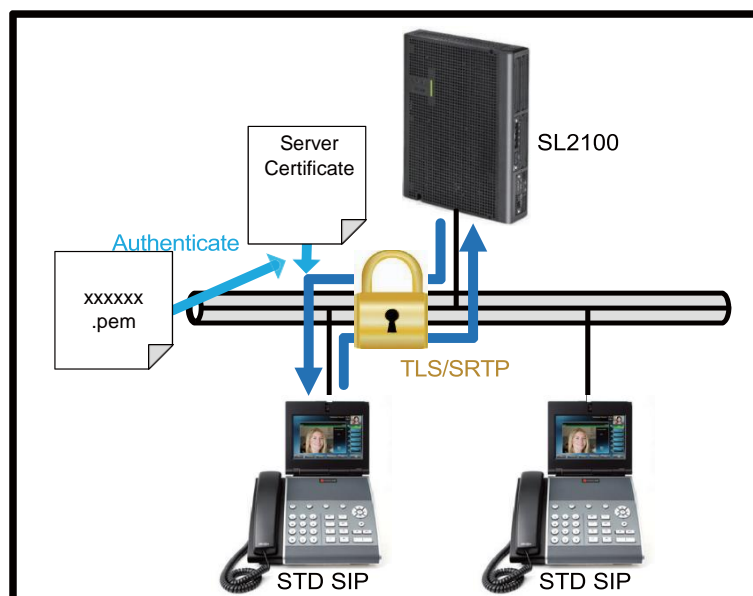


Figure 10-2 SL2100 Authentication

3.1.1 Conditions

- NAT Traversal is supported.
- The TLS STD SIP terminals do not support P2P call even if P2P settings are programmed.
- This feature does not support the terminals which do not use REGISTER method (Program 15-24-04).
- DNS server is required for ST500, to match with the Domain when the Certificate is authenticating.

TLS Support

- The SIP signaling packets between SL2100 and TLS STD SIP terminals are encrypted.
- The TLS connection is disconnected in the following cases:
 - Reset of SL2100
 - Restart of STD SIP terminal
 - Network Disconnection

SL2100 Certificate and Private Key

- Registered certificate on SL2100 is used when SL2100 receives a request for TLS connection.
- If no certificate file name is set in Program 10-72 then SL2100 uses self-signed certificate for TLS connection.
- Only the certificate consisting of "PEM format" is supported. The extension of certificate file is ".pem" or ".PEM" and the extension of private key file is ".key" or ".KEY".
- The file name can consist of a maximum of 32 characters including the extension.
- The SL2100 system retains "certificate" and "key" files even after system restart and cold start.
- The "Certificates" and "Keys" will be saved in "CERT" folder.
- The certificates can be checked, added and deleted by Web Programming under "Certificate Registration".
- Only IN level User can use "Certificate Registration" on Web Programming.
- "Certificate Registration" is not displayed when SA level or UA level user logs in.
- If server certificate and private key file name set in Program 10-72 are not registered with SL2100, the STD SIP cannot be registered using TLS.
- If the private key file name set in Program 10-72-02 is invalid, the STD SIP cannot be registered using TLS.
- The import of CA certificate on different terminal are explained in the following table:

Table 10-1 Import CA Certificate Requirements

| Terminal | Certificate Format | Certificate Import Method |
|-------------------|---|--|
| Polycom | <ul style="list-style-type: none"> • Format PEM • Extension .pem or.crt | A certificate is downloaded from a file server (FTP, TFTP and Web Server) and the certificate is imported. |
| ST500 (iOS) | <ul style="list-style-type: none"> • Format PEM • Extension .pem or.crt | <ul style="list-style-type: none"> • A certificate is attached to a mail and the certificate of attached file is imported. • A link of a certificate file is chosen from a web browser and the certificate is imported. • Android only - Its possible to import a certificate from microSD. |
| ST500 (Android) | <ul style="list-style-type: none"> • Format PEM • Extension .crt | |
| ITX-3370-1[]()TEL | <ul style="list-style-type: none"> • Format PEM | <ul style="list-style-type: none"> • A certificate is imported via the Web GUI. • Open a certificate on your PC with text editor. • Copy CA certificate / private key / private key password, and paste it to the Web GUI. |

- The conditions of a TLS connection:

Table 10-2 TLS Certificate Conditions

| Terminal | CA Certificate of a Terminal | Note |
|---------------------------|---|---|
| Polycom/ST500/ITX-3370-1W | <ul style="list-style-type: none"> • A valid and relevant CA Certificate should be used. | If Domain Name is used for CN in a certificate then DNS server is required. |

SRTP Support

- Voice Encryption acts between SL2100 and STD SIP terminals.
- To encrypt voice packets, it is necessary to enable SRTP Mode (Program 84-27-03 "enable").
- The behavior of SIP message encryption and voice packets encryption depend on both Program 15-05-51 and Program 84-27-03 shown below:

Table 10-3 SIP Message Encryption

| Program 15-05-51 (Read Only) Transport Protocol | Program 84-27-03 SRTP Mode | Behavior |
|---|----------------------------|--|
| TLS | 1: Enable | SIP messages and the voice packets are encrypted. |
| | 0: Disable | SIP messages are encrypted, voice packets are not. |

| Program 15-05-51 (Read Only) Transport Protocol | Program 84-27-03 SRTP Mode | Behavior |
|---|----------------------------|---|
| UDP/TCP | 1: Enable | SIP messages and voice packets are not encrypted. |
| | 0: Disable | |

- When a TLS registered STD SIP terminal calls to a non TLS registered STD SIP terminal or SIP MLT, the call will be non-P2P even if P2P settings are programmed and SRTP is enabled.
- If SRTP mode is enabled, the VoIP resource is not decreased.
- Alarm 74 display on Multiline terminal set in program 90-50-01 are shown below:

Table 10-4 Example of Terminal Display Alarms

| Error | Cause | Correction | Display |
|--|--|---|--|
| License not available | Encryption license not installed. | License check passes. (When the TLS connection is re-established or a TLS SIP call is initiated.) | TLS (P:00) No Lic TLS ALM (P:00) No License |
| Error in Server Certificate (Program 10-72-01) | Failed to validate Server Certificate. | When SL2100 receives offer of TLS connection, after setting correct file in Program 10-72-01. | TLS (P:00) Cert02 TLS ALM (P: 00) Cert Err02 |
| Error in Private Key (Program 10-72-02) | Failed to validate Private Key. | When SL2100 receives offer of TLS connection, after setting correct file in Program 10-72-01. | TLS (P: 00) Cert03 TLS ALM (P: 00) Cert Err03 |

3.1.2 Default Settings

None

3.2 System Availability

3.2.1 Terminals

- Polycom VVX500 (v5.9.7.12459)
- ST500
- ITX-3370-1W

3.2.2 Required Component(s)

- Built-In VoIP (CPU) /VoIPDB
- SL2100 Ver 2.00
- SL2100 ENCRYPTION LIC

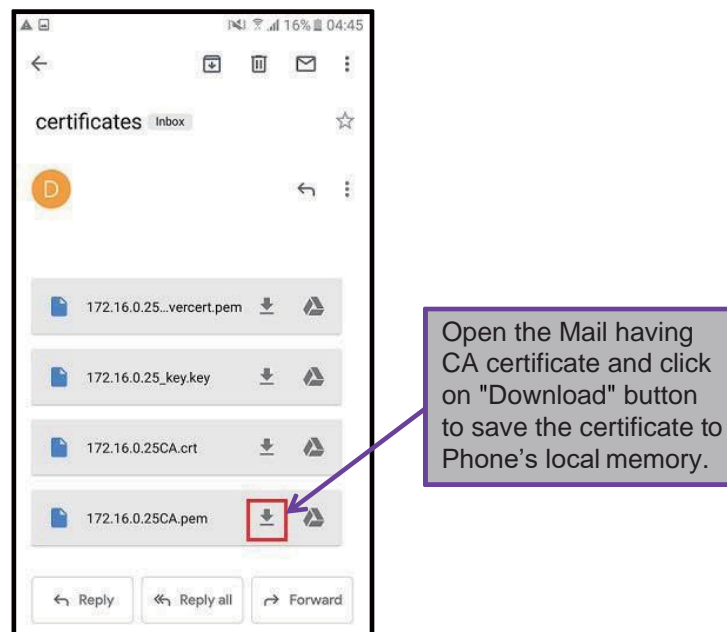
3.2.3 CA Certificate import on ST500 (Smartphone)

The CA certificate on ST500 can be imported by using:

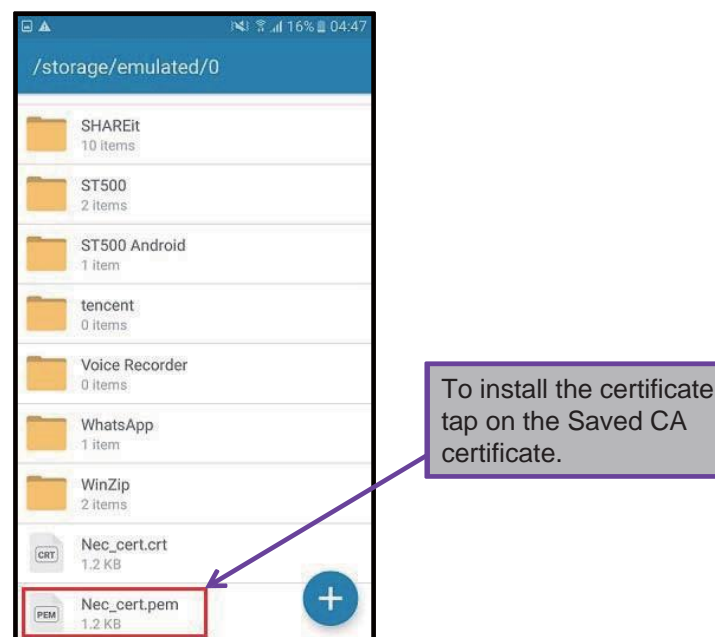
- Mail Server
- Micro SD card
- Webserver

Using Mail Server

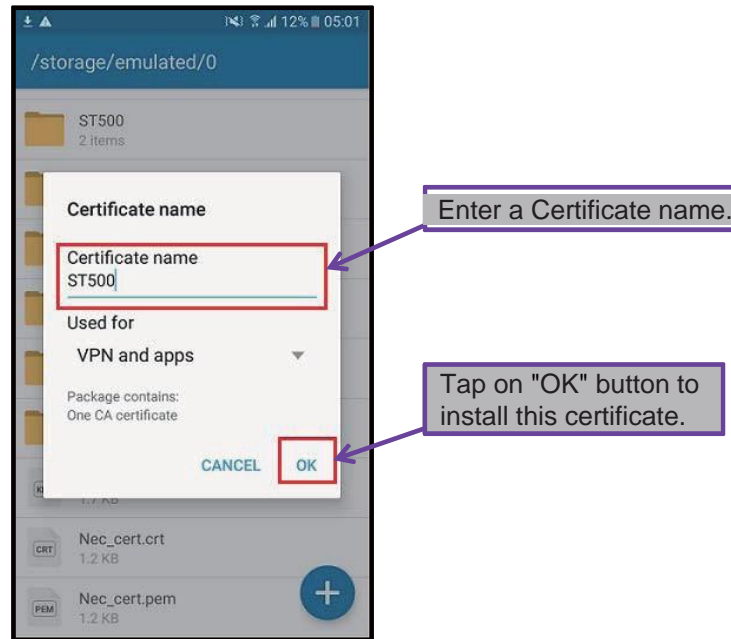
- Attach the CA certificate to a mail and send it to ST500 user's mail address.
- ST500 users access the mail account on android device where ST500 is installed, and check the mail having CA certificate.
- Now Click on the attached CA Certificate to download the CA certificate.



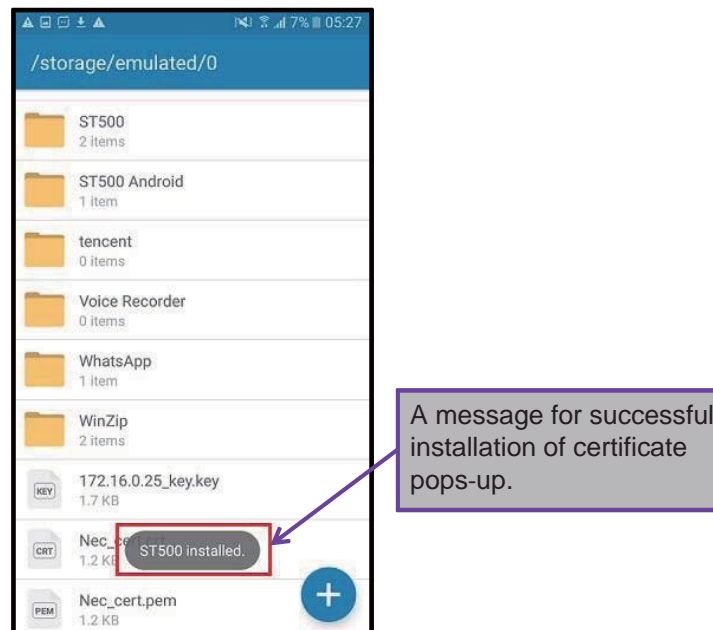
- CA certificate is downloaded in local memory space in device of ST500.



- Enter a Certificate name and click on "OK" button to install the certificate.



- After successful installation of certificate, a success message POPs-UP.

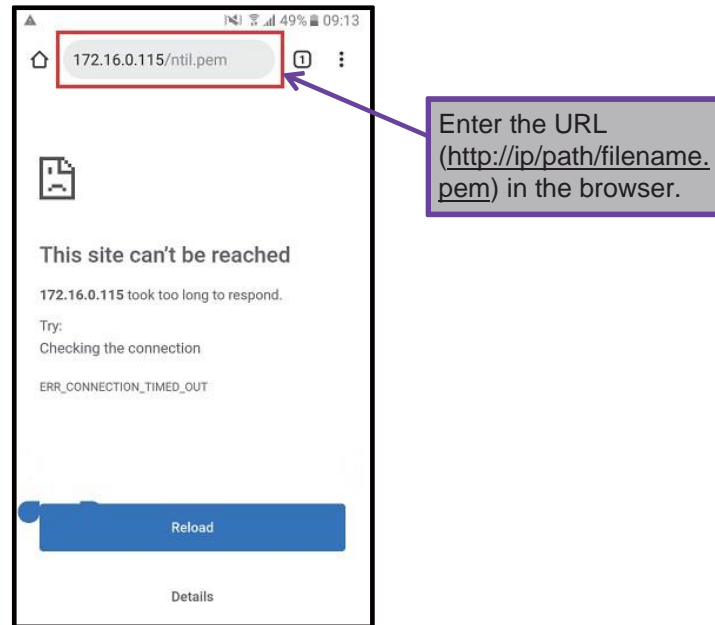


The installed certificates can be viewed under User of "Security Certificates".

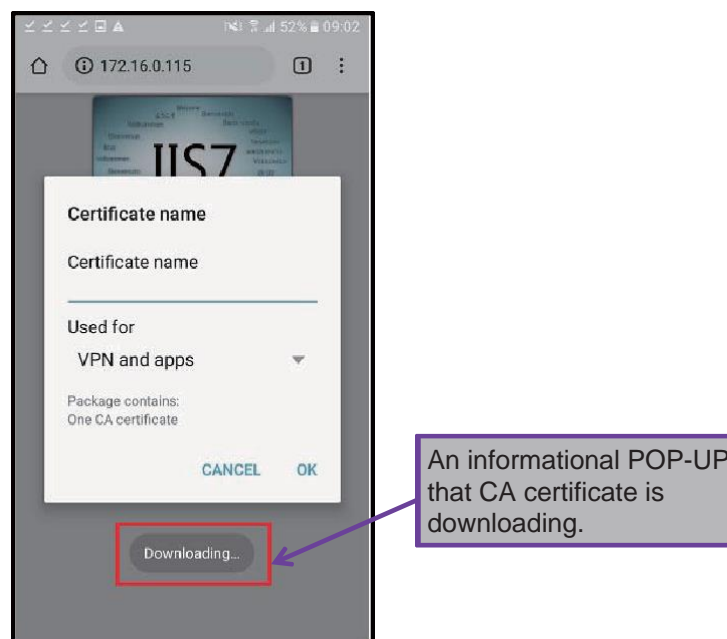
Using WEB-Server

- Place the CA certificate in root directory of WEB-Server.
ST500 users access the WEB-Server on android/iOS device's browser where ST500 is installed.

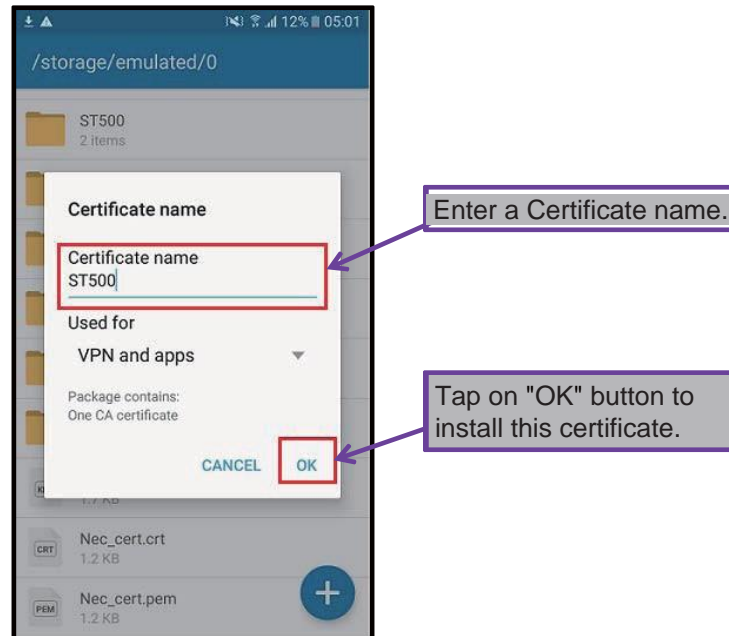
- Now enter the URL to access the certificates.



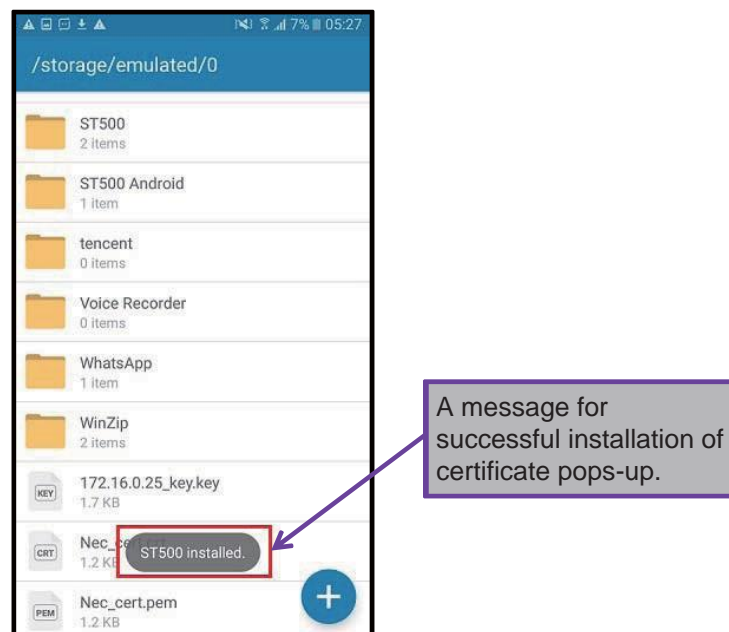
- CA certificate is downloaded to the device.



- Enter a Certificate name and click on "OK" button to install the certificate.



- After successful installation of certificate, a success message POPs-UP.

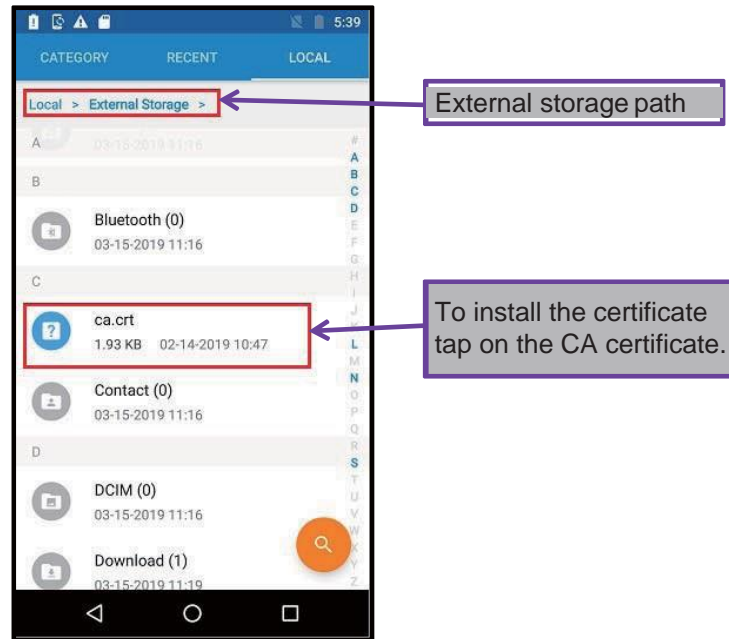


The installed certificates can be viewed under User of "Security Certificates (Android)".

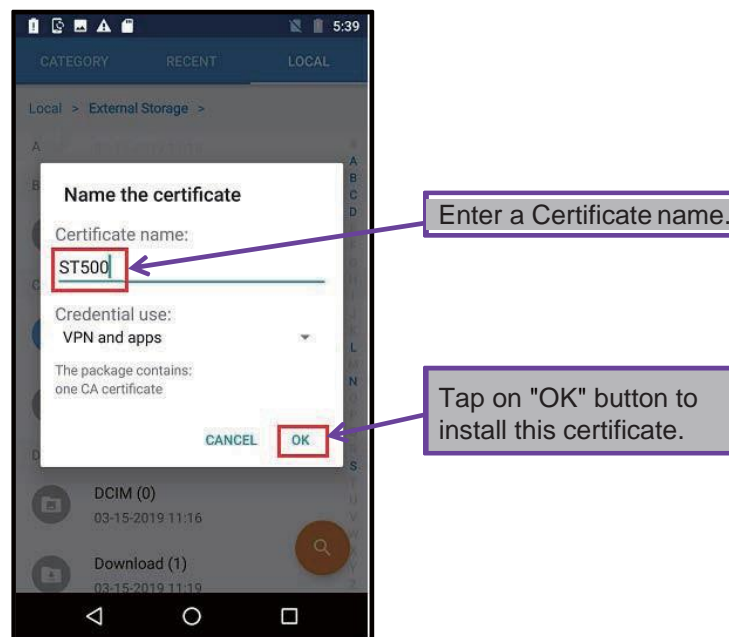
Using SD Card

- Go to the External storage path where the SD card Placed.

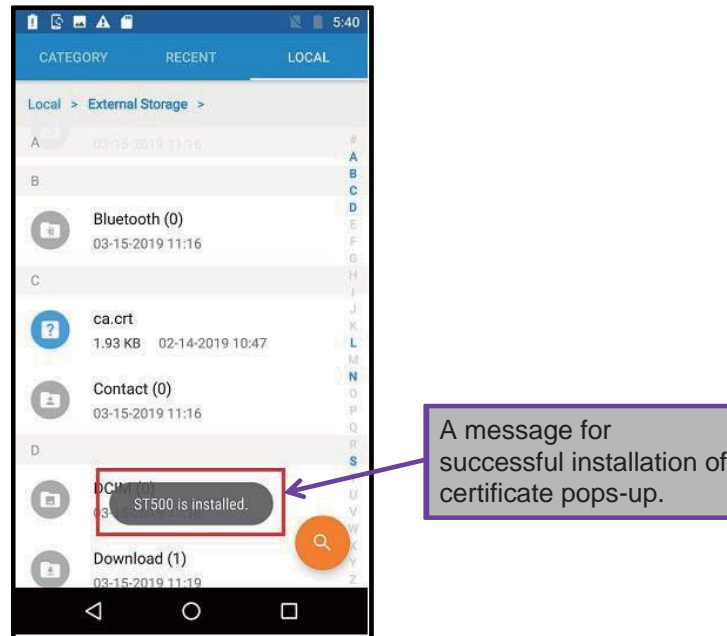
- Tap on the CA certificate to install it.




- Enter a Certificate name and click on "OK" button to install the certificate.



- After successful installation of certificate, a success message POPs-UP.



 The installed certificates can be viewed under User of "Security Certificates (Android)".

3.2.4 Import CA Certificate on Polycom VVX500

- The CA certificate on Polycom VVX500 can be imported using:
 - FTP Server
 - TFTP Server
 - Web Server
- Open the web management interface of the Polycom VVX500 and browse to the **Settings** -> **Network** -> **TLS** menu.
- Select the desired Application CA container and enter the URL to the certificate file using either FTP, TFTP or HTTP.
- The **Install** button should activate once a properly formatted URL is entered. Click **Install**, the device should report that the SSL certificate was successfully installed.

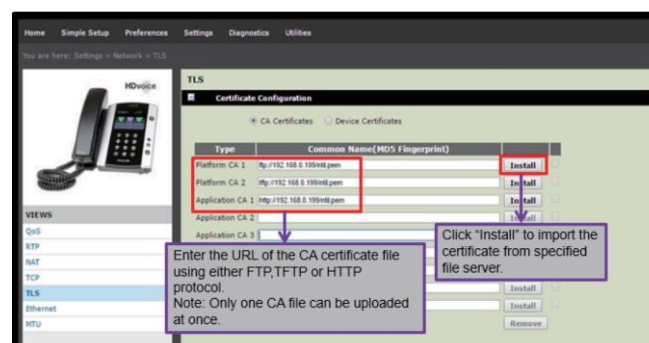


Figure 10-3 TLS Certificate Configuration Screen

After successful import of CA certificate the following screen appears:

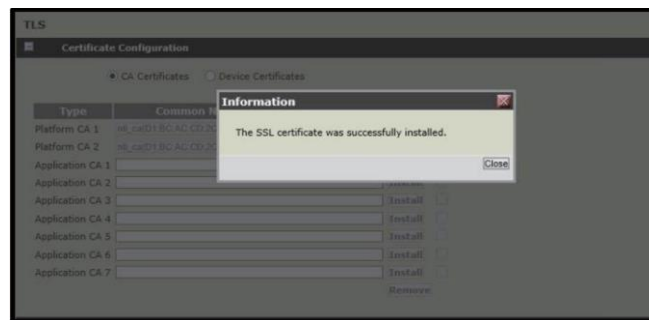


Figure 10-4 SSL Certificate Successfully Installed

The imported certificate is displayed.

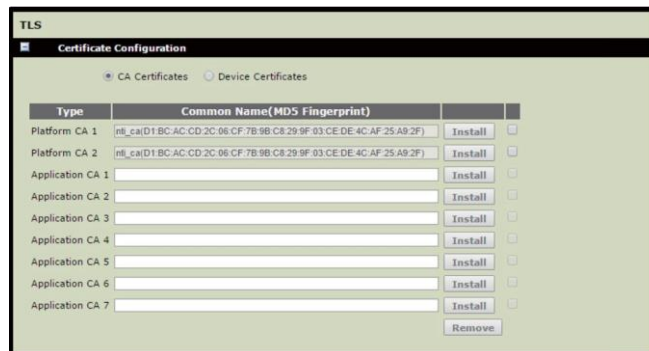


Figure 10-5 Example of Imported CA Certificates

3.2.5 Import CA Certificate on ITX-3370-1W

- The CA certificate on ITX-3370-1W can be imported using the Web GUI.
- Refer to the IP Phone GT890 Administration Guide (ST500) - SIP-TLS section.

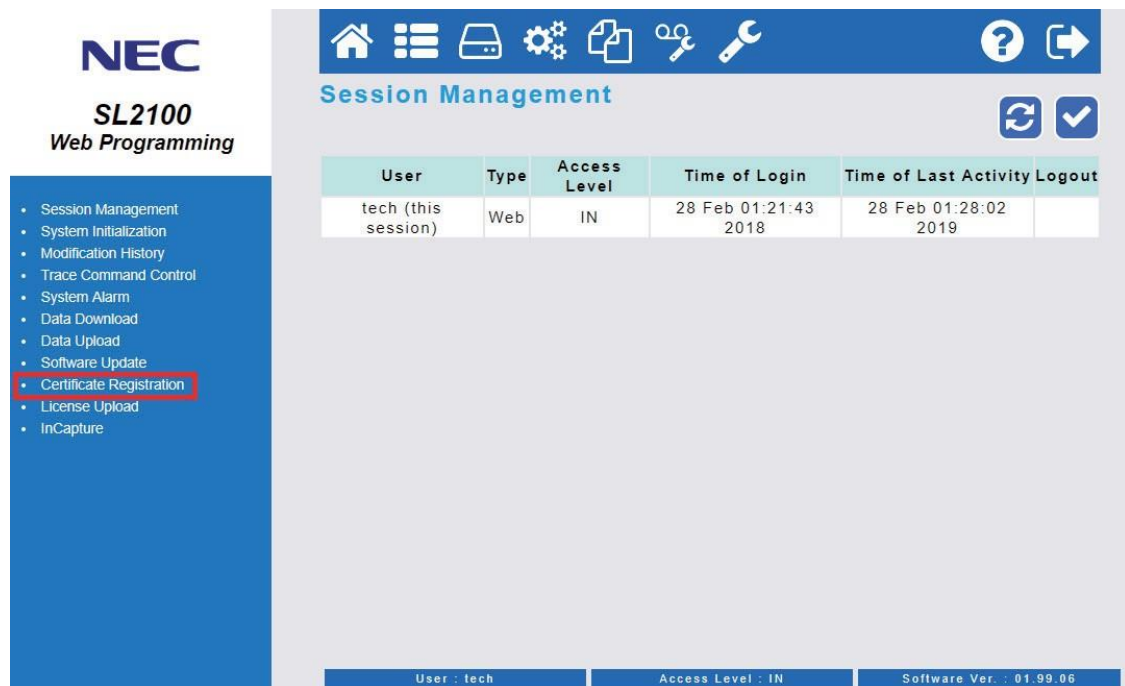
3.3 Guide to Feature Programming

| Program No. | Program Name | Input Data | Default |
|-------------|---|---|------------|
| 10-72-01 | Network Security Setup - Server Certificate | Maximum of 32 characters | No Setting |
| 10-72-02 | Network Security Setup - Private Key | Maximum of 32 characters | No Setting |
| 15-05-51 | IP Phone Basic Setup - Transport Protocol | 0 = UDP 1 = TCP 2 = TLS | 0 |
| 84-27-03 | VOIPDB Basic Setup - SRTP Mode Setup | 0 = Disable 1 = Enable | 0 |
| 84-27-04 | VOIPDB Basic Setup - SRTP Mode Select | 0 = Mode1 | 0 |
| 84-20-07 | SIP Extension Basic Setup - TLS Registrar/Proxy Port | 0 ~ 65535 | 0 |
| 90-10-01 | System Alarm Setup - Alarm Type Alarm 74 | 0 = Not Set 1 = Major Alarm 2 = Minor Alarm | 0 |
| 90-50-01 | System Alarm Display Setup - System Alarm Display Telephone | Up to eight digits | No setting |

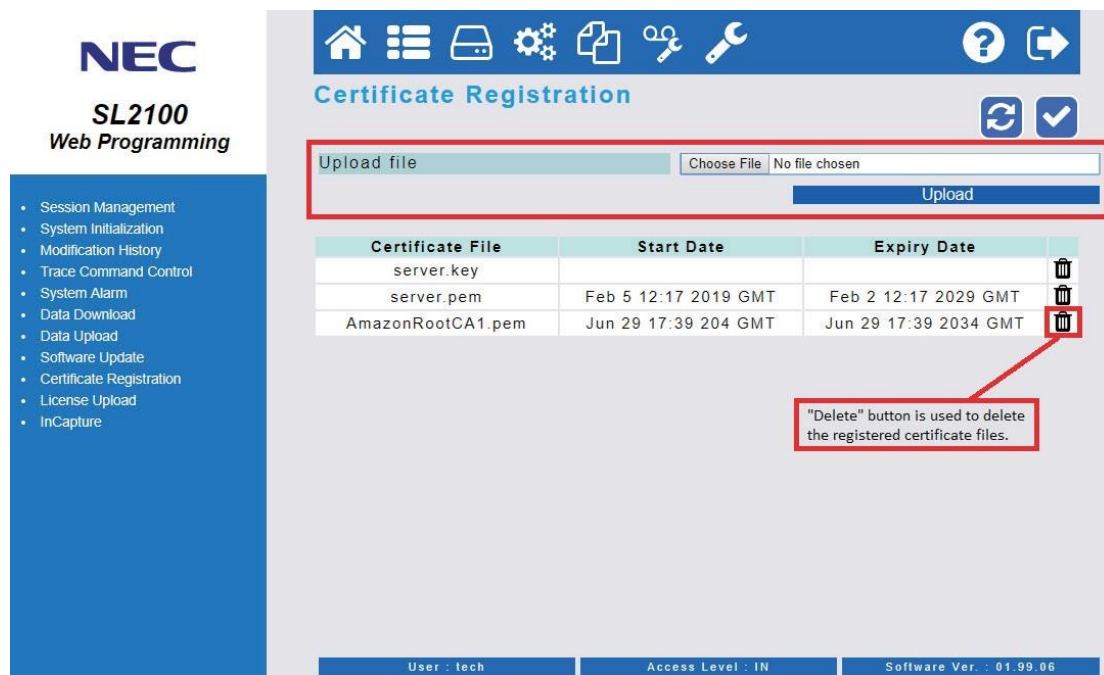
3.4 Operation

3.4.1 Certificate Registration in Web Programming:

1. Login to Web Programming at IN level.



2. Click on **Certificate Registration**.



3. Browse or Select the certificate file and click on **Upload** button.
If no file is selected or invalid extension is selected the error is displayed.
4. A maximum of ten files can be registered and if registered files are 10 or more, an error message "There is no available space in the storage" is displayed.
5. There is a list of the Certificate and Private Key files registered with SL2100 on the Certificate Registration page. "Start Date" and "Expiry Date" columns of Private Key file are displayed as blank.

6. The page is reloaded if the certificate has been uploaded and the certificate file's name is displayed in the list.
7. To delete, find the certificate file and select the **Delete** button located on the same line.

NAPT

Section 1 NAPT

1.1 Introduction

NAPT, or Network Address Port Translation, is a method by which a private address (or addresses) and their TCP/UDP ports are translated into a single public address and its TCP/UDP ports. In the case of IP phones with the SL2100 it allows their connection to a public (Internet) IP address which is then converted back to the private (non-Internet) IP address on the customer's network. The translation is available at the SL2100 end as well as at the remote IP Phone end of the connection if required.

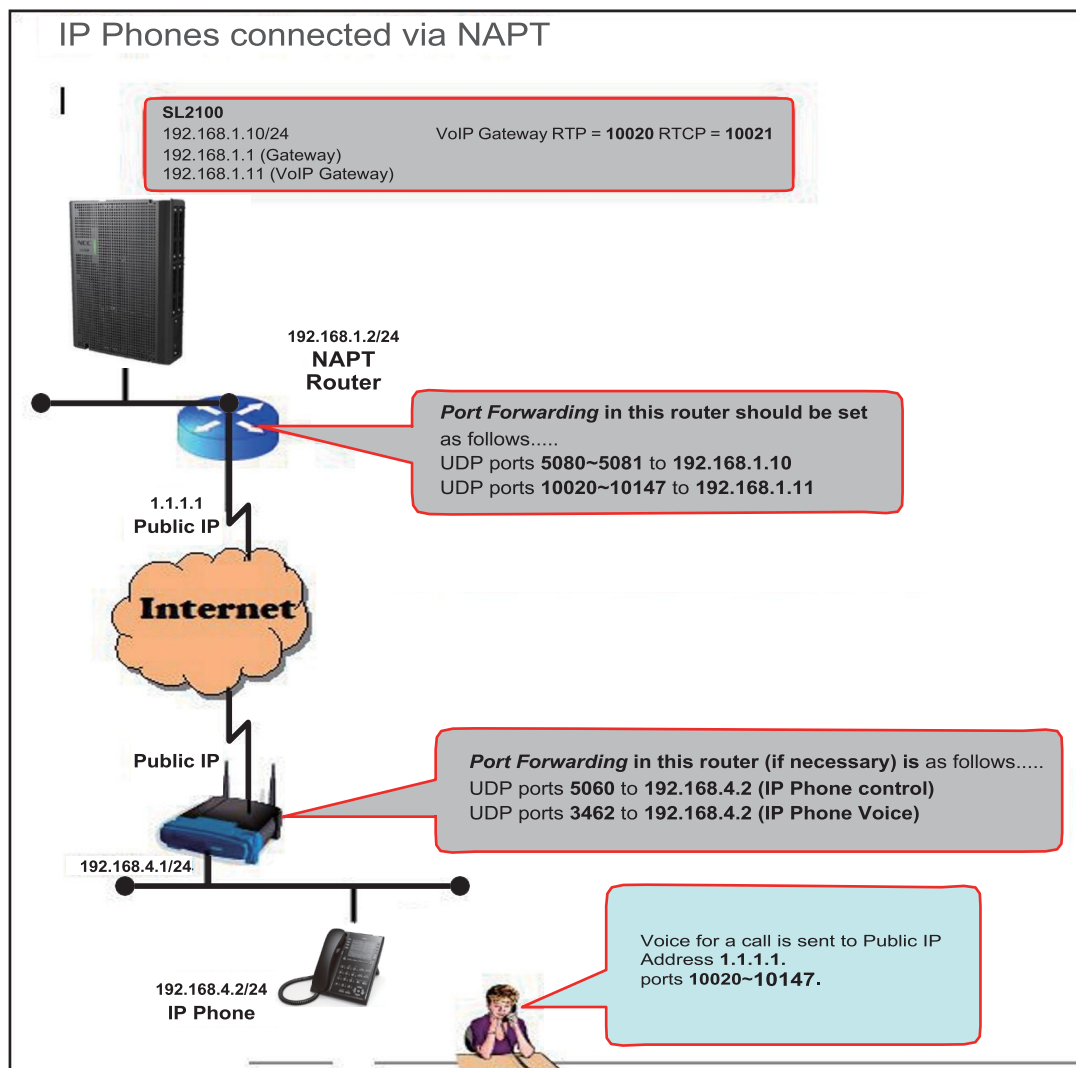


Figure 11-1 NAPT Configuration Example

1.2 SL2100 Requirements

The following information provides requirements for NAPT.

Main Software

NAPT is supported with the V1.00 or higher release of the SL2100.

Hardware

The SL2100 requires the following hardware:

- CPU (V1.00 or higher)
- VoIP Daughter Board

Capacity

A maximum of 112 IP phones can be connected.

1.3 Installation

The following settings have been added for NAT traversal in the IP Multiline Terminal.



*To enter IP phone programming at the terminal, press **HOLD, TRANSFER, *, #**. User name is **ADMIN** and password is **6633222 (NEC)**.*

1.3.1 Settings for terminals

Setting location: 0. Config/ 2. SIP Settings/ 8. NAT Traversal

| Number and Name of Setting | Setting Value | Default Value | Factory Value | Auto Config | Description |
|----------------------------|---------------------------------------|---------------|---------------|-------------|---|
| 1. NAT Traversal Mode | 1. Disable 2. Dynamic 3. Static | 1. Disable | Available | Available | Settings for NAT Traversal: Disable: Disables NAT Traversal Dynamic: Performs NAT Traversal using a dynamic conversion table. Static: Performs NAT Traversal using a static conversion table and requires the WAN IP Address to be entered into the IP Terminal. |
| 2. Network Area Name | A character string | No value | Available | Available | The name of the network area to which terminals belong. Specify up to 32 alphanumeric characters. Step is optional. |
| 3. WAN Settings | | | | | See table below. |

Setting location: 0. Config/ 2. SIP Settings/ 8. NAT Traversal/ 3. WAN Settings

| Number and Name of Setting | Setting Value | Default Value | Factory Value | Auto Config | Description |
|----------------------------|---------------|---------------|---------------|-------------|--|
| 1. WAN Mate IP Address | IP Address | 0.0.0.0 | Available | Available | WAN Address of the router that the SL2100 resides behind. This setting must match what is programmed in PRG 10-12-07. |
| 2. WAN SIP Mate Port | 1024~65535 | 5060 | Available | Available | Port number the SL2100 uses for SIP registration in PRG 10-46-06. |
| 3. WAN Self IP Address | IP Address | 0.0.0.0 | Available | Available | Only used when Static NAT is enabled. This setting is the WAN address of the router that the NAPT Terminal resides behind. |

Setting location: 0. Config/ 1. Network Settings/ 6. Advanced Settings/ 5. Self Port Settings

| Number and Name of Setting | Setting Value | Default Value | Factory Value | Auto Config | Description |
|----------------------------|-----------------------------------|---------------|---------------|---------------|---|
| 1. RTP Self Port | 1024~65528 (Even numbers only) | 3462 | Available | Not available | The number of the port receiving RTP data. At default this is assigned to port 3462. The First IP phone on this local LAN can use this port. The second IP phone would need to be changed to port 5062. |
| 2. SIP Self Port | 1024~65535 | 5060 | Available | Not available | The number of the port receiving SIP data. At default this is assigned to port 5060. The first IP phone on this local LAN can use this port. The second IP phone would need to be changed to port 3464. |

Section 2 CONDITIONS

- The NAPT feature supports CPU software V1.00 or higher.
- Terminals using NAPT must be at firmware V1.0.0.0 or higher.
- IP terminals can be connected via NAT router or WAN (direct connection).
- The NAT router on the SL2100 side must have a static WAN IP address.
- The software change Programs 15-05-47 and 15-05-48 to a shorter interval. These programs are changed on a per station basis. Non NAPT phones will still use Programs 84-23-01 and 84-23-02 while only NAPT phones will use Programs 15-05-47 and 15-05-48.
- It is necessary to set Program 10-46-14 to **OFF** when the VoIPDB is assigned a global (public) IP address.
- When Program 10-46-14 is set to **ON**, it references programs 10-58-01 and 10-58-02. These programs are used to define any destination networks that do not get sent through the NAPT translations.
- UDP ports in the remote routers may be required to be forwarded to the IP Terminals.
- NAPT can be used for SIP trunks and terminals on the same system.

Section 3 RESTRICTIONS

- With static NAT, the terminal needs a static IP Address assigned to it, or entries in the DHCP must be made to provide the same IP Address to the terminal.
- The NAT router on the terminal side must have the function for setting up static NAT.
- A conversion table must be manually set up for the NAT router on the terminal side.
- If installing multiple terminals in the domain of the NAT router on the terminal side, the RTP Self port and SIP Self port for each terminal must be specified so as to avoid overlapping.
- The SIP server cannot be switched. (Only one address can be registered as the SIP server.)

Dynamic NAT

- The NAT router on the terminal side must have the function for setting up dynamic NAT.
- It is assumed that port numbers are not changed by the NAT router on the terminal side. If a port number is changed by NAT router, NEC does not guarantee proper operation.
- If installing multiple terminals in the domain of the NAT router on the terminal side, the RTP Self port and SIP Self port for each terminal must be specified so as to avoid overlapping.
- The SIP server cannot be switched. (Only one address can be registered as the SIP server.)

Section 4 MINIMUM REQUIRED PROGRAMMING

The **Level 1**, **Level 2** and **Level 3** columns indicate the programs that are assigned when programming this feature in the order they are most commonly used. These levels are used with PCPro and WebPro wizards for feature programming.

- Level 1 – these are the most commonly assigned programs for this feature.
- Level 2 – these are the next most commonly assigned programs for this feature.
- Level 3 – these programs are not often assigned and require an expert level working knowledge of the system to be properly assigned.

| Program Number | Program Name | Description/Comments | Assigned Data | Level | | |
|----------------|--|--|--|-------|---|---|
| | | | | 1 | 2 | 3 |
| 10-12-01 | CPU Network Setup - IP Address | Assign the IP Address for the CPU's NIC card. If a VoIPDB is installed in the system it is recommended to set this to 0.0.0.0 and all connections to the system will be made through the VoIPDB. | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 (default = 192.168.0.10) | ✓ | | |
| 10-12-03 | CPU Network Setup – Default Gateway | Assign the default gateway IP address for the CPU. | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 (default = 0.0.0.0) | ✓ | | |
| 10-12-07 | CPU Network Setup – NAPT Router IP Address (Default Gateway [WAN]) | Define the IP Address of the WAN side of the router. | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 (default = 0.0.0.0) | ✓ | | |
| 10-12-09 | CPU Network Setup – IP Address | Assign the IP Address for the VoIPDB. If a VoIPDB is installed in the system it is recommended to set PRG 10-12-01 to 0.0.0.0 and all connections to the system will be made through the VoIPDB. | 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 (default = 172.16.0.10) | ✓ | | |

| Program Number | Program Name | Description/Comments | Assigned Data | Level | | |
|----------------|--|---|---|-------|---|---|
| | | | | 1 | 2 | 3 |
| 10-12-10 | CPU Network Setup – Subnet Mask | Define the Media Gateway Subnet Mask Address. | 128.0.0.0 192.0.0.0 224.0.0.0 240.0.0.0 248.0.0.0 252.0.0.0 254.0.0.0 255.0.0.0 255.128.0.0 255.192.0.0 255.224.0.0 255.240.0.0 255.248.0.0 255.252.0.0 255.254.0.0 255.255.0.0 255.255.128.0 255.255.192.0 255.255.224.0 255.255.240.0 255.255.248.0 255.255.252.0 255.255.254.0 255.255.255.0 255.255.255.128 255.255.255.192 255.255.255.224 255.255.255.240 255.255.255.248 255.255.255.252 255.255.255.254 255.255.255.255 (default = 255.255.0.0) | ✓ | | |
| 10-46-06 | Register Port Assign the port number to which the SIP messages are sent to on the VoIPDB. This same port number must be assigned in the SIP Multiline terminals. If this command is changed, it requires a CPU reset. | 0~65535 | 5080 | | ✓ | |
| 10-46-13 | Subscribe Session Port | 0~65535 | 5081 | | ✓ | |
| 10-46-14 | NAT Mode | Turns On/Off the NAT mode of the system. | 0 = Off 1 = On Default = Off | ✓ | | |
| 15-05-45 | NAT Plug&Play | Effective when PRG 10-46-14 is set to NAT Mode. Select sending RTP port number to remote router, use from negotiation result (0) or received RTP packet (1). | 0 = Off 1 = On Default = Off | | ✓ | |
| 15-05-47 | IP Telephone Terminal Basic Data Setup - Registration Expire Timer for NAT | On a per station basis, this setting defines the SIP registration expiry timer. This setting only applies to IP Multiline Terminals connected via NAPT. If this value is set to 0, for a NAPT terminal, the value in PRG 84-23-01 is applied. | 0 = Disable 60 ~ 65535 (sec) Default = 180 | | ✓ | |
| 15-05-48 | IP Telephone Terminal Basic Data Setup - Subscriber Expire Timer for NAPT | On a per station basis, this setting defines the SIP subscribe expiry timer. This setting only applies to IP Multiline Terminals connected via NAPT. If this value is set to 0, for a NAPT terminal, the value in PRG 84-23-02 is applied. | 0 = Disable 60 ~ 65535 (sec) Default = 180 | | ✓ | |

| Program Number | Program Name | Description/Comments | Assigned Data | Level | | |
|----------------|--|--|-----------------------|-------|---|---|
| | | | | 1 | 2 | 3 |
| 84-26-01 | VoIPDB Basic Setup (DSP)– IP Address | Assign the IP address for the VoIP Gateway. | Default = 172.16.0.20 | ✓ | | |
| 84-26-02 | VoIPDB Basic Setup (DSP)– RTP Port Number | Assign the RTP port number to be used for the DSP on the VoIPDB. | VoIP GW = 10020~10147 | ✓ | | |
| 84-26-03 | VoIPDB Basic Setup (DSP)– RTCP Port Number (RTP Port Number + 1) | Define the port number for RTCP to use for each DSP. | VoIP GW = 10021 | | ✓ | |

All DSP Busy Indication

Section 1 INTRODUCTION

The All DSP Busy feature is used to alert users via telephone displays and/or Alarm reports when all DSP (VoIP) resources in the system are being used. This can be used to trouble shoot issues or to alert when the current hardware might need to be upgraded to a higher capacity.

Alarm Message Format:

The Alarm message for will vary depending on what type of resource is unavailable, and will be displayed on display telephones and included in reports.

Table 12-1 Alarm Types

| Parameters | Description |
|------------|--|
| STA | DSP for IP Station Calls Were All Busy. |
| TRK | DSP for Trunk Calls Were All Busy, includes SIP. |

Alarm Report Example:

The report example below shows an alarm for all busy Station and Trunk DSPs.

| << ALARM REPORT >> | | | | | | | 01/22/2009 23:37 | | | |
|--------------------|------|------|----------|-------|-------------------|--------|------------------|-----|-----------|--|
| LVL | NO | STAT | DATE | TIME | ITEM | UNIT | SLT | PRT | PARAMETER | |
| MAJ | 0068 | ERR | 01/22/09 | 09:30 | VoIP All DSP Busy | VoIPDB | 01 | 00 | STA | |
| MAJ | 0068 | ERR | 01/22/09 | 09:31 | VoIP All DSP Busy | VoIPDB | 01 | 00 | TRK | |
| MIN | 0002 | REC | 01/22/09 | 09:32 | PKG Installation | PRIU | 02 | 00 | | |

LCD Display

| LCD Display Indication | Note |
|-------------------------|--------------------------------------|
| Clock/Calendar | XX = Slot number of CPU with VoIPDB. |
| VoIP All Busy XX | |
| Soft Key | |

Section 2 SERVICE CONDITIONS

- When using IP Phones, the alarm is shown on both terminals involved in that call if they are both on the same system.
- The alarm cannot be displayed on Standard SIP Phones or Single Line phones.
- If a call from a Standard SIP telephone to a Multiline telephone cannot be established due to an All DSP Busy condition, the Multiline telephone will not display the "All DSP" busy message.
- If a SIP trunk call is sent to the SL2100 when all DSP resources are busy, the call is rejected but the alarm is not displayed on any system telephone.

- The default alarm setting is Minor.

Section 3 RELATED FEATURES

- IP Multiline Station (SIP)
- IP Trunk – (SIP) Session Initiation Protocol

Section 4 GUIDE TO FEATURE PROGRAMMING

| Program Number | Program Name | Description/Comments | Assigned Data | Level | | |
|----------------|---------------------------|---|--|-------|---|---|
| | | | | 1 | 2 | 3 |
| 20-13-52 | VoIP All DSP Busy Display | Set on a per station class of service basis, whether the "All DSP Busy" alarm displays on the LCD when the caller makes an IP call and there is no VoIP DSP resource. | 0 = Off 1 = On (default = 1) | ✓ | | |
| 90-10-01 | System Alarm Setup | Alarm Number 68 is used for All DSP Busy. | 0 = Not set 1 = Major 2 = Minor (default = 2) | | ✓ | |

MEMO

SL2100

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