IP Multiline Station (SIP)

8

IP Multiline Station (SIP)

SECTION 1 INTRODUCTION[1]

The UNIVERGE SV8100 system supports IP extensions using a variety of multiline terminals. These telephones have the same look and functionality as typical multiline telephones, but they are connected to the CCPU via IP rather than by a hardwired connection to an ESI port.

The following DT700 IP Multiline Telephones (ITL) support IP extensions.

- ITL-2E-1 (BK) TEL
- ITL-6DE-1 (BK) TEL
- ITL-8LD-1 (BK) TEL / ITL-8LD-1 (WH) TEL
- ITL-12D-1 (BK) TEL / ITL-12D-1 (WH) TEL
- ITL-12PA-1 (BK) TEL
- ITL-24D-1 (BK) TEL / ITL-24D-1 (WH) TEL
- ITL-32D-1 (BK) TEL / ITL-32D-1 (WH) TEL
- ITL-320C-1 (BK) TEL
- ITL-12CG-3 (BK) TEL -- Note: CPU software 7000 or higher required
- ITL-12DG-3 (BK) TEL -- Note: CPU software 7000 or higher required

1. 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 DT300 multiline telephone, single line telephone or trunk, the speech must be converted from IP to TDM (Time Division Multiplexing) technology. The PZ-32IPLA, PZ-64IPLA or PZ-128IPLA daughter board provides this function.

Each PZ-32IPLA, PZ-64IPLA, PZ-128IPLA has a number of DSP resources; each can convert a speech channel from IP to TDM and vice versa.

It is possible for DT700 IP Phones to talk directly to other DT700 IP Phones without using a PZ-32IPLA, PZ-64IPLA, PZ-128IPLA DSP resource. For more information, refer to Section 6 Peer-to-Peer on page 8-8.

2.1 DT700 IP Multiline Telephones (ITL)

The IP multiline telephone operates in the same way as a DT300 (DTL) digital multiline telephone. It has all features and flexibility you expect from a DT300 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 CD-8DLCA/CD-16DLCA.

Figure 8-1 DT700 IP Telephone (ITL)

2.2 Conditions

V1100 system software (V1.10) has been optimized to improve the performance related to the following condition. However, it is still not recommended to assign the following features to a large number of phones.

When using DT700 IP phones, it is not recommended to assign the following features to a large number of phones (30 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 location CAP 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

SECTION 3 POWER FAIL ADAPTER [PSA-L ( ) UNIT]

The power fail adapter is an add-on module for the IP (DT700) multiline telephones and digital (DT300) multiline telephones. It allows connection to an analog trunk if the power or system connection fails, or the IP telephone loses connection to the UNIVERGE SV8100 system.

No programming is required on the UNIVERGE SV8100 to support this adapter.

3.1 Connecting to an IP Telephone

The Power Fail Adapter connects to an analog PSTN (Public Switched Telephone Network) line. At a small branch office, for example, this may be the same line that is used for faxes/modems/etc. The handset is also connected to the Power Fail Adapter. It is necessary to unplug it from the IP telephone and reconnect to the adapter. This allows the speech path to be redirected to the handset during a power/network failure.

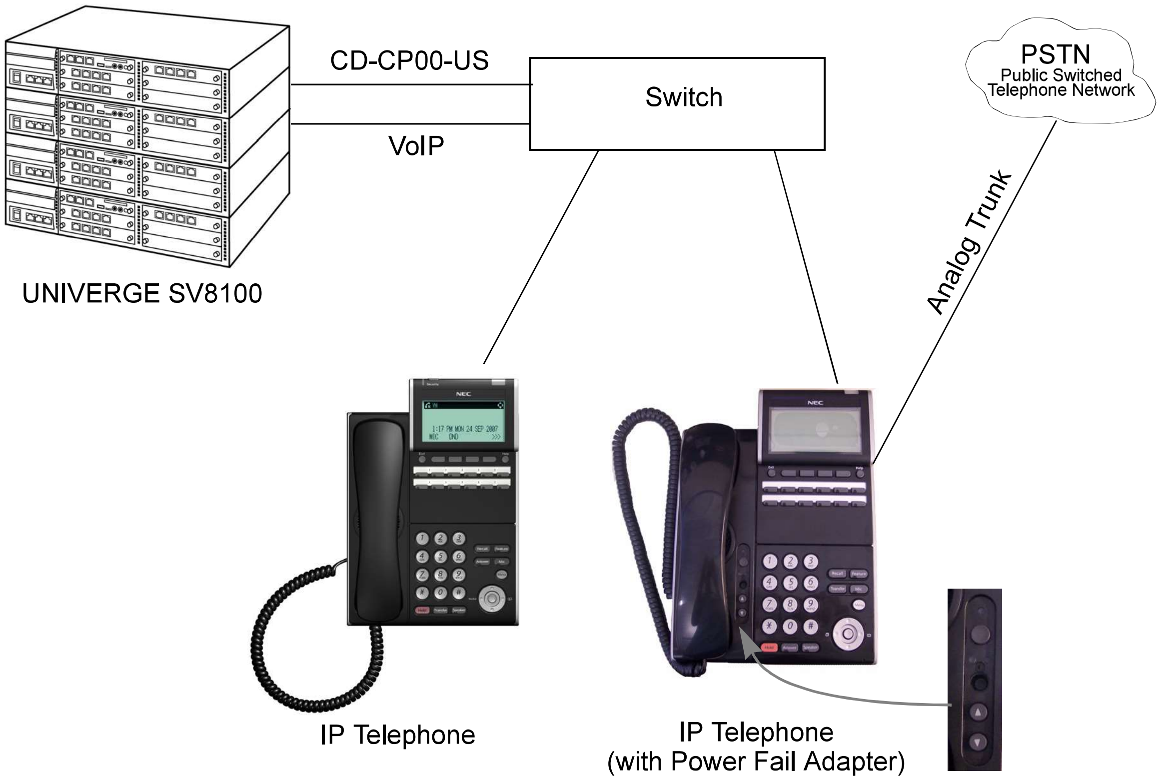


Figure 8-2 Power Fail Adapter Connection

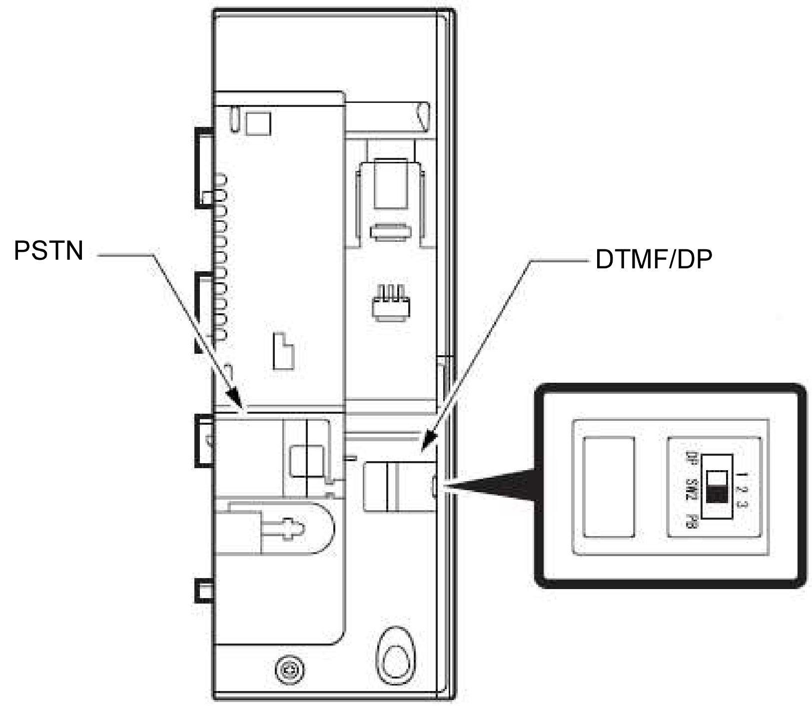


Figure 8-3 IP Telephone Connection

3.2 Operation During Power Failure

If the telephone becomes disconnected from the power supply (e.g., power loss) the telephone display is blank.

To make a call, lift the handset to receive dial tone from the analog line. Dial as normal.

If a call is received on the analog line, the Power Fail Adapter rings. Lift the handset to answer.

If the telephone is connected to the power supply, but disconnected from the UNIVERGE SV8100 system (e.g., data network failure), the IP telephone attempts to reconnect. If this fails, press the button on the top of the adapter. Refer to Figure 8-2 Power Fail Adapter Connection on page 8-4. This puts the IP telephone in analog mode. The telephone display shows *LINE -> PSTN*.

To make a call, lift the handset to receive dial tone from the analog line. Dial as normal.

If a call is received on the analog line, the Power Fail Adapter rings. Lift the handset to answer.

✎ Handsfree (Speaker) mode is not supported on calls made to or from the Power Fail Adapter. The handset must always be used.

✎ A special dial pad option is supplied with the PSA-L ( ) UNIT.

SECTION 4 LAN CONNECTION

As illustrated in Figure 8-4 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 port.

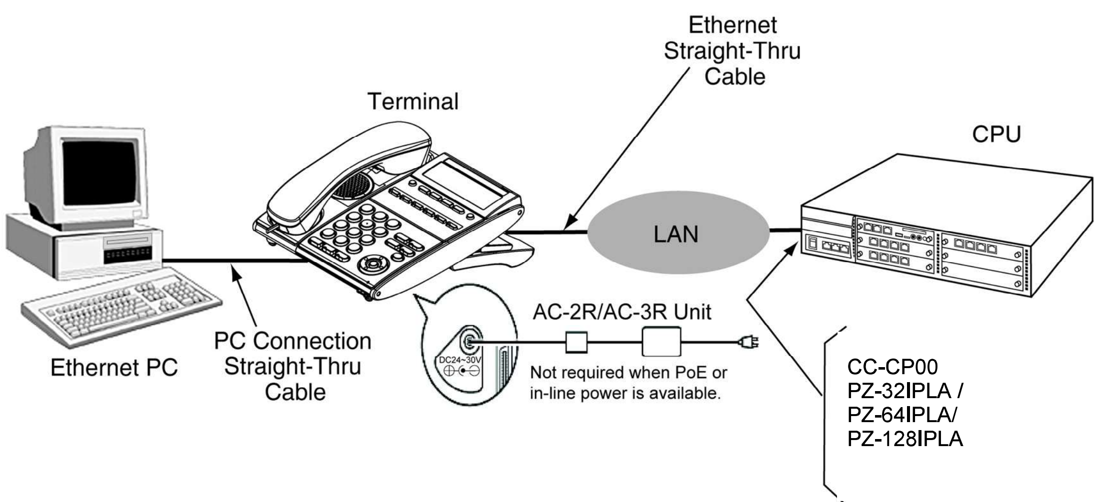


Figure 8-4 Typical Network IP Connection

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 5 PROVIDING POWER

IP telephones require power to function. This can be provided in various ways.

5.1 Local Power

The IP telephone has a connector for external power. This is supplied by an AC adapter that outputs 27V DC. This means that a power outlet is required in the vicinity of each IP Phone, and loss of power in the building prevents the telephones from working.

✎ Use only the NEC supplied power supply.

5.2 802.3af Power Over Ethernet (PoE)

A 802.3af PoE switch is a data switch that also 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).

SECTION 6 PEER-TO-PEER

An IP telephone can send and receive RTP packets to or from another IP telephone without using DSP resources on a PZ-32IPLA, PZ-64IPLA, PZ-128IPLA. This operation allows only Intercom calls between the IP telephones.

If a DT700 IP multiline telephone or trunk line is required, a DSP resource is needed and a PZ-32IPLA, PZ-64IPLA, PZ-128IPLA 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 PZ-32IPLA, PZ-64IPLA, PZ-128IPLA. 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 UNIVERGE SV8100 chassis must still have a registered PZ-32IPLA, PZ-64IPLA, PZ-128IPLA 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 10-26-04 – DT700 Peer to Peer.

SECTION 7 MISCELLANEOUS

7.1 System Tones and Ring Tones

IP Phones do not use Program 80-01 : Service Tone Setup entries. The tones are generated locally by the IP telephone. When a Door Box chime rings an IP telephone, the system activates the chimes using a ring command. Because of this, if the volume is adjusted while the door chime is sounding, the ringing volume of the IP Phone is adjusted.

7.2 Music on Hold

In addition, 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 8 CONFIGURATION EXAMPLES

8.1 IP Addressing

When using a PZ-(X)IPLA, for every 16 DSPs used, one valid IP address must be assigned. In addition to the DSP IP address, another IP address is required for registration and signaling. For full functionality of the IPLA 32, three valid IP addresses must be assigned, all in the same subnet.

When using a PZ-(X)IPLB, only 1 IP Address is needed for all DSP's. In addition to the one DSP IP address, another IP address is required for registration and signaling. For full functionality of the IPLB(32/64/128), two valid IP addresses must be assigned, all in the same subnet.

The following chart shows the minimum and maximum number of IP addresses used with different IPLA/IPLB card configurations.

Card	Minimum IP Addresses	Maximum IP Addresses	Notes
PZ-32IPLA	2	3	With a PZ-(XX)IPLA One IP address is required for 16 DSP channels. One additional IP Address is required for registration and signaling. With a PZ-(XX)IPLB One IP Address is required for ALL DSP Channels and one additional IP address is required for registration and signaling.
PZ-64IPLA	2	5	
PZ-128IPLA	2	9	
PZ-32IPLB/ PZ-64IPLB/ PZ-128IPLB	2	2	

✎ When assigning the IP addresses to the IPLA/IPLB card, the addresses must be in the same network (subnet). If the CPU is also to be connected to the network, it requires a separate IP address in a different network (subnet). When a IPL() card is installed it is recommended to not use the CPU NIC and to change PRG 10-12-01 to 0.0.0.0

When an IPLA/IPLB card is attached to the CPU, using the CPU NIC is no longer required. All connections that previously terminated to the CPU NIC card can now be terminated to the IPLA/IPLB NIC. E.g. PCPro, Web Pro, ACD, etc. all terminate to the IPLA/IPLB NIC card when installed.

The IPLA/IPLB and the CPU NIC share the same gateway assignment. The default gateway command in Program10-12-03 is used by both NICs allowing only one device, or CPU, to route outside of its own network.

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).

8.2 Example Configuration 1 - Static IP Addressing, One LAN

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

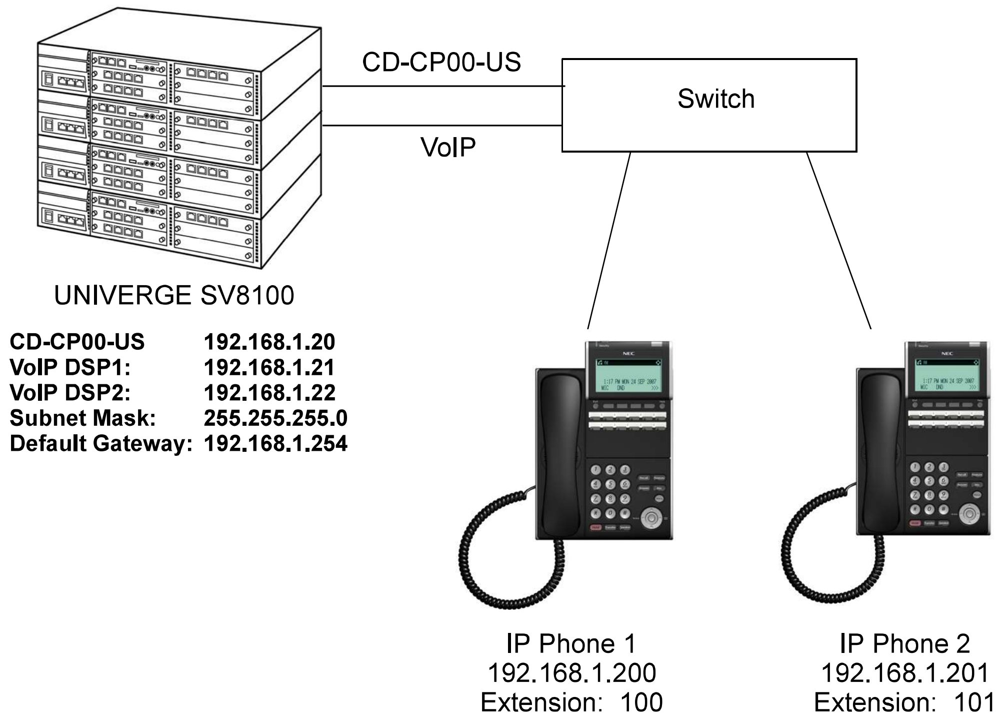


Figure 8-5 Example Configuration 1 - Static IP Addressing, One LAN

Programming - CD-CP00-US:

10-12-01 : CD-CP00-US Network Setup - IP Address (for CD-CP00-US)	0.0.0.0
10-12-02 : CD-CP00-US Network Setup - Subnet M ask	255.255.255.0
10-12-03 : CD-CP00-US Network Setup - Default Gateway	192.168.1.254
10-12-09 : CD-CP00-US Network Setup - IP Address	192.168.1.20

(This assignment is for CD-CP00-US when the PZ-32IPLA, PZ-64IPLA, PZ-128IPLA daughter board is installed.)

Programming - PZ-32IPLA, PZ-64IPLA, PZ-128IPLA:

84-26-01 : IPL 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

8.3 Example Configuration 2 - Dynamic IP Addressing, One LAN

This 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 Chapter 3 General IP Configuration.

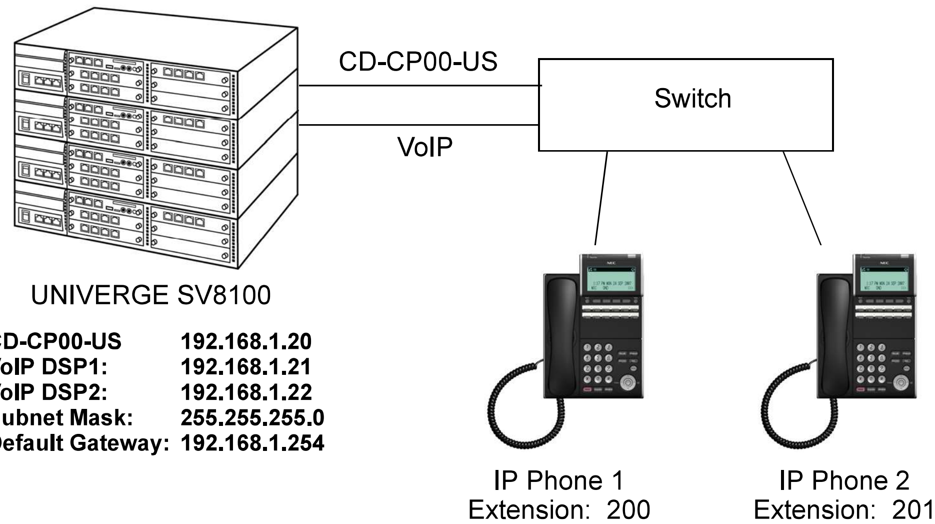


Figure 8-6 Example Configuration 2 - Dynamic IP Addressing, One LAN

Programming - CD-CP00-US:

10-12-01 : CD-CP00-US Network Setup - IP Address (for CD-CP00-US)	0.0.0.0
10-12-10 : CD-CP00-US Network Setup - Subnet Mask	255.255.255.0
10-12-03 : CD-CP00-US Network Setup - Default Gateway	192.168.1.254
10-12-09 : CD-CP00-US Network Setup - IP Address	192.168.1.20
(This assignment is for CD-CP00-US when the PZ-32IPLA, PZ-64IPLA,	
PZ-128IPLA daughter board is installed.)	On
10-13-01 : In-DHCP Server Setup - DHCP Server Mode	Enable
10-14-01 : Managed Network Setup	192.168.1.200 Min
10-14-02: Managed Network Setup	192.168.1.250 Max
10-16-16: SIP Server Address	192.168.1.20
10-16-27: SIP Server Port	5080

Programming - PZ-32IPLA, PZ-64IPLA, PZ-128IPLA:

84-26-01 : IPL Basic Setup - IP Address	192.168.1.21
	192.168.1.22

Programming - IP Phones:

DHCP Mode (Ext. 200):	Enabled

8.4 Example Configuration 3 - Static IP Addressing, Routed WAN

This example shows IP Phones connected to an UNIVERGE SV8100 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.

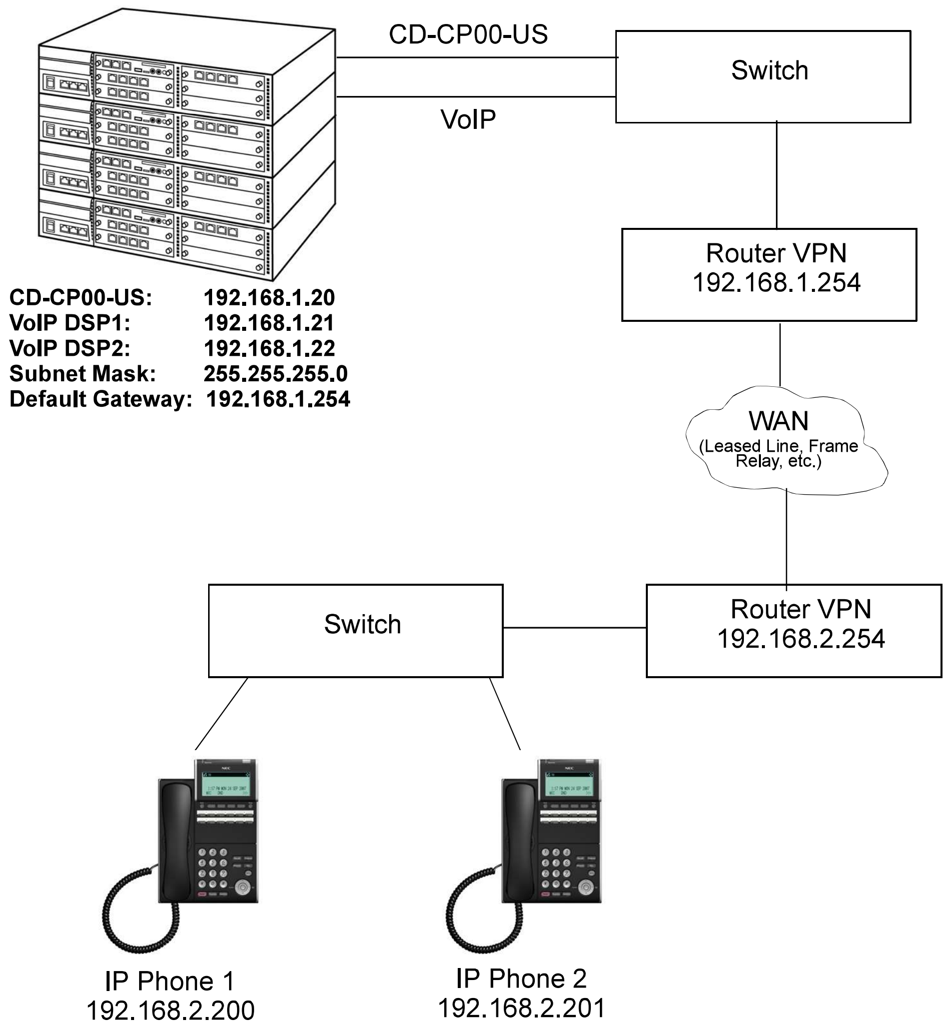


Figure 8-7 Example Configuration 3 - Static IP Addressing, Routed WAN

Programming - CD-CP00-US:

10-12-01	: CD-CP00-US Network Setup - IP Address (for CD-CP00-US)	0.0.0.0
10-12-10	: CD-CP00-US Network Setup - Subnet Mask	255.255.255.0
10-12-03	: CD-CP00-US Network Setup - Default Gateway	198.168.1.254
10-12-09	: CD-CP00-US Network Setup - IP Address	192.168.1.20

(This assignment is for CD-CP00-US when the PZ-32IPLA, PZ-64IPLA, PZ-128IPLA daughter board is installed.)

Programming - PZ-32IPLA, PZ-64IPLA, PZ-128IPLA:

84-26-01	: IPL Basic Setup - IP Address (Slot # - DSP)	192.168.1.21 192.168.1.22

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 9 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 DT700 telephone, press the **Menu** button to enter program mode. The IP User Menu is displayed.
2. On the IP User Menu, select **Config (0)** for the IP Phone. Settings are listed in Table 8-1 IP Phone Programming Options User Menu.

Table 8-1 IP Phone Programming Options User Menu

Programming Option	Default
UserName	ADMIN
Password	6633222

SECTION 10 DHCP SERVER CONFIGURATION

It is possible to use either an external DHCP server (e.g., Windows Server) or the UNIVERGE SV8100 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 8.3 Example Configuration 2 - Dynamic IP Addressing, One LAN on page 8-11.

When using an external DHCP server, you must add a new Option Code to the DHCP scope for the PZ-(X)IPLA/IPLB address. The method for adding this service varies depending on the DHCP server used.

SECTION 11 AUTO CONFIG FOR IP TERMINALS SV8100

11.1 Required Equipment

- IP Phone Manager and software to create the config file for the IP terminals. Software is available for download free on the NEC website.
- FTP server and free software available on the web.
- DHCP Server supporting ability to define the following:
  - Vendor Class
  - Option Codes

11.2 Building Config File

1. Launch the IP Phone Manager software.
2. Once the software is launched, click **Auto Config**.

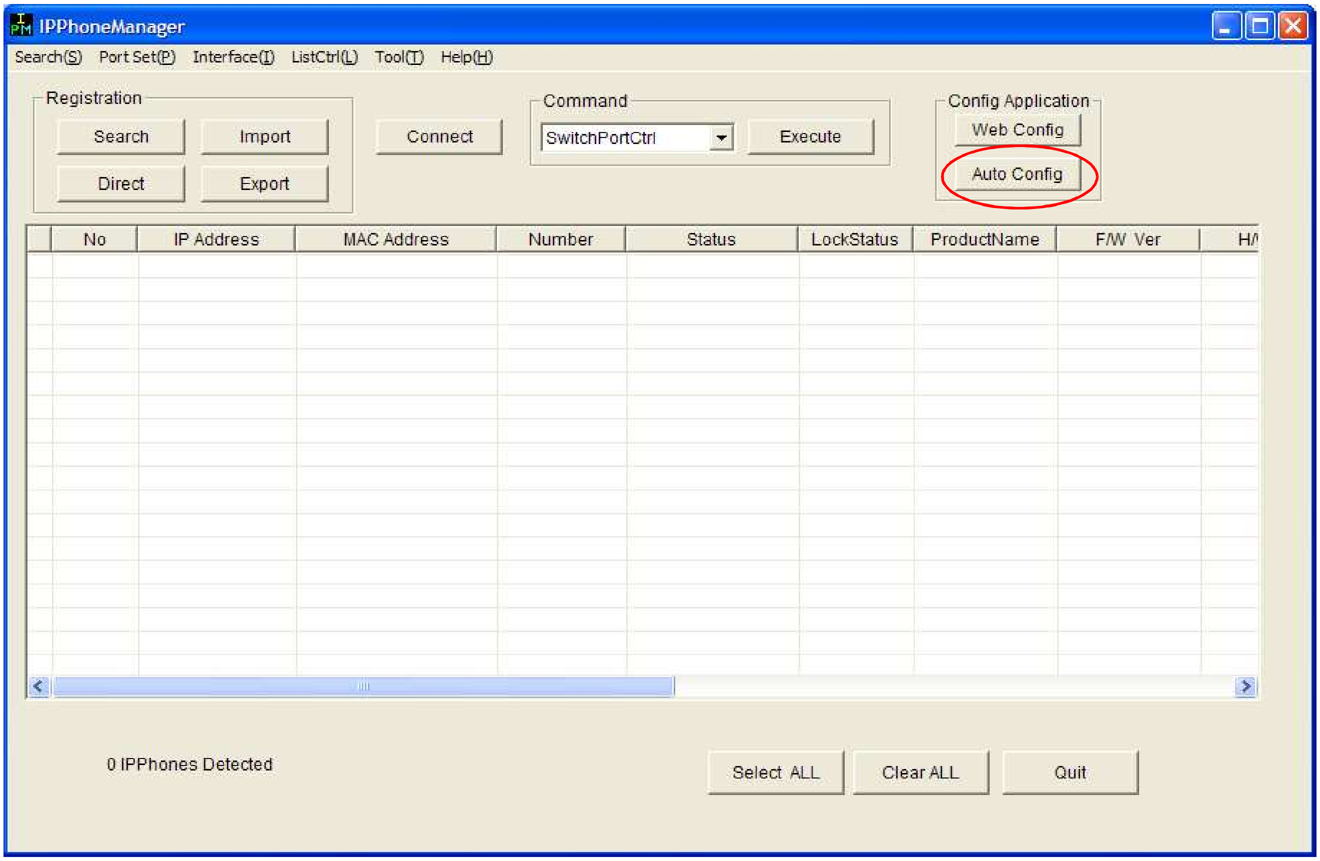


Figure 8-8 IP Phone Manager

3. Click **Terminal**.

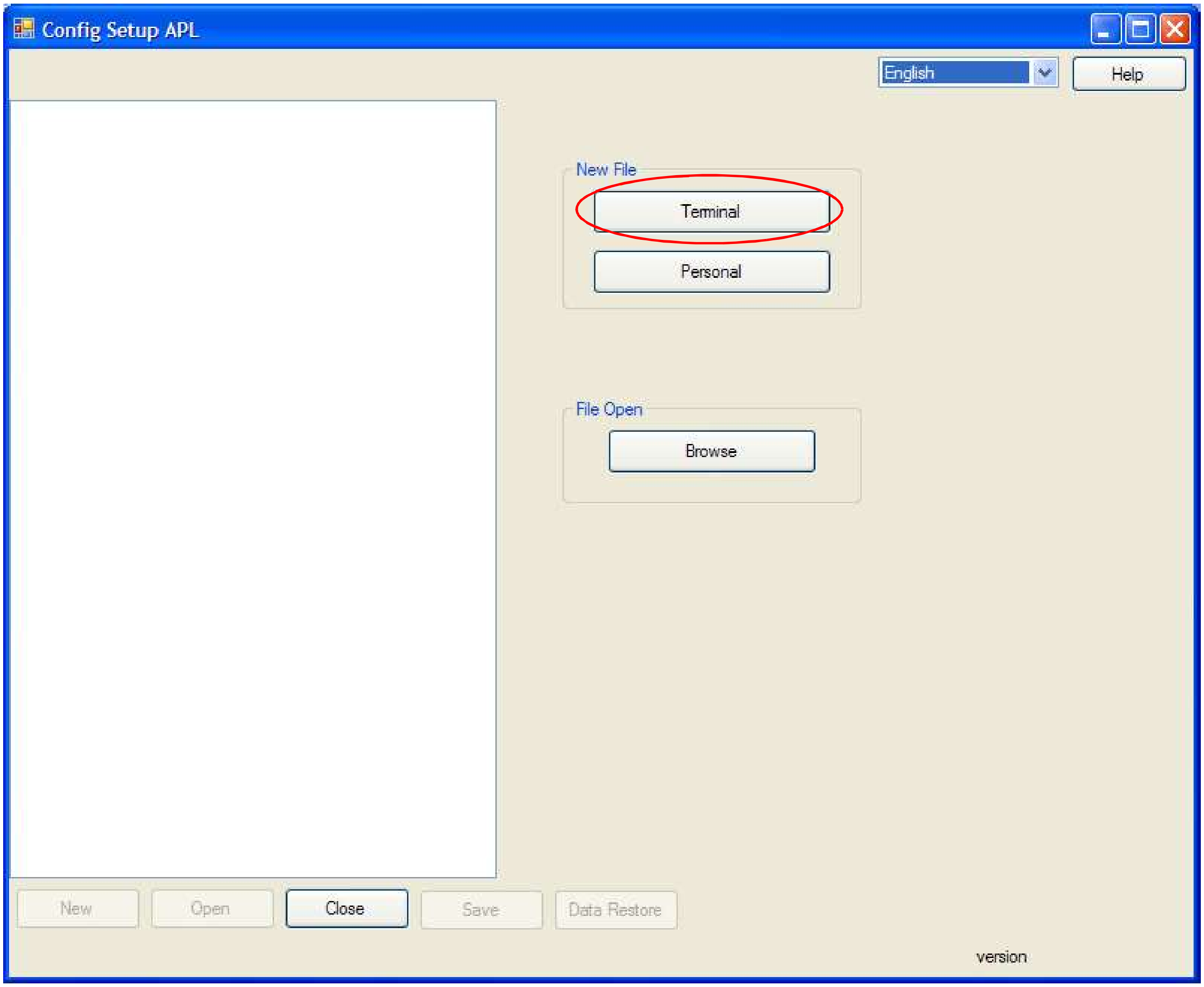


Figure 8-9 Config Setup APL

4. Click **1st Server Address**.
5. Assign the 1st Server Address using the IP address programmed in command 10-12-09.

6. Click **OK**.

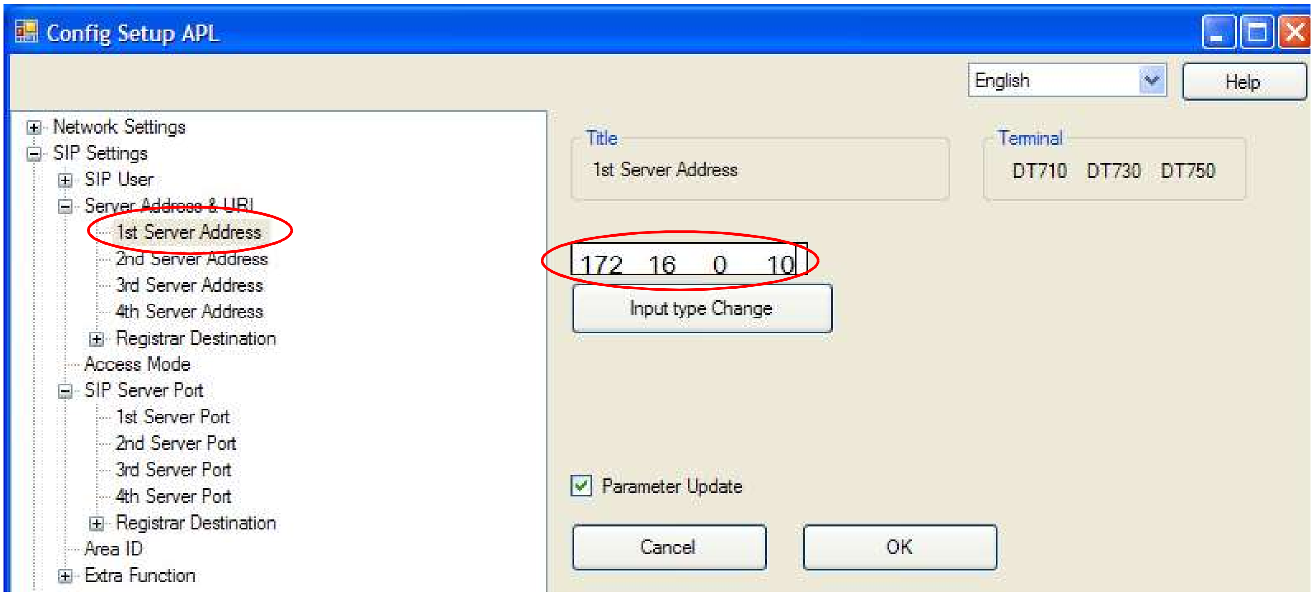


Figure 8-10 Assign IP Address

7. Click **1st Server Port**. Assign port number 5080.
8. Click **OK**.

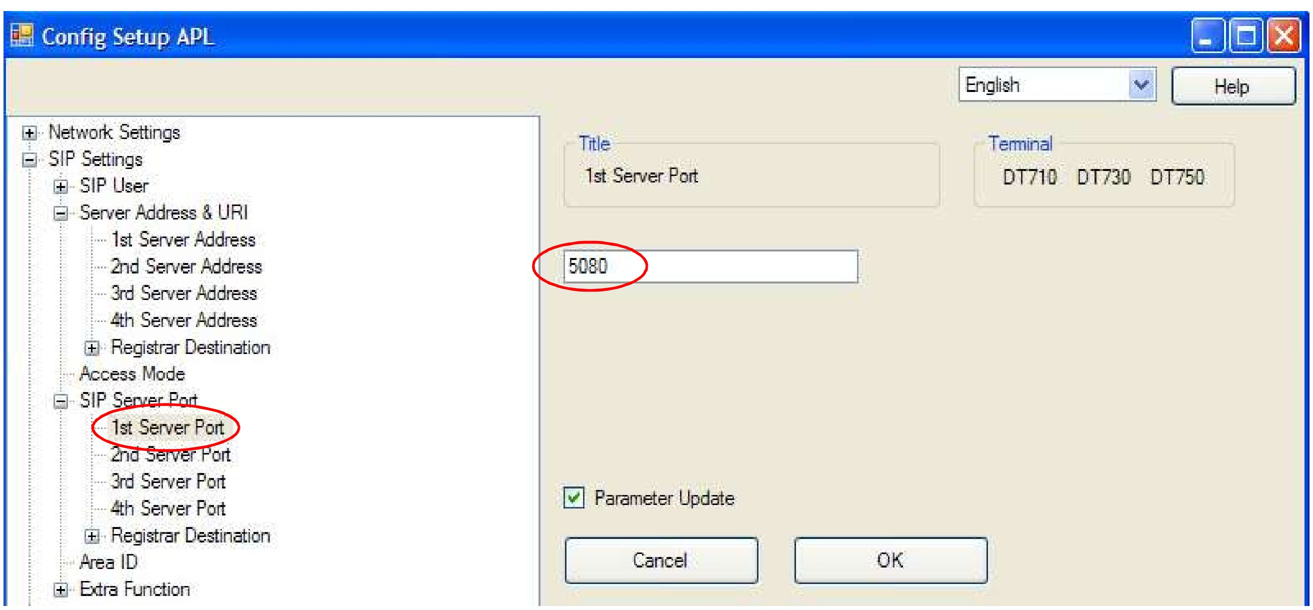


Figure 8-11 Network Settings - 1st Server Port

9. After the changes are made, click **Save**.
10. When the Save window opens, click **Save as...**

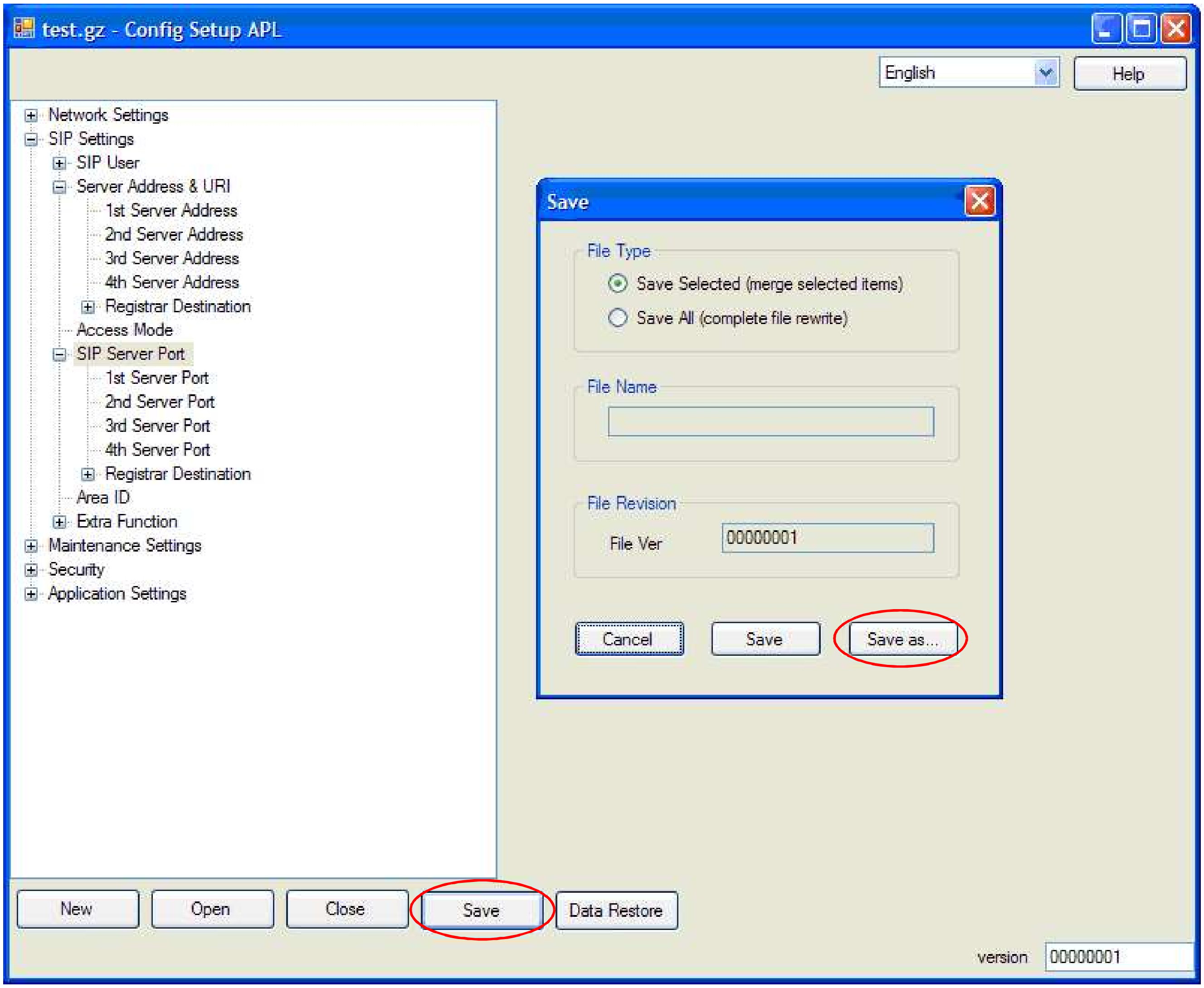


Figure 8-12 Save As/Name File Window

11. In the Save As window, name the file xxx.gz.

    Example: To name the file test, enter **test.gz**
12. Place this file in the FTP server.

11.3 Configuring an FTP Server

The file generated in the IP Phone Manager must be placed in an anonymous login folder. The FTP server must be configured with an anonymous login account.

The following procedure is an example using Quick and Easy FTP server.

1. Click **Configure Settings**.

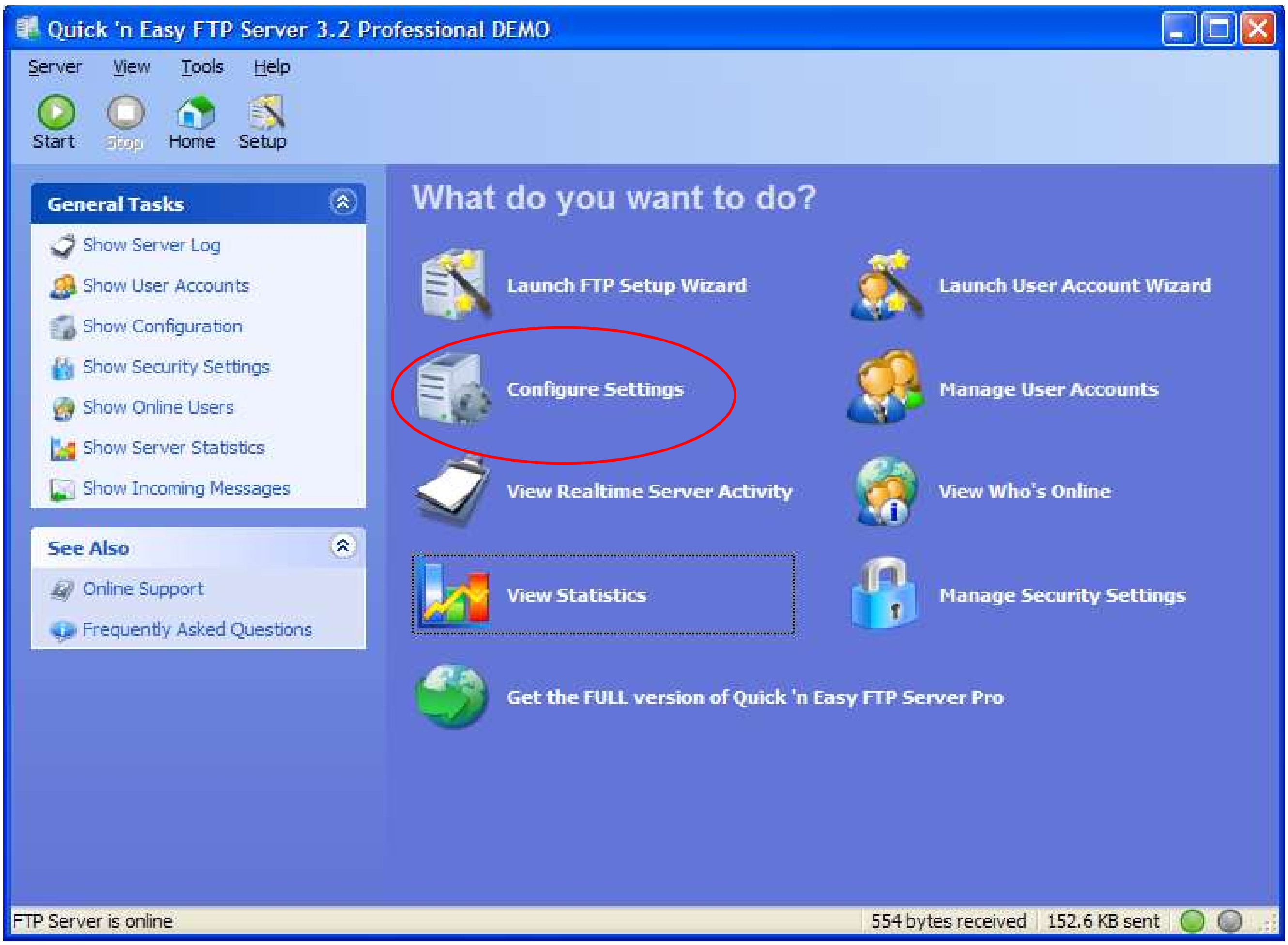


Figure 8-13 Quick 'n Easy FTP Server

2. Click on the **Advanced** tab to specify the directory to store files.

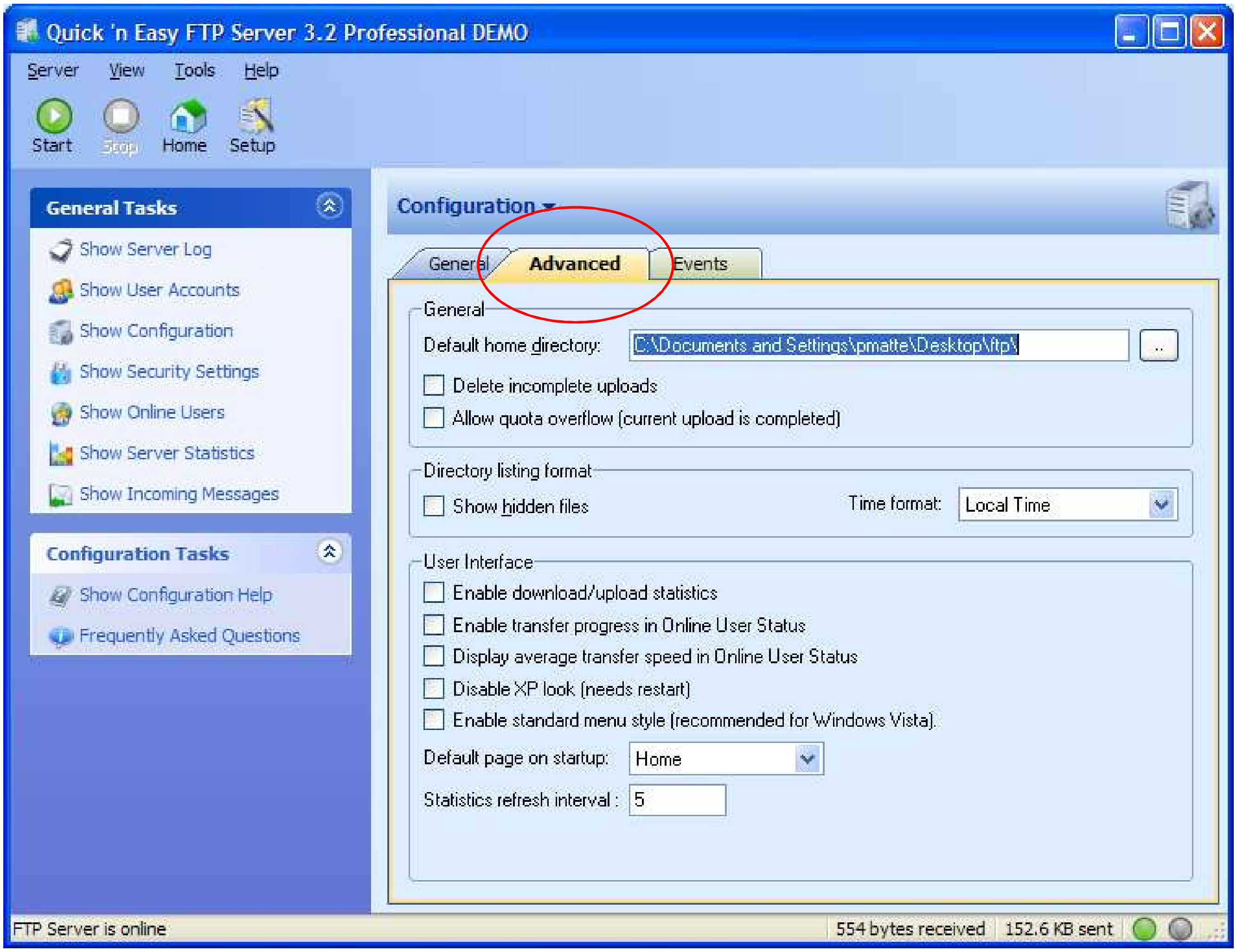


Figure 8-14 Quick 'n Easy FTP Server Configuration

3. Place the file (for example: test.gz) in the Default home directory.
4. After the file is loaded to the proper directory, click **Start** to start the FTP server.
5. At this point the FTP server can be minimized to run in the background.

11.4 DHCP Server Setup Windows Server

Defining Vendor Classes

1. In the DHCP server highlight the server machine on the left side. Right click on the server and select **Define Vendor Classes**.
2. Click **ADD**.
3. Display Name = NECDT700
4. Description = auto config
5. In the same window down below there is a section that shows ID, Binary, and ASCII. Click under ASCII.
6. Enter **NECDT700**. This should have also added 4E 45 43 44 54 37 30 30 under the binary section.
7. Click **OK** and close.

Setting Predefined Options

1. Highlight the server again. Then right click and select **Set Predefined Options**.
2. Change the option class to **NECDT700**.
3. Click **ADD,** and provide the following information:
   - ❒ Name = FTP Address
   - ❒ Data Type = IP address
   - ❒ Code = 141
   - ❒ Click **OK,** and start the process over again.
4. Click **ADD,** and provide the following information:
   - ❒ Name = Auto Config File Name
   - ❒ Data Type = String
   - ❒ Code = 151

5. Click on **ADD**, and give the following information:

    - ❒ Name = Download Protocol
    - ❒ Data Type = Byte
    - ❒ Code = 163

6. Click **OK**.

Configuring Options

1. Highlight scope options on the left side. Then right click and choose **Configure Options**.
2. Click **Advanced** and change the vendor class to NECDT700.
3. Place a check mark next to 141 FTP Address. Down below assign the IP address of the FTP server. Then click **Apply**.
4. Place a check mark next to 151 auto config file name. Enter the name of the config file created using IP Phone Manager. Then click **Apply**.
5. Place a check mark next to 163 download protocol. Down below change the HEX address to be 0x1.
6. Click on **Apply** and **OK**.

SECTION 12 IP TELEPHONE REGISTRATION AND DELETION

When an IP Phone connects to the UNIVERGE SV8100 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 PZ-32IPLA/PZ-64IPLA/PZ-128IPLA.
- ❍ 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:

Via Telephone Programming:

Enter Program 90-23-01, and enter the extension number of the IP Phone. Press 1 and Transfer to delete the registration.

Via Web Pro:

Enter Program 90-23-01, and place a check next to the extension number of the IP Phone. Click on Apply to delete the registration.

SECTION 13 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.

With **Version 7000 and higher** Program 90-68-01 can be used to automatically test the lines and auto assign the proper values in Program 81-07. It is recommended to use this program whenever analog trunks are involved.

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.

SECTION 14 FIRMWARE UPGRADE PROCEDURE

A new version of NEC firmware for the IP Phones can be applied automatically or manually.

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.)

14.1 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 **itlisip(e,s,v).tgz** to the default FTP/TFTP directory.

3. To enter Programming Mode, press the **Help** button on the IP Phone.

4. To enter Maintenance Mode, press **0** (Config) and **#3** (Maintenance).

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 **3** (Boot Program).

10. Press the softkey.

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

14.2 Checking the Firmware Version

To check the IP Phone firmware version:

1. Press and hold the **Help** button on the System IP Phone.

2. To access information, press **8** (System Information).

3. To display the firmware version, press the **Up** softkey.

14.3 Upgrading Automatically

This procedure causes all IP Phones to attempt firmware upgrade the next time they connect to the CD-CP00-US. This can make the upgrade procedure easier, as it is not necessary to visit every telephone to perform the upgrade.

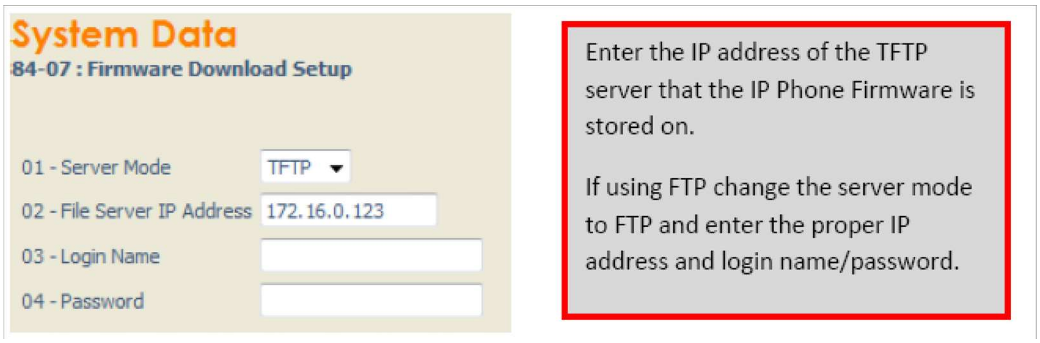
This can cause problems if, for example, a PoE (Power over Internet) switch is used. When the PoE switch is powered up, all telephones connect to the FTP/TFTP server at the same time. This causes a large amount of data for the FTP/TFTP server to transfer over the data network.

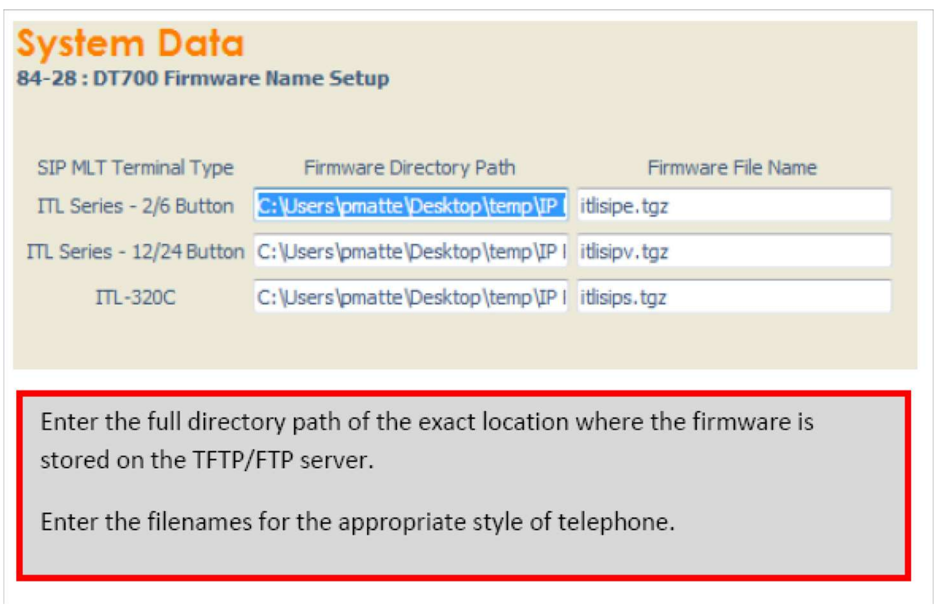
To avoid this, connect the telephones to the PoE switch gradually, to allow time for each telephone to upgrade before connecting the next.

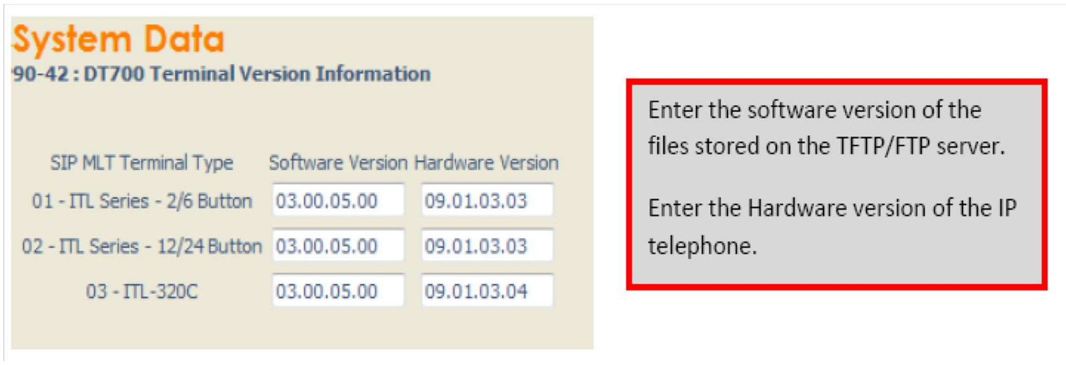
To enable automatic upgrade:

When the IP phone boots up and connects to the SV8100 it receives download information from the system that includes firmware information. When the SV8100 reports a version that is different than the version the IP phone is currently utilizing, the IP phone will initiate the upgrade procedure.

Below is an example of this setup. *** Note *** This is just an example, you must enter your own local information.







SECTION 15 IP STATION (SIP MULTILINE TELEPHONE)

15.1 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 use the following operation:

Press the "Down Arrow" Soft Key, press the "Prog" soft key, and then press the "LOGOFF" soft key.

Multiple Login

With **Version 3000 or higher** software, the same user name and password can be assigned to multiple extensions when using Automatic or Manual Registration. This makes it easier on the user by only having to remember one password. For example, if a user has an IP Multiline terminal, MH240 handset, and uses Desktop Applications with the Enhancement bundle controlling the IP Multiline, three different ports are used in the system. Prior to Version 3000, each IP port required a unique user name and password. With Version 3000 all three can be assigned the same user name and password.

Encryption

This feature is supported with main software Version 2.5 or above, and terminal firmware 2.1.1.0 or above. The SV8100 Supports AES 128-bit encryption between DT700 terminals and the IPLA/IPLB. This feature requires the LK- SYS-ENCRYPTION-LIC which is a system license. Once installed, any of the DT700 IP terminals can use the encryption feature.

Table 8-2 DT700 Supported Encryption

Source	Destination	SRTP	Comment
DT700	DtermIP	S	DT700 VoIPDB Encryption between DT700 and VoIPDB
DT700	SDT SIP (P2P)	N	
DT700	STD SIP (Non P2P)	S	DT700 VoIPDB Encryption between DT700 and VoIPDB
DT700	DT700	S	MH240 without MH240
DT700	PSTN	S	DT700 VoIPDB Encryption between DT700 and VoIPDB
DT700	IP Network (SIP/H323/ CCIS)	S	DT700 VoIPDB Encryption between DT700 and VoIPDB
DT700 Encryption On	DT700 Encryption Off	N‘	
S = Supported N = Not Supported			

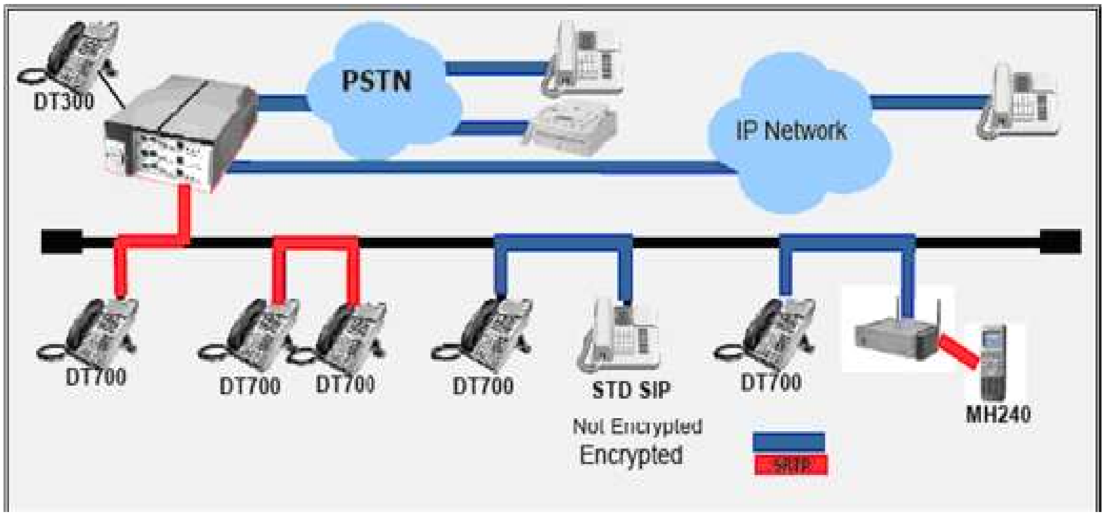


Figure 8-15 DT700 Encryption

Conditions

- ❍ Encryption is not supported on DT700 series phones that are connected via NAPT.
- ❍ Encryption is not supported on DT700 series phones that are registered to a secondary NetLink system.
- ❍ With Version 4000 or higher main software, DT700 series phones that are registered to a primary NetLink can fail over to a secondary system regardless of encryption settings.
- ❍ With Version 4000 or higher main software, if the encryption feature is enabled in terminal programming but not licensed, the terminal display will no longer display "Invalid Info", however the encryption feature will not work until licensed.

15.2 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 15.5 Bandwidth on page 8-47.

Depending on how QOS policies are built in the network, assignments may be needed in both the CPU and IP terminal. The UNIVERGE SV8100 supports the flagging of packets at layer 2 (VLAN tagging 802.1Q/802.1P) and at layer 3 levels.

15.3 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.

Currently the IPLA/IPLB and CPU do not support the tagging of VLAN packets. These devices also do not support receiving a frame with a VLAN tag. If either device receives a packet with a VLAN tag, it is treated as an illegal frame and discarded. Therefore when the CPU/IPLA/IPLB is plugged into a data switch supporting VLANS, the VLAN tag must be removed before passing the frame onto the CPU/IPLA/IPLB.

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.

15.3.1 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**.

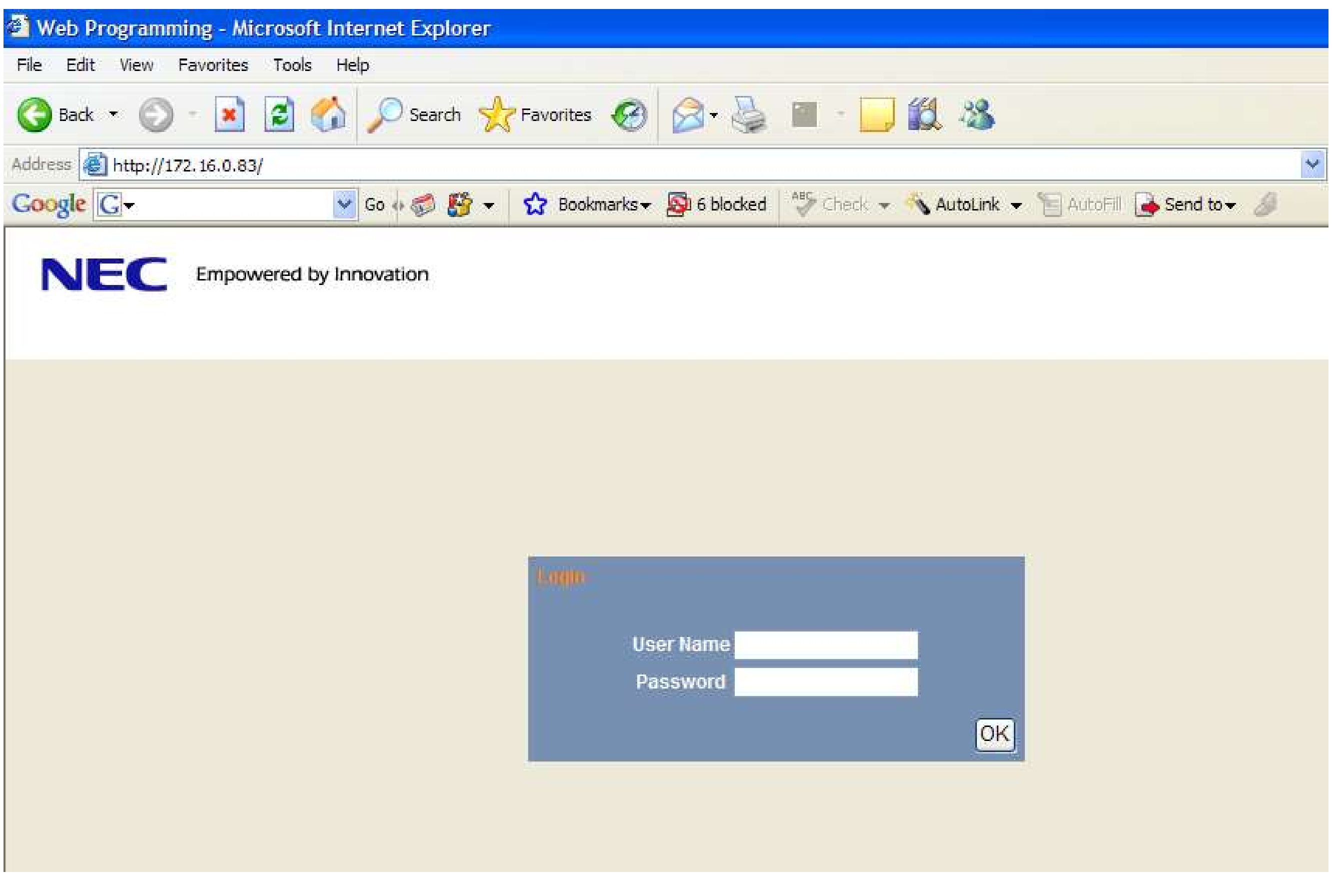


Figure 8-16 Log In to IP Phone

15.3.2 Tagging Voice Packets Using IP Phone

1. Log in. Refer to Section 15.3.1 Logging In on the PC on page 8-32.

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

3. Access the following three menus to select options for LAN Port Settings:

   - VLAN Mode
   - VLAN ID
   - VLAN Priority

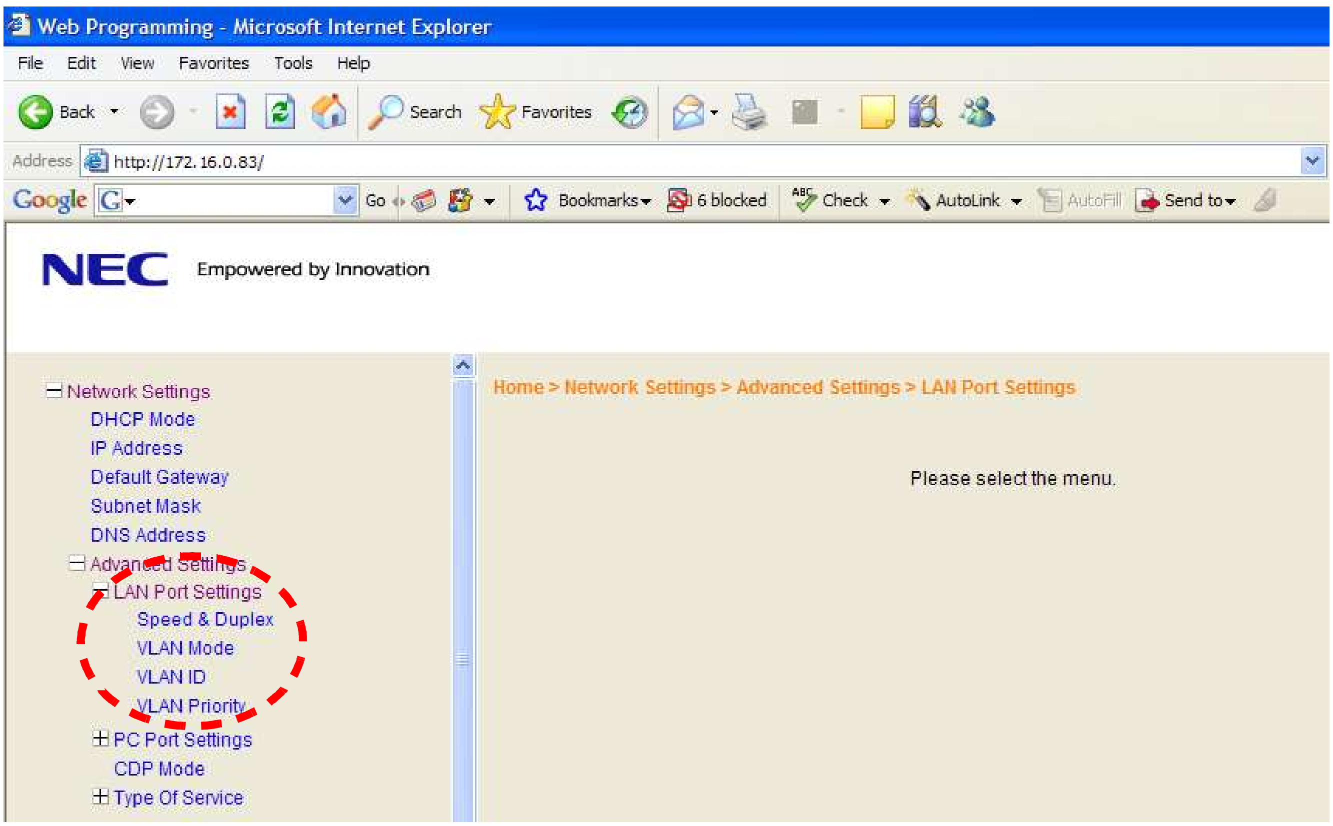


Figure 8-17 LAN Port Settings Window

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

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

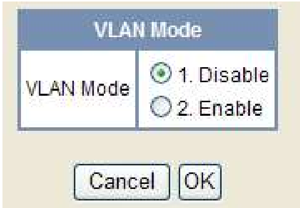


Figure 8-18 VLAN Mode

6. 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**.

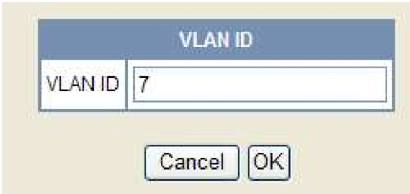


Figure 8-19 VLAN ID

7. 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**.

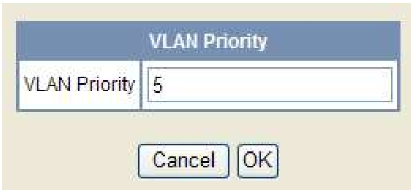


Figure 8-20 VLAN Priority

15.3.3 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 15.3.1 Logging In on the PC on page 8-32.

2. Access the following three menus to select options for PC Port Settings:

   - Port VLAN Mode
   - Port VLAN ID
   - Port VLAN Priority**.**

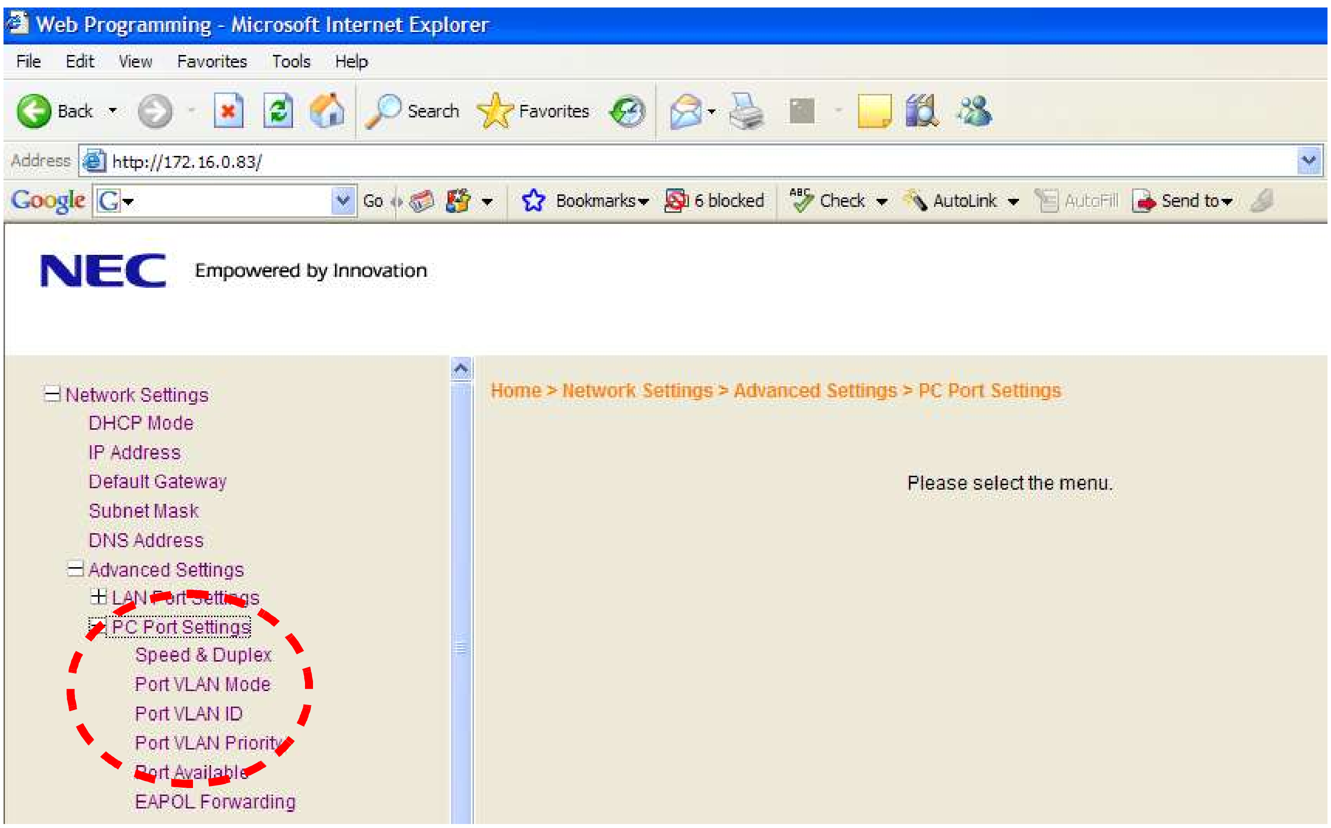


Figure 8-21 PC Port Settings Window

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

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

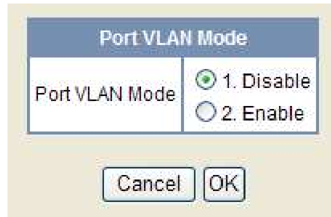


Figure 8-22 Port VLAN Mode

✎ 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**.

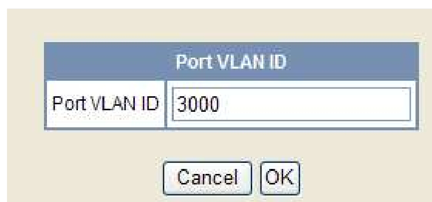


Figure 8-23 Port VLAN ID

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**.

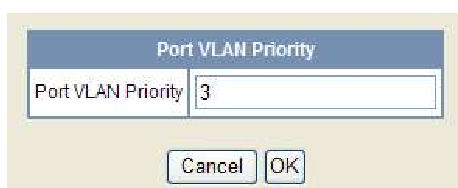


Figure 8-24 Port VLAN Priority

7. Click **Save**.

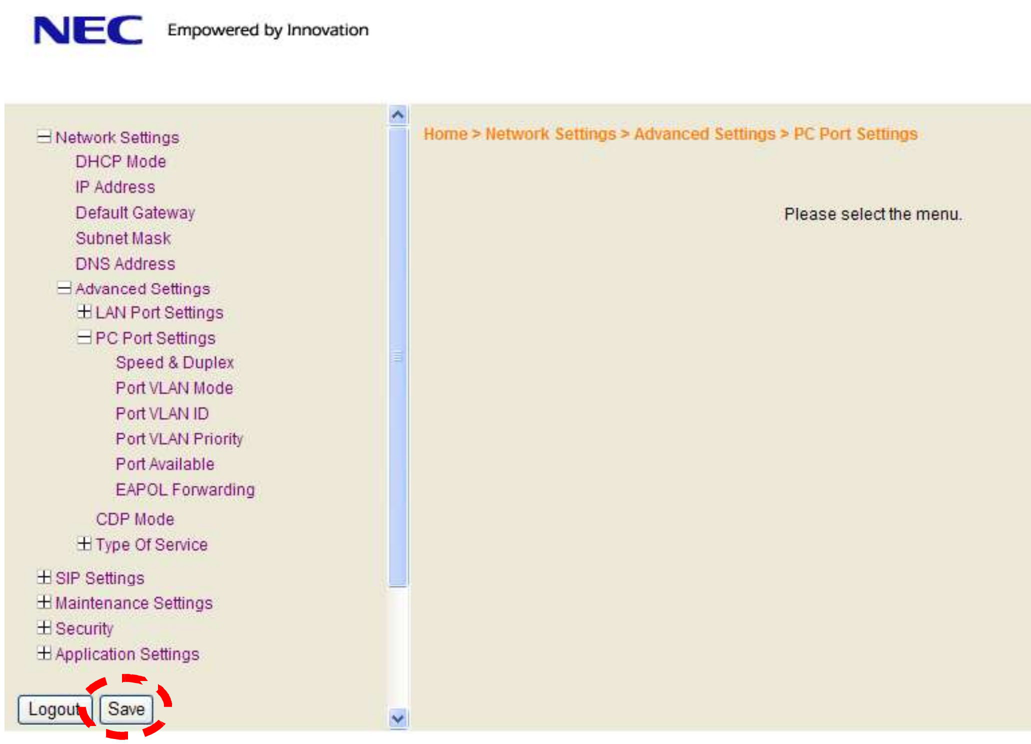


Figure 8-25 Save Network Settings

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

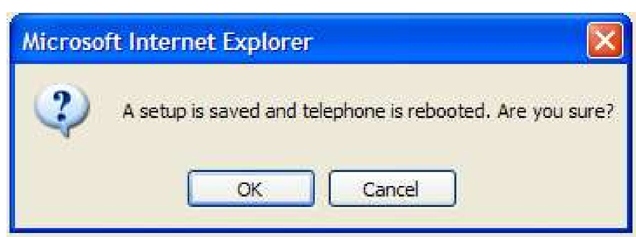


Figure 8-26 Save Confirmation Window

15.3.4 Entering VLAN Settings by Phone (Voice Packets Only)

1. Log in. Refer to Section 15.3.1 Logging In on the PC on page 8-32.
2. Press and hold the phone **Menu** button until the screen changes.
3. Press **0** on the dial pad to access Configuration.
4. Press the **OK** soft key. No login or password is associated with the IP terminal in the default state.
5. Press **1** on the dial pad for Network Settings.
6. Press **6** on the dial pad for Advanced Settings
7. Press **1** on the dial pad for LAN Port Settings (VLAN for the voice packets only)
8. Press **2** on the dial pad for VLAN Mode
9. Press **1** or **2** to disable/enable the VLAN for the voice packets. Press the OK soft key after the setting is changed.
10. Press **3** on the dial pad for VLAN ID
11. Enter a valid VLAN ID of 1~4094. Press the **OK** soft key after the setting is changed.
12. Press **4** on the dial pad for VLAN Priority
13. Enter the VLAN priority of 0~7. Press the **OK** soft key after the setting is changed.
14. If no more changes are made, press the **Exit** soft key three times. Then press the **Save** soft key, and the phone reboots.

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

1. Log in. Refer to Section 15.3.1 Logging In on the PC on page 8-32.
2. Access the main menu.
3. Press **6** on the dial pad for Advanced Settings.
4. Press **2** on the dial pad for PC Port Settings.

5. Press **2** on the dial pad for Port VLAN Mode.
6. Press **1** or **2** to disable/enable the VLAN for the data packets. Press the **OK** soft key after the setting is changed.
7. Press **3** on the dial pad for Port VLAN ID.
8. Enter a valid VLAN ID of 1~4094. Press the **OK** soft key after the setting is changed.
9. Press **4** on the dial pad for Port VLAN Priority.
10. Enter the VLAN priority of 0~7. Press the **OK** soft key after the setting is changed.
11. If no more changes are made, press the **Exit** soft key three times. Then press the **Save** soft key, and the phone reboots.

15.4 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 UNIVERGE SV8100 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 Precedence	IP Precedence	IP Precedence	Delay	Throughput	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)

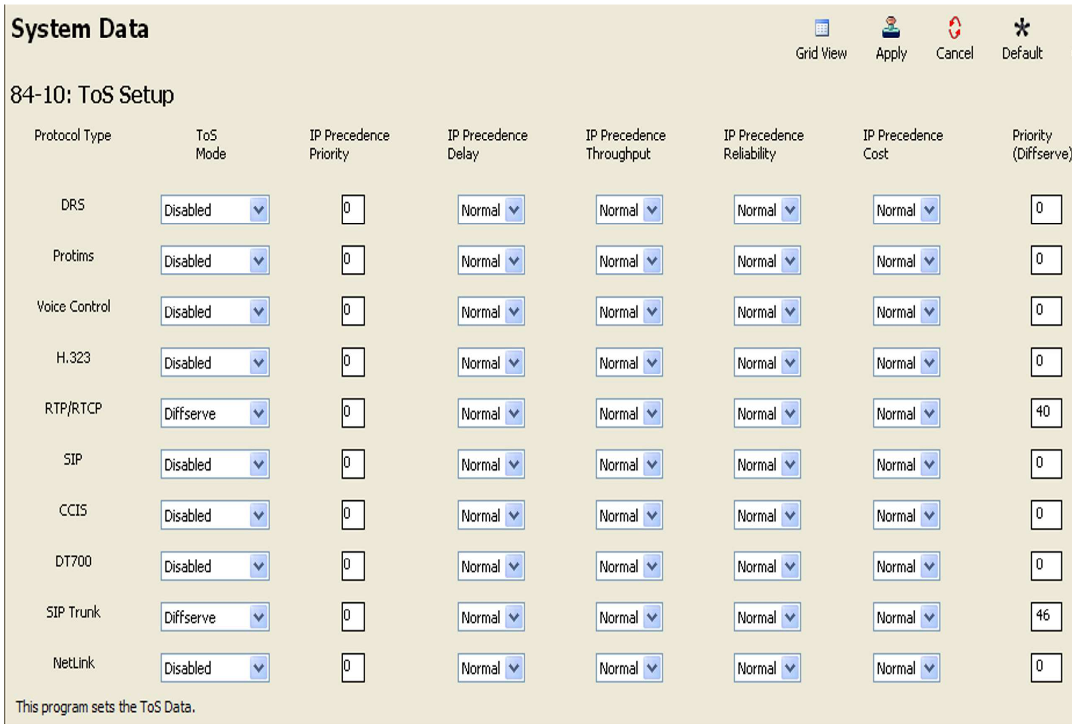
DSCP Code Points	Binary Values	Names
100000	32	Class Selector 4 (CS4)
100001	34	Assured Forwarding 41 (AF41)
100100	36	Assured Forwarding 42 (AF42)
100110	38	Assured Forwarding 43 (AF 43)
101110	46	Expedited Forwarding (EF)
110000	48	Class Selector 6 (CS6)
111000	56	Class Selector 7 (CS7)

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 IPLA/IPLB card. This does not affect the packets sent from the IP terminals.



Protocol Type	ToS Mode	IP Precedence Priority	IP Precedence Delay	IP Precedence Throughput	IP Precedence Reliability	IP Precedence Cost	Priority (Diffserve)
DRS	Disabled	0	Normal	Normal	Normal	Normal	0
Protims	Disabled	0	Normal	Normal	Normal	Normal	0
Voice Control	Disabled	0	Normal	Normal	Normal	Normal	0
H.323	Disabled	0	Normal	Normal	Normal	Normal	0
RTP/RTCP	Diffserve	0	Normal	Normal	Normal	Normal	40
SIP	Disabled	0	Normal	Normal	Normal	Normal	0
CCIS	Disabled	0	Normal	Normal	Normal	Normal	0
DT700	Disabled	0	Normal	Normal	Normal	Normal	0
SIP Trunk	Diffserve	0	Normal	Normal	Normal	Normal	46
NetLink	Disabled	0	Normal	Normal	Normal	Normal	0

Figure 8-27 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 8-28 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

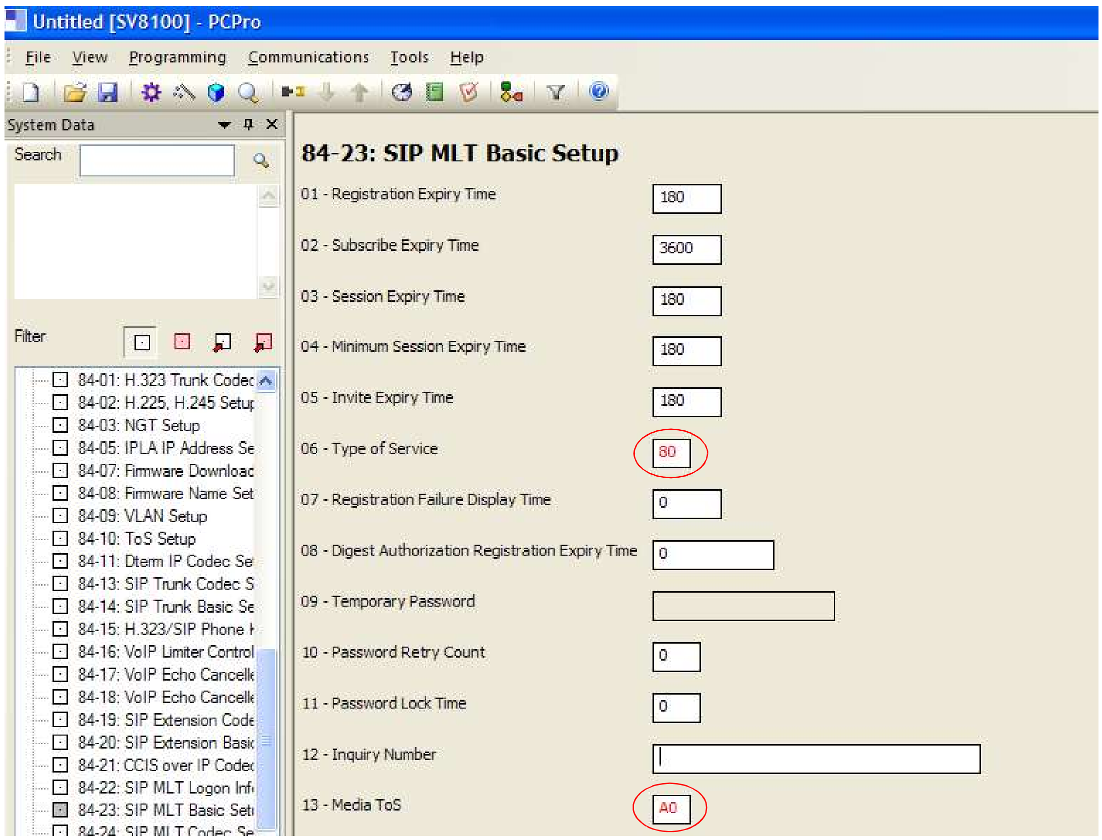


Figure 8-28 SIP MLT Basic Setup

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

Table 8-3 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

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 15.3.1 Logging In on the PC on page 8-32.

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

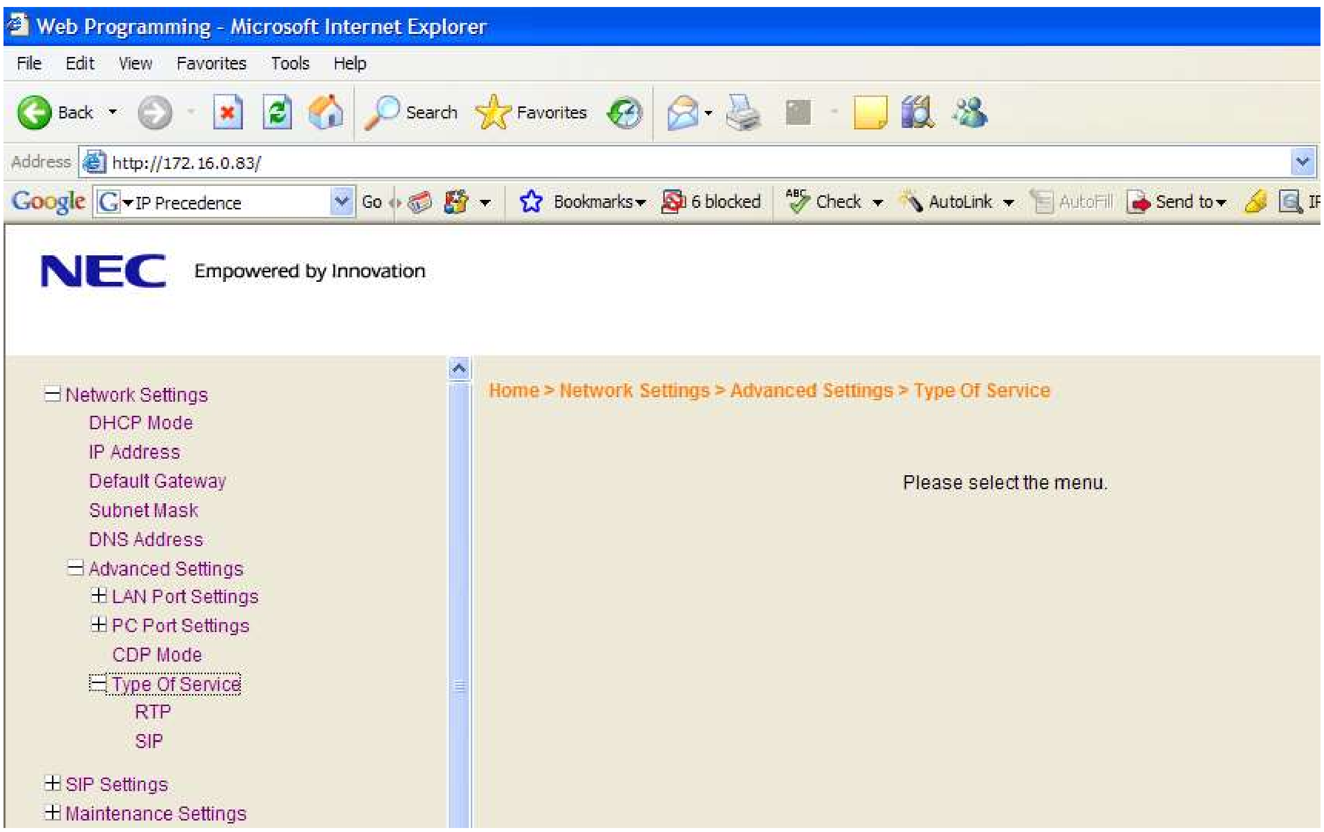


Figure 8-29 Type of Service Window

3. 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 8-3 Common IP Precedence/Diffserv Values and Hexadecimal Equivalent on page 8-44.

   Access the following menus to select options:

   - RTP - Voice Packets
   - SIP - Signalling Packets

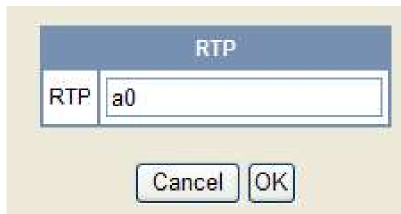


Figure 8-30 RTP - Voice Packets

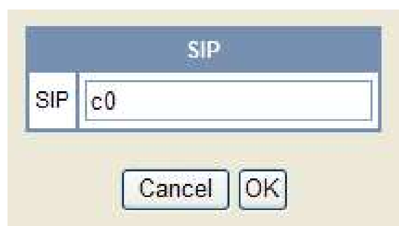


Figure 8-31 SIP - Signalling Packets

4. After selecting **Save**, the following message appears.

5. Select **OK** and the phone reboots. Once online, the phone tags all packets with the applied settings.

Assign Values on IP Terminal

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

1. Press and hold the **Menu** button until display changes.

2. Press **0** on the dial pad for Configuration.

3. At a default state there is no login or password associated to the IP terminal. Press the **OK** soft key.

4. Press **1** on the dial pad for Network Settings.

5. Press **6** on the dial pad for Advanced Settings.

6. Press **4** on the dial pad for Type of Service.

7. There are two options (1) is RTP and (2) is SIP.

8. Press **1** on the dial pad for RTP (voice packets), enter the hexadecimal value, and then press the **OK** soft key.

9. Press **2** on the dial pad for SIP (Signaling packets), enter the hexadecimal value, and then press the OK soft key.

10. 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.

15.5 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
- Possibly add encryption after research.

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 compressio**n**

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

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

([Layer 2 overhead + IP/UDP/RTP header + Voice Payload] /
Voice Payload) * Default CODEC speed = 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 Bytes for Voice Payload

([18 + 40 + 160] / 160) * 64000 = 87200bps

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

When using Video (H.263) soft phone, add the bandwidth for the video portion to the call. The estimated bandwidth per video stream is as follows:

<Receive> Approximately 40k ~ 120k

<Send> Approximately 40k ~ 120k

15.6 Some Network Considerations

When adding the SV8100 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.

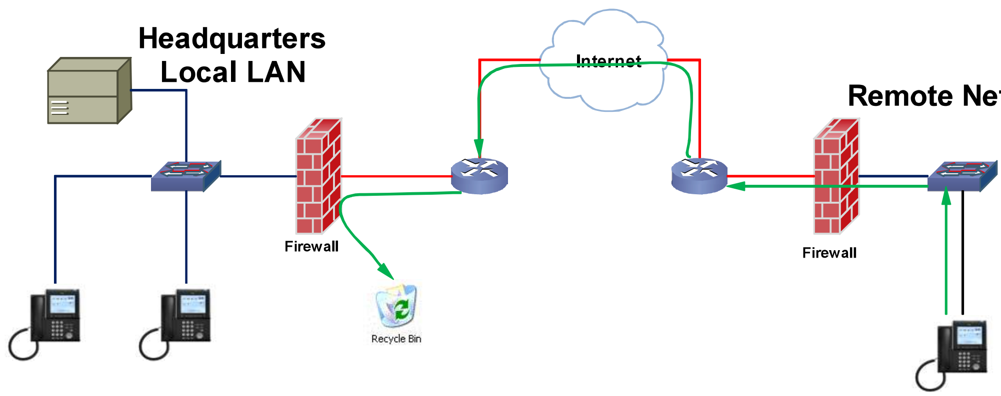


Figure 8-32 SV8100 Network Example No. 1

The green arrow in the diagram above represents the data packets leaving the IP phone destined for the SV81000 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 from 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 SV8100.

The ports that are required open on the Headquarters location are **5080** (UDP) for signaling and the voice ports which depend on how many IPLA/ IPLB ports are installed.

- IPLA/IPLB 32 open UDP ports **10020 ~ 10083**
- IPLA/IPLB 64 open UDP ports **10020 ~ 10147**
- IPLA/IPLB 128 open UDP ports **10020 ~ 10275**

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 SV8100. 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.

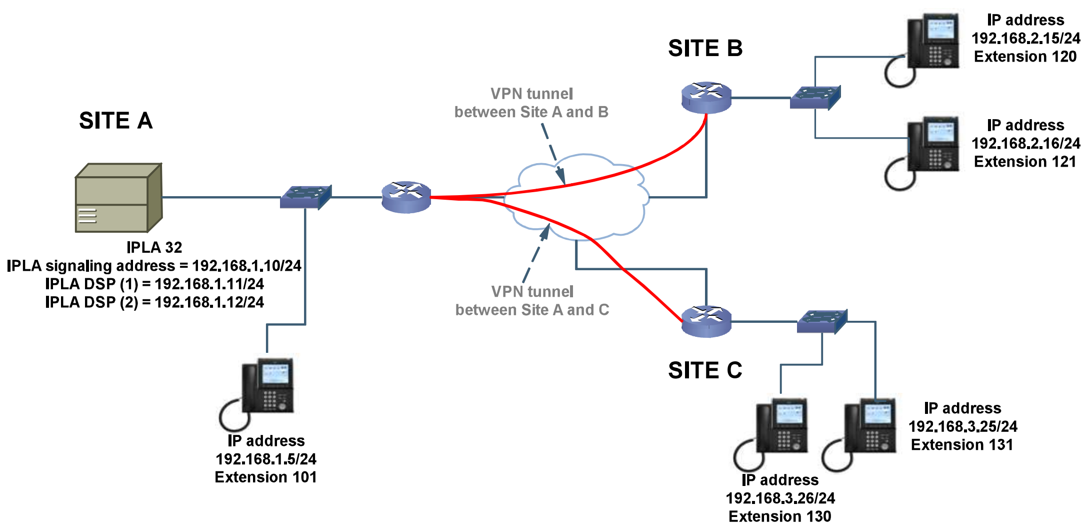


Figure 8-33 SV8100 Network Example No. 3

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 (IPLA/IPLB).
- 192.168.1.10 (IPLA/IPLB) 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 (IPLA/IPLB). In the 200 OK message is the SDP field reporting the IP address of 192.168.3.26 (ext 130).
- 192.168.1.10 (IPLA/IPLB) 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 (IPLA/IPLB).
- 192.168.1.10 (IPLA/IPLB) 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 SV8100. With Peer-to-Peer disabled, all packets (Signaling and Voice) route through the IPLA/IPLB 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 10-26-04.

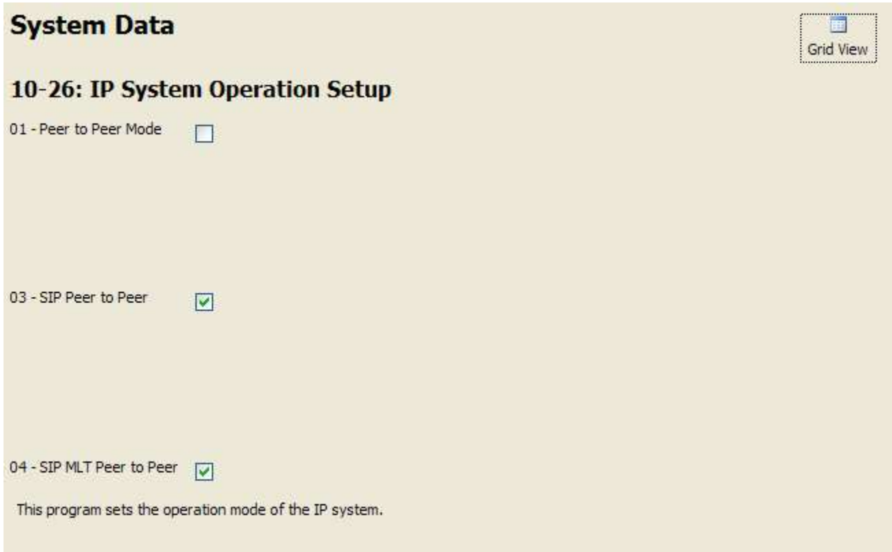


Figure 8-34 IP System Operation Setup

15.7 Guide to Feature Programming

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
10-12-03	CD-CP00-US Network Setup – Default Gateway	IPLA/IPLB uses the Default Gateway that is assigned here.	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	X		
10-12-09	CD-CP00-US Network Setup – IP Address	Assign Layer 3 IP Address to the IPLA/IPLB connected to CCPU.	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 172.16.0.10	X		
10-12-10	CD-CP00-US Network Setup – Subnet Mask	Assign Subnet Mask to the IPLA/IPLB connected to CCPU.	Default is 255.255.0.0	X		

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
84-26-01	IPL Basic Setup – IP Address	When using a PZ-XXIPLA assign one IP address per 16 channels (e.g. a PZ-32IPLA will require 2 addresses, a PZ-64-IPLA will require 4 addresses, and a PZ-128IPLA will require 8 addresses) to each gateway. When using a PZ-XXIPLB assign only one IP address. All channels will utilize this one address. With both the PZ-XXIPLA and the PZ-XXIPLB assign any unused gateway's to IP Address 0.0.0.0	Default Values: Slot 1 = 172.16.0.20~ Slot 4 = 172.16.0.44 VoIP GW Number 1~8: 172.16.0.20~172.16.0.27	X		
84-26-02	IPL Basic Setup – RTP Port Number		Range: 0 ~ 65534 Default Values: VoIP GW1 = 10020 VoIP GW2 = 10052 VoIP GW3 = 10084 VoIP GW4 = 10116 VoIP GW5 = 10148 VoIP GW6 = 10180 VoIP GW7 = 10212 VoIP GW8 = 10244			X
84-26-03	IPL Basic Setup – RTP Port Number (RTP Port Number +1)		Range: 0 ~ 65534 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

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
10-46-01	DT700 Server Information Setup – Register Mode	Define which of the three Registration modes you wish the SIP MLTs to use. Normal 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 = Normal 1 = Automatic 2 = Manual Default is 0		X	
10-46-04	DT700 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	
10-46-06	DT700 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

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
10-46-07	DT700 Server Information Setup – Encryption Mode	Enable or disable encryption mode.	0 = Off 1 = On 1 = Default		X	
10-46-08	DT700 Server Information Setup – Encryption Type	Assign the encryption type.	0 = Mode 1 Default is 0		X	
10-46-09	DT700 Server Information Setup – One-Time Password	Password used when Program 10-46-07 is set to ALL. Assign a character string of 10 characters or less.	Valid Characters (0~9, *, #) Default Not assigned		X	
10-46-10	DT700 Server Information Setup – Start Port	With Automatic logon the starting port number for automatic port allocation.	Range = (1 ~ 512) Default = 1		X	
15-05-01	IP Telephone Terminal Basic Data Setup – Terminal Type	Type of IP terminal registered with the specified extension number.	0 = NGT 1 = H.323 2 = SIP 3 = MEGACO 4 = SIP MLT Default is 0 **READ ONLY**			X
15-05-02	IP Telephone Terminal Basic Data Setup – IP Phone Fixed Port Assignment	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			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	
15-05-19	IP Telephone Terminal Basic Data Setup – Side Option Information	Read Only CM showing type of Line Key unit installed on the ITH-style telephone.	0 = No Option 1 = 8LK Unit 2 = 16LK Unit 3 = 24ADM Default is 0 **READ ONLY**			X

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
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
15-05-21	IP Telephone Terminal Basic Data Setup – Handset Option Information	Read Only CM showing type of Handset installed on the ITH-style telephone.	0 = Normal Handset 1 = Handset for Power Failure (PSA/PSD) 2 = BCH Default is 0 **READ ONLY**			X
15-05-22	IP Telephone Terminal Basic Data Setup – Side Option Additional Information	DSS console number when installed to a ITH-style telephone.	0 = No Setting 1~32 = DSS Console Number Default is 0 **READ ONLY**			X
15-05-23	IP Telephone Terminal Basic Data Setup – Handset Option Additional Information		0 = No Setting 1-16 = Terminal equipment number (TEN) of Bluetooth Cordless Handset (BCH) Default is 0			X
15-05-24	IP Telephone Terminal Basic Data Setup – Protection Service	When enabled allows SIP Multi-Line phones to use the Security button located at top of the SIP MLT display. When disabled, the Security key has no effect.	0 = Not Used 1 = Used Default is 0		X	
15-05-26	IP Telephone Terminal Basic Data Setup – DT700 Terminal Type	Assign type of SIP MLT terminal connected.	0 = Not Set 1 = ITL-**E-1D/IP-*E-1 2 = ITL-**D-1D/ITL-24BT-1D/ITL- 4PA-1D [without 8LKI(LCD)-L] 3 = ITL-**D-1D/ITL-24BT-1D/ITL-24PA-1D [with 8LKI(LCD)-L] 4 = ITL-320C-1 5 = Softphone 6 = CTI 7 = AGW Default is 0			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 1-512 = 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 – Terminal WAN Side IP Address	Future use with NAT	0.0.0.0 ~ 255.255.255.255 Default is 0.0.0.0		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
15-05-32	IP Telephone Terminal Basic Data Setup – Key Pad Talkie		0 = Disable 1 = Enable Default is 0			X
15-05-33	IP Telephone Terminal Basic Data Setup – LAN Side IP Address of Terminal		0 = Disable 1 = Enable Default is 0.0.0.0 **READ ONLY**			X
15-05-34	IP Telephone Terminal Basic Data Setup – Terminal Touch Panel On/Off	Whether the touch screen used on ITL-320C-1 (BK) TEL can be used (1) or cannot be used (0).	0 = Off 1 = On Default is 1			X

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
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 = Voice Control 4 = H.323 5 = RTP/RTCP 6 = SIP 7 = CCISoIP 8 = DT700 MLT 9 = SIP Trunk 10 = NetLink		X	
84-22-01	DT700 Multiline 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	DT700 Multiline 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	DT700 Multiline 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	DT700 Multiline 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	DT700 Multiline Logon Information Setup – Nickname		Assign up to 32 Alpha/ Numeric Characters Default is No Setting		X	
84-23-01	DT700 Multiline 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	DT700 Multiline 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	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
84-23-03	DT700 Multiline Basic Information Setup – Session Expire Timer	At half the value of this timer the IP terminal sends a re-invite message.	Range: 60~65535 Sec. Default is 180		X	
84-23-04	DT700 Multiline Basic Information Setup – Minimum Session Expire Timer	Minimum time the CPU accepts a session timer for a new call.	Range: 60~65535 Sec. Default is 180		X	
84-23-05	DT700 Multiline 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	DT700 Multiline 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	DT700 Multiline 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	DT700 Multiline 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

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
84-23-10	DT700 Multiline 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	DT700 Multiline 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	DT700 Multiline 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	DT700 Multiline 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	DT700 Multiline Basic Information Setup – Refer Expire Timer	The valid period of the REFER subscription.	Range: 0 ~ 65535 Sec. Default is 60			X
84-24-01	DT700 Multiline 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	--Not Used--					
84-24-03	DT700 Multiline CODEC Basic Information Setup – G.711 Type	μ-law used in N.A.	0 = A-law 1 = μ-law Default is 1		X	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
84-24-04	DT700 Multiline CODEC Basic Information Setup – G.711 Jitter Buffer Minimum	Minimum value of the dynamic jitter buffer.	Range: 0 ~ 160ms Default is 20		X	
84-24-05	DT700 Multiline CODEC Basic Information Setup – G.711 Jitter Buffer Average	Average value of the dynamic jitter buffer.	Range: 0 ~ 160ms Default is 40		X	
84-24-06	DT700 Multiline CODEC Basic Information Setup – G.711 Jitter Buffer Maximum	Maximum value of the dynamic jitter buffer.	Range: 0 ~ 160ms Default is 80		X	
84-24-07	DT700 Multiline 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	--Not Used--					
84-24-09	DT700 Multiline CODEC Basic Information Setup – G.729 Jitter Buffer Minimum	Minimum value of the dynamic jitter buffer.	0~270ms Default is 20		X	
84-24-10	DT700 Multiline CODEC Basic Information Setup – G.729 Jitter Buffer Average	Average value of the dynamic jitter buffer.	0~270ms Default is 40		X	
84-24-11	DT700 Multiline CODEC Basic Information Setup – G.729 Jitter Buffer Maximum	Maximum value of the dynamic jitter buffer.	0~270ms Default is 80		X	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
84-24-17	DT700 Multiline CODEC Basic Information Setup – Jitter Buffer Mode		1 = Static 2 = Adaptive during silence 3 = Adaptive immediate Default is 3 (Auto)		X	
84-24-18	--Not Used¸					
84-24-19	DT700 Multiline CODEC Basic Information Setup – Idle Noise Level		5000~7000 (-5000dbm ~ -7000dbm) Default is 7000			X
84-24-20	DT700 Multiline CODEC Basic Information Setup – Echo Canceller Mode		0 = Disable 1 = Enable Default is 1			X
84-24-21	DT700 Multiline CODEC Basic Information Setup – Echo Canceller Tail Size		1 = 4 ms 2 = 8 ms 3 = 16 ms 4 = 32 ms 5 = 64 ms 6 = 128 ms Default is 6			X
84-24-22	DT700 Multiline CODEC Basic Information Setup – Echo Canceller NLP Mode		0 = Disable 1 = Enable Default is 1			X
84-24-24	DT700 Multiline CODEC Basic Information Setup – Echo Canceller CNG Configuration		0 = Adaptive 1 = Fixed Default is 0			X

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
84-24-26	DT700 Multiline CODEC Basic Information Setup – TX Gain	Gain setting for IPLA/IPLB on transmit side of connection to phone.	Range 0~40 (-14dBm~+14dBm) 0 = -20dBm 1 = -19dBm : 20 = 0dBm : 39 = 19dBm 40 = 20dBm Default is 20		X	
84-24-27	DT700 Multiline CODEC Basic Information Setup – RX Gain	Gain setting for IPLA/IPLB on received side of connection to phone.	Range 0~40 (-14dbm~+14dBm) 0 = -20dBm 1 = -19dBm : 20 = 0dBm : 39 = 19dBm 40 = 20dBm Default is 20		X	
84-24-28	DT700 Multiline 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 3		X	
84-24-29	DT700 Multiline CODEC Basic Information Setup – Echo Canceller Configuration Type		Range: 0 ~ 3 0 = Auto 1 = Type 1 2 = Type 2 3 = Type 3 Default is 0			X
84-24-30	DT700 Multiline CODEC Basic Information Setup – Auto Gain Control		Range: 0 ~ 5 Default is 0			X

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
84-24-32	DT700 Multiline 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 2		X	
84-24-33	--Not Used--					
84-24-34	DT700 Multiline CODEC Basic Information Setup – G.722 Jitter Buffer Minimum	Minimum value of the dynamic jitter buffer.	Range: 0 ~ 160ms Default is 30		X	
84-24-35	DT700 Multiline CODEC Basic Information Setup – G.722 Jitter Buffer Average	Average value of the dynamic jitter buffer.	Range: 0 ~ 160ms Default is 60		X	
84-24-36	DT700 Multiline CODEC Basic Information Setup – G.722 Jitter Buffer Maximum	Maximum value of the dynamic jitter buffer.	Range: 0 ~ 160ms Default is 120		X	
84-28-01	DT700 Multiline Firmware Name Setup – Firmware Directory	Maximum 64 characters.	Default is No Setting		X	
84-28-02	DT700 Multiline Firmware Name Setup – Firmware File Name	Maximum 30 characters.	Default is No Setting		X	

15.8 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

15.8.1 Plug and Play

1. Program 10-12
   Assign the IPLA/IPLB 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.

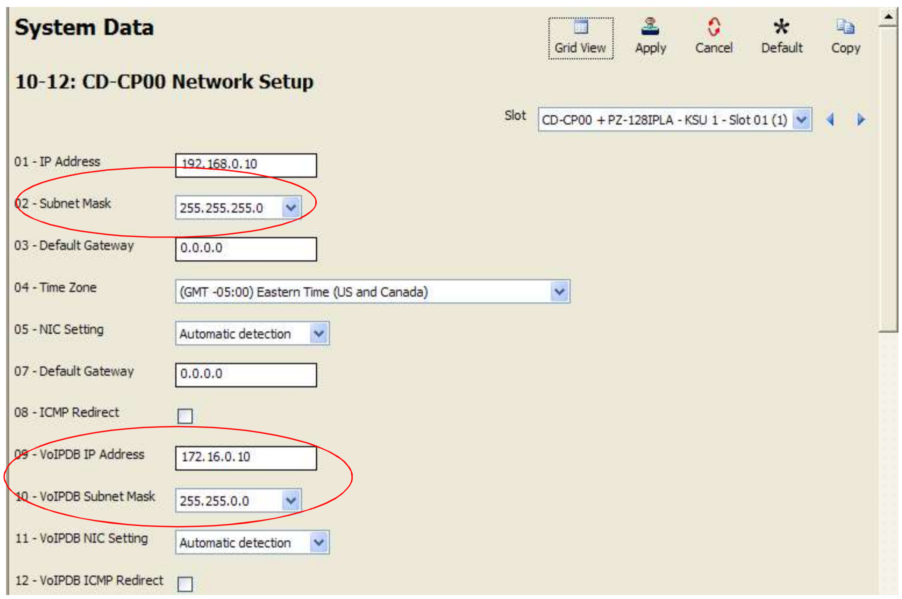


Figure 8-35 System Data 10-12: CD CP00 Network Setup

2. Program 84-26
   Assign IP addresses to the DSPs that are to be used. The IP addresses 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.

Assign an IP address only to the DSP channels that are used. With an IPLA Each DSP provides 16 paths from IP to TDM. With an IPLB only the first DSP is used, set the remaining 7 DSP's to 0.0.0.0.

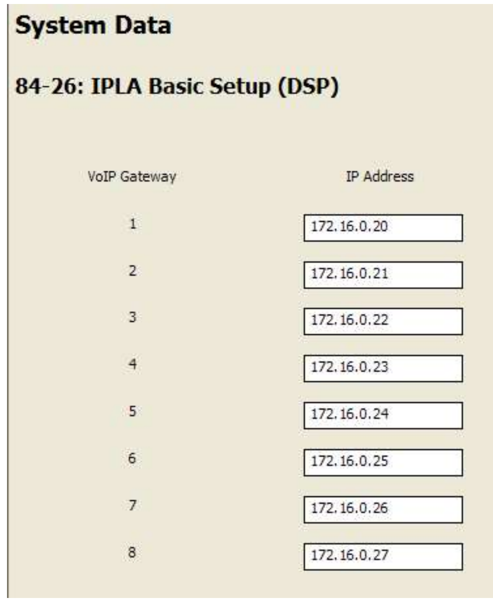


Figure 8-36 System Data 84-26: IPLA/IPLB Basic Setup (DSP)

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 SV8100. 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 to the left if port 504 (Extension 3505) is used for a SIP MLT Station that group of 2 ports (Ports 503 and 504) 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.

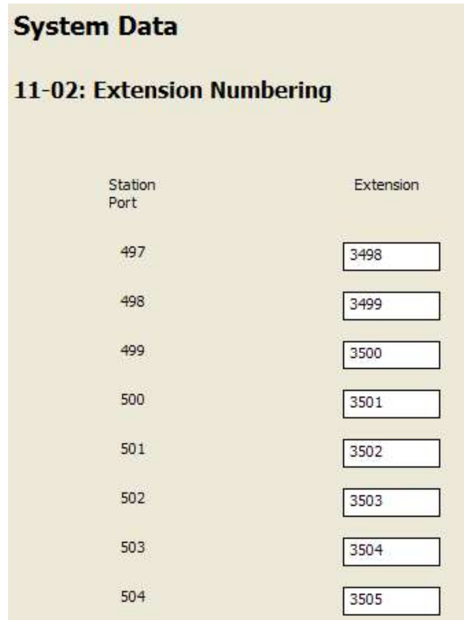


Figure 8-37 System Data 11-02: Extension Numbering

4. 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 IP Phone List in PCPro, the extensions are available for selection in Program 15 and other station related Programs.

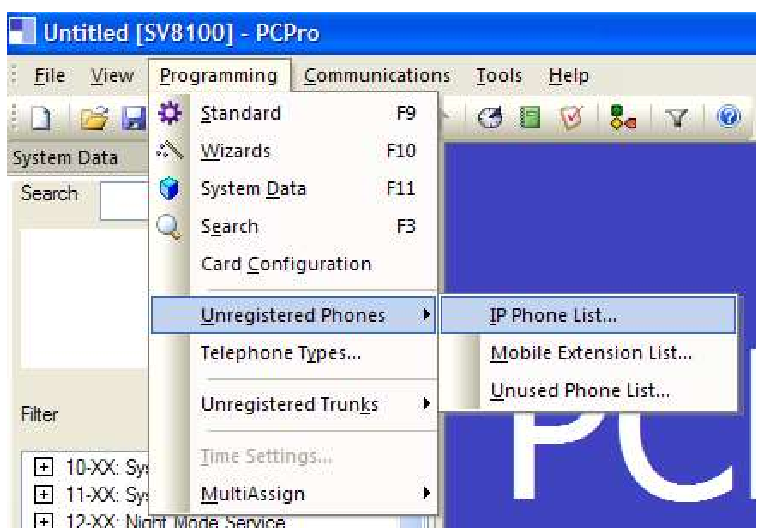


Figure 8-38 PCPro Programming Unregistered Phones

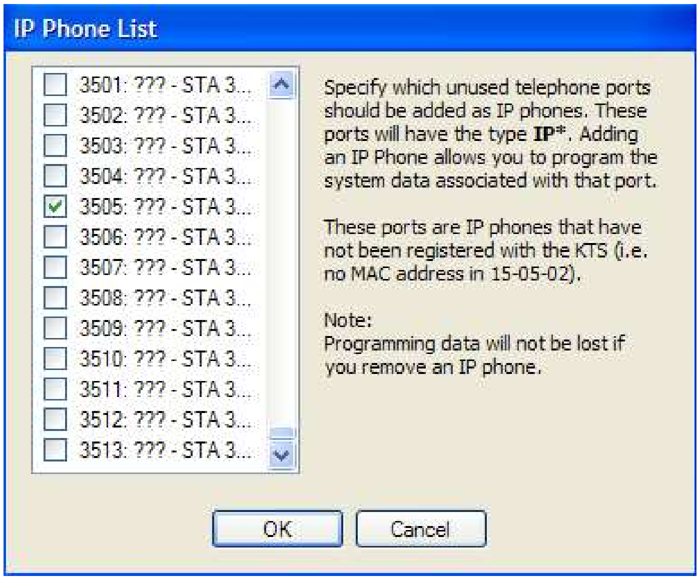


Figure 8-39 IP Phone List

5. 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:

❍ Press **0** on the dial pad for configuration mode.

❍ 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.

15.8.2 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.

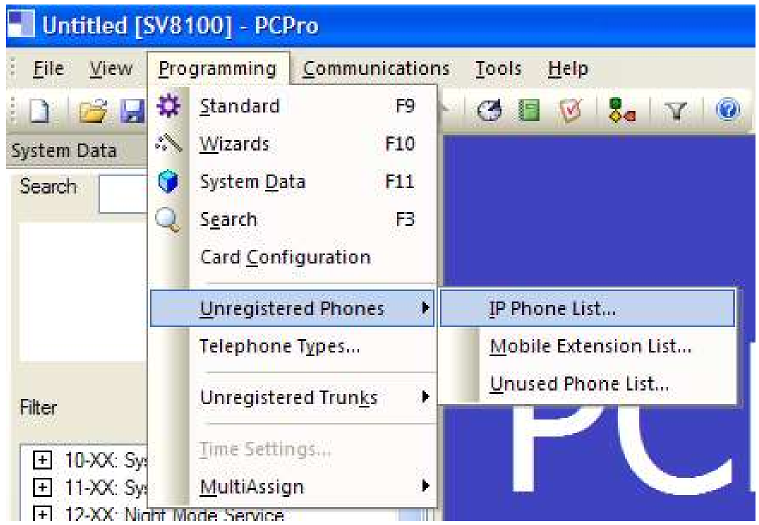


Figure 8-40 PCPro Programming Unregistered Phones

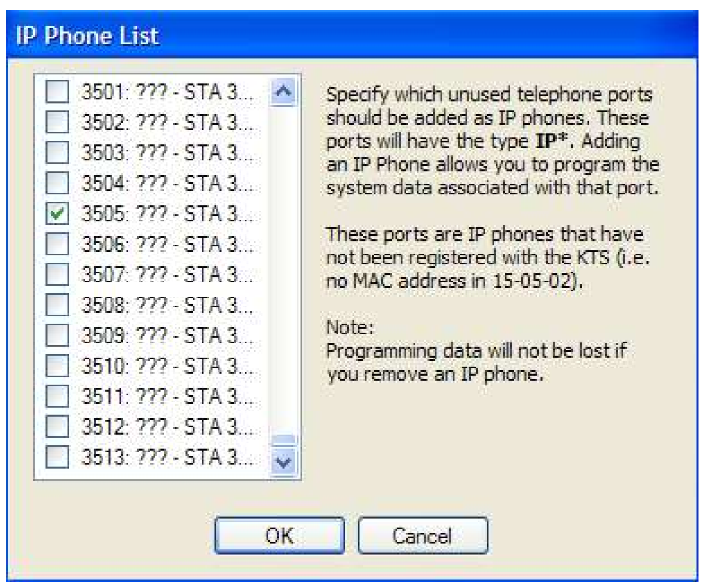


Figure 8-41 IP Phone List

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

Figure 8-42 Automatic Registration Setting

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

Figure 8-43 Automatic Registration Basic Setup

Figure 8-44 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.

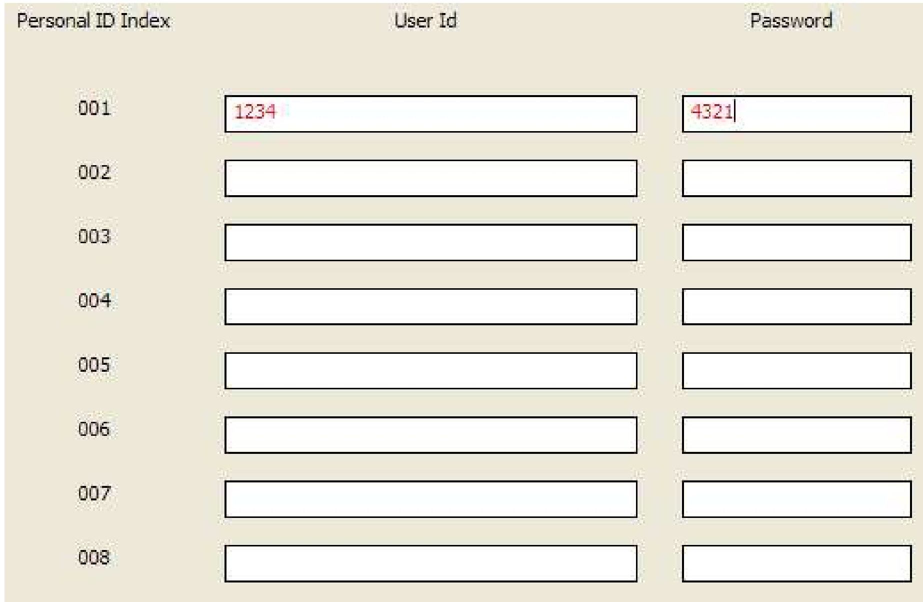


Figure 8-45 Automatic Registration User Name and Password Assignment

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:

❍ Press **0** on the dial pad for configuration mode.

- 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.

15.8.3 Manual Registration

Steps 1~4 are the same as for Section 15.8.2 Automatic Registration on page 8-71.

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 8-46 Manual Registration

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.

❍ Press **0** on the dial pad for configuration mode.

- 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.

15.9 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.

❍ 2-button Phones cannot use the IP Phone Relocation feature.

❍ 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.

❍ The system sees terminal types 1 (Economy), 2 (Value), 3 (Desi-Less), 4 (Sophisticated), and 5 (Softphone) as the same terminal type.

❍ When using Multiple Login, the same Personal ID index can be assigned to an ITL/Softphone, a CTI (Desktop), and an MH240 terminal type.

❍ Two ports of the same terminal type (Program 15-05-26) cannot be assigned to the same Personal ID index (Program 15-05-27).

❍ Program 10-46-01 must be set to 1 (Auto), or 2 (Manual) for Multiple Login to work.

- When three ports are assigned the same Personal ID index, in Program 15-05-27, if Program 15-05-26 is not set for those ports, the terminal types will be assigned based on order of login. If Program 15-05-26 is set, the login order does not matter and they will assign the correct port.
- Prior to **Version 3000**, Override is not supported in a SV8100 system that had a 3rd Party CTI connection to the CPU (i.e., Desktop Apps Shared Services, UCB), or to a terminal with a 1st Party CTI connection (i.e., PC Assistant/Attendant and Softphone or 1st Party TAPI driver), and would show Rejected Override >>>CTI Link... in the display.
- With **Version 3000** or higher, Override with CTI is supported on a per station basis using Program 15-05-39 with certain restrictions.
- 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 8-4 IP Phone Relocation

Program/ Item No.	Description/ Selection	Assigned Data	Comments
10-46-01	DT700 Server Information Setup – Register Mode	0 = Normal 1 = Auto 2 = Manual Default is 0	Set up the information of the SIP Multiline (DT700 series) Server. This PRG is a system-wide setting.
15-05-27	IP Telephone Terminal Basic Data Setup – Personal ID Index	0~512 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	DT700 Multiline 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	DT700 Multiline 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	DT700 Multiline 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 Phone Relocation Flow Chart

The following flow chart can be used to enable the IP Phone Relocation feature. Every user must enter both login ID and Password.

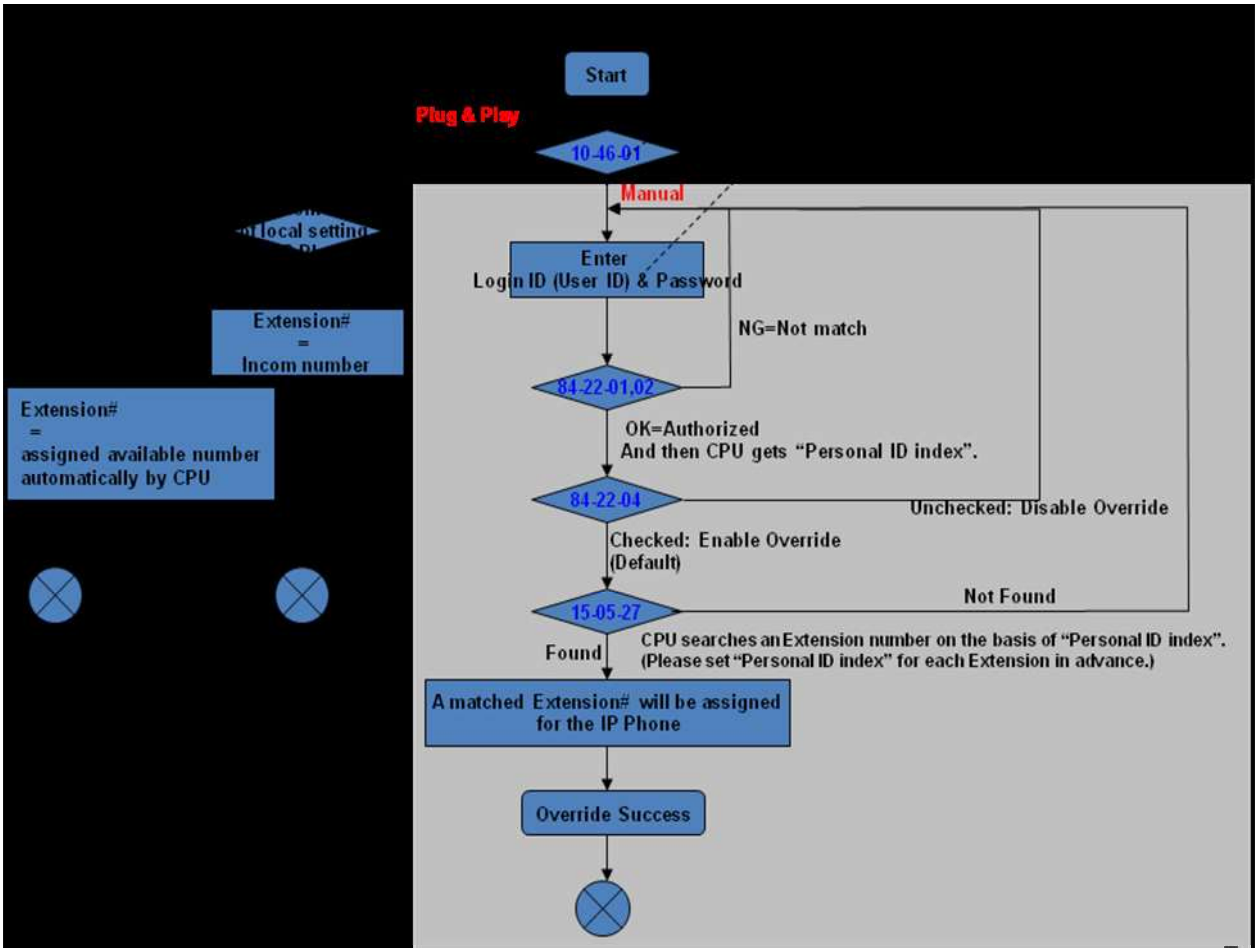


Figure 8-47 IP Phone Relocation Flow Chart