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UNIVERGE® SV8100

Networking Manual

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Introduction

SECTION 1   GENERAL OVERVIEW

This manuals provides information for networking for the UNIVERGE SV8100 systems. Networking can be accomplished using one of the following methods:

- ❍   SV8100 K-CCIS
- ❍   SV8100 IP K-CCIS
- ❍   SV8100 NetLink

SECTION 2   MANUAL ORGANIZATION

This manual is divided into two books to describe the networking functions: the K-CCIS networking/K-CCIS IP networking and IP networking.

SECTION 3   COMMON TERMS

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

Table 1-1 Common Terms and Associated Abbreviations

Term	Abbreviation	Description
CD-CP00-US	CD-CP00 SV8100 CPU CPU	SV8100 CPU
PZ-32IPLA PZ-64IPLA PZ-128IPLA	VOIPDB VOIP Daughter Board	VoIP Daughter Board, mounted on CD-CP00-US
CD-2BRIA	BRIU 2BRIU	2-port Basic Rate Interface blade
PZ-2BRIA	2BRIUDB BRIUDB	2-port basic rate daughter board, mounted on the CD-2BRIA

Table 1-1 Common Terms and Associated Abbreviations

Term	Abbreviation	Description
CD-CCTA	CCHU CCIS CCHU	CCIS Trunk Interface blade
CD-4COTB	4COIU	4-port loop/ground start trunk blade
PZ-4COTF	4COIUDB 4COIU Daughter Board	4-port loop/ground start daughter board, mount on CD-4COTB blade
CD-8DLCA CD-16DLCA	8ESIU 16ESIU	8 digital station interface blade 16 digital station interface blade
PZ-8DLCB	8ESIUDB 8ESIU Daughter Board	8 digital station interface daughter board, mounted on the CD-8DLCA
CD-PVAA	PVAA	Conference bridge blade
DT300 Series	DT300 Digital Multiline Terminal DTL Telephones Digital Multiline Telephones	SV8100/SV8300 Digital Multiline Terminals. Telephone equipment names begin with DTL. For example: DTL-12D-1 TEL, DTL-24D-1 TEL, DTL-32D-1 TEL, etc.
DT700 Series	DT700 IP Digital Multiline Terminal ITL Telephones IP Digital Multiline Telephones	SV8100/SV8300 IP Digital Multiline Terminals. Telephone equipment names begin with ITL. For example: ITL-12D-1 TEL, ITL-24D-1 TEL, ITL-32D-1 TEL, etc.
CD-4DIOPA	DOIPU 4DOIPU	4-port DID/OPX blade
CD-4ODTA	TLIU 4TLIU	4-port E&M tie line blade

Book 1 – SV8100 K-CCIS

General Information

1

This chapter provides a system outline, the name and functions of the circuit cards required, system capacity, time slot assignments, system specifications and network structure considerations for UNIVERGE SV8100 K-CCIS.

SECTION 1   SYSTEM OUTLINE

Key-Common Channel Interoffice Signaling (K-CCIS) can interface this system with a Public Network. The system is configured with the (CD-CCTA) for CCIS Trunk Interface/CCH. The CD-CCTA is the Digital Trunk Interface required for receiving/transmitting common signaling data from/to the distant office. A Phase Locked Oscillator (PLO) for digital network synchronization is built into the CD-CP00-US.

The system can provide a variety of interoffice service features such as Link Reconnect, Virtual Look Ahead Routing, Centralized Voice Mail/Auto Attendant Integration, Call Forwarding, Voice Call with Hands free Answerback and Caller ID Display. For a more detail description refer to Chapter 4 Features and Specifications. For a diagram of the system outline, refer to Figure 1-1 K-CCIS System Outline.

General Information
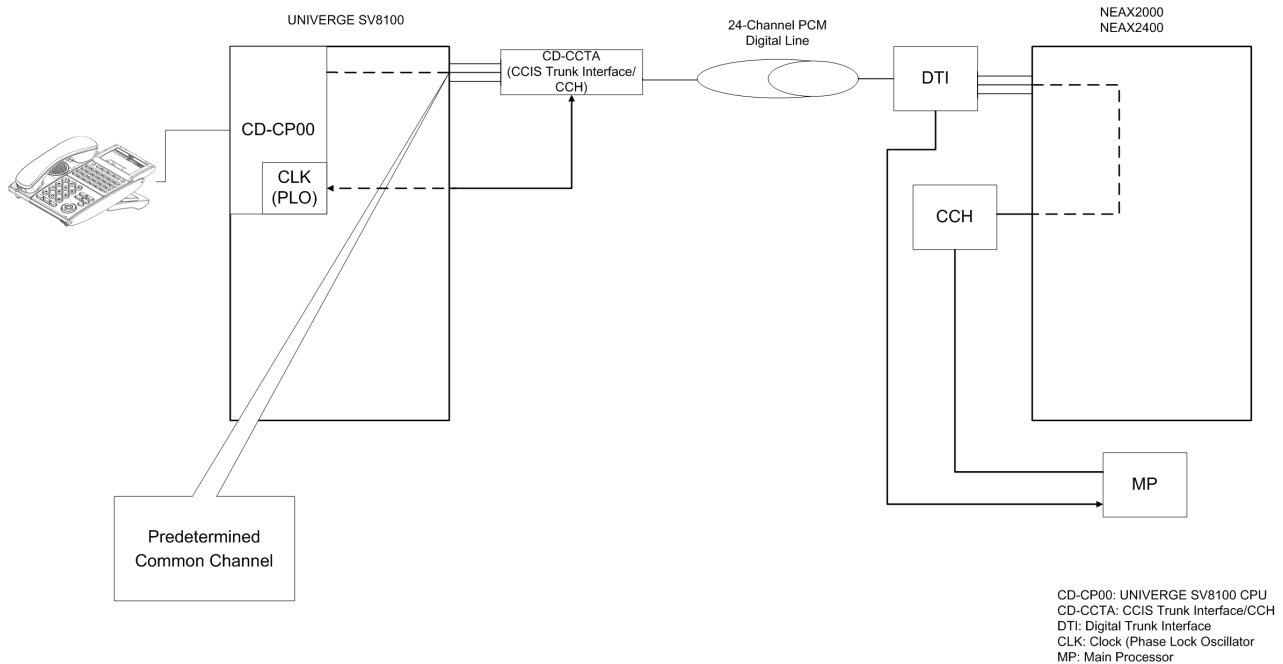
Figure 1-1 K-CCIS System Outline

1.1   Common Channel Handler (CD-CCTA)

The Common Channel Handler (CD-CCTA) blade provides a common channel signal to a UNIVERGE SV8100 network. It is responsible for signaling between the Key Telephone System (KTS) and the UNIVERGE SV8100 network under control of the CD-CP00-US.

1.2   Phase Locked Oscillator (PLO)

The Phase Locked Oscillator (PLO) provides synchronization between the TDSW and other offices and is built into the CD-CP00-US. The clock generates a synchronized clock signal according to the source clock signals supplied from the source office in the network, and supplies the generated clock signal to the TDSW. The clock is supplied with clock signals extracted from the CD-CCTA.
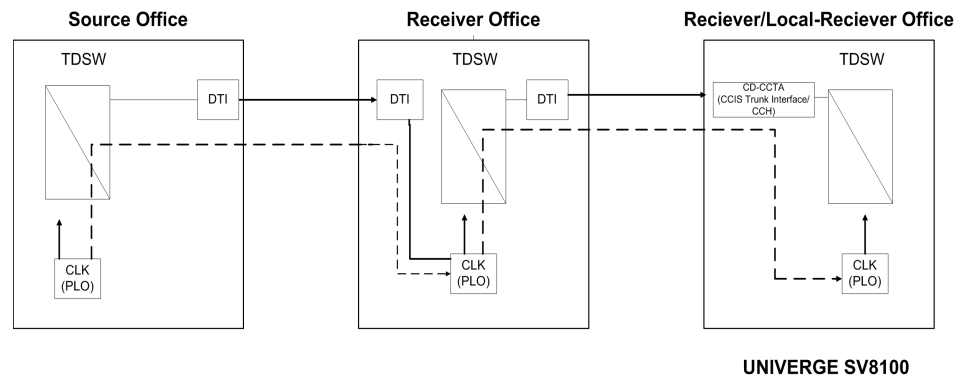
Figure 1-2 Clock Supply Route

1.3   Common Channel Interoffice Signaling (K-CCIS)

The PBX can provide connection to the UNIVERGE SV8100 using a digital
network. The network requires the Common Channel Handler (CD-CCTA) to
control the common signaling between offices.

Digital Network

When UNIVERGE SV8100 is provided through a digital network, the CCH and
T-1 functionality is combined on the CD-CCTA to provide a fixed path in the
TDSW and to transmit and receive common signaling data to/from the distant
office through a predetermined channel. Voice signals are transmitted and
received per line through other channels.

Figure 1-3 K-CCIS System Configuration shows the system configuration of K-CCIS provided using a digital network.

CD-CP00: UNIVERGE SV8100 CPU
CD-CCTA: DTI/Common Channel Handler
DTI: Digital Trunk Interface
CLK: Clock (Phase Lock Oscillator
MP: Main Processor
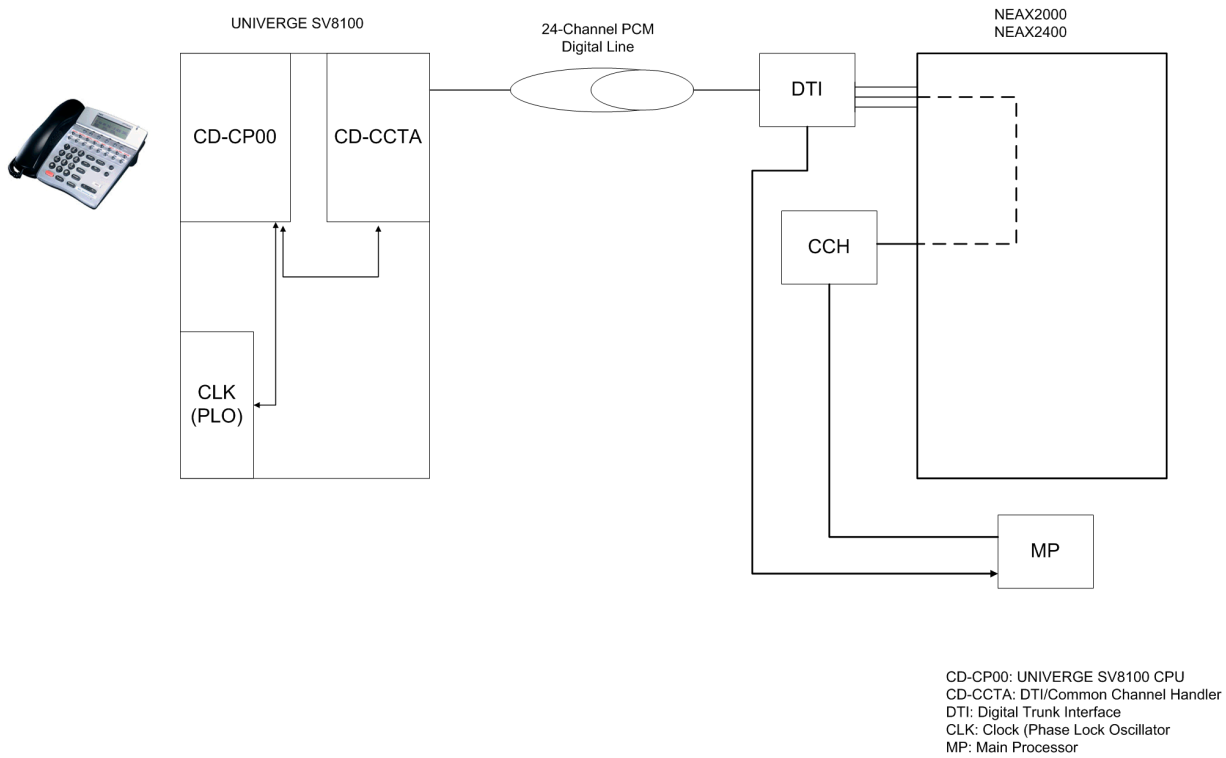
Figure 1-3 K-CCIS System Configuration

SECTION 2   DTI SPECIFICATIONS

The following specifications apply to the CD-CCTA.

2.1   Characteristics

Output

Line Rate	1.544 Mbps + 50 ppm
Line Code	AMI with ZCS/B8ZS ∗
Line Impedance	100 Ω
Pulse Amplitude (Base to Peak)	3 volts ± 0.6 volts
Pulse Width	3214 ns ± 30 ns

Input

Line Rate	1.544 Mbps ± 200 ppm (130 ppm)
Pulse Amplitude (Base to Peak)	1.5 volts ~ 3 volts
Frame Synchronization Pattern	100011011100
Input Jitter	ITU-T Fig.1/G743
Wander	±138UI, −193UI or −138UI, +193UI
Cable Length from SV8100 to MDF or External Equipment	Maximum 196m (655 ft.) with (22 AWG) twisted-pair cable

✱ AMI: Alternate Mark Inversion
ZCS: Zero Code Suppression
B8ZS: Bipolar Eight Zero Substituting

2.2   Frame Configuration for 24 DTI

According to the AT&T Specifications for 24-channel transmission, there are two frame configurations: 12 Multiframe (D4) and 24 Multiframe (ESF).

12 Multi-Frame (D4)

This frame has 12 Multiframes, and each Multiframe has a 24-channel PCM signal (8 bits/channel) and an S (Superframe) bit. Figure 1-4 Frame Configuration of 24-DTI (12 Multiframe) shows the frame configuration, and Figure 1-5 Frame Configuration of 24-DTI (24 Multiframe) shows the frame bit assignment.

S: Superframe Bit
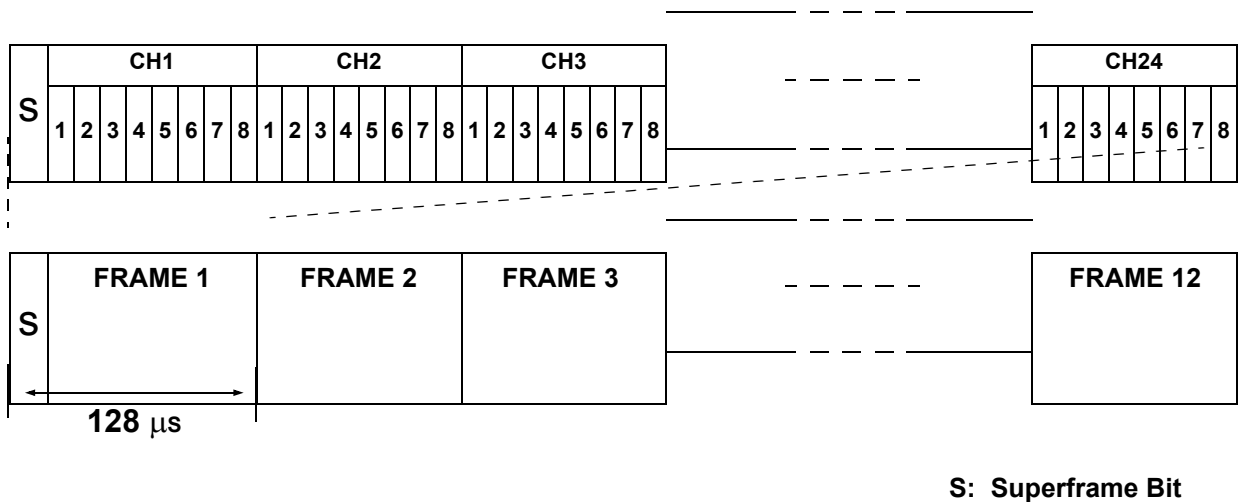
Figure 1-4 Frame Configuration of 24-DTI (12 Multiframe)

Table 1-1 12-Multiframe Bit Assignment

Frame Number	S-Bit	
	Terminal Synchronization (FT)	Signal Synchronization (FS)
1	1	
2		0
3	0	
4		0
5	1	
6		1
7	0	
8		1
9	1	
10		1
11	0	
12		0

✎   The S-bit is the first bit in each frame.

Frames are repeated in the order shown in this table.

24 Multiframe (Extended Superframe – ESF)

This frame has 24 Multiframes and each Multiframe has a 24-channel PCM signal (8 bits/channel) and an S (Superframe) bit.
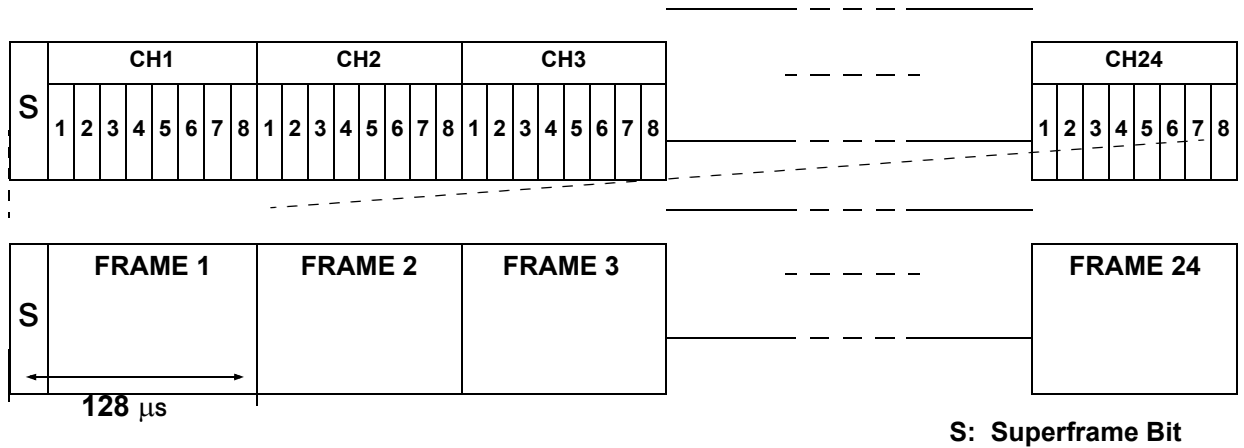
Figure 1-5 Frame Configuration of 24-DTI (24 Multiframe)

Table 1-2 24-Multiframe Bit Assignment

Frame Number	S-Bit		
	Frame Synchronization	4 kbps Data Link	CRC
1		m	
2			CB1
3		m	
4	0		
5		m	
6			CB2
7		m	
8	0		
9		m	
10			CB3

Table 1-2 24-Multiframe Bit Assignment (Continued)

Frame Number	S-Bit		
	Frame Synchronization	4 kbps Data Link	CRC
11		m	
12	1		
13		m	
14			CB4
15		m	
16	0		
17		m	
18			CB5
19		m	
20	1		
21		m	
22			CB6
23		m	
24	1		

✎   The S-bit is the first bit in each frame.

Frames are repeated in the order shown in this table.

The letter m in the 4 kbps Data Link column indicates the frame is usually assigned to 1.

SECTION 3   NETWORK STRUCTURE CONSIDERATIONS

3.1   Determining System Configurations

The configuration of the network and the number of lines (channels) is determined by the traffic between each office.

The topologies listed in this section are supported in the UNIVERGE SV8100 system KTS-to-KTS structure.

❍   Star Topology – Refer to Figure 1-6 Star Topology (KTS-to-KTS or PBX-to-KTS).

❍   Tree Topology – Refer to Figure 1-7 Tree Topology (KTS-to-KTS or PBX-to-KTS).

❍   Mesh Topology is supported only when the KTS is the end-point in a PBX-to-KTS network. Refer to Figure 1-8 Mesh Topology (PBX-to-KTS).

Main Hub
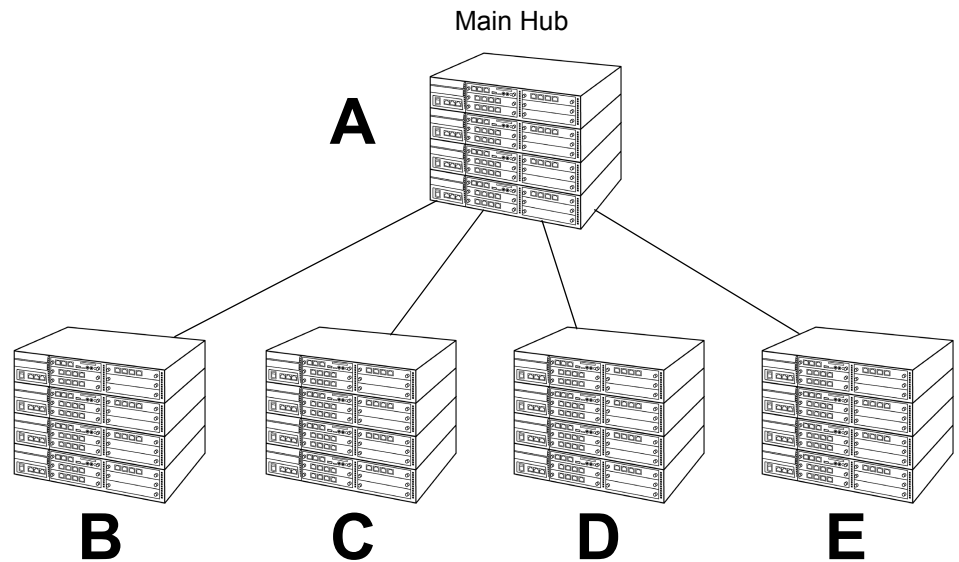
Figure 1-6 Star Topology (KTS-to-KTS or PBX-to-KTS)
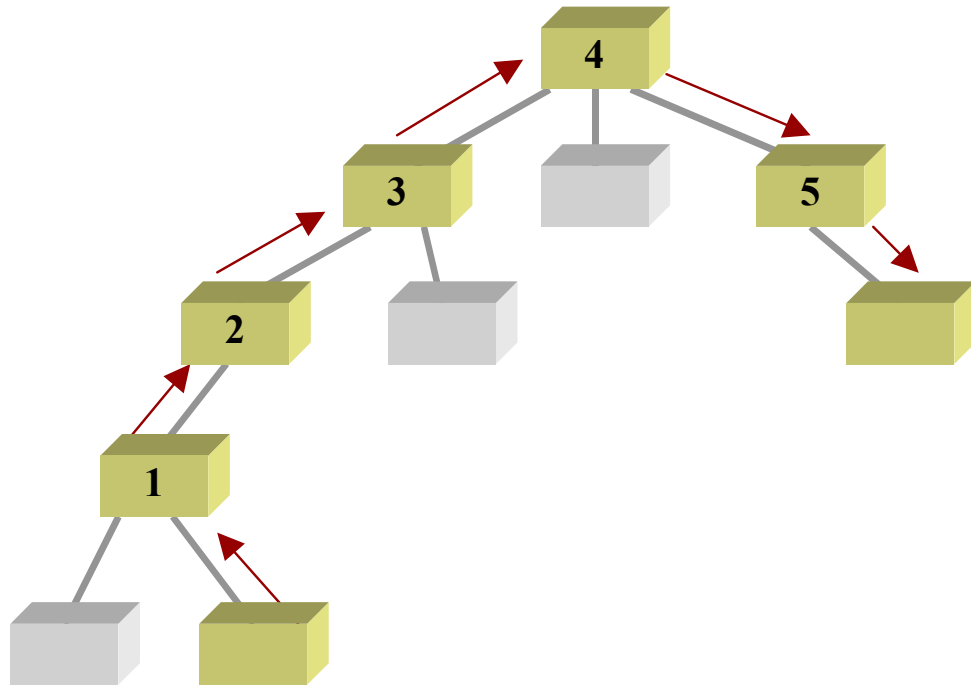
Tree Topology supports a total of 255 systems. Even though 255 systems are allowed, only five hops* are permitted. Software does not limit the number of hops. The limitation is due to the CCH message delay through each tandem system.

✱ Hops - Tandem through another system

Figure 1-7 Tree Topology (KTS-to-KTS or PBX-to-KTS)
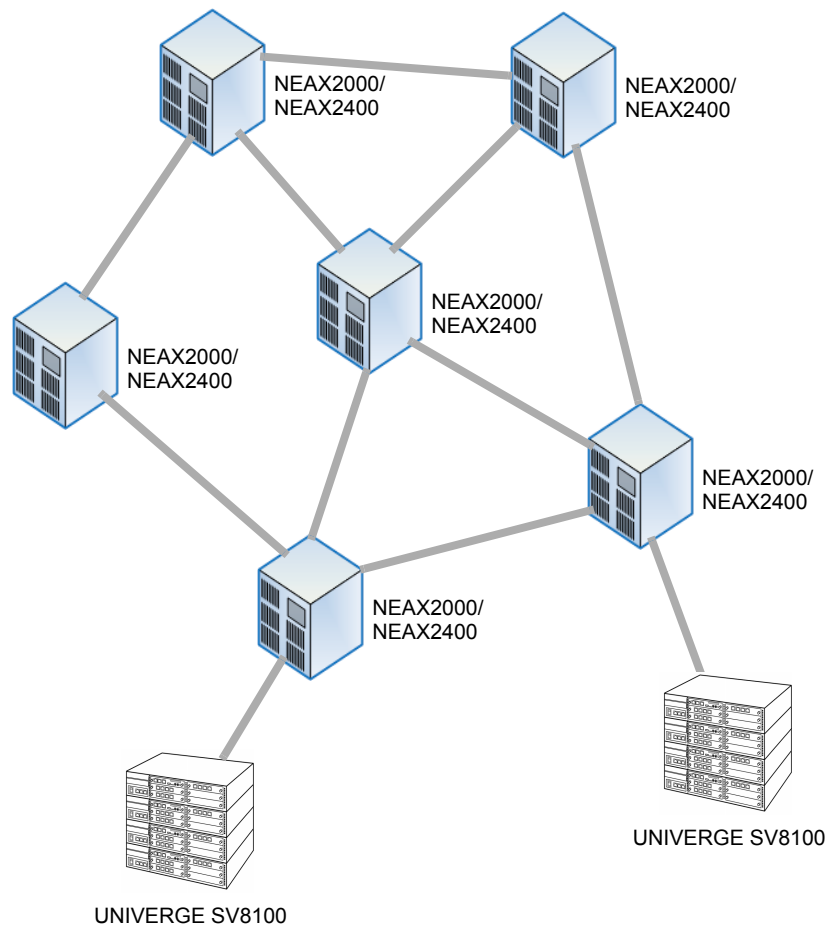
A UNIVERGE SV8100 can be connected to a PBX only as a remote office. The UNIVERGE SV8100 software can register a maximum of 255 point codes in the UNIVERGE SV8100 network. The network must consist of a PBX-to-KTS structure with the KTS programmed as a remote office. The KTS must be located as the end-point in the UNIVERGE SV8100 network.

Figure 1-8 Mesh Topology (PBX-to-KTS)

3.2   Determining Number of K-CCIS Routes

When the system is a Central Office or Tandem Office, two or more routes to other offices are required. Each CD-CCTA can support one K-CCIS link. Up to eight CD-CCTA blades can be installed in a UNIVERGE SV8100 system. The KTS requires the CD-CCTA to support a K-CCIS interface.

One Common Signaling Channel (CSC) can support up to 127 voice channels, if needed.

One CD-CCTA provides 24 channels. One Common Signaling Channel port must be assigned on the Digital Trunk Interface in case two systems are connected by one Digital Link.
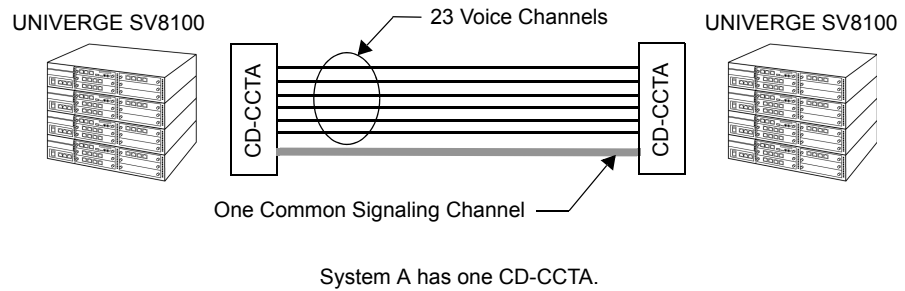
Figure 1-9 K-CCIS Routes

One Common Signaling Channel can be assigned even if two Digital Links are connected between two systems. Only one Common Signaling Channel can be assigned per CD-CCTA in each system.
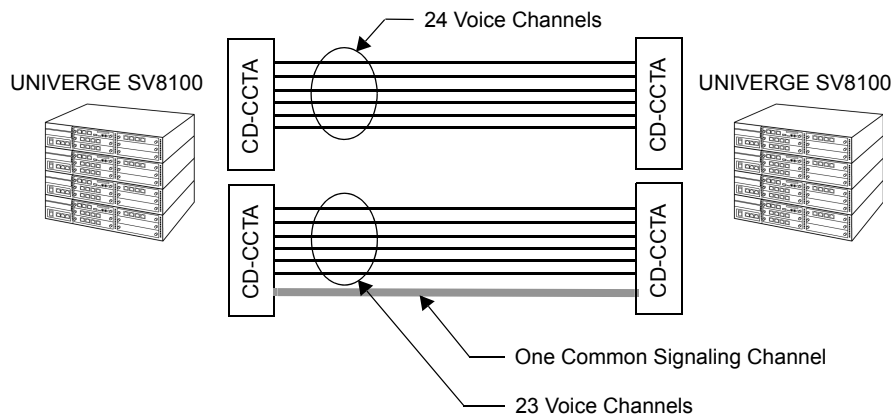
Figure 1-10 Assigning One Common Signaling Channel Between Two Systems

One CD-CCTA can be assigned to one Common Signaling Channel. Tandem K-CCIS connections require one CD-CCTA for every connection as shown in Figure 1-11 One CD-CCTA Assigned per Common Signaling Channel.
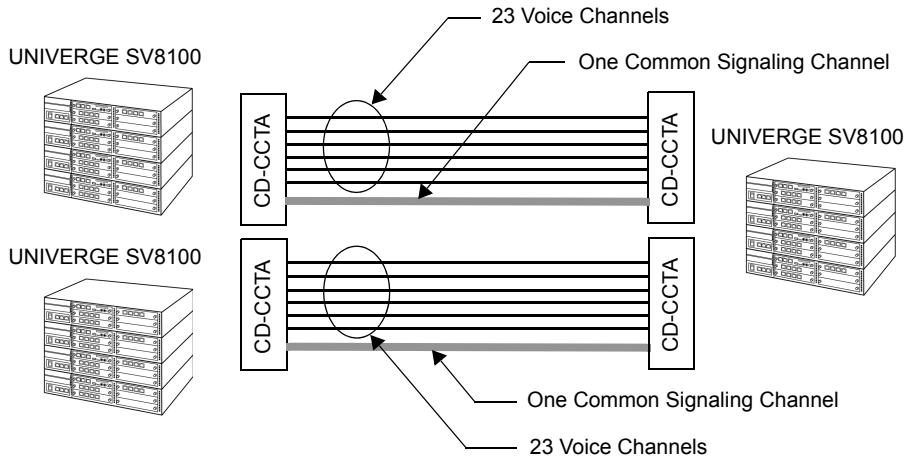
Figure 1-11 One CD-CCTA Assigned per Common Signaling Channel

Common Signaling Channels cannot be connected to another CD-CCTA. They must be assigned to their own physical link.
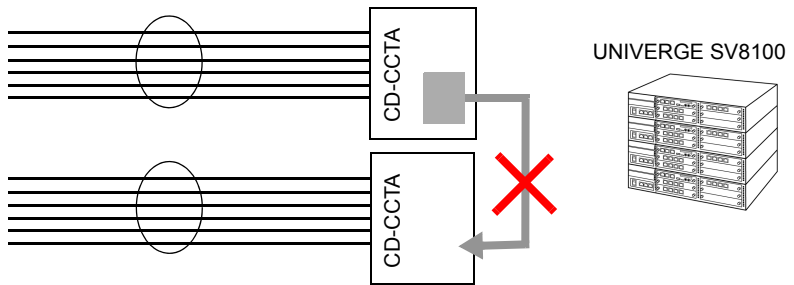
Common Signaling Channels cannot be connected to another CD-CCTA.

Figure 1-12 Common Signaling Channel - Example of Incorrect Connection

3.3   Determining the Type of Transmission Lines

The transmission lines, available on the UNIVERGE SV8100 system, are digital only (CD-CCTA).

3.4   Determining which Systems Should be the Central Office

If using a KTS-to-KTS only network and features such as Voice Mail Integration – K-CCIS are used, the key system that has the voice mail system installed, must be programmed as the Central/Originating Office. All other key systems must be programmed as Remote/Destination offices.

3.5   Determining Point Codes

Point Codes are used to distinguish an originating office from a destination office in the K-CCIS network. A Point Code is assigned to each office in the K-CCIS network. The following guidelines apply when determining the Point Codes:

❍   The Point Code cannot be assigned to more than one office.

❍   The same origination Point Code must be assigned to each K-CCIS channel in the same system.

❍   The maximum number of Point Codes that can be assigned is 255 (a maximum of 256 offices can be connected in the same network).
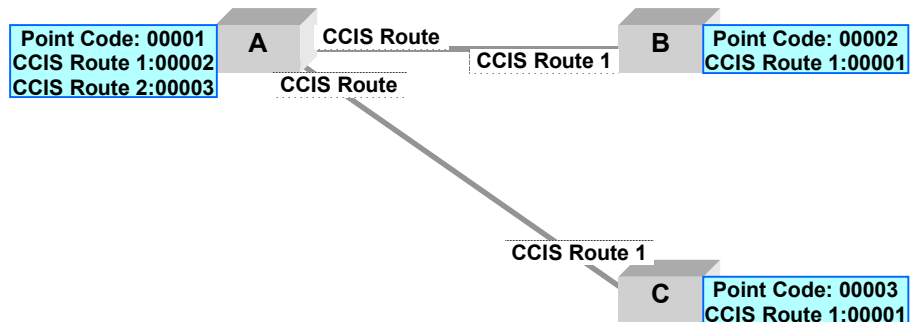
Figure 1-13 Point Code Determinations

❍   **Data Assignment for System A**

PRG	CCIS Route ID	Point Code	Remarks
50-02-03	1	00001	Assign the Originating Point Code for System A
	2	00001	
50-02-04	1	00002	Assign the Destination Point Code for CCH 1
	2	00003	Assign the Destination Point Code for CCH 2

❍   **Data Assignment for System B**

PRG	CCIS Route ID	Point Code	Remarks
50-02-03	1	00002	Assign the Originating Point Code for System B
50-02-04	1	00001	Assign the Destination Point Code for CCH 1

❍   **Data Assignment for System C**

PRG	CCIS Route ID	Point Code	Remarks
50-02-03	1	00003	Assign the Originating Point Code for System C
50-02-04	1	00001	Assign the Destination Point Code for CCH 1

3.6   Determining CCH Link to Send Messages

The tandem office must be programmed with the proper information to indicate how the CCH (in its own system) is connected to other offices in the network. Every office linked to the CCH must be identified by assigning the Point Codes that it accepts in the network. Refer to Figure 1-14 CCH Linking. When using features such as the Link Reconnect – K-CCIS, this must be programmed using Program 50-06-01. Program 50-03-01 allows the links between each system and the CCH, through which it is directed, to be assigned.
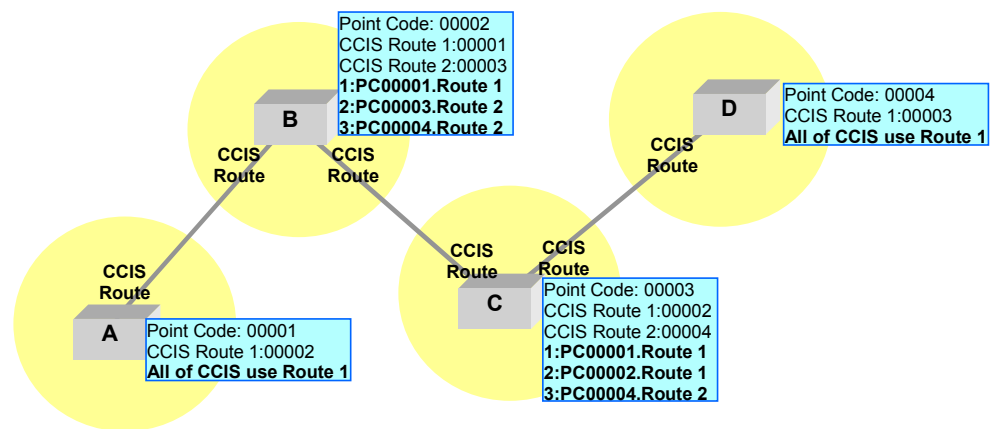
Figure 1-14 CCH Linking

❍ **Data Assignment for System A**

PRG	CCIS Route ID	Point Code	Remarks
50-03-01	None	None	It is unnecessary to assign system data for this office.

❍ **Data Assignment for System B**

PRG	CCIS Route ID	Point Code	Remarks
50-03-01	1	00001	Assign the data to create a link between Office A and Office B using the CCH1 and a link between Office B and C using CCH2.
	2	00003	
	2	00004	

❍ **Data Assignment for System C**

PRG	CCIS Route ID	Point Code	Remarks
50-03-01	1	00001	Assign the data to create a link between Office B and Office C using the CCH1 and a link between Office C and D using CCH2.
	1	00002	
	2	00004	

❍ **Data Assignment for System D**

PRG	CCIS Route ID	Point Code	Remarks
50-03-01	None	None	It is unnecessary to assign system data for this office.

3.7 Determining Circuit Identification Code (CIC)

The CD-CCTA trunk must distinguish between Voice Path and Common Signaling channel. The trunks using Voice Path are assigned a CIC number for each T1 trunk. The CIC numbers must match those in the connected system. Refer to Figure 1-15 Circuit Identification Codes (CIC). The maximum value of a CIC that can be assigned is 127.

System A

CD-CCTA (2)

TK25
TK26
TK27

CD-CP00

CLK
(PLO)

TK47
TK48

CD-CCTA (1)

TK01
TK02
TK03

TK23
TK24

**Common Signaling
Channel**

System B

CD-CCTA (1)

TK01
TK02
TK03

CD-CP00

CLK
(PLO)

TK23
TK24

System C

CD-CCTA (1)

TK01
TK02
TK03

CD-CP00

CLK
(PLO)

TK23
TK24

**Common Signaling
Channel**
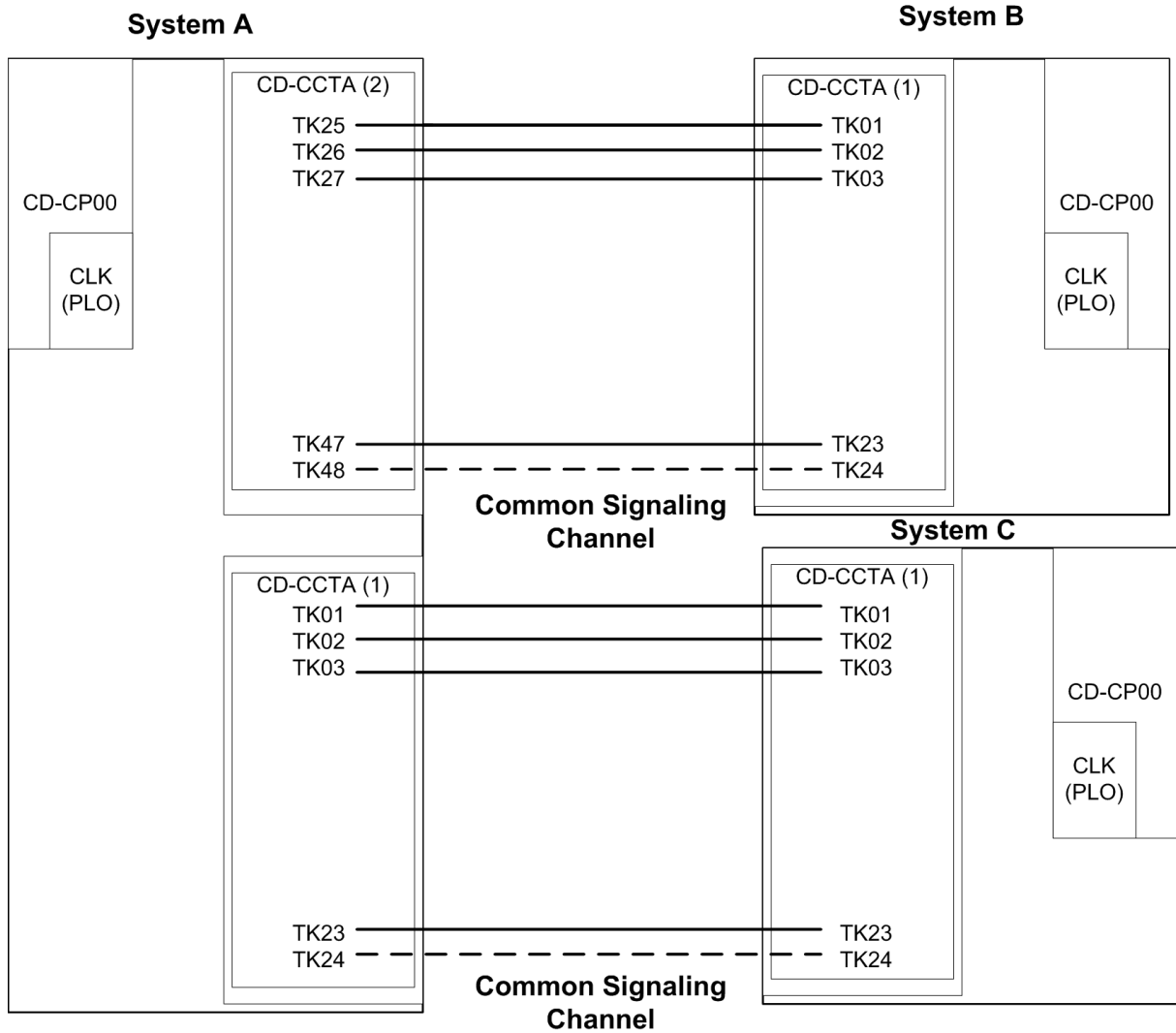
System A:
CIC Numbers for CD-CCTA(1) TK 01~23 = 001~023
CIC Numbers for CD-CCTA(2) TK 25~47 = 001~023

System B:
CIC Numbers for CD-CCTA(1) TK 01~23 = 001~023

System C:
CIC Numbers for CD-CCTA(1) TK 01~23 = 001~023

☐The Common Channel Signaling (CCH) cannot have a CIC
number assigned.

Figure 1-15 Circuit Identification Codes (CIC)

3.8 Determining Numbering Plan

The Uniform Numbering Plan is the numbering plan in the K-CCIS network. The F-Route (Flexible Route Selection) and the Automatic Route Selection (ARS) feature provide the Open Numbering Plan.

When an outgoing call is placed through a K-CCIS link, the originating station number (Office Code and Station Number) and a terminating Station Number are transmitted across the link to the destination office. The originating station number consists of the office number assigned in Program 50-04-01 and the station number assigned in 11-02-01 for the station. These features can be used to edit the dialing string and specify the dialing digits to add to the programmed data to specify the maximum dialing digits in an F-Route table setting.

In a Closed Numbering Plan network, the user can make a call to another station by dialing the station number of the distant system extension. When a Closed Numbering plan is used the extension in the network cannot have the same prefix numbering. Example: System A extensions are 100, System B extensions are 200.

Figure 1-16 Closed Numbering Plan Example and Figure 1-17 Open Numbering Plan Example provide examples of Station Numbering (Closed Numbering) and Office Code and Station Numbering (Open Numbering).

KTS B

KTS A

KTS C

2500   2501

2100   2101   2800   2801
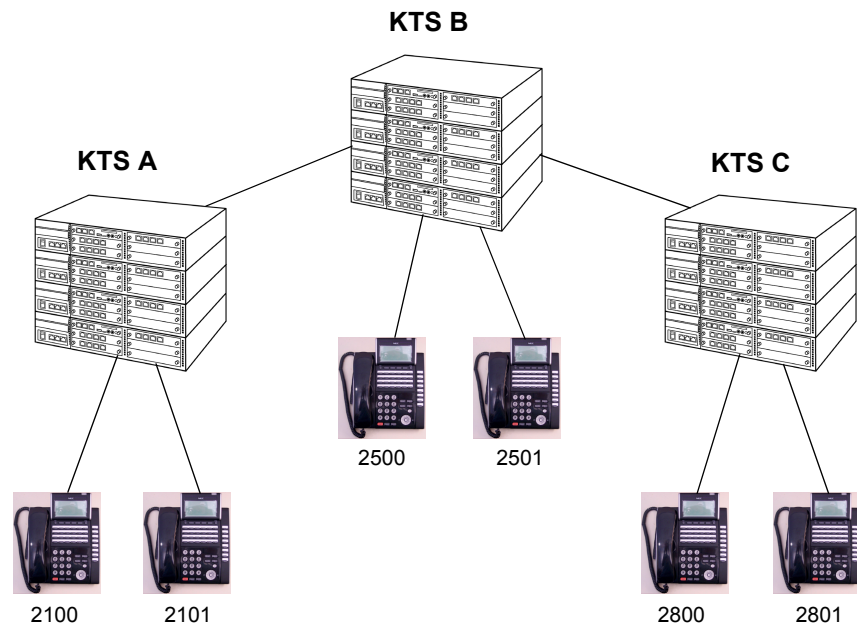
When a call is originated from Office A to Office C, 2800 is dialed.

28 00

Station Number

Office Location (Access Code
analyzed by Program 44-02-01)

✎   When using a Closed Numbering Plan, the station numbers can have two to
       eight digits.

Figure 1-16 Closed Numbering Plan Example
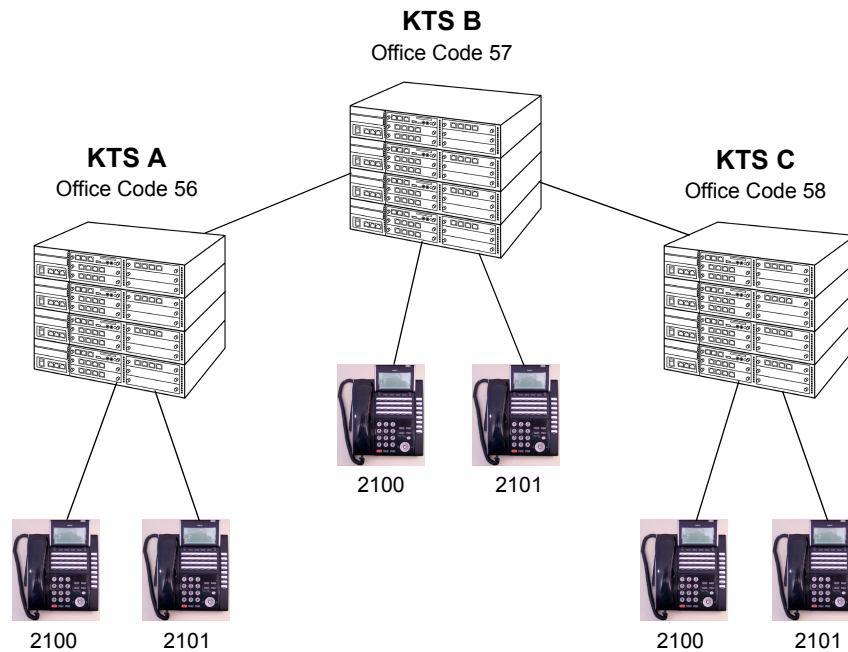
When a call is originated from Office A to Office C, 8 + 58 + 2100 is dialed.

8 = Access Code for F-Route in PRG 11-01-01

858 = Office Code for System C in PRG 44-02-01 (F-Route Dial Analyze Table)

2100 = Station Number

The Station Number can have two to eight digits.

Program 50-04-01 allows a maximum of four digits, including the Access Code and the Office Code.

When using the Open Numbering Plan, the following combination of digits can be used:

❏ When the Access Code is set for two digits, the Office Code can have only two digits.
Access Code = XX
Office Code = XX
Station Number = XXXX

❏ When the Access Code is set for one digit, the Office Code can have two or three digits.
Access Code = X
Office Code = XX or XXX
Station Number = XXXXX

Figure 1-17 Open Numbering Plan Example

Hardware Installation

SECTION 1   INSTALLATION PRECAUTIONS

Pre-installation planning is essential. Advanced planning minimizes installation time, cost, and disruption of the customer business activities.

> *Observe the following precautions when installing the blades to avoid static electricity damage to hardware or exposure to hazardous voltages.*

❍   Never install telephone wiring during a lightning storm.

❍   Never install telephone jacks in wet locations unless the jack is specifically designed for wet locations.

❍   Never touch uninsulated telephone wires or terminals unless the telephone line is disconnected at the network interface.

❍   Use caution when installing or modifying telephone lines.

❍   Ground the Controlling and Expansion chassis before installing or removing the blades.

❍   The Expansion Chassis must be installed with the system power Off.

❍   **Do not touch** the blade components.

❍   Carry the blade in a conductive polyethylene bag to prevent static electricity until ready to install the blade.

❍   When installing or removing the blades from the chassis, the installer must wear a grounded wrist strap to protect the blade from static electricity.

❍   Although it is recommended to install the blades with the ***system power* off**, most blades can be installed hot.

   Ensure the power is turned OFF before installing the following blades:

   ❑   CD-CP00-US

   ❑   PZ-BS10 and PZ-BS11

1.1   Busying Out Extension/Line Blades

The extension/trunk blades may "busy-out" idle circuits. Extensions/lines cannot make or receive calls during this condition. Calls in progress before the blade is "made-busy" are not affected. The blade can be pulled out without interrupting a call in progress.

An extension/line blade LED status:

❍   Normally flashes

❍   Lights steady when "made-busy" when an extension/line is in use

❍   Goes out when the all extensions/lines are "made-busy" (idle)

SECTION 2   INSTALLING AN EXTENSION OR TRUNK BLADE

2.1   Installing the Blades

To install an extension/trunk blade with the system running:

1.   Insert the blade in the guide rail and push the blade securely into position. Tighten the thumb screw on each side of the blade.

2.   The Status LED starts flashing when the blade starts processing (15 seconds).
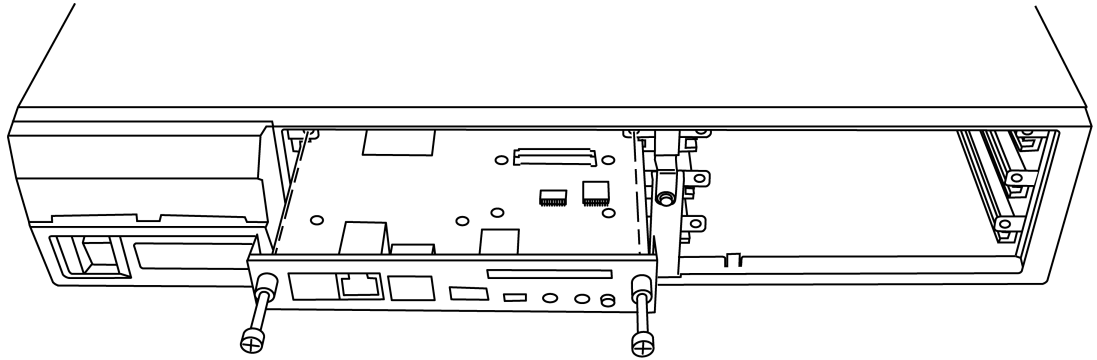
Figure 2-1 Inserting Blades in the 19" Chassis

2.2   Order of Installing Extension Blades

The order in which the station blades (ESIU and SLIU) are physically inserted determines the numbering plan.

> *To avoid unexpected extension/trunk numbering if the VoIP or Voice Mail Daughter Board register with the system first, install the CD-CP00-US blade <u>after</u> the other types of extension and trunk blades are installed.*

For example, when a digital station blade (CD-16DLCA) is in Slot #1 (ext. 301~316) and three additional digital station blades are installed *in the following order,* the numbering plan below applies:

Table 2-1 Extension Blade Installation Order Example

Order of Installation	Blade Slot Number	Blade	Extension Numbers
1	1	CD-16DLCA	101~116
2	2	CD-16DLCA	117~132
3	4	CD-8DLCA PZ-8LCE	133~148
4	3	CD-8DLCA	149~164

After the initial power up of the system, subsequent power ups or resets do not change the slot identification. System programming (Program 90-05) must be performed to change the slot identification.

Adding any daughter board to increase the available ports or going to a higher capacity blade (e.g., CD-16DLCA) may require that the slot be deleted in programming and the blade reinstalled. In the following example, to add a daughter board to slot 2, the blade must be removed, deleted in Program 90-05-01, then reinstalled with the daughter board attached, otherwise the additional ports are not recognized. This however, uses new ports for the combined blade – the initial ports (ports 17~24 using the example below) are not used.

Table 2-2 Adding Daughter Board to Chassis Example

Initial Blade			Updated Blade		
Blade Slot #	**Blade**	**Extension Numbers**	**Blade Slot #**	**Blade**	**Extension Numbers**
1	CD-16DLCA	101~116	1	CD-16DLCA	101~116
2	CD-8DLCA (no daughter board)	117~124	2	—	—
3	CD-16DLCA	125~140	3	CD-16DLCA	125~140
			4	CD-8DLCA (with daughter board)	141~156

The system automatically recognizes each Blade installed in the system. *If a Blade was previously installed* in a slot and another type of Blade is to be installed in that same slot, the Blade must be removed from the chassis and then the slot definition removed using Program 90-05 prior to installing the new Blade.

This same condition applies to extensions and other devices connected to the system. For example, if a port was previously used for a telephone and a DSS Console is to be installed in that same port, the telephone must first be undefined in Program 10-03 before the console is connected.

2.3   Order of Installing Trunk Blades

2.3.1   Installing COIU-LS1/LG1, CD-4ODTA, CD-4DIOPA, or CD-2BRIA Blades

The order in which trunk blades are physically inserted determines the numbering plan.

To avoid unexpected extension/trunk numbering if the VoIP or Voice Mail blades register with the system first, install CD-CP00-US blade after the other types of extension and trunk blades are installed.

For example, if four blades are installed *in the following order,* the numbering plan below would apply.

Table 2-3 Trunk Blade Installation Order Example

Order of Installation	Blade Slot Number	Blade	Line Circuits
1	4	4COIU	1~4
2	5	4COIU	5~8
3	7	4TLIU	9~12
4	6	4TLIU	13~16

SECTION 3   INSTALLING THE CD-CCTA (CCIS Trunk Interface)
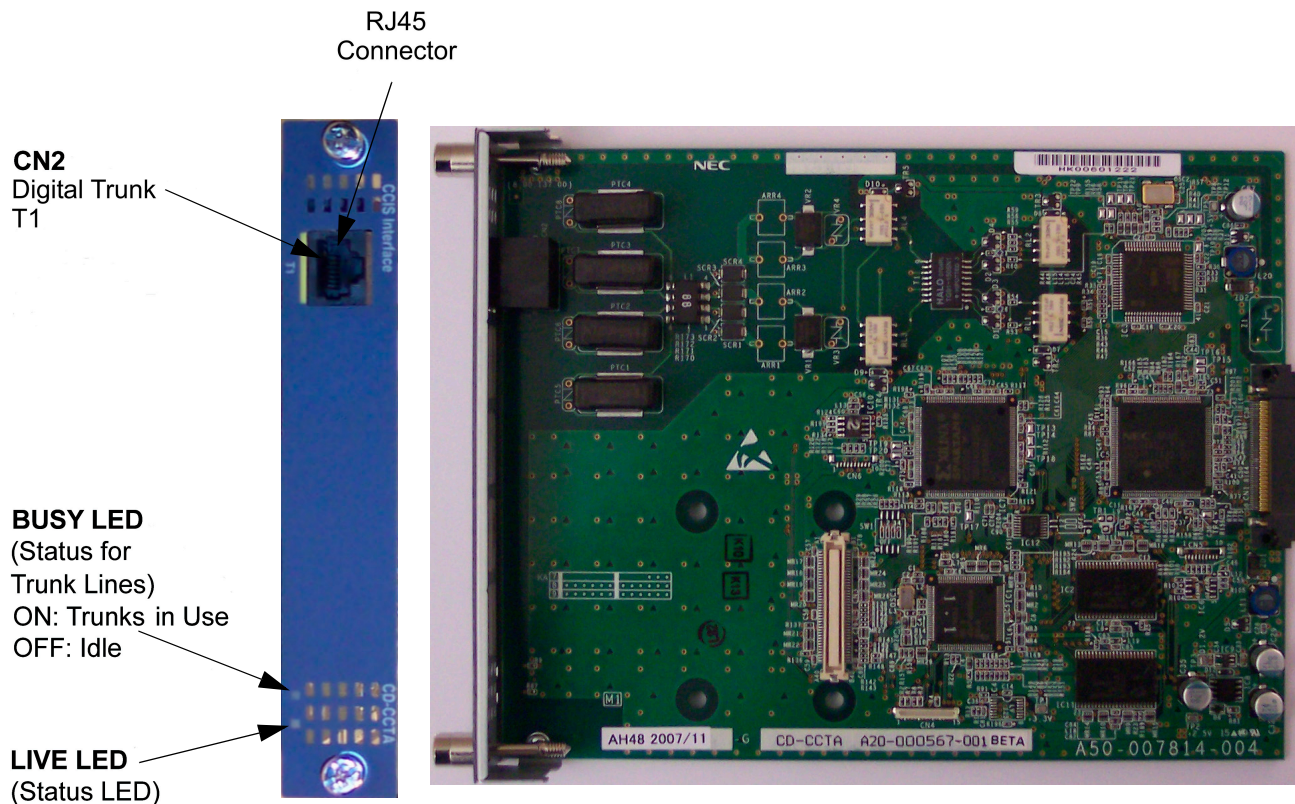
Figure 2-2 CD-CCTA Blade

3.1   Description

The CD-CCTA blade is common to both UNIVERGE SV8100 and SV8300 systems.

The Common Channel Handler Interface blade is a digital trunk blade that terminates FT1 trunks (up to 24 DS-0 channels) providing a common channel signal interface.

The CD-CCTA (Common Channel Handler) is an optional blade that provides a common channel signal through the CD-CCTA to a K-CCIS network and controls the signaling between the KTS and the CP00. Each CD-CCTA blade supports one K-CCIS link. Eight CD-CCTA blades can be installed per system.

The T1 interface has a single 24 channel 64kb/s digital signal circuit that can be configured for T1 trunking.

3.2   Installation

Install the CD-CCTA in any universal slot.

3.3   LED Indications

LED indications for the CD-CCTA are listed in Table 2-4 CD-CCTA LED Indications. Each LED is listed with its associated function and LED and Operational status. Refer to Figure 2-3 CD-CCTA LED Indication Pattern of Layer 1 on T1 Unit on page 2-8 for LED pattern information.

Table 2-4 CD-CCTA LED Indications

Alarm	Details of the Alarm	LED Indication Pattern
LOS	Loss of Signal (Red Alarm) No Signal (Analog Interface)	Following an alarm blink (red, green, red, green) a Red LED will light.
AIS	Alarm Indication Signal (Blue Alarm)	Following an alarm blink (red, green, red, green) a Red LED slowly flashes On and Off twice.
OOF	Out Of Frame (Red Alarm)	Following an alarm blink (red, green, red, green) a Red LED and Green LED flash On and Off three times simultaneously.
RAI	Remote Alarm Indication (Yellow Alarm)	Following an alarm blink (red, green, red, green) a Green LED flashes On and Off twice.
No Alarm	The system does the LED control.	

✎   The order of priority is set up to alarm in the order LOS>AIS>OOF>RAI.
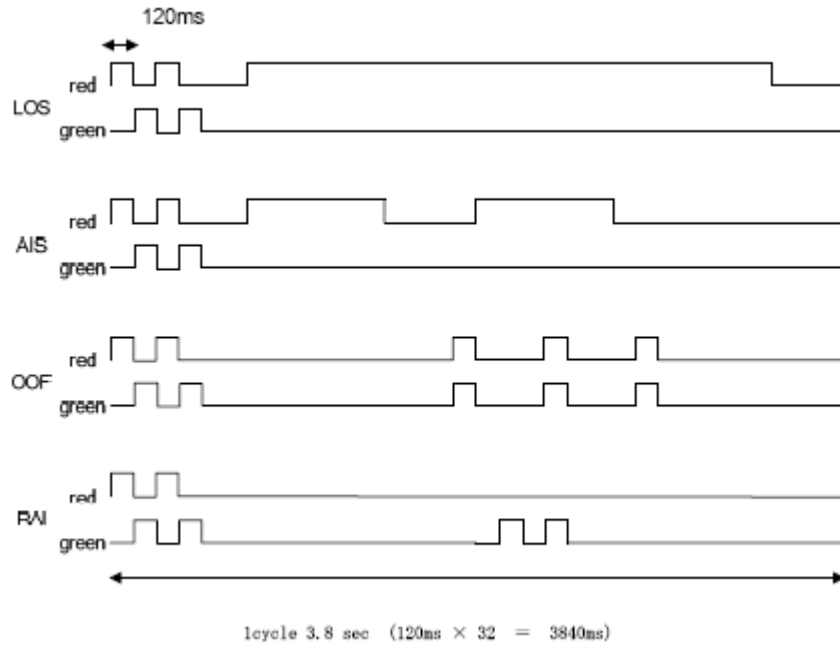
Figure 2-3 CD-CCTA LED Indication Pattern of Layer 1 on T1 Unit

3.4   Connectors

Table 2-5 CD-CCTA RJ-45 Cable Connector Pin-Outs shows the pin-outs for the RJ-45 connector. Refer to Figure 2-2 CD-CCTA Blade on page 2-6 for an illustration showing the location of the connectors on the CD-CCTA blade.

Table 2-5 CD-CCTA RJ-45 Cable Connector Pin-Outs

RJ-45 Cable Connector – CN2		
	Pin No.	**Connection**
	1	RA
	2	RB
	3	—
	4	TA
	5	TB
	6	—
	7	—
	8	—

System Data Programming

SECTION 1   K-CCIS PROGRAMMING

This chapter lists the Programs that must be assigned to support K-CCIS. The Programming used depends on the K-CCIS features that are used. The tables provided in this section provide a complete list of the required Programs that support the function (e.g., Digital Trunk Assignment, CCH Assignment, Numbering Plan Assignment).

At the end of this section, programming samples are provided for Open and Closed Numbering Plans.

1.1   Digital Trunk Data Assignment

Use these Programming assignments to indicate to the system where (which slot) the CD-CCTA blade is located, the signaling format the CD-CCTA blade uses, and to assign other information relating to the trunks.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
10-03-01	Blade Setup – Logical Port Number	0 ~ 200	The CD-CP00-US automatically defines each blade during installation.
10-03-02	Blade Setup – Frame Type Setup	0 = D4 (default) (12 Multiframe) 1 = ESF (24 Multiframe)	If 56 K K-CCIS is used. 24 Multiframe (ESF) must be assigned.
10-03-03	Blade Setup – Zero Code Suppression Setup ZCS_B8ZS	0 = B8ZS (default) 1 = AMI/ZCS	

Program/ Item No.	Description/ Selection	Assigned Data	Comments
10-03-04	Blade Setup – DTI<-> CSU Distance Setup	0 = 0~ 133 feet (default) 1 = 133 ~ 266 feet 2 = 266 ~ 399 feet 3 = 399 ~ 533 feet 4 = 533 ~ 655 feet	Maximum distance back-to-back T1s can be connected without CSU/DSU service.
10-03-05	Blade Setup – T1 Clock Source Master/Slave	0 = Internal 1 = External (default)	Define the Master (Internal) or Slave (External) clock source.
10-03-06	Blade Setup – Number of Ports	Auto 4 Ports 8 Ports 12 Ports 16 Ports 20 Ports	
10-03-07	Blade Setup – Wiring Type	Auto Cross Straight	
14-05-01	Trunk Group Assign Trunk to Trunk Groups/Outbound Priority	Trunk Group = 1~100 Default is 1 Priority = 1 ~ 200	Default priorities for trunks 1 ~ 200 is 1 ~ 200.
22-02-01	Incoming Call Trunk Setup	0 = Normal (default) 1 = VRS (second dial tone if no VRS installed) 2 = DISA 3 = DID 4 = DIL 5 = E&M Tie Line 6 = Delayed VRS 7 = ANI/DNIS 8 = DID(DDI) Mode Switching	Set the feature type for the trunk you are programming.
34-01-01	E&M Tie Line Basic Setup – DID/E&M Start Signaling	0 = 2nd Dial Tone 1 = Wink 2 = Immediate 3 = Delay Default is 1	Set the signaling mode for DID and Tie trunks. DID and Tie trunks can use either Immediate start or Wink start signaling.

1.2   CCIS Assignment

Use these Programming assignments to set the availability of CCIS.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
50-01-01	CCIS System Setting – CCIS Availability	0=Disable 1=Enable Default is 0	Any CCIS settings lose functionality if this setting is set to 0.
50-02-01	Connecting System Settings – Port Number of Common Signaling Channel (T1)	0~200 Default is 0	Specify the Trunk port to send D-channel information. This program is available for using DTI package.
50-02-02	Connecting System Settings – Common Signaling Channel Data Speed Assignment (T1)	0=64Kbps 1=56Kbps (Default) 2=48Kbps(1) 3=48Kbps(2) Default is 1	If 56K K-CCIS is used (1), 24 Multi-Frame (ESF) must be assigned in PRG 10-03-02.
50-02-03	Connecting System Settings – Originating Point Code	0 ~ 16367 Default is 0	Default settings for each CCIS Route ID is not assigned. This Program is for DTI Only and must be the same for all CCIS Route IDs.
50-02-04	Connecting System Settings – Destination Point Code (T1)	0 ~ 16367 Default is 0	DPC must be what the OPC is on the opposite side of the link.
50-02-05	Connecting System Settings – Calling Name Indication (T1)	0=Disable 1=Enable Default is 1	Enable to receive name display across K-CCIS.
50-04-01	CCIS Office Code Assignment – CCIS Office Code	Assign up to four digits. Default is No Setting	Only used in an open numbering plan network. This should include the Trunk Access Code and Office Code number.
14-14-01	CCIS Trunk CIC Assignment – Trunk Group Number 1~200	0 = Not Assigned 1~127 = CIC Numbers Default is 0	Assign a Circuit Identification Code (CIC) number to each trunk number used for voice channel.

1.3   Numbering Plan Assignment

Use these Program assignments to indicate to the system the number of digits that are assigned to stations, the number of digits assigned to Access Codes, and to assign stations to ports.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-01-01	System Numbering	0 = Not Used1 = Service Code 2 = Extension Number 3 = Trunk Access 4 = Special Trunk Access 5 = Operator Access 6 = Flexible Routing 9 = Dial Extension Analyze	Default for 1X, 2X, and 3X is 2.
11-02-01	Extension Numbering	Assign Station Numbers to Port Numbers	Defaults for Ports 1 ~ 512: 101 ~ 199 3101 ~ 3513
11-04-01	Virtual Extension Numbering	Assign Station Numbers to Port Number	Defaults for Ports 1 ~ 99 = 201~ 299 100 ~ 256 = 3601 ~ 3757

1.4   Programming for Closed Numbering Plan

Use these Program assignments to assign an Access Code to the Intercom Function Numbers, to assign F-Routes to the appropriate Trunk Routes, and specify the digits that are added to the dialed number.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-01-01	System Numbering	2 = Extension Number	Default for 1X, 2X, and 3X is 2.
44-02-01	Dial Analysis Table for ARS/F-Route Access – Dial Number of digits to be analyzed by the system.	Up to eight digits can be assigned. Default is No Setting.	Assign the digits to be dialed across the K-CCIS link. These digits were assigned as F-Route in Program 11-01-01. (Use line key I for "Don't Care" digit @.)
44-02-02	Dial Analysis Table for ARS/F-Route Access – Service Type	2 = ARS/F-Route Table Default is 0	Service type 2 assigns the digits to be dialed to an F-Route. Program 44-02-03 assigns the F-Route to be used.
44-02-03	Dial Analysis Table for ARS/F-Route Access – Additional Data	2 = 0 ~ 500 (0 = No Setting) Default is 0	When setting data is 2, refer to Program 44-05 for further routing options.
44-05-01	ARS/F-Route Table – Trunk Group Number	1 ~ 100, 255 0 = No Setting 255 = Extension Call Default is 0	Select the trunk group to be used for the outgoing K-CCIS call.

1.5   Programming for Open Numbering Plan

Use these Programs to assign the number of digits to Access Code and to make ARS assignments.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-01-01	System Numbering	2 = Extension Number	Default for 1X, 2X, and 3X is 2
44-02-01	Dial Analysis Table for ARS/F-Route Access – Number of digits to be analyzed by the system	Up to eight digits can be assigned. Default is No setting	Assign the digits to be dialed across the K-CCIS link. These digits were assigned as F-Route in Program 11-01-01. Use line key I for "Don't Care" digit @.
44-02-02	Dial Analysis Table for ARS/F-Route Access – Service Type	2 = ARS/F-Route Table Default is 0	Service type 2 assigns the digits to be dialed to an F-Route. Program 44-02-03 assigns the F-Route to be used.
44-02-03	Dial Analysis Table for ARS/F-Route Access – Additional Data	2 = 0 ~ 500 (0 = No Setting) Default is 0	When setting data is (2), refer to Program 44-05.
44-05-01	ARS/F-Route Table – Trunk Group Number	1 ~ 100, 255 0 = No Setting 255 = Extension Call Default is 0	Select the trunk group to be used for the outgoing K-CCIS call.
44-05-02	ARS/F-Route Table – Delete Digits	0 = No Setting 1 ~ 255 = Number of digits to delete (255 = Delete All) Default is 0	Enter the number of digits to be deleted from the dialed number.
44-05-03	ARS/F-Route Table – Additional Dial Number Table	0 = No Setting 1 ~ 1000 = Add Table in Program 44-06-01 Default is 0	Enter the table number defined in Program 44-06 for additional digits to be dialed.
44-05-04	ARS/F-Route Table – Beep tone	0 = Off 1 = On Default is 0	Select whether or not a beep is heard if a lower priority trunk group is used to dial out.
44--05-05	ARS/F-Route Table – Gain Table Number for Internal Calls	0 = No Setting 1 ~ 500 Default is 0	Select the gain table number to be used for the internal call defined in Program 44-07.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
44-05-06	ARS/F-Route Table – Gain Table Number for Tandem Connections	0 = No Setting 1 ~ 500 Default is 0	Select the gain table number to be used for the internal call defined in Program 44-07.
44-05-07	ARS/F-Route Table – ARS Class of Service	0 = No Setting 1 ~ 16 Default is 0	Select the ARS Class of Service to be used for the table. An extension ARS COS is determined in Program 26-04-01.
44-05-08	ARS/F-Route Table – Dial Treatment	0 = No Setting 1 ~ 15 Default is 0	If a Dial Treatment is selected, Programs 44-05-02 and 44-05-03 are ignored and the Dial Treatment defined in Program 26-03-01 is used instead.
44-05-09	ARS/F-Route Table – Maximum Digit Assignment	0 = No Maximum 1 ~ 24 Default is 0	Assign the Maximum Digits for K-CCIS or ISDN Calls.

1.6   Closed Number Programming Example

This section provides the steps needed to program a closed numbering plan.

Step 1: T1 Tie Lines

The following diagram is an example of Programs that should be assigned for T1 Tie lines. The example assumes that the SV8100 system is defaulted with the CD-CCTA card installed.

Abbreviations used in the diagram:

MB = Memory Block   TRK = Trunk   TG = Trunk Group

CC = Clear Channel   TT = Trunk Type

Univerge SV8100	Univerge SV8100	Electra Elite IPK II
PRG 10-03-03 CC = B8ZS	**PRG 10-03-03** CC = B8ZS	**PRG 10-03-03** CC = B8ZS
PRG 10-03-05 Internal	**PRG 10-03-05** External Internal	**PRG 10-03-05** External
PRG 22-02-01 TT = E&M	**PRG 22-02-01** TT = E&M	**PRG 22-02-01** TT = E&M
PRG 14-05-01 TRK 001~024 = TG 10	**PRG 14-05-01** TRK 001~024 = TG 10 TRG 025~048 = TG 11	**PRG 14-05-01** TRK 001~024 = TG 10
PRG 14-01-13 TRK 001~024 = Yes	**PRG 14-01-13** TRK 001~048 = Yes	**PRG 14-01-13** TRK 001~024 = Yes
PRG 34-01-01 TRK 001~024 = Wink	**PRG 34-01-01** TRK 001~048 = Wink	**PRG 34-01-01** TRK 001~024 = Wink
PRG 11-01-01 1X = 3 digit; Intercom 2X = 3 digit; F-Route 3X = 3 digit; F-Route	**PRG 11-01-01** 1X = 3 digit; F-Route 2X = 3 digit; Intercom 3X = 3 digit; F-Route	**PRG 11-01-01** 1X = 3 digit; F-Route 2X = 3 digit; F-Route 3X = 3 digit; Intercom
PRG 11-02-01 Extensions = 1XX	**PRG 11-02-01** Extensions = 2XX	**PRG 11-02-01** Extensions = 3XX
PRG 11-04-01 Unassign Virtuals = 2XX	**PRG 11-04-01** Unassign Virtuals = 2XX	**PRG 11-04-01** Unassign Virtuals = 2XX

Step 2: Closed Numbering Plan

The following diagram provides an example of Programs that should be assigned for Closed Numbering. The example assumes that Step 1: T1 Tie Lines was completed.

AC = Access Code   BLK = Closed Numbering Block   MB = Memory Block

ACG = Access Item Code   TT = Trunk Type

Univerge SV8100	Univerge SV8100	Electra Elite IPK II
——Tie Lines—— TG 10	TG 10 ——Tie Lines—— TG 11	TG 10 (300)
100 ~ 130	200 ~ 230	300 ~ 330
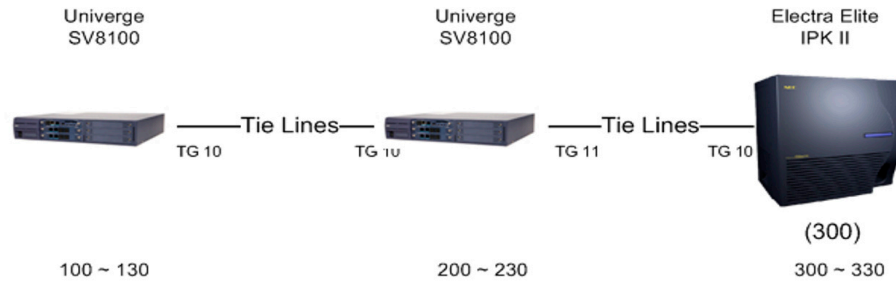
<u>**Calling**</u> 200~230 300~330	<u>**Calling**</u> 100~130 300~330	<u>**Calling**</u> 100~130 200~230
<u>**PRG 11-01-01**</u> Dial 1X = 3 digit; Intercom Dial 2X = 3 digit; F-Route Dial 3X = 3 digit; F-Route	<u>**PRG 11-01-01**</u> Dial 1X = 3 digit; F-Route Dial 2X = 3 digit; Intercom Dial 3X = 3 digit; F-Route	<u>**PRG 11-01-01**</u> Dial 1X = 3 digit; F-Route Dial 2X = 3 digit; F-Route Dial 3X = 3 digit; Intercom
<u>**PRG 44-02-01**</u> TBL 1 = 2 TBL 2 = 3	<u>**PRG 44-02-01**</u> TBL 1 = 1 TBL 2 = 3	<u>**PRG 44-02-01**</u> TBL 1 = 1 TBL 2 = 2
<u>**PRG 44-02-02**</u> TBL 1 = F-Route TBL 2 = F-Route	<u>**PRG 44-02-02**</u> TBL 1 = F-Route TBL 2 = F-Route	<u>**PRG 44-02-02**</u> TBL 1 = F-Route TBL 2 = F-Route
<u>**PRG 44-02-03**</u> TBL 1 = F-Route 1 TBL 2 = F-Route 1	<u>**PRG 44-02-03**</u> TBL 1 = F-Route 1 TBL 2 = F-Route 2	<u>**PRG 44-02-03**</u> TBL 1 = F-Route 1 TBL 2 = F-Route 1
<u>**PRG 44-05-01**</u> F-Route TBL 1 = TG 10	<u>**PRG 44-05-01**</u> F-Route TBL 1 = TG 10 F-Route TBL 2 = TG 11	<u>**PRG 44-05-01**</u> F-Route TBL 1 = TG 10
<u>**PRG 44-05-02**</u> F-Route TBL 1 Delete Digit = 0	<u>**PRG 44-05-02**</u> F-Route TBL 1 Delete Digit = 0 F-Route TBL 2 Delete Digit = 0	<u>**PRG 44-05-02**</u> F-Route TBL 1 Delete Digit = 0
<u>**PRG 44-05-03**</u> F-Route TBL 1 Add Dial = 0	<u>**PRG 44-05-03**</u> F-Route TBL 1 Add Dial = 0 F-Route TBL 2 Add Dial = 0	<u>**PRG 44-05-03**</u> F-Route TBL 1 Add Dial = 0
<u>**PRG 44-05-09**</u> F-Route TBL 1 Max. Digit = 3	<u>**PRG 44-05-09**</u> F-Route TBL 1 Max. Digit = 3 F-Route TBL 2 Max. Digit = 3	<u>**PRG 44-05-09**</u> F-Route TBL 1 Max. Digit = 3

✎   *Before moving on to the next step, test the T1 Tie lines and the Closed Numbering Plan.*

Step 3: K-CCIS Activation

The following diagram provides an example of Programs that should be assigned for K-CCIS. The example assumes that Step 1: T1 Tie Lines and Step 2: Closing Number Plan are completed.

DSTCCH = Destination Point Code   CCH = Control Channel Handler   TRK = Trunk

ORGCCH = Originating Point Code   TG = Trunk Group

UNIVERGE SV8100	UNIVERGE SV8100	Electra Elite IPK II
TG 10 TG 10	TG 11 TG 10	
TG 00 TG 00	TG 00 TG 00	
CCH	CCH	
(101s)	(201s)	(301s)
PC00001	PC00002	PC00003
101~130	201~230	301~330

PRG 14-13-01
TG10 = CCIS Route ID 1

PRG 14-14-01
TRK 001~023 = 1~23

PRG 14-05-01
TRK 001~024 = 10
TRK 024 = 00
Priority = TRK 001~024 = 024~001

PRG 22-02-01
TRK 001~024 = Tie

PRG 50-01-01
CCIS Availability = Yes

PRG 50-02-01
CCIS Route ID1 = TRK 024

PRG 50-02-02
CCIS Route ID1 = 56K

PRG 50-02-03
CCIS Route ID1 = Org. Point Code 1

PRG 14-13-01
TG10 = CCIS Route ID 1
TG11 = CCIS Route ID 2

PRG 14-14-01
TRK 001~023 = 1 ~ 23
TRK 025~047 = 1 ~ 23

PRG 14-05-01
TRK 001~023 = 10
TRK 024 = 00
Priority = TRK 001~024 = 024~001
TRK 025~048 = 11
TRK 048 = 00
Priority = TRK 025~048 = 048~025

PRG 22-02-01
TRK 001~024 = Tie
TRK 025~048 = Tie

PRG 50-01-01
CCIS Availability = Yes

PRG 50-02-01
CCIS Route ID1 = TRK 024
CCIS Route ID2 = TRK 048

PRG 50-02-02
CCIS Route ID1 = 56K
CCIS Route ID2 = 56K

PRG 50-02-03
CCIS Route ID1 = Org. Point Code 1
CCIS Route ID2 = Org. Point Code 1

PRG 14-13-01
TG10 = CCIS Route ID 1

PRG 14-14-01
TRK 001~023 = 1~23

PRG 14-05-01
TRK 001~023 = 10
TRK 024 = 00
Priority = TRK 001~024 = 024~001

PRG 22-02-01
TRK 001~024 = Tie

PRG 50-01-01
CCIS Availability = Yes

PRG 50-02-02
CCIS Route ID1 = 56K

PRG 50-02-03
CCIS Route ID1 = Org. Point Code 3

PRG 50-02-04
CCIS Route ID1 = Dest. Point Code 2

PRG 50-02-04
CCIS Route ID1 = Dest. Point Code 1
CCIS Route ID2 = Dest. Point Code 3

PRG 50-02-04
CCIS Route ID1 = Dest. Point Code 2

PRG 50-02-06
CCIS Route ID1 = CCH1
CCIS Route ID2 = CCH2

PRG 50-03-01
N/A

PRG 50-03-01
System ID1 = 00001
System ID2 = 00003

PRG 50-03-01
N/A

PRG 50-04-01
Vacant

PRG 50-04-01
Vacant

PRG 50-04-01
Vacant

1.7   Open Number Programming Example

This sections provides the steps needed to program an open numbering plan.

Step 1: T1 Tie Lines

The following diagram provides an example of Program and item numbers that should be assigned for T1 Tie lines. The example assumes that each UNIVERGE SV8100 and Electra Elite IPK II system is defaulted with the CD-CCTA blades installed.

Abbreviations used in the diagram:

MB = Memory Block   TRK = Trunk   TG = Trunk Group

CC = Clear Channel   TT = Trunk Type
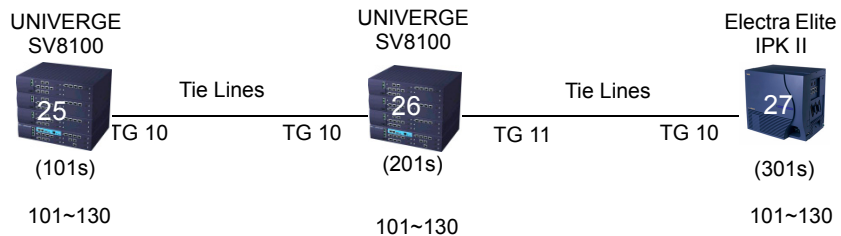
PRG 10-03-05	**PRG 10-03-05**	**PRG 10-03-05**
Internal	External Internal	External
PRG 10-03-03 CC = B8ZS	**PRG 10-03-03** CC = B8ZS	**PRG 10-03-03** CC = B8ZS
PRG 22-02-01 TRK 001~024 = Tie	**PRG 22-02-01** TRK 001~024 = Tie TRK 025~048 = Tie	**PRG 22-02-01** TRK 001~024 = Tie
PRG 14-05-01 TRK 001~024 = TG 10	**PRG 14-05-01** TRK 001~024 = TG 10 TRK 025~048 = TG 11	**PRG 14-05-01** TRK 001~024 = TG 10
PRG 14-02-13 TRK 001~024 = Yes	**PRG 14-02-13** TRK 001~024 = Yes TRK 025~048 = Yes	**PRG 14-02-13** TRK 001~024 = Yes
PRG 22-02-01 TRK 001~024 = Tie	**PRG 22-02-01** TRK 001~024 = Tie TRK 024~048 = Tie	**PRG 22-02-01** TRK 001~024 = Tie
PRG 34-01-01 TRK 001~024 = Wink	**PRG 34-01-01** TRK 001~024 = Wink TRK 025~048 = Wink	**PRG 34-01-01** TRK 001~024 = Wink

Step 2: Open Numbering Plan

The following diagram provides an example of Programs and item numbers that should be assigned for Open Numbering. The example assumes that Step 1: T1 Tie Lines was completed.

Abbreviations used in the diagram:

AC = Access Code   ACG = Access Item Code   MB = Memory Block

RT = Route   TB = ARS Table   PRG =Program

Calling	Calling	Calling
101 ~ 130	101 ~ 130	101 ~ 130
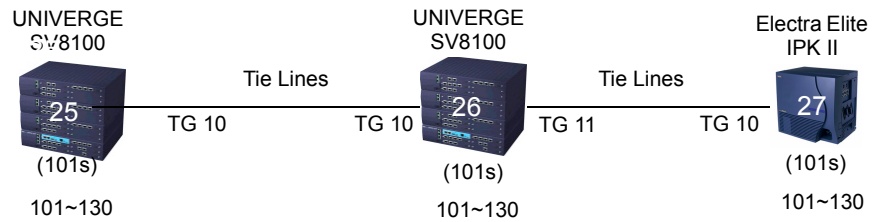
PRG 11-01-01	**PRG 11-01-01**	**PRG 11-01-01**
Dial 1X = 3 Digits; Intercom	Dial 1X = 3 Digits; Intercom	Dial 1X = 3 Digits; Intercom
Dial 8X = 1 Digit, F-Route Access	Dial 8X = 1 Digit, F-Route Access	Dial 8X = 1 Digit, F-Route

PRG 11-02-01	**PRG 11-02-01**	**PRG 11-02-01**
Port 1 = 101	Port 1 = 101	Port 1 = 101

PRG 44-02-01	**PRG 44-02-01**	**PRG 44-02-01**
F-Route TBL1 = 825	TBL1 = 825	F-Route TBL1 = 825
F-Route TBL2 = 826	TBL2 = 826	F-Route TBL2 = 826
F-Route TBL3 = 827	TBL3 = 827	F-Route TBL3 = 827

PRG 44-02-02	**PRG 44-02-02**	**PRG 44-02-02**
TBL1 = F-Route	TBL1 = F-Route	TBL1 = F-Route
TBL2 = F-Route	TBL2 = F-Route	TBL2 = F-Route
TBL3 = F-Route	TBL3 = F-Route	TBL3 = F-Route

PRG 44-02-03	**PRG 44-02-03**	**PRG 44-02-03**
TBL1 = F-Route TBL1	TBL1 = F-Route 1	TBL1 = F-Route 1
TBL2 = F-Route TBL2	TBL2 = F-Route 2	TBL2 = F-Route 2
TBL3 = F-Route TBL2	TBL3 = F-Route 3	TBL3 = F-Route 2

PRG 44-05-01	**PRG 44-05-01**	**PRG 44-05-01**
TBL1 = TG 255	TBL1 = TG 10	TBL1 = TG 10
TBL2 = TG 10	TBL2 = TG 255	TBL2 = TG 10
	TBL3 = TG 11	TBL3 = TG 255

PRG 44-05-02
F-Route TBL1; Delete Digit = 3

PRG 44-05-02
F-Route TBL2: Delete Digit = 3

PRG 44-05-02
F-Route TBL2; Delete Digit = 3

PRG 44-05-09
F-Route TBL2; Max Digit = 6

PRG 44-05-09
F-Route TBL1: Max Digit = 6
F-Route TBL3: Max Digit = 6

PRG 44-05-09
F-Route TBL2; Max Digit = 6

✎   *Before moving to the next step, test the T1 Tie lines and the Open Numbering Plan.*

Step 3: K-CCIS Activation

The following diagram provides an example of Programs and item numbers
that are assigned for K-CCIS. The example assumes that Step 1: T1 Tie
Lines and Step 2: Open Number Plan are completed.

DSTCCH = Destination Point Code   CCH = Control Channel Handler   TRK = Trunk

ORGCCH = Originating Point Code   TG = Trunk Group   PRG =Program

PRG 14-13-01
TG10 = CCIS Route ID 1

PRG 14-14-01
TRK 001~023 = 1~23

PRG 14-05-01
TRK 001~024 = 10
TRK 024 = 00
Priority = TRK 001~024 = 024~001

PRG 22-02-01
TRK 001~024 = Tie

PRG 50-01-01
CCIS Availability = Yes

PRG 50-02-01
CCIS Route ID1 = TRK 024

PRG 50-02-02
CCIS Route ID1 = 56K

PRG 50-02-03
CCIS Route ID1 = Org. Point Code 1

PRG 14-13-01
TG10 = CCIS Route ID 1
TG11 = CCIS Route ID 2

PRG 14-14-01
TRK 001~023 = 1 ~ 23
TRK 025~047 = 1 ~ 23

PRG 14-05-01
TRK 001~023 = 10
TRK 024 = 00
Priority = TRK 001~024 = 024~001
TRK 025~048 = 11
TRK 048 = 00
Priority = TRK 025~048 = 048~025

PRG 22-02-01
TRK 001~024 = Tie
TRK 025~048 = Tie

PRG 50-01-01
CCIS Availability = Yes

PRG 50-02-01
CCIS Route ID1 = TRK 024
CCIS Route ID2 = TRK 048

PRG 50-02-02
CCIS Route ID1 = 56K
CCIS Route ID2 = 56K

PRG 50-02-03
CCIS Route ID1 = Org. Point Code 1

PRG 14-13-01
TG10 = CCIS Route ID 1

PRG 14-14-01
TRK 001~023 = 1~23

PRG 14-05-01
TRK 001~024 = 10
TRK 024 = 00
Priority = TRK 001~024 = 024~001

PRG 22-02-01
TRK 001~024 = Tie

PRG 50-01-01
CCIS Availability = Yes

PRG 50-02-02
CCIS Route ID1 = 56K

PRG 50-02-03
CCIS Route ID1 = Org. Point Code 3

CCIS Route ID2 = Org. Point Code 1

PRG 50-02-04	**PRG 50-02-04**	**PRG 50-02-04**
CCIS Route ID1 = Dest. Point Code 2	CCIS Route ID1 = Dest. Point Code 1	CCIS Route ID1 = Dest. Point Code 2
	CCIS Route ID2 = Dest. Point Code 3	

PRG 50-02-06
CCIS Route ID1 = CCH1
CCIS Route ID2 = CCH2

PRG 50-03-01	**PRG 50-03-01**	**PRG 50-03-01**
N/A	System ID1 = 00001	N/A
	System ID2 = 00003	

PRG 50-04-01	**PRG 50-04-01**	**PRG 50-04-01**
825	826	827

1.8   Dedicated Tandem CO Trunk Calls

The following diagram provides an example of SV8100 Programs that should be assigned when all Local and Long Distance CO calls, from the REMOTE site, are routed through the MAIN site using Flexible Routing (F-Routes) and Automatic Route Selection (ARS).

The example assumes that the UNIVERGE SV8100 systems are connected via K-CCIS and using Closed Numbering Plan. All local calls are 10-digit dial and all 1+ calls are 11-digit dial. Only one CO Trunk is set for 911 in the REMOTE system.

MB = Memory Block	TRK = Trunk	TG = Trunk Group
RT = ARS Route	DN = Dial Number	PRG = Program
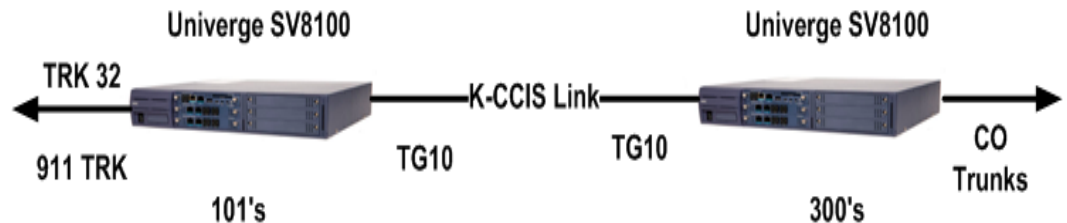
PRG 11-01-01
Dial 8 = 1 Digit: 2nd Trunk Access
Dial 9 = 1 Digit; Trunk Access
Dial 3 = 1 Digit; F-Route
Dial 0 = 1 Digit; F-Route

PRG 11-01-01
Dial 9 = 1 Digit; Trunk Access
Dial 3 = 3 Digit; Intercom
Dial 0 = 1 Digit; Operator Access

PRG 11-09-01
Trunk Access Code = 9

PRG 11-09-02
2nd Trunk Access Code = 8

PRG 26-01-01
ARS Service = On

PRG 26-02-01
ARS TBL1 = @

PRG 26-02-02
ARS TBL1 = F-Route

PRG 26-02-03
ARS TBL1 = F-Route TBL 1

PRG 44-02-01
Analysis TBL1 = 0
Analysis TBL2 = 3

PRG 44-02-02
Analysis TBL1 = F-Route
Analysis TBL2 = F-Route

PRG 44-02-03
Analysis TBL1 = Data F-Route 2
Analysis TBL2 = Data F-Route 3

PRG 44-05-01
F-Route TBL1 = TG 10
F-Route TBL2 = TG 10
F-Route TBL3 = TG 10

PRG 44-05-08
F-Route TBL1 = ARS Treatment TBL1

PRG 44-05-09
F-Route TBL1 = Max Digit 10
F-Route TBL2 = Max Digit 1
F-Route TBL3 = Max Digit 3

PRG 26-03-01
ARS Treatment TBL1 = D019RE

1.9   Shared Tandem CO Trunk Calls

The following diagram provides an example of Programs and item numbers that should be assigned when two sites share CO lines for reducing Long Distance calls using Automatic Route Selection (ARS).

The example assumes that the UNIVERGE SV8100 systems are connected via K-CCIS and using Closed Numbering Plan. All local calls are 10-digit dial and all 1+ calls are 11-digit dial.

MB = Memory Block   TRK = Trunk   TG = Trunk Group

TB = ARS Table   DN = Dial Number   RT = ARS Route

UNIVERGE
SV8100
Remote

UNIVERGE
SV8100
Main

CO Trunks

K-CCIS Link

CO Trunks

TG10   TG10

Area Code 817
Area Code 940

Area Code 214
Area Code 972

101s   301s
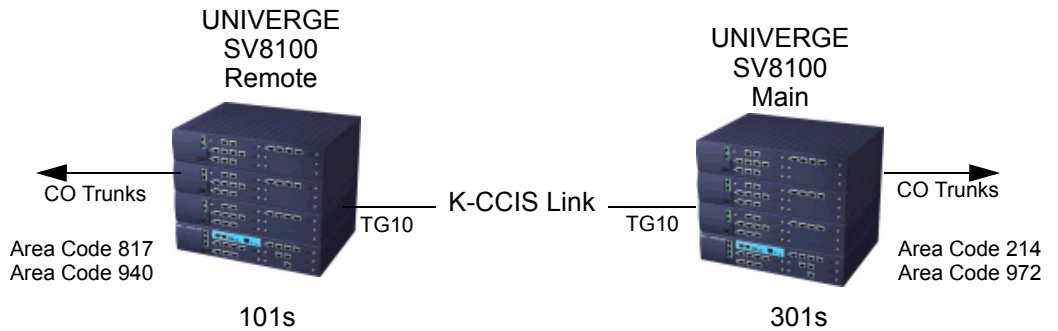
PRG 11-01-01   **PRG 11-01-01**
Dial 9 = 1 digit: Trunk Access   Dial 9 = 1 digit: Trunk Access

PRG 11-09-01   **PRG 11-09-01**
Trunk Access Code = 9   Trunk Access Code = 9

PRG 11-09-02   **PRG 11-09-02**
2nd Trunk Access = 8   2nd Trunk Access = 8

PRG 26-01-01   **PRG 26-01-01**
ARS Service = On   ARS Service = On

PRG 26-02-01   **PRG 26-02-01**
ARS Analysis TBL1 = 1214   ARS Analysis TBL1 = 1817
ARS Analysis TBL2 = 1972   ARS Analysis TBL2 = 1940
ARS Analysis TBL3 = 1817   ARS Analysis TBL3 = 1214

PRG 26-02-02
ARS Analysis TBL1 = F-Route
ARS Analysis TBL2 = F-Route
ARS Analysis TBL3 = TRG (Trunk Group)

PRG 26-02-03
ARS Analysis TBL1 = F-Route TBL1
ARS Analysis TBL2 = F-Route TBL1
ARS Analysis TBL3 = Add Data TG 01

PRG 44-05-01
F-Route TBL1 = TG 10

PRG 44-05-08
F-Route TBL1 = ARS Edit TBL1

PRG 44-05-09
F-Route TBL1 = Max Digit 11

PRG 26-03-01
ARS treatment TBL1 = D019RE

Features and Specifications

4

SECTION 1   GENERAL INFORMATION

Key-Common Channel Interoffice Signaling (K-CCIS) allows multiple systems to be connected together to provide additional feature compatibility, above what normal Tie Lines provide. The system is configured with the 24-channel CD-CCTA (T-1/CCH) for receiving or transmitting common signaling data from/to a distant office. The system can provide a variety of interoffice service features such as Calling Name display, Centralized Voice Mail Integration, or Link Reconnect.

The following features are provided:

❍   Automatic Recall – K-CCIS on page 4-7

❍   Brokerage Hotline – K-CCIS on page 4-11

❍   Call Forwarding – All Calls – K-CCIS on page 4-13

❍   Call Forwarding – Busy/No Answer – K-CCIS on page 4-17

❍   Call Park Retrieve – K-CCIS on page 4-21

❍   Call Transfer – All Calls – K-CCIS on page 4-27

❍   Calling Name Display – K-CCIS on page 4-31

❍   Calling Number Display – K-CCIS on page 4-33

❍   Calling Party Number (CPN) Presentation from Station – K-CCIS on page 4-35

❍   Centralized Billing – K-CCIS on page 4-39

❍   Centralized BLF (K-CCIS) on page 4-43

❍   Centralized Day/Night Mode Change – K-CCIS on page 4-49

❍   Centralized E911 – K-CCIS on page 4-55

❍   Dial Access to Attendant – K-CCIS on page 4-59

Features and Specifications

❍ Direct Inward Dialing – K-CCIS on page 4-65

❍ Dual Hold – K-CCIS on page 4-67

❍ Elapsed Time Display – K-CCIS on page 4-69

❍ Flexible Numbering of Stations – K-CCIS on page 4-71

❍ Handsfree Answerback – K-CCIS on page 4-75

❍ Hot Line – K-CCIS on page 4-77

❍ Multiple Call Forwarding – All Calls – K-CCIS on page 4-85

❍ Multiple Call Forwarding – Busy/No Answer - K-CCIS on page 4-91

❍ Paging Access – K-CCIS on page 4-97

❍ Quick Transfer to Voice Mail – K-CCIS on page 4-107

❍ Station-to-Station Calling – K-CCIS on page 4-111

❍ Uniform Numbering Plan – K-CCIS on page 4-113

❍ Voice Call – K -CCIS on page 4-117

❍ Voice Mail Integration – K-CCIS on page 4-119

SECTION 2   SYSTEM AVAILABILITY

2.1   Terminal Type

All stations

2.2   Required Components

CD-CCTA

- OR -

PZ-32/64/128IPLA

2.3   Operating Procedures

Normal call handling procedures apply.

The following table shows the chassis system software compatibility with CD-CCTA firmware.

Chassis Software	CD-CCTA
SV8100 V1.00	X

X = Compatible
– = Not compatible

SECTION 3   SERVICE CONDITIONS

General:

❍   Each UNIVERGE SV8100 system can have up to eight K-CCIS routes.

❍   One CD-CCTA is required to support each K-CCIS link. A maximum of eight K-CCIS links are supported.

❍   The Basic Port System can have up to 60 DTI trunks for K-CCIS voice path.

❍   The Unlimited Port System can have up to 96 DTI trunks for K-CCIS Voice path.

❍   The K-CCIS feature shares the CO/PBX/Tie/DID trunks available for the system.

❍   When assigning a Closed Numbering Plan and DID conversion across K-CCIS is required, the UNIVERGE SV8100 uses the ARS/F-Route Tables.

❍   The UNIVERGE SV8100 uses the F-Route Tables to assign an Open Numbering Plan.

❍   When all K-CCIS voice channels are busy, the UNIVERGE SV8100 originator of a K-CCIS call hears a busy tone from the system.

❍   Outgoing CO calls in a K-CCIS network can be routed over the K-CCIS link and use the distant system CO lines.

❍   Distant system extension numbers in the K-CCIS network can be assigned to One Touch keys and Speed Dial buffers.

❍   When a K-CCIS trunk is on hold, the Specified Line Seizure access codes can be used to retrieve the call from its held state.

❍ FoIP (Fax over Internet Protocol) with T.38 protocol is supported only between an SV8100 to SV8100.

Restrictions:

❍ The UNIVERGE SV8100 can support only 1~8-digit station numbers.

❍ Station Numbers are assigned by the 10s group for 4-digit station numbers, 100s group for 5-digit station numbers, 1000s group for 6-digit station numbers, 10000s group for 7-digit station numbers.

❍ When Voice Mail Message Waiting status must be sent across the K-CCIS to a remote system, F-Routes must be used.

❍ For a Closed Numbering Plan network using F-Routes, a maximum of 120 F-Route Tables are available allowing a maximum of 121 connected systems per K-CCIS network.

❍ When a Closed Numbering Plan Network is used, a user can call another station by dialing the distant extension number, but extensions in the network cannot have the same prefix.

❍ For an Open Numbering Plan network, a user can dial another station by dialing the office location number plus an extension number and the extension number can have the same prefix, but the office location cannot be the same.

❍ When an UNIVERGE SV8100 system is a tandem system (in the middle) between systems with higher K-CCIS feature support (including NEAX PBXs), only the K-CCIS features supported by the UNIVERGE SV8100 tandem system are passed through and supported.

❍ An UNIVERGE SV8100 K-CCIS network should never have more than five hops (tandem connections) because of the message delay through each tandem system.

❍ A Star topology network supports up to eight systems.

❍ A Tree topology network is supported. The maximum number of systems depends on the Numbering Plan used and the maximum number of hops (tandem connections).

❍ A Mesh topology network is not supported.

❍ K-CCIS requires assigning a point code for each office. Point codes differentiate between an originating office and a destination office in the K-CCIS Network. Assigning point codes requires the following considerations:

   ❏ The point code must be unique in the network.

❑   When a system has two or more CCH channels, the same originating point code must be assigned to all channels in the system.

❑   The UNIVERGE SV8100 can have a maximum of 255 codes assigned to distant systems.

❍   Using an UNIVERGE SV8100-to-UNIVERGE SV8100 network, centralized voice mail is not supported when an Open Numbering Plan is used.

❍   Centralized E911 – K-CCIS is supported.

❍   When Centralized E911 – K-CCIS is not used, each UNIVERGE SV8100 system in a K-CCIS network must have at least one trunk for Emergency 911 calls.

❍   Using a NEAX-to-UNIVERGE SV8100 network, the PBX must supply centralized voice mail.

❍   Multiline terminals must have an available Call Appearance (CAP) key to originate or answer a K-CCIS trunk call.

❍   Direct access of K-CCIS voice or data channels using Line keys or Specified Line Seizure access codes is prohibited.

❍   The Recall key or Drop key does not function on K-CCIS calls. When either key is pressed, operation is ignored, and the call continues.

❍   Trunk queuing is prohibited on a K-CCIS trunk route.

❍   The ability to route an incoming DID call directly across a K-CCIS link (Direct Inward Dialing – K-CCIS) is supported only when a Closed Numbering Plan using F-Routes is used.

❍   This feature is not supported by the CD-4ODTA Analog Line interface.

❍   Eight CD-CCTA can be assigned per system.

❍   Extension numbers cannot start with 0 or 9.

❍   Internal Calls, transferred calls, and K-CCIS calls do not provide Caller ID to single line telephones.

❍   Caller ID Call Return feature is not supported with K-CCIS calls.

SECTION 4   RELATED FEATURE LIST

 ❍ T1 Connections

 ❍ Universal Slots

SECTION 5   K-CCIS FEATURES

The remainder of this chapter provides detailed information for the available K-CCIS features.

Automatic Recall – K-CCIS

FEATURE DESCRIPTION

This feature allows a call to be release transferred to another station in another office in the K-CCIS network and recall back to the originator of the transfer after a programmed time.

SYSTEM AVAILABILITY

All Terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

Using a multiline terminal with a call in progress (Closed Numbering Plan):

1. Press Transfer. Internal dial tone is heard. The call is placed on Non-Exclusive Hold.

2. Dial the distant K-CCIS station number where the call is to be transferred.

3. Wait for the ringback tone.

4. Hang up.

- OR -

1. When the party answers, announce the transfer.

2. Restore the handset (transfer is completed).

Using a multiline terminal with a call in progress (Open Numbering Plan):

1.   Press Transfer, and receive internal dial tone. The call is placed on
      Non-Exclusive Hold.

2.   Dial the trunk Access Code (normally 8).

3.   Dial the Office Code number.

4.   Dial the distant K-CCIS station number where the call is to be transferred.

5.   Wait for the ringback tone.

6.   Hang up.

SERVICE CONDITIONS

❍   If PRG 34-07-05 is left at default (30) the transferred call recalls to the station
     that performed the transfer when not answered.

❍   A UNIVERGE SV8100 station can receive a K-CCIS transferred call as a
     camp-on call if allowed by Class of Service.

Restrictions:

❍   PRG 34-07-05 cannot be set based on a Timer Class of Service in PRG 20-31.

❍   Trunk-to-Trunk Transfer must be allowed in Program 14-01-13 (Trunk-to-Trunk
     Transfer Yes/No Selection).

❍   A blind transfer across a K-CCIS link cannot be completed until ringback tone is
     received at the transferring station.

RELATED FEATURE LIST

❍   Link Reconnect – K-CCIS

❍   Station-to-Station Calling – K-CCIS

❍   Uniform Numbering Plan – K-CCIS

❍   Call Transfer – All Calls – K-CCIS

GUIDE TO FEATURE PROGRAMMING

Program/ Item No.	Description/Selection	Assigned Data	Comments
34-07-05	E&M Tie Line Timer – Trunk Answer Detect Timer for E&M	0 ~ 64800 seconds Default is 30	Determine the amount of time the call should ring a station in another office before recalling back to the originator of the transfer.

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Brokerage Hotline – K-CCIS

FEATURE DESCRIPTION

This feature provides a ringdown connection between two stations, each using a multiline terminal, in different offices in the CCIS network.

SYSTEM AVAILABILITY

All Terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

To use this feature at any terminal:

1.   Lift the handset or press **Speaker**.

2.   Press the line/feature key associated with the pre-assigned station.

      The destination station is automatically dialed, ring back tone is heard and the destination station answers.

3.   After completion of conversation, hang up or press **Speaker**

      To make another Brokerage-Hot Line-CCIS call immediately, press another line/ feature key without going on hook and off hook.

SERVICE CONDITIONS

❍   Either multiline terminal in a Brokerage - Hot Line – (K-CCIS) pair may transfer a Hot Line call to another station in the K-CCIS network using the Call Transfer – All Calls - (K-CCIS) feature.

Restrictions:

None

RELATED FEATURE LIST

❍   Call Transfer – All Calls - K-CCIS

❍   Station-to-Station Calling – K-CCIS

❍   Uniform Numbering Plan – K-CCIS

GUIDE TO FEATURE PROGRAMMING

Program/ Item No.	Description/Selection	Assigned Data	Comments
15-07-01	Programmable Function Keys	01 = DSS / One-Touch	Assign a DSS/One Touch key of the station in the Distant Office.

Call Forwarding – All Calls - K-CCIS

FEATURE DESCRIPTION

This feature allows all calls destined for a particular station to be routed to another station or to an Attendant, in another office in the K-CCIS network, regardless of the status (busy or idle) of the called station. The activation and cancellation of this feature may be accomplished by either the station user or an Attendant position if allowed by Class of Service (COS). Attendant Positions can be used to cancel Call Forward – All Call system-wide.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

To set Call Forward – All Calls – K-CCIS from a multiline telephone (Closed Numbering plan):

1.   Press the Call Forward – All ON/OFF key.

2.   Dial 1 to set, then enter the remote K-CCIS station number.

3.   Press Speaker.

- OR -

1.   Lift the handset or press Speaker.

2.   Dial Access Code 741 (default), then 1 to set.

3.   Dial the remote K-CCIS station number.

4.   Restore handset or press Speaker.

To set Call Forward – All Calls – K-CCIS from a Multiline Telephone (Open Numbering Plan):

---

1.   Press the Call Forward – All ON/OFF key.

2.   Dial 1 to set.

3.   Dial the trunk Access Code (normally 8).

4.   Dial the Office Code number.

5.   Dial the distant K-CCIS station number.

6.   Restore handset or press Speaker.

- OR -

1.   Lift the handset or press Speaker.

2.   Dial Access Code 741 (default), then 1 to set.

3.   Dial the trunk Access Code (normally 8).

4.   Dial the Office Code number.

5.   Dial the distant K-CCIS station number.

6.   Restore handset or press Speaker.

To cancel Call Forward – All Calls – K-CCIS from a Multiline Telephone:

---

1.   Press Call Forward – All ON/OFF key.

2.   Dial 0 to cancel.

3.   Press Speaker.

- OR -

1. Lift the handset or press Speaker.

2. Dial Access Code 741(default), then dial 0.

3. Restore the handset or press Speaker.

SERVICE CONDITIONS

General:

❍   Any station or Call Arrival (CAR) key can be set for Call Forwarding – All Calls – K-CCIS.

Restrictions:

❍   Call Forward – Off-Premise must be allowed in PRG 20-11-12 (Class of Service External Call Forward) to set call forwarding to a remote K-CCIS station number.

❍   Trunk-to-Trunk Transfer must be allowed in PRG 14-01-13 (Trunk-to-Trunk Transfer Yes/No Selection).

❍   A Single Line Telephone user can transfer a trunk call to another internal station that is set for Call Forwarding – All Calls – K-CCIS, however, when the distant party answers the call, a conference cannot be established.

❍   The destination station in the distant system is the only station that can call a station with Call Forwarding – All Calls – K-CCIS set.

❍   Call Forwarding with Both Ringing (All Calls) is not supported.

❍   Call Forward Split Internal/External is not supported.

❍   Forwarding to Voice Mail is not Included in the Maximum Hop Count.

❍   Call Forward continues to operate to a D^{term} that has been removed.

RELATED FEATURE LIST

❍   Call Forwarding – Busy/No Answer – K-CCIS

❍   Multiple Call Forwarding – All Calls – K-CCIS

❍   Multiple Call Forwarding – Busy/No Answer – K-CCIS

❍   Link Reconnect – K-CCIS

❍   Uniform Numbering Plan – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
20-11-12	Class of Service Options (Hold Transfer Service) – Call Forwarding Off-Premise (External Call Forwarding	0 = Off 1 = On Default is 0	To enable per class of service.
20-06-01	Class of Service for Extensions	0~15	Ext.101 is in Class 15. All others are in Class 1.
20-09-07	Class of Service Options (Incoming Call Service) – Call Queuing	0 = Off 1 = On Default is 1	Must be Off for Call Forward – Busy to operate.
20-13-06	Class of Service Options (Supplementary Service) – Automatic Off Hook Signaling (Automatic Override)	0 = Off 1 = On	Must be Off for Call Forward – Busy to operate.
15-07-01	Programmable Function Keys	10 = Call Forward – Immediate 11 = Call Forward – Busy 12 = Call Forward – No Answer 13 = Call Forward – Busy No Answer	Service Codes: 741 742 743 744
14-01-13	Basic Trunk Data Setup – Trunk-to-Trunk Transfer Loop Supervision	0 = Disable 1 = Enable Default is 1	Must be enabled for Trunk-to-Trunk Transfer, Call Forward – Off-premise, and tandem trunking.
50-06-01	CCIS Feature Availability – Link Reconnect	0 = Not Available 1 = Available Default is 1	To enable or disable Link Reconnect.
50-05-01	CCIS Maximum Call Forwarding Hop Counter	0~7 Hops Default is 5	Sets Maximum Hops allowed in a CCIS network.

Call Forwarding – Busy/No Answer – K-CCIS

FEATURE DESCRIPTION

This feature permits a call to a Busy or unanswered station to be forwarded to another station or an Attendant, in another office in the K-CCIS network. The activation and cancellation of this feature may be accomplished by either the station user or an Attendant position, if allowed by Class of Service (COS).

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

To set Call Forward – Busy/No Answer – K-CCIS from a Multiline Telephone (Closed Numbering Plan):

1. Press the Call Forward – Busy/No Answer ON/OFF key.

2. Dial 1 to set.

3. Dial the remote K-CCIS station number.

4. Press Speaker.

- OR -

1. Lift the handset or press Speaker.

2. Dial Access Code 744 (default), then 1 to set.

3. Dial the remote K-CCIS station number.

4.   Restore handset or press Speaker.

To set Call Forward – Busy/No Answer - K-CCIS from a Multiline Telephone
(Open Numbering Plan):

1.   Press the Call Forward – Busy/No Answer ON/OFF key.

2.   Dial 1 to set.

3.   Dial the trunk Access Code (normally 8).

4.   Dial the Office Code number.

5.   Dial the distant K-CCIS station number.

6.   Press Speaker.

- OR -

1.   Lift the handset or press Speaker.

2.   Dial Access Code 744 (default), then Dial 1 to set.

3.   Dial the trunk Access Code (normally 8).

4.   Dial the Office Code number.

5.   Dial the distant K-CCIS station number.

6.   Restore handset or press Speaker.

To cancel Call Forward – Busy/No Answer – K-CCIS from a Multiline Telephone:

1.   Press Call Forward – Busy/No Answer On/Off key.

2.   Dial 0 to cancel.

3.   Press Speaker.

- OR -

1.   Lift the handset or press Speaker.

2.   Dial Access Code 744 (default), then Dial 0 to cancel.

3.   Restore handset or press Speaker.

SERVICE CONDITIONS

General:

❍   Any station or Call Arrival (CAR) key can be set for Call Forwarding – Busy/No Answer – K-CCIS.

Restrictions:

❍   Call Forward – Off-Premise must be allowed in Class of Service (PRG 20-11-12) External Call forward to set call forwarding to a remote K-CCIS station number.

❍   Trunk-to-Trunk Transfer must be allowed in PRG 14-01-13 for each trunk (Trunk-to-Trunk Transfer Yes/No Selection).

❍   A Single Line Telephone user can transfer a trunk call to another internal station that is set for Call Forwarding – All Calls - K-CCIS, however, when the distant party answers the call, a conference cannot be established.

❍   The destination station in the distant systems is the only station that can call a station with Call Forwarding – All Calls – K-CCIS set.

❍   Call Forwarding with Both Ringing (All Calls) is not supported.

❍   Call Forward Split Internal/External is not supported.

❍   Forwarding to Voice Mail is not Included in the Maximum Hop Count.

❍   Call Forward continues to operate to a D^{term} that has been removed.

RELATED FEATURE LIST

❍   Call Forwarding – Busy/No Answer – K-CCIS

❍   Multiple Call Forwarding – All Calls – K-CCIS

❍   Multiple Call Forwarding – Busy/No Answer – K-CCIS

❍   Link Reconnect – K-CCIS

❍   Uniform Numbering Plan – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
20-11-12	Class of Service Options – Call Forwarding Off-Premise (External Call Forwarding)	0 = Off (default) 1 = On Default is 0	Enable per class of service.
20-06-01	Class of Service for Extensions	0~15	Ext.101 is in Class 15. All others are in Class 1.
20-09-07	Class of Service Options (Incoming Call Service) – Call Queuing	0 = Off 1 = On Default is 1	Must be Off for Call Forward – Busy to operate.
20-13-06	Class of Service Options (Supplementary Service) – Automatic Off Hook Signaling (Automatic Override)	0 = Off 1 = On Default is 1	Must be Off for Call Forward – Busy to operate.
15-07-01	Programmable Function Keys	10 = Call Forward – Immediate 11 = Call Forward – Busy 12 = Call Forward – No Answer 13 = Call Forward – Busy No Answer	Service Codes: 741 742 743 744
14-01-13	Basic Trunk Data Setup – Trunk-to-Trunk Transfer Loop Supervision	0 = Disable 1 = Enable Default is 1	Must enable for Trunk-to-Trunk Transfer, Call Forward – Off-Premise, and tandem trunking.
50-06-01	CCIS Feature Availability – Link Reconnect	0 = Not Available 1 = Available Default is 1	Enable/Disable Link Reconnect.
50-05-01	CCIS Maximum Call Forwarding Hop Counter	0~7 Hops Default is 5	Set Maximum Hops allowed in a CCIS network.

Call Park Retrieve – K-CCIS

FEATURE DESCRIPTION

This feature allows a station user to retrieve Parked Calls at remote sites across K-CCIS. Locally parked calls can be retrieved from a remote system, connected via K-CCIS, by dialing the Call Park Hold Group Number, plus the park orbit location.

SYSTEM AVAILABILITY

Terminal Type:

All Terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

Retrieve Park Call at remote location:

1. Go off-hook, and wait for internal dial tone.

2. Dial the Call Park Retrieve Access Code *6 (default) + Park Hold Group Number (1~64) + Park Orbit Number (1~64) of the call to be retrieved. Call Park Retrieve access codes cannot be the same at all locations in the K-CCIS Network.

3. Talk with party.

SERVICE CONDITIONS

General:

❍   A different Call Park Retrieve Access Code must be programmed for each system in the K-CCIS network.

❍   The Park Group Number and Park Orbit Number must be dialed immediately following the Park Retrieve Service Code.

❍   When two or more stations attempt to retrieve the parked call, only one station can retrieve the call.

❍   A station connected to a PBX can retrieve a parked call in an UNIVERGE SV8100, but the station connected to the UNIVERGE SV8100 system cannot retrieve a parked call in a PBX.

❍   A Park Hold key cannot be used to retrieve a parked call from a distant system.

❍   F-Routes are required to route Call Park Retrieve Access Code to proper system in the K-CCIS network.

❍   When a remote location retrieves a call from another location, the call is treated as if it were transferred from the distant location.

❍   SMDR reports the retrieved call from the distant location as if it were a transferred call.

❍   When a call that has Caller ID Information is retrieved at the distant location the Caller ID information is treated as if it were a transferred call.

❍   Link Reconnect operates when the trunk is retrieved back to the origination system.

Restrictions:

❍   A Call cannot be placed into remote systems Call Park Location.

❍   Call Park Retrieve – K-CCIS is only a Key System-to-Key System supported feature.

❍   The digit (# or *) cannot be used in conjunction with IP K-CCIS.

❍   When the UNIVERGE SV8100 is connected to the Electra Elite IPK II, the maximum digits assignment in the UNIVERGE SV8100 is determined by Program 44-05-09.

❍   Call Park Searching is supported in the local system only.

RELATED FEATURE LIST

❍   Call Park – System

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Park Originate System

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-12-32	Answer for Park	*6 (default)	This varies based on the K-CCIS network configuration and PRG 11-01-01.
20-14-12	Class of Service for DISA/ E&M – Retrieve Park Hold	0 = Off (default) 1 = On	Enable Retrieve Park Hold feature per Class of Service.
24-03-01	Park Group	01~64 1 = Default	Assigns an extension to a Park Group.

Remote System (Call Park Retrieve)

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-01-01	System Numbering		
11-12-32	Service Code Setup (for Service Access) – Answer for Park Hold	Default is ✳6	
20-14-12	Class of Service Options for DISA/E&M – Retrieve Park Hold	0 = Off 1 = On Default is 1	Enable Retrieve Park Hold feature per Class of Service.
24-03-01	Park Group	01~64 Default is 1	Assigns an extension to a Park Group.
44-02-01	Dial Analysis Table for ARS/F-Route Access – Dial	Analysis Table 1 = *6 Default is No Setting	Assigns access code used to retrieve the parked call.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
44-02-02	Dial Analysis Table for ARS/F-Route Access – Service Type	0 = No setting (None) 1 = Extension Call (Own) 2 = ARS/F-Route Table (F-Route) 3 = Dial Extension. Analyze Table (Option) Default is 2	To assign the service type used.
44-02-03	Dial Analysis Table for ARS/F-Route Access – Additional Data	0 = No setting 1 = Delete Digits = 0~255 (255 = delete all digits) 2 = 0~500 3 = Dial Extension Analyze Table Number = 0~4 Default is 0	Enter additional data required for the Service Type selected in PRG 44-02-02.
44-05-01	ARS/F-Route Table – Trunk Group Number	Trunk Groups (1~254)	Select trunk group number used for outgoing ARS calls. Setting of 255 = Internal Extension Call.
44-05-02	ARS/F-Route Table – Delete digits	0 = No setting 1~254 255 = Delete all Digits Default is 0	Enter number of digits to delete from the dialed number.
44-05-03	ARS/F-Route Table – Additional Dial Number Table	0 = No setting 1~1000 Default is 0	Enter table number (defined in 44-06) for additional digits to be dialed.
44-05-09	ARS/F-Route Table – Max Digit	0 = No Max 1~24 Default is 0	Assign Max digits for the Call Park Retrieve Access Code.

Programming Example:

For the following example, to retrieve a call which is parked, use the following access codes from any system:
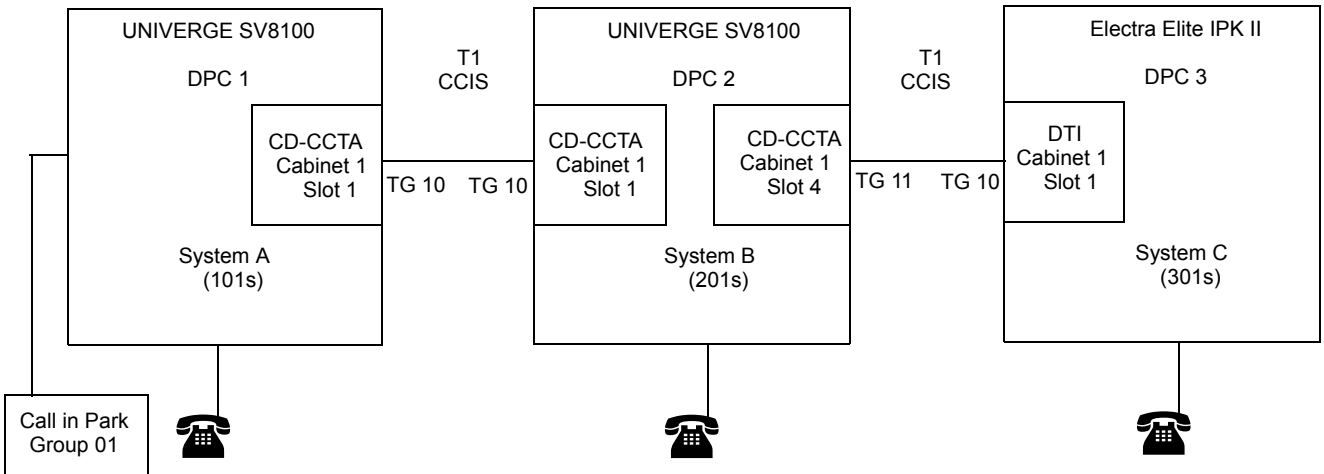
Call Parked At	Call Park Retrieve Access Codes
System A	501+01
System B	502+01+01
System C	503+01+01

System A (101s)	System B (201s)	System C (301s)
PRG 11-01-01 50 = 2 digits; F-Route	PRGPRG 11-01-01 50 = 2 digits; F-Route	PRG 11-01-01 50 = 2 digits; F-Route
PRG 11-12-32 Ans Park Hold = *6 (Park Retrieve System A)	PRG 11-12-32 Ans Park Hold = *6 (Park Retrieve System B)	PRG 11-12-32 Ans Park Hold = *6 (Park Retrieve System C)
PRG 44-02-01 Table 1 = Dial 501 (Park Retrieve System A)	PRG 44-02-01 Table 1 =Dial 502 (Park Retrieve System B)	PRG 44-02-01 Table 1 =Dial 503 (Park Retrieve System C)
PRG 44-02-01 Table 2 = Dial 502 (ARS Table #4) (Park Retrieve System B)	PRG 44-02-01 Table 2 = Dial 501 (Park Retrieve System A)	PRG 44-02-01 Table 2 = Dial 501 (Park Retrieve System A)
PRG 44-02-01 Table 3 = Dial 503 (Park Retrieve System C)	PRG 44-02-01 Table 3 = Dial 503 (Park Retrieve System C)	PRG 44-02-01 Table 3 = Dial 502 (Park Retrieve System B)
PRG 44-02-02 Table 1 = F-Route Table 2 = F-Route Table 3 = F-Route	PRG 44-02-02 Table 1 = F-Route Table 2 = F-Route Table 3 = F-Route	PRG 44-02-02 Table 1 = F-Route Table 2 = F-Route Table 3 = F-Route
PRG 44-02-03 Table 1 = F-Route 1 Table 2 = F-Route 2 Table 3 = F-Route 3	PRG 44-02-03 Table 1 = F-Route 1 Table 2 = F-Route 2 Table 3 = F-Route 3	PRG 44-02-03 Table 1 = F-Route 1 Table 2 = F-Route 2 Table 3 = F-Route 3
PRG 44-05-01 Table 1 = TRK GP 255 Table 2 = TRK GP 10 Table 3 = TRK GP 10	PRG 44-05-01 Table 1 = TRK GP 255 Table 2 = TRK GP 10 Table 3 = TRK GP 11	PRG 44-05-01 Table 1 = TRK GP 255 Table 2 = TRK GP 10 Table 3 = TRK GP 10
PRG 44-05-02 Table 1 = Delete Digits 3 Table 2 = Delete Digits 0 Table 3 = Delete Digits 0	PRG 44-05-02 Table 1 = Delete Digits 3 Table 2 = Delete Digits 0 Table 3 = Delete Digits 0	PRG 44-05-02 Table 1 = Delete Digits 3 Table 2 = Delete Digits 0 Table 3 = Delete Digits 0
PRG 44-05-03 Table 1 = Add Dial 1 Table 2 = Add Dial 0 Table 3 = Add Dial 0	PRG 44-05-03 Table 1 = Add Dial 1 Table 2 = Add Dial 0 Table 3 = Add Dial 0	PRG 44-05-03 Table 1 = Add Dial 1 Table 2 = Add Dial 0 Table 3 = Add Dial 0
PRG 44-05-09 Table 1 = Max Digit 0 Table 2 = Max Digit 7 Table 3 = Max Digit 7	PRG 44-05-09 Table 1 = Max Digit 0 Table 2 = Max Digit 7 Table 3 = Max Digit 7	PRG 44-05-09 Table 1 = Max Digit 0 Table 2 = Max Digit 7 Table 3 = Max Digit 7
	* PRG 44-05-10 Table 2 = DPC Table1 = 1 Table 2 = DPC Table2 = 3	* PRG 44-05-10 Table 2 = DPC Table1 = 1 Table 2 = DPC Table2 = 2
PRG 44-06-01 Table 1 = Dial *6	PRG 44-06-01 Table 1 = Dial *6	PRG 44-06-01 Table 1 = Dial *6

* For CCISoIP Programming only.

Call Transfer – All Calls – K-CCIS

FEATURE DESCRIPTION

This feature allows a station user to transfer incoming or outgoing Central Office, intraoffice, and interoffice calls to another station in the K-CCIS network without Attendant assistance.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

Using a Multiline Terminal with a call in progress (Closed Numbering Plan):

1. Press Transfer, and internal dial tone is heard. The call is placed on Non-Exclusive Hold.

2. Dial the distant K-CCIS station number where the call is to be transferred.

3. Wait for the ringback tone.

4. Hang up.

- OR -

1. When the party answers, announce the transfer.

2. Restore the handset (transfer is completed).

Using a Multiline Terminal with a call in progress (Open Numbering Plan):

1.   Press Transfer, and receive internal dial tone. The call is placed on Non-Exclusive Hold.

2.   Dial the trunk Access Code (normally 8).

3.   Dial the Office Code number.

4.   Dial the distant K-CCIS station number where the call is to be transferred.

5.   Wait for the ringback tone.

6.   Hang up.

- OR -

1.   When the party answers, announce the transfer.

2.   Restore the handset (transfer is completed).

SERVICE CONDITIONS

General:

❍   A UNIVERGE SV8100 station can receive a K-CCIS transferred call as a camp-on call if allowed by Class of Service.

Restrictions:

❍   Trunk-to-Trunk Transfer must be allowed in Program 14-01-13 (Trunk-to-Trunk Transfer Yes/No Selection).

❍   A blind transfer across a K-CCIS link cannot be completed until ringback tone is received at the transferring station.

RELATED FEATURE LIST

❍   Link Reconnect – K-CCIS

❍   Station-to-Station Calling – K-CCIS

❍   Uniform Numbering Plan – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
20-09-07	Class of Service Options (Incoming Call Service) – Call Queuing	0 = Off 1 = On Default is 1	To enable receiving Call Queuing.
20-06-01	Class of Service for Extensions	0~ 15	Extension. 101 default is Class 15. All other extensions are in Class 1.
20-09-01	Class of Service Options (Incoming Call Service) – Second Call for DID/DISA/ DIL/E&M Override	0 = Off 1 = On Default is 1	Turn Off or On the extension ability to receive the second call.
20-13-06	Class of Service Options (Supplementary Service) – Automatic Off Hook Signaling (Automatic Override)	0 = Manually 1 = Automatically Default is 1	Allow a busy extension user to manually or automatically receive off-hook signals.
14-01-13	Basic Trunk Date setup – Trunk-to-Trunk Transfer Loop Supervision	0 = Disable 1 = Enable Default is 1	Must enable for Trunk-to-Trunk Transfer, Call-Forward – Off-Premise, or tandem trunking.
50-06-01	CCIS Feature Availability – Link Reconnect	0 = Not Available 1 = Enable Default is 1	Enable Link Reconnect.

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Calling Name Display – K-CCIS

FEATURE DESCRIPTION

This feature permits the station name of a calling or called party at another switching office to be displayed on a multiline terminal, through the K-CCIS network.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

Normal call handling procedures apply.

SERVICE CONDITIONS

General:

❍   Both the caller/calling station number name and number can be displayed on an UNIVERGE SV8100 station if allowed by Class of Service.

❍   For incoming or outgoing K-CCIS calls, the Calling/Called Name and Number are displayed for the entire length of the call including the Elapsed Call Time.

RESTRICTIONS:

❍   In the UNIVERGE SV8100 system, only 12 digits/characters can be entered for each station name.

RELATED FEATURE LIST

❍   Calling Number Display – K-CCIS

❍   Station-to-Station Calling – K-CCIS

❍   Uniform Numbering Plan – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
14-02-10	Analog Trunk Date Setup – Caller ID	0 = No 1 = Yes Default is 0	Enable/Disable a trunk ability to receive Caller ID.
20-09-02	Class of Service Options (Incoming Call Service) – Caller ID Display	0 = Off 1 = On Default is 0	Control the Caller ID Display at an extension.
20-09-08	Class of Service Options (Incoming Call Service) – Calling Party Information	0 = Off 1 = On Default is 1	Enable receiving Calling Party Information from K-CCIS.
20-06-01	Class of Service for Extensions	0~15	Extension 101 default is Class 15. All other extensions are in Class 1.
50-02-05	Connecting System Settings – Calling Name Indication (T1)	0 = Disable 1 = Enable Default is 1	Enable receiving Calling Name indication from K-CCIS.
15-01-01	Basic Extension Data Setup – Extension Name	Up to 12 Characters Default: Sta 101 = Ext 101 Sta 102 = Ext 102 etc.	Set the extension/Virtual extension name.

Calling Number Display – K-CCIS

FEATURE DESCRIPTION

This feature permits the number of a calling or called party at another switching office, to be displayed on a multiline terminal through the K-CCIS network.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

Normal call handling procedures apply.

SERVICE CONDITIONS

General:

❍   Both the caller/calling station number and name can be displayed on an UNIVERGE SV8100 station if allowed by Class of Service.

❍   For incoming or outgoing K-CCIS calls, the Calling/Called Name and Number are displayed for the entire length of the call including the Elapsed Call Time.

❍   For an open numbering plan the Office Code number and station number are displayed for caller/calling station number.

Restrictions:

❍ The UNIVERGE SV8100 supports 2~8-Digit station numbers.

❍ When calling over a K-CCIS tandem connection, the calling party number (CPN) is transferred to the ISDN network.

RELATED FEATURE LIST

❍ Calling Name Display – K-CCIS

❍ Station-to-Station Calling – K-CCIS

❍ Uniform Numbering Plan – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
14-02-10	Analog Trunk Date Setup – Caller ID	0 = Disable (No) 1 = Enable (Yes) Default is 0	Enable/Disable a trunk ability to receive Caller ID.
20-09-02	Class of Service Options (Incoming Call Service) – Caller ID Display	0 = Off 1 = On Default is 0	Control the Caller ID Display at an extension.
20-09-08	Class of Service Options (Incoming Call Service) – Calling Party Information	0 = Off 1 = On Default is 1	Enable receiving Calling Party Information from K-CCIS.
20-06-01	Class of Service for Extensions	0~15	Extension 101 default is Class 15. All other extensions are in Class 1.
50-02-05	Connecting System Settings – Calling Name Indication (T1)	0 = Disable 1 = Enable Default is 1	Enable receiving Calling Name indication from K-CCIS.
15-01-01	Basic Extension Data Setup – Extension Name	Up to 12 Characters Default: Sta 101 = Ext 101 Sta 102 = Ext 102 etc.	Set the extension/Virtual extension name.

Calling Party Number (CPN) Presentation from Station – K-CCIS

FEATURE DESCRIPTION

Calling Party Number (CPN) Presentation from Station K-CCIS feature allows each station of the remote systems a unique 10-digit number (the DID number of the originating station) to be sent out over the PRI circuit of the main system.

SYSTEM AVAILABILITY

Terminal Type:

All Terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

Placing a call with CPN:

1.   Lift the handset or press Speaker.

2.   Dial 9 + the number.

3.   Converse with caller.

4.   Hang up the handset.

SERVICE CONDITIONS

Restrictions:

❍ A maximum of 16 digits can be assigned as the Calling Party Number (CPN) in Program 21-12-01 and Program 21-13-01.

❍ The PRI provider must provision for the CPN used for E911. The CPN must be within the allowable range. For more information please contact your local ISDN provider regarding allowable ranges.

❍ The Calling Party Number (CPN) is sent only to the network when the calling party from the remote system dials a trunk access code of *9* when making an outbound call.

❍ The Calling Party Number (CPN) is not sent to the network when the originating station of the remote system calls a station in the main system that is call forwarded off site.

RELATED FEATURE LIST

❍ ISDN Compatibility

❍ Automatic Route Selection

❍ Central Office Calls, Placing

❍ T1 Trunking (with ANI/DNIS Compatibility)

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
14-01-24	Basic Trunk Data Setup – Trunk-to-Trunk Outgoing Caller ID through Mode	0 = Off 1 = On Default is 0	Enable Outgoing Caller ID through Mode for each CCIS trunk to enable CPN information to pass through the Tandem Office. Tandem System Only
20-08-13	Class of Service Options (Outgoing Call Service) – ISDN CLIP	0 = Off 1 = On Default is 0	Determine if the ISDN calling line identity presentation and screening indicators are to be allowed.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
21-12-01	ISDN Calling Party Number Setup for Trunks	Up to 16 digits max. Default is No Setting	Assign each trunk a Calling Party Number. When a call is made by an extension which does not have an Extension Calling Number assigned (Program 21-13), the system sends the calling number for the ISDN trunk defined in 21-12.
21-13-01	ISDN Calling Party Number Setup for Extensions	Up to 16 digits max. Default is No Setting	Assign each extension a Calling Party Number.

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Centralized Billing – K-CCIS

FEATURE DESCRIPTION

This feature sends the billing information from local systems to a billing center office for central management of all billing information in the network. The UNIVERGE SV8100 can send billing information to a billing center office (NEAX2000/2400), but cannot receive the billing information as the billing center office.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

Not Applicable

SERVICE CONDITIONS

General:

❍   The Station Message Detail Recording (SMDR) feature and Centralized Billing Feature can be used at the same time.

❍   Centralized Billing – K-CCIS feature supports the following calls:

❑   Incoming CO Calls (using the main system and another system trunk/ K-CCIS trunk)

❑   Outgoing CO Call (using the main system and another system trunk/ K-CCIS trunk)

❏   Call Transfer of CO calls (using the main system and another system trunk/K-CCIS trunk)

❏   Conference Calls (using the main system and another system trunk/ K-CCIS trunk)

Restrictions:

❍   In a K-CCIS network, the PBX must be the main system where billing information is sent. Centralized billing cannot be used in a KTS-to-KTS network.

❍   Station-to-station calls in their own system are not reported to the billing center office with UNIVERGE SV8100.

❍   The information storage capacity of the local UNIVERGE SV8100 office is approximately 300 calls. If the K-CCIS link is down due to network trouble, the billing information is stored by the CD-CP00-US. When the maximum calls exceed this amount, the oldest call information is overwritten by the latest (newest) call.

❍   With the UNIVERGE SV8100, trunk type (e.g., analog or ISDN) information is not reported to the billing center office.

❍   When the K-CCIS link is down due to network trouble, the UNIVERGE SV8100 system does not provide an SMDR alarm indication.

❍   Account Codes cannot exceed 10 digits.

RELATED FEATURE LIST

❍   Account Code – Forced/Verified/Unverified

❍   Account Code Entry

❍   Authorization Code

❍   Centralized Day/Night Mode Change – K-CCIS

❍   Station Message Detail Recording (SMDR)

❍   Voice Mail Integration – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

For Centralized Billing Installation

Program/ Item No.	Description/ Selection	Assigned Data	Comments
14-01-24	Basic Trunk Data Setup – Trunk-to-Trunk Outgoing Caller ID through Mode	0 = Off 1 = On Default is 0	Enable Outgoing Caller ID through Mode for each CCIS trunk to enable CPN information to pass through the Tandem Office. Tandem System Only
20-08-13	Class of Service Options (Outgoing Call Service) – ISDN CLIP	0 = Off (Default) 1 = On Default is 0	Determine if the ISDN calling line identity presentation and screening indicators are to be allowed.
21-12-01	ISDN Calling Party Number Setup for Trunks	Up to 16 digits max. Default is No Setting	Assign each trunk a Calling Party Number. When a call is made by an extension which does not have an Extension Calling Number assigned (Program 21-13), the system sends the calling number for the ISDN trunk defined in 21-12.
21-13-01	ISDN Calling Party Number Setup for Extensions	Up to 16 digits max. Default is No Setting	Assign each extension a Calling Party Number.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
50-01-01	CCIS System Setting – CCIS Availability	0 = Disable 1 = Enable Default is 0	All CCIS settings lose functionality when 0 is selected.
50-03-01	CCIS Destination System Settings – Destination Point Code	0~16367 Default is 0	Assign the destination transfer point code for Tandem KTS.
50-07-01	CCIS Centralized Billing Center Office – Destination Point Code	0~16367 Default is 0	Assign point code for the Centralized Billing Office PBX.

For Station Message Detail Recording (SMDR)

Program/ Item No.	Description/ Selection	Assigned Data	Comments
21-01-03	System Options for Outgoing Calls – Trunk Interdigit Time (External)	0~64800 seconds Default is 5	The system waits for this time to expire before placing the call.
35-01-04	SMDR Options – Omit Digits	0 = Not Applied 1~24 Default is 0	The number of leading digits that do not print on the SMDR report.

For Centralized Billing K-CCIS

Program/ Item No.	Description/ Selection	Assigned Data	Comments
35-01-06	SMDR Options – Minimum Call Duration	0 = All 0~65535 seconds Default is 0	A call must be longer than this duration to be included in the SMDR report.
35-02-08	SMDR Output Options – Incoming Call	0 = Not Displayed 1 = Displayed Default is 1	Select whether or not incoming calls are displayed on the SMDR report.
15-01-03	Basic Extension Data Setup – SMDR Printout	0 = Not printed on SMDR report 1 = Included on SMDR Report Default is 1	Use to include the extension being programmed in the SMDR report.

Centralized BLF (K-CCIS)

FEATURE DESCRIPTION

This feature provides a busy indication for another station across the K-CCIS network on programmed Direct Station Selection/Busy Lamp Field (DSS/BLF) keys. The busy indication is a red LED associated with a Feature Access or One-Touch key programmed for Centralized BLF (K-CCIS). Pressing the Centralized DSS/BLF key allows direct access to the station through the K-CCIS network. Do Not Disturb and Voice Mail Message Waiting on Line key indication are also supported.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Attendant Add-On Console

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

To program a Feature Access or One-Touch key for Centralized DSS/BLF:

1.   Press Speaker.

2.   Dial 751.

3.   Press the key to be programmed.

4.   Dial 01 to assign a DSS/One Touch Key.

5.   Dial the station number.

6.   Press Hold.

Using a Feature Access or a One-Touch key programmed for Centralized DSS/BLF:

1.   Press the programmed Feature Access or One-Touch key. Hear ringback
      tone.

2.   When the called party answers, lift the handset or talk using handsfree if
      allowed.

SERVICE CONDITIONS

General:

❍   Voice Mail Message Waiting on Line Key indication is supported for Centralized
     DSS/BLF keys if PRG 20-13-41 [Class of Service Options (Supplementary
     Service)] – VM Message Indication on DSS/BLF key (VMS Message Indication)
     is allowed.

❍   If Voice Mail Message Waiting on Line Key indication is allowed and a VM
     Message Waiting indication is provided (a new message is stored), pressing the
     Centralized DSS/BLF key performs the following;

     ❏   At system with Voice Mail installed, the user is logged into the owner mail
          box.

     ❏   At remote systems, the station is called.

BLF Sending Service Conditions:

❍   The maximum number of destination offices for sending BLF messages is eight
     per system.

❍   Up to 120 Extension Numbers (entries into the table) can be assigned for
     sending BLF messages. With each assigned Extension Number, up to eight
     destination offices can be selected until a maximum of 240 total sending
     Extension Numbers are assigned.

❍   A maximum of 240 total sending Extension Numbers (BLF messages) can be
     assigned. If 30 Extension Numbers (entries into the table) are assigned with
     each set for all eight groups (systems), the 240 limit is reached and no more
     Extension Numbers can be entered.

❍   The BLF messages are sent in a four-second cycle (at default), so some delay
     occurs to change the indication in the destination office. In the network
     configured with two systems, it can take about four to five seconds (at default) to
     change the BLF indication in the destination office.

❍ This feature is provided with DSS/One-Touch Keys on multiline terminals and with an Attendant Add-On Console.

❍ When the button on the Attendant Add-On Console has a Mail Box button of a remote user programmed and the button is pressed the call is placed to the station on the remote side.

BLF Receiving Service Conditions:

❍ BLF information can be received for up to 120 remote extensions per system.

❍ All multiline terminals in the system can assign Centralized DSS/BLF keys for the supported remote extensions.

❍ The LED indication of the DSS/BLF button on a multiline terminal is as follows:

❑ Idle – No lamp indication

❑ Busy – Steady red lamp

❑ Do Not Disturb – Flashing red lamp

❑ VM Message Waiting – Fast flashing red lamp

✎ The LED indication of the DSS/BLF button on the Attendant Add-on Console is as follows:

❍ Idle   No lamp indication

❍ Busy   Steady red lamp

❍ Do Not Disturb   Flashing red lamp

❍ VM Message Waiting   Fast flashing red lamp

❍ The BLF Information expels when data cannot be sent because of link disconnect. Status changes of BLF information while the system could not send data are not indicated on restoration.

✎ The Voice Mail MSG Waiting has priority over any other state of the flashing line key or One-Touch key.

Restrictions:

❍ This feature is not supported between UNIVERGE SV8100 and NEAX PBXs.

❍ This feature is supported with a Closed Numbering Plan only (not available with an Open Numbering plan).

❍ The same extension line from a remote site can be assigned to multiple DSS/One Touch keys.

❍ The BLF information is expelled when data cannot be sent if the K-CCIS link is down. The UNIVERGE SV8100 does not send BLF information again when the K-CCIS link is restored.

❍ BLF messages can be forwarded up to eight times in the network. When designing the K-CCIS network, this should be a consideration.

❍ When a Centralized DSS/BLF key is first programmed on a Feature Access or One Touch key, the BLF status does not change (update) until new BLF information is received from the remote system.

RELATED FEATURE LIST

❍ Do Not Disturb (DND)

❍ Feature Access – User Programmable

❍ Voice Mail Message Indication on Line Keys

GUIDE TO FEATURE PROGRAMMING (FOR MAIN SYSTEM)

This guide provides a list of associated Programs that support this feature.

(For Sending System)

Program/ Item No.	Description/ Selection	Assigned Data	Comments
30-01-01	DSS Console Operating Mode	0 = Business Mode 1 = Hotel Mode 2 = ACD Monitor Mode 3 = Business/ACD Mode	
30-02-01	DSS Console Extension Assignment	Up to eight digits Default is No Setting.	Identify extensions with DSS Consoles attached.
30-03-01	DSS Console Key assignment	Keys No. 001~114 00~99 = General Functional Level *00~*99 = Appearance Functional Level Default = extensions. 101~160	Customize key assignments for DSS Consoles 1~32.
50-08-01	CCIS Centralized BLF Sending Group Assignment – Destination Point Code	0~16367 1~8+Destination Point Code Default is 0	Define the Point Code of Billing Center Office.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
50-08-02	CCIS Centralized BLF Sending Group Assignment – CCIS Route ID	0~8 Default is 0	Define the CCIS Route ID to send Billing Center Office. ✎ Not used with IP K-CCIS.
50-09-01	CCIS Centralized BLF Sending Extension Number Assignment – Extension number	Up to eight digits Default is No setting.	BLF message is indicated when the status of the specified extension number is changed.
50-09-02	CCIS Centralized BLF Sending Extension Number Assignment – Send to Sending Group 1	Select Tables 1~ 120 0 = Disable 1 = Enable Default is 0	Enable sending BLF to Send Group 1 assigned in PRG 50-08-XX.
50-10-01	CCIS Centralized BLF Interval Time Assignment – Type of Interval Time	0 = 4 seconds 1 = 8 seconds 2 = 12 seconds 3 = 16 seconds Default is 0	Assign BLF sending interval to each sending system.

(For Receiving System)

Program/ Item No.	Description/ Selection	Assigned Data	Comments
20-13-41	Class of Service Options (Supplementary Service) – Voice Mail Message Indication on DSS key	0 = Off 1 = On Default is 0	Allow the DSS/BLF to indicate when the extension has a new message waiting in VM.
15-07-01	Programmable Function keys	LK01 = ✱01(Trunk Line Key)	Enter extension number up to eight digits.
20-06-01	Class of Service for Extensions	0~15	Extension 101 is in Class 15 by default. All other extensions are in Class 1.
30-03-01	DSS Console Key Assignment	1~114	0~99 (General Functional) ✱00~✱99 (Appearance Functional Level) 95 = Page Switching

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Centralized Day/Night Mode Change – K-CCIS

FEATURE DESCRIPTION

This feature switches the Day/Night mode of a remote office that is linked to a main office using K-CCIS, in accordance with the Day/Night mode switching from an Attendant Position at the main office.

When a UNIVERGE SV8100 system is connected to another UNIVERGE SV8100 system, the main office *can* control remote offices.

When connected to a NEAX2400, the UNIVERGE SV8100 can be used only as a remote office.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

To set or cancel Night Transfer system-wide from an Attendant Position:

Main Office:

1. Press Speaker.

2. Dial Access Code 718.

3. Dial the Night service Code:
    1   Day 1 mode
    2   Night 1 mode
    3   Midnight 1 mode

      4   Rest 1 mode

      5   Day 2 mode

      6   Night 2 mode

      7   Midnight 2 mode

      8   Rest 2 mode

4.   Press Speaker or hang up.

- OR -

1.   Press the Night Transfer key on the Attendant Add-On Console.

Remote Office:

No manual operation is required.

SERVICE CONDITIONS

General:

❍   A maximum of 16 remote offices can be controlled by one main office.

❍   If Automatic Day/Night Mode Switching is assigned in the main office, all remote offices change the mode, if assigned.

❍   If the remote office is to be restricted from overriding the Day/Night Mode setting, the following Memory Blocks should be assigned:

❑   12-01-01 Night Mode Function Setup – Manual Night Service enable

❑   20-07-01 Class of Service Options (Administrator Level) – Manual Night Service enable

❍   When the remote office is in Night Mode (as assigned in the Centralized Day/Night Mode - K-CCIS feature), normal Night Mode indications are provided.

❍   The Night Mode indication is the first word (Night) on the second row of the multiline terminal LCD. The LED for any Feature Access key assigned for Night Mode transfer and the Night Mode key on the Attendant console are On.

❍   If the K-CCIS link is not available due to network trouble, the UNIVERGE SV8100 main office resends the K-CCIS Day/Night Mode switch command every 16 minutes.

Restrictions:

❍ Centralized Day/Night Mode switching from a main office can send a system-wide K-CCIS Day/Night mode switch command only. Individual Night Service Groups Mode switching is not supported.

❍ When an UNIVERGE SV8100 receives the K-CCIS Day/Night Mode switch command from a main office, the remote office changes all Night Service Groups to the requested mode.

❍ Program 50-03-01 (Destination Point Code Transfer Assignment) must be set for all offices for the Centralized Day/Night Mode feature.

❍ A NEAX2000 cannot be used as a main office or a tandem office for the Centralized Day/Night Mode Change (K-CCIS) feature.

RELATED FEATURE LIST

❍ Assigned Night Answer (ANA)

❍ Authorization Code

❍ Automatic Day/Night Mode Switching

❍ Centralized Billing – K-CCIS

❍ Code Restriction

❍ Dial Access to Attendant – K-CCIS

❍ Direct Inward Termination (DIT)

❍ Flexible Ringing Assignment

❍ Night Call Pickup

❍ Night Chime

❍ Night Transfer

❍ Voice Mail Integration – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Centralized Day/Night Mode Change Installation

Program/ Item No.	Description/ Selection	Assigned Data	Comments
50-01-01	CCIS System Setting – CCIS Availability	0 = Disable 1 = Enable Default is 0	All CCIS settings lose functionality when this setting is 0.
50-03-01	CCIS Destination System Settings – Destination Point Code	0~16367 Default is 0	Enable for all offices for Centralized Day/Night mode change.
50-06-02	CCIS Feature Availability – Centralized Day/Night Switching (for message receiver side)	0 = Disable 1 = Enable Default is 1	If this data is set to 0, Link Reconnect does not run.
50-11-01	CCIS Centralized Day/ Night Switching Sending Group Assignment – Destination Point Code	Send Group (1~16) Point Code (1~16367) 0 = No Setting Default is 0	Select the Remote Office to send Day/Night Switching control message.
50-11-02	CCIS Centralized Day/ Night Switching Sending Group Assignment – CCIS Route ID	Send Group (1~16) CCIS Route ID (0~8) 0 = No Setting Default is 0	Select the Remote Office to send Day/Night Switching control message.
50-12-01	CCIS Centralized Day/ Night Mode to System Mode Assignment – Switching Synchronized Day/Night Mode Group	Day Mode = Mode 1~8 Default is 1 Night Mode = Mode 1~8 Default is 2	Set the mode for Day/Night Switching Synchronized Day/Night Group.

For Night Transfer Feature

Program/ Item No.	Description/ Selection	Assigned Data	Comments
12-01-01	Night Mode Function Setup – Manual Night Mode Switching	0 = Off 1 = On Default is 1	Allow user to activate Night Service by dialing a service code.
12-02-01	Automatic Night Service Patterns	Night Mode Group (1~32) Time Pattern Number (01~10) Set Time Number (01~20) Default = All Groups, All patterns: 00:00~00.00 for Mode 1	Set to define the daily pattern for the auto night mode switch setting.
12-03-01	Weekly Night Service Switching	Night Mode Service Group Number = 01~32 Default: 01 Sunday = Pattern 2 02 Monday = Pattern 1 03 Tuesday = Pattern 1 04 Wednesday = Pattern 1 05 Thursday = Pattern 1 06 Friday = Pattern 1 07 Saturday = Pattern 2	Set to define a weekly schedule of night switch settings.
12-04-01	Holiday Night Service Switching	Night Mode Service Group Number = 1~32 Default is No Setting	Set to define a yearly schedule for holiday.
30-03-01	DSS Console Key Assignment	Key Number (001~114) 00~99 = General Functional Level *00~*99 = Appearance Functional level Default is extensions. 101~160	Customize key assignments for DSS Consoles 1~32.
20-07-01	Class of Service Options (Administrator Level) – Manual Night Service Enabled	0 = Off 1 = On Default is 1 for COS 15, 0 for COS 1~14	
15-07-01	Programmable Function Keys	09 = Day/Night Mode Switch	Enter 0 to toggle night mode.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
20-06-01	Class of Service for Extensions	0~15	Extension 101 is in Class 15 by default. All other extensions are in Class 1.

Centralized E911 – K-CCIS

FEATURE DESCRIPTION

This feature allows a remote system to transmit a Calling Party Number to the 911 Emergency System over a K-CCIS direct or tandem connection.

SYSTEM AVAILABILITY

Terminal Type:

All Stations

Required Components

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

To use this feature at any terminal:

1.   Lift the handset, and wait for internal dial tone.

2.   Dial 911.

- OR -

1.   Dial 9 911.

SERVICE CONDITIONS

General:

❍   If you want to send your phone number via CCIS, please refer to Calling Party Number (CPN) Presentation from Station – K-CCIS on page 4-35.

❍   The Calling Party Number (CPN) is sent only to the network when the remote system accesses an ISDN – PRI trunk in the distant system and the ISDN – PRI trunk has Calling Party Number (CPN) Presentation and Screening service enabled from the network.

❍   If Program 21-01-10 is programmed with an entry other than 0, a call does not have a talk path unless the user dials at least the number of digits entered in this option when placing an out going call. This means that an entry of 4 or higher in this program causes a problem when dialing 911. Since it is only a 3-digit number, the call does not have a talk path preventing the emergency dispatcher from hearing the caller. It is recommended that this option be kept at its default setting of 0 to prevent any problems with dialing 911.

❍   The attendant receives a notification each time a co-worker dials an emergency 911 call. This notification is the co-worker name and number display optionally accompanied by an audible alarm. Notification occurs regardless of whether the attendant is idle or busy on a call. You can optionally extend this ability to other supervisory extensions as well.

❍   The PRI provider must provision for the CPN used for E911. The CPN must be within the allowable range. For more information please contact your local ISDN provider regarding allowable ranges.

❍   Virtual Extensions notify the attendant with the stations name and number when an emergency 911 call is originated from the Virtual Extension.

Restrictions:

❍   Centralized E911 (outgoing with CES-ID) is not supported.

❍   A maximum of 16 digits can be assigned as the Calling party Number (CPN) in Program 21-13-01.

❍   CAMA trunks are not supported.

❍   If Virtual Extensions are used to make E911 calls, they provide the information for the VE key.

RELATED FEATURE LIST

❍ ISDN Compatibility

❍ Calling Party Number (CPN) Presentation from Station K-CCIS

❍ E911 Compatibility

❍ Automatic Route Selection

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
20-08-13	Class of Service Options (Outgoing Call Service) – ISDN CLIP	0=Off 1=On Default is 0	Determine if the ISDN calling line identity presentation and screening indicators are to be allowed.
21-12-01	ISDN Calling Party Number Setup for Trunks	Up to 16 digits max. Default is No Setting	Assign each trunk a Calling Party Number. When a call is made by an extension which does not have an Extension Calling Number assigned (Program 21-13), the system sends the calling number for the ISDN trunk defined in 21-12.
21-13-01	ISDN Calling Party Number Setup for Extensions	Up to 16 digits max. Default is No Setting	Assign each extension a Calling Party Number.
14-01-24	Basic Trunk Data Setup – Trunk-to-Trunk Outgoing Caller ID Through Mode	0 = Disable (Off) 1 = Enable (On) Default is 0	Enable CPN information to pass through the Tandem Office.

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Dial Access to Attendant – K-CCIS

FEATURE DESCRIPTION

This feature allows a station user to call an Attendant by dialing a call code through the K-CCIS network.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

To call an Attendant Position:

1. Lift the handset or press Speaker.

2. Dial 0 or the operator call code if it is different from 0.

SERVICE CONDITIONS

General:

❍ The operator call code must be for an individual Attendant Access Code number.

❍ When calling to a UNIVERGE SV8100 Attendant Position, the SV8100 sends Operator to the call originator as the name.

✎ Shows Office Code if using Open Numbering Plan.

❍ If using an Open Numbering Plan, and a call is made to an UNIVERGE SV8100 Attendant Position, the operator office code is included with the name.

❍ When making a call from a UNIVERGE SV8100 Attendant Position across a K-CCIS network, the Caller ID Name and Number display is the same as for a station-to-station call.

❍ This feature is also available when the Attendant Console is in a NEAX2000 IVS2 or NEAX2400 in the CCIS network.

❍ When an UNIVERGE SV8100 station calls a NEAX Desk Console Attendant Position, OPERATOR is displayed on the LCD during the incoming ring. If using an Open Number Plan, the office code of the Desk Console is also displayed.

❍ In a Closed Numbering Plan network, a station can call an Attendant in the K-CCIS network by dialing 0.

❍ In an Open Numbering Plan network, a station user can call an Attendant within the K-CCIS network by dialing: Access Code + Office Code + 0.

Restrictions:

❍ When a PBX is in Night Mode, calls to a NEAX Desk Console are restricted. When an UNIVERGE SV8100 station calls a NEAX Desk Console Attendant Position that is set to Night Mode, ERROR is displayed in the calling station LCD and the call is rejected.

❍ When a NEAX Desk Console Attendant Position calls an UNIVERGE SV8100 station, the UNIVERGE SV8100 station does not store the call in the Caller ID Scrolling feature. This call record can be printed on SMDR without Caller ID information.

❍ Operator Calling, PRG 20-14-05, does not keep a Tie Line caller from dialing 0 for the operator.

RELATED FEATURE LIST

❍ Attendant Positions

❍ Calling Name Display – K-CCIS

❍ Calling Number Display – K-CCIS

❍ Centralized Day/Night Mode Change – K-CCIS

❍ Voice Calls – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

For Main System

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-01-01	System Numbering	Default is 0 = 1 digit Intercom	Refer to UNIVERGE SV8100 Programming Manual for all options and default settings.
20-17-01	Operator Extension – Operator Extension Number	Up to eight digits Default is 101.	Define extension numbers that are used as operators. Assign only in KTS-to-KTS network.

For Remote System

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-01-01	System Numbering	Default is 0 = 1 digit; F-Route	Refer to UNIVERGE SV8100 Programming Manual for all options and default settings.
44-02-01	Dial analysis Table for ARS/F-Route Access – Dial	Analysis Table 1 = 0 Default is No Setting.	Assign access code (up to eight digits) to dial the Attendant.
44-02-02	Dial analysis Table for ARS/F-Route Access – Service Type	0 = No Setting (None) 1 = Extension Call (Own) 2 = ARS/F-Route Table (F-Route) 3 = Dial Extension Analyze Table (Option) Default is 0	Assign the service type to be used.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
44-02-03	Dial analysis Table for ARS/F-Route Access – Additional Data	0 = No Setting 1 = Delete Digit = 0~255 (255: Delete all digits) 2 = 0~500 3 = Dial Extension Analyze Table Number (0~4) Default is 0	Enter additional data for Service Type selected in PRG 44-02-02.
44-05-01	ARS/F-Route Table – Trunk Group Number	0~100, 255 0 = No Setting 255 = Extension Call Default is 0	Select trunk group number for outgoing calls.
44-05-02	ARS/F-Route Table – Delete Digits	0 = No Setting 0~255 (255 = Delete all digits) Default is 0	Enter number of digits to delete from the dialed number.
44-05-03	ARS/F-Route Table – Additional Dial Number Table	0~1000 Default is 0	Enter Table Number defined in PRG 44-06.
44-05-09	ARS/F-Route Table – Maximum Digit	0~24 Default is 0	Assign Max. digits for Call Park Retrieve Access Code.
44-05-10	ARS/F-Route Table – CCIS over IP Destination Point Code	0~16367 Default is 0	Assign remote IP Destination Point Code.

Programming Example (Dest PC)
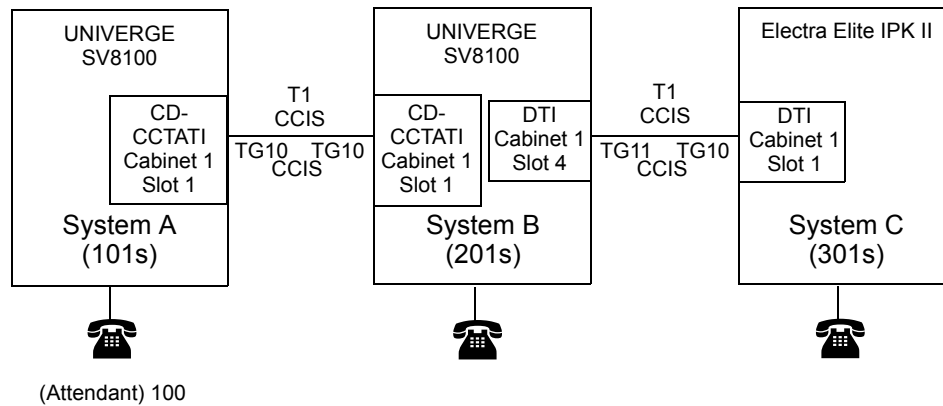
System A (101s)	System B (201s)	System C (301s)
Program 11-01-01 0 = 1 digits; Intercom	Program 11-01-01 0 = 1 digits; F-Route	Program 11-01-01 0 = 1 digits; F-Route
Program 20-17-01 Operator1 = Oper.Ext.No. 101	Program 44-02-01 Table 1 =Dial 0 (Dial Attendant System A)	Program 44-02-01 Table 1 =Dial 0 (Dial Attendant System A)
	Program 44-02-02 Table 1 = F-Route	Program 44-02-02 Table 1 = F-Route
	Program 44-02-03 Table 1 = F-Route 1	Program 44-02-03 Table 1 = F-Route 1
	Program 44-05-01 Table 1 = TRK GP 10	Program 44-05-01 Table 1 = TRK GP 10
	Program 44-05-02 Table 1 = Delete Digits 0	Program 44-05-02 Table 1 = Delete Digits 0
	Program 44-05-03 Table 1 = Add Dial 0	Program 44-05-03 Table 1 = Add Dial 0
	Program 44-05-09 Table 1 = Max Digit 1	Program 44-05-09 Table 1 = Max Digit 1
	* Program 44-05-10 F-Route Table 1 DPC = 1	*Program 44-05-10 F-Route Table 1 DPC = 1

* For CCISoIP Programming only.

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Direct Inward Dialing – K-CCIS

FEATURE DESCRIPTION

This feature allows an incoming DID call (centralized DID) to be routed directly across a K-CCIS link to reach a station in the remote system without Attendant assistance.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All Stations

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

Normal call handling procedures apply.

SERVICE CONDITIONS

General:

❍   Call billing to the outside party starts when the incoming call connects to the K-CCIS trunk.

❍   When an incoming DID call from the CD-PRTA card with Caller ID information is transferred to the station in K-CCIS network, the Caller ID Name and Number follow across the K-CCIS network to the distant system.

❍   This feature is supported when a Closed Numbering Plan or Open Numbering is used.

❍   The UNIVERGE SV8100 system supports DID Digit Conversion when using station numbers with 2~8 digits.

❍   An extension on a remote system can be the destination for the DID Received Vacant Number Operation Assignment (Program 22-09-02).

Restrictions:

❍   Program 20-02-15 (Caller ID Display Mode) must be set to 0 to display the DID Name on incoming DID calls.

Refer to the Key-Common Channel Interoffice Signaling (K-CCIS) feature for more details related to Single Line Telephone and IP (K-CCIS) support.

RELATED FEATURE LIST

❍   Key-Common Channel Interoffice Signaling (K-CCIS)

❍   Flexible Numbering of Stations – K-CCIS

❍   Uniform Numbering Plan – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
22-09-01	DID Basic data Setup – Expected Number of Digits	1~8 Default is 4	Assign number of digits the Table expects to receive from Telco. Use this program to make the system compatible with three- and four- digit DID service.
22-10-01	DID Translation Table Setup – Conversion Table Area Number	0 = No setting 1~2000 Default is 0	Program the Translation Table size.
22-11-01	DID Translation Number Conversion – Received Number	Maximum eight digits Default is No Setting	Assign the received number to the Conversion Table number.
22-11-02	DID Translation Number Conversion – Target Number	Maximum 24 digits Default is No Setting.	Assign the destination extension based on the digits received.
22-11-03	DID Translation Number Conversion – DID Name	Maximum 12 digits Default is No Setting	Assign the DID Name based on the digits received.

Dual Hold – K-CCIS

FEATURE DESCRIPTION

This feature allows two connected Multiline Telephones to be placed on hold simultaneously over the K-CCIS link. This enables the held parties to answer or originate a call from a secondary line or intercom path.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

Normal call handling procedures apply.

SERVICE CONDITIONS

General:

❍   This feature is available for interoffice calls through K-CCIS.

❍   Both Non-Exclusive Hold and Exclusive Hold can be used for Dual Hold – K-CCIS.

❍   The K-CCIS call is held on a Call Appearance key.

Restrictions:

None

RELATED FEATURE LIST

❐   Station-to-Station Calling – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
24-01-01	System Options for Hold – Hold Recall Time	0~64800 seconds Default is 90	A call on Hold recalls to the extension that placed the call on hold after this time expires.
24-01-03	System Options for Hold – Exclusive Hold Recall Time	0~64800 seconds Default is 90	A call left on Exclusive Hold recalls to the extension that placed it on hold after this time expires.
20-29-01	Timer Class for Extensions – Day/Night Mode 1~8, Class Number	0~15 0 = Not Assigned Default is 0	Assign Timer Class (0~16) to each extension for night mode. Virtual extension numbers are included.
20-30-01	Timer Class for Trunks – Day/Night Mode 1~8, Class Number	0~15, #, ✱ 0 = Not Assigned Default is 0	Assign Timer Class (0~16) to each trunk for night mode.
20-31-01~23	Timer Class Timer Assignment	Refer to Flexible Timeouts in the UNIVERGE SV8100 Features and Specifications Manual for more Flexible Time details.	Assign times. These timers are referred when a class is set to any number from 1 to 16 in PRG 20-29-01/20-30-01.

Elapsed Time Display – K-CCIS

FEATURE DESCRIPTION

This feature provides an Elapsed Call Time on the LCD which shows the duration of time that a multiline terminal is connected to any call through the K-CCIS network.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All Display multiline terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

No manual operation is required.

SERVICE CONDITIONS

General:

❍ When a call is retrieved from Exclusive Hold and/or Non-Exclusive Hold from the same station, the elapsed call timer begins at 0.

❍ When a call is transferred, the elapsed time of the party receiving the transfer begins at zero.

Restrictions:

❍   For calls across a K-CCIS link, the Elapsed Call timer begins only after receiving answer supervision from the distant system.

❍   For Voice Calls across the K-CCIS link, the Elapsed Call timer does not begin until the distant station answers.

❍   For conference calls established across a K-CCIS link, the elapsed call timer does not start during an active conference call.

RELATED FEATURE LIST

❍   Station-to-Station Calling – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
20-13-36	Class of Service Options (Supplementary Service) – Call Duration Timer Display	0 = Off 1 = On Default is 1	Turn off the extension Call Timer. The system waits for the interdigit time (21-01-01) before this time begins.
20-09-06	Class of Service Options (Incoming Call Service) – Incoming Time Display	0 = Off 1 = On Default is 0	Turn on the Incoming Time and Date display on the LCD while the Telephone is ringing.

Flexible Numbering of Stations – K-CCIS

Station numbers can be assigned by the 10's group for 4-digit station numbers, 100's group for 5-digit, 1,000's group for 6-digit station numbers, and 10,000's group for 7-digit station numbers.

Example:

Station Numbering Plan	Site A	Site B	Site C
4-digit station numbers	1000 ~ 1009	1010 ~ 1019	1020 ~ 1029
5-digit station numbers	10000 ~ 10009	10010 ~ 10019	10020 ~ 10029
6-digit station numbers	100000 ~ 100009	100010 ~ 100019	100020 ~ 100029
7-digit station numbers	1000000 ~ 1000009	1000010 ~ 1000019	1000020 ~ 1000029
8-digit station numbers	10000000 ~ 10000009	10000010 ~ 10000019	10000020 ~ 10000029

FEATURE DESCRIPTION

This feature allows telephone numbers to be assigned to any stations in the K-CCIS network, based solely upon numbering plan limitations.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All Stations

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

Normal call handling procedures apply.

SERVICE CONDITIONS

General:

❍   Give careful consideration to the network numbering plan to avoid needless loss of Access Codes or duplication of telephone numbers.

❍   The first digit or first two digits of a telephone number distinguishes one system from another system.

❍   Station Numbering Plan can have 2~8 digits.

Restrictions:

❍   Tenant service is not provided, i.e., numbers cannot be duplicated for different tenants.

❍   Extension numbers should not start with 0,9,* or #.

For non-K-CCIS feature support, refer to the UNIVERGE SV8100 Features and Specifications Manual, Flexible Numbering Plan feature.

RELATED FEATURE LIST

❍   Key-Common Channel Interoffice Signaling (K-CCIS)

❍   Station-to-Station Calling – K-CCIS

❍   Uniform Numbering Plan – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-11-01	System Numbering	Default is 1 = 3-digit; Intercom	Refer to the UNIVERGE SV8100 Programming Manual for all options and default settings.
11-02-01	Extension Numbering – Dial (up to eight digits)	Default : Port 1~99 = 101~199 Port 100~512 = 3101~3513	Assign up to eight digits for Extension Numbers.
11-20-01	Dial Extension Analyze Table – Dial (up to eight digits) Use tables 01~128 to assign the digits to be dialed using the Dial Extension Analyze Tables. These tables are used when Program 11-01-01 is set to option 9 = Dial Extension Analyze. (Up to eight digits can be assigned with valid entries: 0, 1~9, #, ✱).	Valid entries: 0, 1~9, #, ✱ Default is not assigned.	
11-20-02	Dial Extension Analyze Table – Type of Dials Assign the type of dial for the Dial Extension Analyze Table from Program 11-20-01. (Svc Code, Intercom, operator, or F-Route)	0 = Not used 1 = Service Code 2 = Intercom 5 = Operator 6 = F-Route Default is Not Set.	

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Handsfree Answerback – K-CCIS

FEATURE DESCRIPTION

This feature allows Multiline Telephone station users to respond to voice calls through a K-CCIS network without lifting the handset.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

To turn the microphone On/Off:

1. Press Feature.

2. Dial 1.

- OR -

Press the programmable line key assigned as the MIC On/Off key.

SERVICE CONDITIONS

Restrictions

❍ Handsfree Answerback – (K-CCIS) can be used only when responding to Voice Calls – (K-CCIS) from a remote user.

❍ After a user changes ring back tone to voice call, it cannot be changed back to ringing.

❍ Voice Call cannot be set as the initial call across K-CCIS. The initial call must be a ringing call.

RELATED FEATURE LIST

❍ Voice Calls – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-16-03	Single Digit Service code Setup – Switching of Voice/Signal Call	Default is 1	Customize the one-digit Service Code used when a busy or ring back signal is heard.
11-12-06	Service Code Setup (for Service Access) – Switching of Voice Call and Signal Call	Default is 712	Toggle an ICM call between Handsfree Answerback or Forced Intercom Ringing for outgoing Intercom Calls.
20-08-10	Class of Service Options (Outgoing Call service) – Signal/Voice Call	0 = Off 1 = On Default is 1	Turn Off or On the ability to force Handsfree Answerback or Forced Intercom Ringing for outgoing Intercom Call on/off.
20-08-11	Class of Service Options (Outgoing Call service) – Protect for the Call Mode Switching from Caller (Internal Call)	0 = Off 1 = On Default is 0	When extension is set for ICM calls, enable this option to prevent callers from changing to voice announce mode.

Hot Line – K-CCIS

FEATURE DESCRIPTION

This feature allows two stations at different nodes in the K-CCIS network to be mutually associated on automatic ringdown through the K-CCIS network.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

To execute at any station programmed for Hot Line:

1.   Lift the handset or press Speaker.

2.   The remote K-CCIS station is called.

SERVICE CONDITIONS

General:

❍   Any multiline terminal (a maximum number of 512 stations) can be assigned for
     Hot Line – (K-CCIS).

❍   Either multiline terminal in a Hot Line – (K-CCIS) pair may transfer a Hot Line
     call to another station in the K-CCIS network using the Call Transfer – All Calls -
     (K-CCIS) feature.

Restrictions:

None

RELATED FEATURE LIST

❍   Call Transfer – All Calls - K-CCIS

❍   Station-to-Station Calling – K-CCIS

❍   Uniform Numbering Plan – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
15-01-02	Basic Extension Data Setup – Outgoing Trunk Line Preference	0 = Off 1 = On Default is 0	If enabled, the extension user gets trunk dial tone when the handset is lifted.
20-06-01	Class of Service for Extensions	1 ~ 15 Default: Extension 101 is Class 15, all other extensions are Class 1	Assign a Class of Service to an extension
20-08-09	Class of Service Options (Outgoing Call Service) – Hotline/ Extension Ringdown	0 = Off 1 = On Default is 0	Turn Off or On Ringdown Extension for extensions with this COS.
20-08-19	Class of Service Options (Outgoing Call Service) – Hotline for SPK	0 = Off 1 = On Default is 0	When Hotline is programmed and 20-08-19 is turned ON (1), the user can press the speaker button and the Hotline destination is dialed.
21-01-09	System Options for Outgoing Calls – Ringdown Extension Timer (Hotline Start)	0~64800 seconds Default is 5	A Ringdown extension automatically calls its programmed destination after this time.
21-11-01	Extension Ringdown (Hotline) Assignment – Hotline Destination Number	Maximum 24 digits (0~9, ✳, #, Pause, Hook Flash, and @(code to wait for answer supervision) Default is No Setting	Define the Hotline Destination number for each extension number.

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Link Reconnect – K-CCIS

FEATURE DESCRIPTION

This feature provides the system that is connected to a K-CCIS network with the ability to release the redundant K-CCIS link connections and reconnect the link with the system for efficient usage of the K-CCIS trunks.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components

PZ-(X)IPLA

OPERATING PROCEDURES

Normal call handling procedures apply.

SERVICE CONDITIONS

General:

❍   The link reconnect ability is provided for a station call over K-CCIS that is transferred or forwarded to another station or trunk in the same office as the call originating station. (Refer to Figure 4-1 Link Reconnect for Station Calls.)
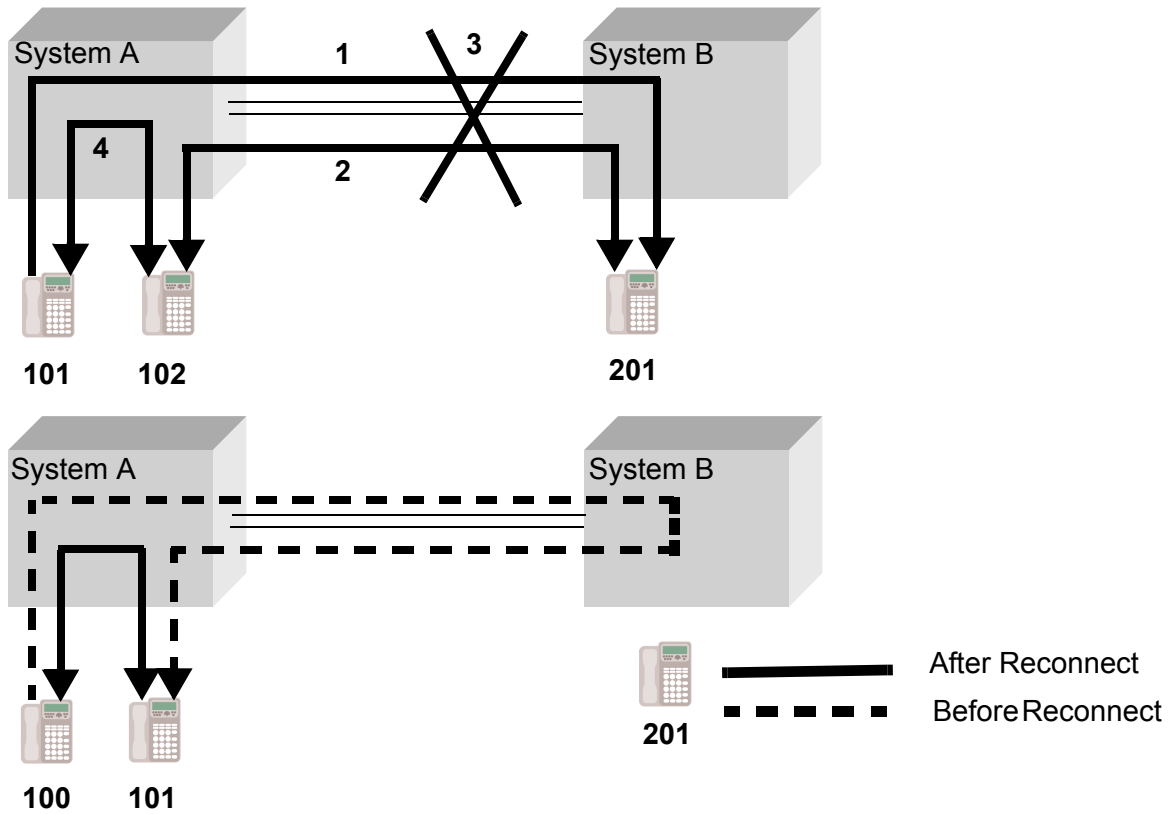
Stations 101, 102, and 201 must be multiline terminals.

✎   When Station A holds the call or is in the conference, Link Reconnect is not provided.

Figure 4-1 Link Reconnect for Station Calls

❍   A trunk call (CO/PBX/TIE/DID/K-CCIS) over a K-CCIS network is transferred or forwarded to another station or trunk within the same office as the original incoming trunk. (Refer to Figure 4-2 Link Reconnect for Trunk Tandem Calls.)

Call Forward to CO Line
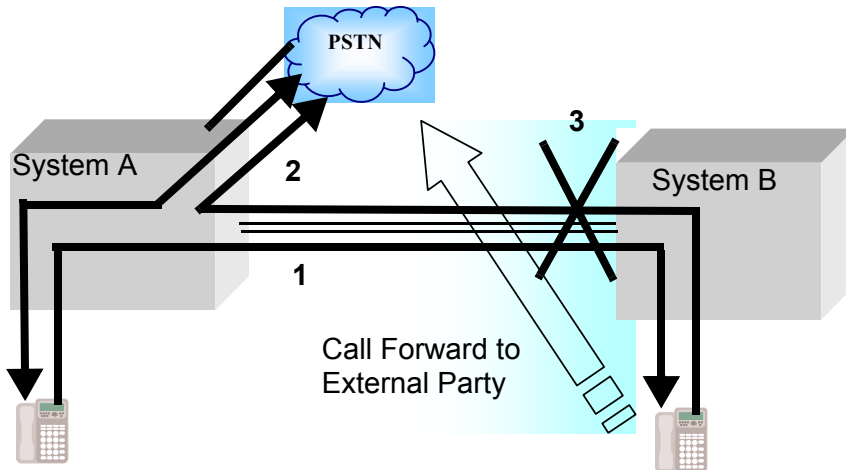
Voice Path after Call Forwarding
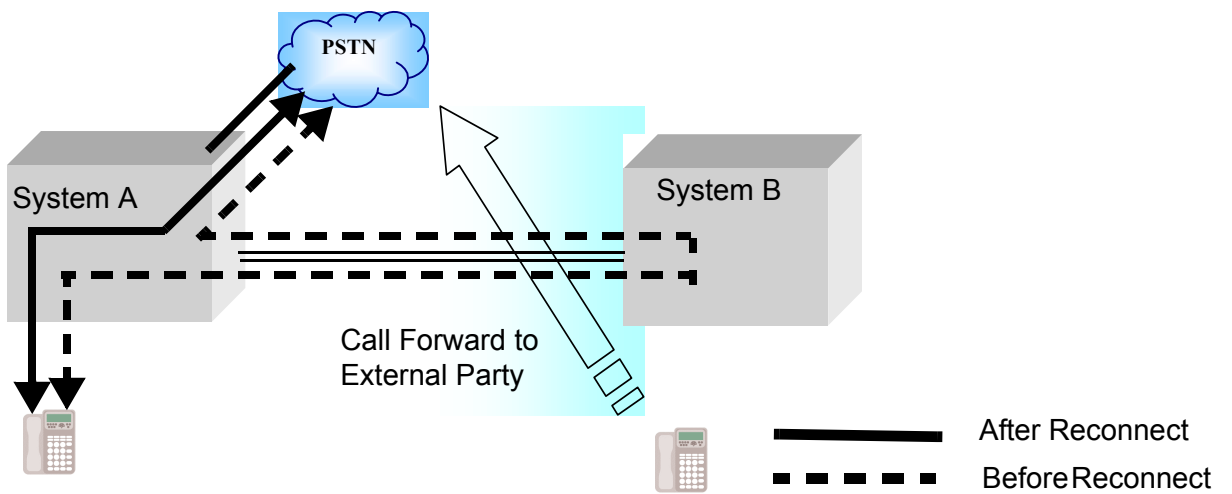
Figure 4-2 Link Reconnect for Trunk Tandem Calls

O   Link reconnect occurs after answering a transferred or forwarded K-CCIS call.

Restrictions:

❍   Answer supervision is required for Link Reconnect to occur. For outgoing calls
     on analog trunks, Answer supervision is based on the Elapsed Call Time -
     Program 21-01-03 (Trunk Interdigit Time).

❍   When a call is on hold, or in a conference, and is transferred back across the
     K-CCIS link, Link Reconnect is not provided.

RELATED FEATURE LIST

❍   Call Forwarding – All Calls – K-CCIS

❍   Call Forwarding – Busy/No Answer – K-CCIS

❍   Call Transfer – All Calls – K-CCIS

❍   Multiple Call Forwarding – All Calls – K-CCIS

❍   Multiple Call Forwarding – Busy/No Answer – K-CCIS

❍   Station-to-Station Calling – K-CCIS

❍   Uniform Numbering Plan – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
50-06-01	CCIS Feature Availability – Link Reconnect	0 = Not Available 1 = Available Default is 1	Enable Link Reconnect.
21-01-03	System Options for Outgoing Calls – Trunk Interdigit Time (External)	0 = Disabled 1~64800 seconds Default is 5	The system waits for this time to expire before placing the call (Call time starts after this time expires).

Multiple Call Forwarding – All Calls – K-CCIS

FEATURE DESCRIPTION

This feature allows a Multiple Call Forwarding – All Calls sequence to be forwarded over a K-CCIS network to a station in another office.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

To set Call Forward – All Calls – K-CCIS from a Multiline Telephone (Closed Numbering Plan):

1. Press the Call Forward – All ON/OFF key.

2. Dial 1 to set. Then enter the K-CCIS station number.

3. Press Speaker.

- OR -

1. Lift the handset or press Speaker.

2. Dial Access Code 741 (set as default). Then Dial 1 to set.

3. Dial the remote K-CCIS station number.

4. Restore handset or press Speaker.

To set Call Forward – All Calls – K-CCIS from a Multiline Telephone (Open Numbering Plan):

1.   Press the Call Forward – All Call ON/OFF key, and Dial 1 to set.

2.   Dial the trunk Access Code (normally 8).

3.   Dial the Office Code number.

4.   Dial the distant K-CCIS station number.

5.   Press Speaker.

- OR -

1.   Lift the handset or press Speaker.

2.   Dial Access Code 741 (set as default), and Dial 1 to set.

3.   Dial the trunk Access Code (normally 8).

4.   Dial the Office Code number.

5.   Dial the distant K-CCIS station number.

6.   Restore handset or press Speaker.

To cancel Call Forward – All Calls – K-CCIS from a Multiline Telephone:

1.   Press Call Forward – All Call On/Off key.

2.   Dial 0 to cancel.

3.   Press Speaker.

- OR -

1.   Lift the handset or press Speaker.

2.   Dial Access Code 741 (set as default), and Dial 0 to cancel.

3.   Restore handset or press Speaker.

SERVICE CONDITIONS

General:

❍ Multiple Call Forwarding – All Calls – K-CCIS can forward a call up to seven times across K-CCIS links (up to seven hops) depending on system data.

❍ Multiple Call Forwarding over a K-CCIS link is combined with Multiple Call Forwarding – All Calls/Busy/No Answer.

❍ If the calling station is set as the destination in a multiple hop scenario, Multiple Call Forwarding – All Calls – K-CCIS is not performed (*i.e.*, an infinite loop does not occur).

❍ For multiple Call Forwarding – All Calls/Busy (Immediate) calls, the display on the calling party Multiline Telephone displays the terminating station user name and the station number for the first station of a distant system in the Multiple Call Forwarding group. For the terminating station, the telephone display indicates the name and the number of the calling party and the trunk number of the incoming call.

❍ When a calling station has been Call Forwarding – All Calls – K-CCIS to the maximum times assigned in Program 50-05-01 (K-CCIS Maximum Call Forwarding Hop Assignment) and encounters another Call Forwarding – All Calls – K-CCIS condition, the calling station is not forwarded and rings at the last destination.

❍ If the destination station in a Multiple Call Forwarding – All Calls – K-CCIS situation is busy and has not set Call Forwarding – Busy and has Call Alert Notification disabled, the calling party receives busy tone.

❍ When combining Call Forwarding – Busy and Call Forwarding – All Calls – K-CCIS, if the destination station is busy and has Call Alert Notification disabled, the calling party hears busy tone after the maximum hops assigned in Program 50-05-01 (K-CCIS Maximum Call Forwarding Hop Assignment).

❍ Multiple Call Forwarding – All Calls -K-CCIS and Call Forwarding – Busy – K-CCIS may be mixed; up to seven combined multiple forwardings may occur.

❍ An example of Multiple Call Forwarding over a K-CCIS link is shown in Figure 4-3 Multiple Call Forwarding over K-CCIS Links for All Calls.

Figure showing Office D, Office C, Office B, Office A connected by K-CCIS links with Counter = , Counter = 1, Counter = 2, and a callout "Number of Call Forwards over K-CCIS = 2"

Multiple Call Forwarding Over K-CCIS	
Office A	Allowed
Office B	Allowed
Office C	Allowed
Office D	Allowed
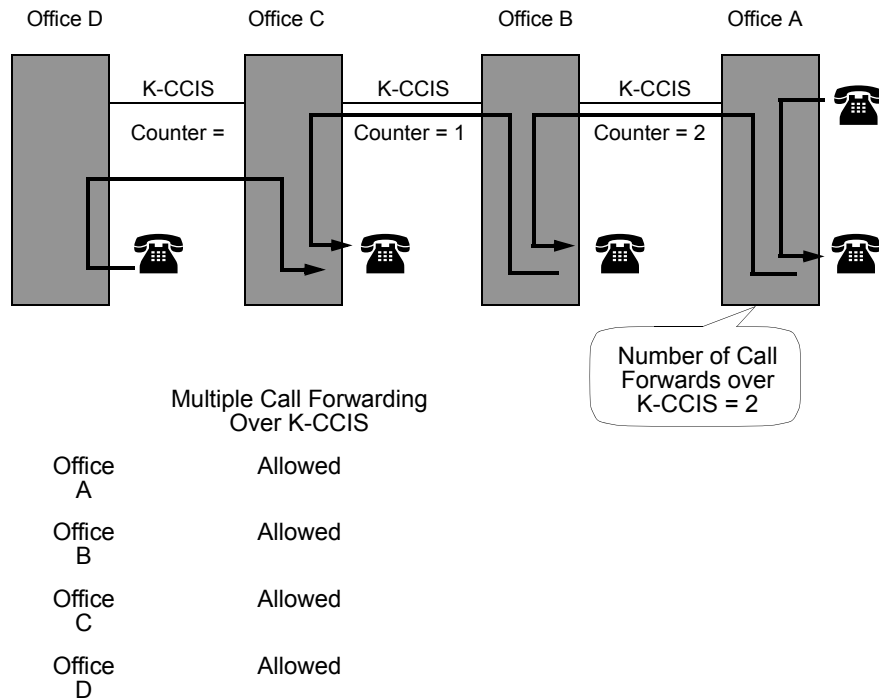
✎   The counter is reduced by one with each hop (tandem connection).

Figure 4-3 Multiple Call Forwarding over K-CCIS Links for All Calls

RELATED FEATURE LIST

❍   Call Transfer – All Calls – K-CCIS

❍   Call Forwarding – All Calls – K-CCIS

❍   Call Forwarding – Busy/No Answer – K-CCIS

❍   Multiple Call Forwarding – Busy/No Answer – K-CCIS

❍   Link Reconnect – K-CCIS

❍   Uniform Numbering Plan – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

| Program/ Item No.	Description/ Selection	Assigned Data	Comments
20-06-01	Class of Service for Extensions	0~15	Default: Extension 101 is in Class 15. All other extensions are in Class 1.
20-09-07	Class of Service Options (Incoming Call Service) – Call Queuing	0 = Off 1 = On Default is 1	Must be Off for Class of Service for Call Forward Busy to operate.
20-11-12	Class of Service Options (Hold/Transfer Service) – Call Forwarding Off Premise (External Call Forwarding	0 = Off 1 = On Default is 0	Enable Call Forward – Off-Premise per Class of Service.
20-13-06	Class of Service Options (Supplementary Service) – Automatic Off Hook Signaling (Automatic Override)	0 = Off 1 = On Default is 1	Must be Off for Call Forward – Busy to operate.
15-07-01	Programmable Function Keys	10 = Call Forward – Immediate 11 Call Forward – Busy 12 Call Forward – No Answer 13 Call Forward – B/NA	Service Codes: 741 742 743 744
14-01-13	Basic Trunk Data Setup – Trunk-to-Trunk Transfer Loop Supervision	0 = Disable 1 = Enable Default is 1	Must be enabled for Trunk-to-Trunk Transfer; Call Forward – Off- Premise, or Tandem Trunking.
50-06-01	CCIS Feature Availability – Link Reconnect	0 = Not Available 1 = Available Default is 1	Enable Link Reconnect.
50-05-01	CCIS Maximum Call Forwarding Hop Counter	0~7 Hops Default is 5	Set Maximum Hops.

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Multiple Call Forwarding – Busy/No Answer - K-CCIS

FEATURE DESCRIPTION

This feature allows a Multiple Call Forwarding – Busy/No Answer sequence to be forwarded over a K-CCIS network to a station in another office.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

To set Call Forward – Busy/No Answer - K-CCIS from a Multiline Telephone (Closed Numbering Plan):

1.   Press the Call Forward – Busy/No Answer ON/OFF key.

2.   Dial 1 to set.

3.   Dial the remote K-CCIS station number.

4.   Press Speaker.

- OR -

1.   Lift the handset or press Speaker.

2.   Dial Access Code 744 (default), then 1 to set.

3.   Dial the remote K-CCIS station number.

4.   Restore handset or press Speaker.

To set Call Forward – Busy/No Answer - K-CCIS from a Multiline Telephone (Open Numbering Plan):

1.   Press the Call Forward – Busy/No Answer ON/OFF key.

2.   Dial 1 to set.

3.   Dial the trunk Access Code (normally 8).

4.   Dial the Office Code number.

5.   Dial the distant K-CCIS station number.

6.   Press Speaker.

- OR -

1.   Lift the handset or press Speaker.

2.   Dial Access Code 744 (default), then dial 1 to set.

3.   Dial the trunk Access Code (normally 8).

4.   Dial the Office Code number.

5.   Dial the distant K-CCIS station number.

6.   Restore handset or press Speaker.

To cancel Call Forward – Busy/No Answer – K-CCIS from a Multiline Telephone:

1.   Press Call Forward – Busy/No Answer On/Off key.

2.   Dial 0 to cancel.

3.   Press Speaker.

- OR -

1.   Lift the handset or press Speaker.

2.   Dial Access Code 744 (default), then Dial 0 to cancel.

3.   Restore handset or press Speaker.

To set for any station for Attendant Positions only (Closed Numbering Plan):

1. Lift the handset or press Speaker.

2. Dial the Call Forward Busy/No Answer for any Extension to Destination Service Code (default: 793).

3. Dial 1 (Set).

4. Dial the extension number to be forwarded and then the destination number.

5. Press Speaker or hang up.

To cancel for any station for Attendant Positions only:

1. Pick up the handset or press Speaker.

2. Dial the Call Forward Busy/No Answer for any Extension to Destination Service Code (default: 793).

3. Dial 0 (Cancel).

4. Dial the station number, which is forwarded.

5. Press Speaker or hang up.

SERVICE CONDITIONS

General:

❍   Multiple Call Forwarding – Busy/No Answer Calls - K-CCIS can forward a call up to seven times across K-CCIS links (up to seven hops) depending on systems data.

❍   Multiple Call Forwarding over a K-CCIS link is combined with Multiple Call Forwarding – All Calls/Busy/No Answer.

❍   If the calling station is set as the destination in a multiple hop scenario, Multiple Call Forwarding – Busy/No Answer Calls - K-CCIS are not performed, i.e., an infinite loop does not occur.

❍   For multiple Call Forwarding – All/Busy (Immediate) calls, the display on the calling party's Multiline Telephone indicates the terminating station user's name and the station number for the first station of a distant system in the Multiple Call Forwarding group. For the terminating station, the telephone display indicates the name and the number of the calling party and the trunk number of the incoming call.

❍   For multiple Call Forwarding – No Answer/Busy (Delay) calls, the display on the calling party's Multiline Telephone indicates the name and number of the first station of a distant systems in the Multiple Call Forwarding group. For the terminating station, the telephone display indicates the name and the number of the calling party and the trunk number of the incoming call.

❍   When a calling station has been Call Forwarding – Busy/No Answer Calls – K-CCIS to the maximum times assigned in Program 50-05-01 (K-CCIS Maximum Call Forwarding Hop Assignment) and encounters another Call Forwarding – Busy/No Answer Calls – K-CCIS condition, the calling station is not forwarded and rings at the last destination.

❍   When the destination station in a Multiple Call Forwarding – Busy/No Answer Calls – K-CCIS situation is busy and has not set Call Forwarding – Busy and has Call Alert Notification disabled, the calling party receives busy tone.

❍   When combining Call Forwarding – Busy and Call Forwarding – All Calls – K-CCIS and the destination station is busy and has Call Alert Notification disabled, the calling party hears a busy tone after the maximum hops assigned in Program 50-05-01 (K-CCIS Maximum Call Forwarding Hop Assignment).

❍   Multiple Call Forwarding – All Calls – K-CCIS can forward a call a maximum of seven times across K-CCIS link (maximum of seven hops) depending on system data assignments.

❍   An example of Multiple Call Forwarding over a K-CCIS link as shown Figure 4-4 Multiple Call Forwarding over K-CCIS Links for Busy/No Answer.
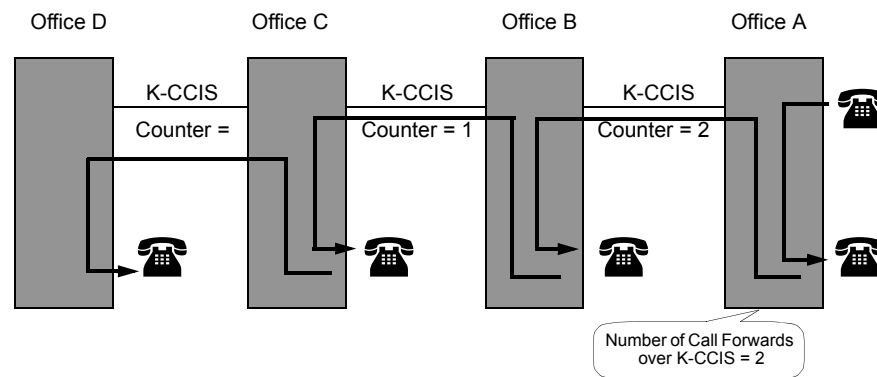
Office D	Office C	Office B	Office A

K-CCIS	K-CCIS	K-CCIS
Counter =	Counter = 1	Counter = 2

Number of Call Forwards over K-CCIS = 2

Multiple Call Forwarding Over K-CCIS

Office A	Allowed
Office B	Allowed
Office C	Allowed
Office D	Allowed

✎   Note: The counter is reduced by one with each hop (tandem connection).

Figure 4-4 Multiple Call Forwarding over K-CCIS Links for Busy/No Answer

Restrictions:

❍ Trunk-to-Trunk Transfer must be allowed in Program 20-11-14 [Class of Service Options (Hold/Transfer Service) Trunk-to-Trunk Transfer Restriction].

RELATED FEATURE LIST

❍ Call Forwarding – All Calls – K-CCIS

❍ Call Forwarding – Busy/No Answer – K-CCIS

❍ Multiple Call Forwarding – All Calls – K-CCIS

❍ Link Reconnect – K-CCIS

❍ Uniform Numbering Plan – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
20-06-01	Class of Service for Extensions	0~15	Default: Extension 101 is in Class 15, All other extensions are in Class 1.
20-09-07	Class of Service Options (Incoming Call Service) – Call Queuing	0 = Off 1 = On Default is 1	Must be Off for Class of Service for Call Forward Busy to operate.
20-11-12	Class of Service Options (Incoming Call Service) – Call Forwarding Off Premise (External Call Forwarding)	0 = Off 1 = On Default is 0	Enable Call Forward – Off-Premise per Class of Service.
20-11-14	Class of Service Options (Incoming Call Service) – Trunk-to-Trunk Transfer Restriction	0 = Off 1 = On Default is 0	Turn Off or On the Trunk-to-Trunk Transfer Restriction. If enabled (turned on), Trunk-to-Trunk Transfer is not possible.
20-13-06	Class of Service Options (Supplementary Service) –Automatic Off Hook Signaling (Automatic Override)	0 = Off 1 = On Default is 1	Must be Off for Call Forward – Busy to operate.
15-07-01	Programmable Function Keys	10 = Call Forward – Immediate 11 = Call Forward – Busy 12 = Call Forward – No Answer 13 = Call Forward – B/NA	Service Codes: 741 742 743 744
14-01-13	Basic Trunk Data Setup – Trunk-to-Trunk Transfer Loop Supervision	0 = Disable (No) 1 = Enable (Yes) Default is 1	Must be enabled for Trunk-to-Trunk Transfer; Call Forward – Off- Premise, or Tandem Trunking.
50-06-01	CCIS Feature Availability – Link Reconnect	0 = Not Available 1 = Available Default is 1	Enable Link Reconnect.
50-05-01	CCIS Maximum Call Forwarding Hop Counter	0~7 Hops Default is 5	Set Maximum Hops.

Paging Access – K-CCIS

FEATURE DESCRIPTION

This feature allows users to access internal or external paging from remote sites across the K-CCIS network. Local stations where the external paging equipment is installed can use the Meet-Me Answer feature to answer the page and establish a station-to-station K-CCIS call.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

To access internal or external paging across a K-CCIS network:

1.   Lift the handset or press Speaker.

2.   Dial the Access Code for the required zone, or press the programmed Feature Access or One-Touch key.

SERVICE CONDITIONS

General:

❍   The single external paging zone output built into the basic B64-U20 KSU can be used for Paging Access – (K-CCIS).

❍   Program 31-01-02 (Paging Announcement Duration) applies to Paging Access – (K-CCIS).

❍   If a user dials during Paging Access – (K-CCIS), DTMF tones are heard from the external paging equipment at the remote site.

❍   Program 31-02-01 (Internal Paging Group Number) applies to Paging Access – (K-CCIS).

❍   Program 31-02-02 (Internal All Call Paging Receiving) applies to Paging Access – (K-CCIS).

Restrictions:

❍   Amplifiers and speakers must be locally provided.

❍   Combined Paging is not supported over K-CCIS.

❍   Internal Paging across K-CCIS is supported only between UNIVERGE SV8100 and UNIVERGE SV8100.

RELATED FEATURE LIST

❍   Background Music Over External Speakers

❍   External Zone Paging (Meet-Me)

❍   Internal Zone Paging (Meet-Me)

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

For Paging Installation

Program/ Item No.	Description/ Selection	Assigned Data	Comments
10-03-01	ETU Setup – Terminal Type (B1)	0 = Not Set 1 = Multiline Terminal 2 = SLT Adapter 3 = Bluetooth Cordless Handset 6 = PGD (Paging) 7 = PGD (Tone Ringer) 8 = PGD (Door Box) 9 = PGD (ACI) 10 = DSS Console 11 = ---Not Used---	Assign ESI port for External Paging.
11-01-01	System Numbering	Default is 51 = 2 digit: F-Route.	This Service Code must be assigned also in PRG 11-12-20.
11-12-20	Service Code Setup (for Service Access) – External Paging	Default is 703	Setting 703 should be changed based on the K-CCIS network configuration.
11-12-22	Service Code Setup (Service Access) – Meet Me Answer to External Paging	Default is 765	The Service Code assigned in this Program is used for Meet Me Answer to External Paging.
15-07-01	Programmable Function Keys	19 = External. Group Paging 1~8 20 = External. All Call Paging 21 = Internal. Group Paging 1~64 22 = Internal. All Call Paging	Set the functions of programmable extension Function Keys.
20-10-06	Class of Service Options (Answer Service) – Meet Me Conference and Paging	0 = Off 1 = On Default is 1	Enable Meet Me Conferencing and Paging.
20-14-07	Class of Service Options for DISA/E&M – External Paging	0 = Off 1 = On Default is 1	Allow DISA or Tie Line Trunk user to use External Paging.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
31-01-01	System Options for Internal/External Paging – All Call Paging Zone Name	Up to 12 Characters Default is Group All	Assign a name to each All Call Internal Paging Zone. The name is displayed to make the announcement.
31-01-02	System Options for Internal/External Paging – Page Announcement Duration	0~64800 seconds Default is 1200	Set the length of Paging Announcements.
31-01-04	System Options for Internal/External Paging – Privacy Release Time	0~64800 seconds Default is 90	After user initiates a Meet Me or Voice Call conference, the system waits this time for the Paged Party to answer the call.
31-04-01	External Paging Zone Group	0 = No Setting Default: Speaker 1 = Group 1 Speaker 2 = Group 2 Speaker 3 = Group 3 Speaker 4 = Group 4 Speaker 5 = Group 5 Speaker 6 = Group 6 Speaker 7 = Group 7 Speaker 8 = Group 8 Speaker 9 = CPU Group 1	Assign each External Paging Zone to an External Paging Group.
31-06-01	External Speaker Control – Broadcast Splash Tone before Paging (Paging Start Tone)	0 = No Tone 1 = Splash Tone 2 = Chime Tone Default is 2	Enable Splash Tone before paging.
31-06-02	External Speaker Control – Broadcast Splash Tone After Paging (Paging End Time	0 = No Tone 1 = Splash Tone 2 = Chime Tone Default is 2	Enable Splash Tone after paging.
31-06-03	External Speaker Control – Speech Path	0 = Both Ways (Duplex) 1 = One Way PGD $\rightarrow$ SPK (Simplex) Default is 1	Establish whether or not the external speaker is used for talkback. (Not Available on CPU page port 9)

For Remote System

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-01-01	System Numbering	Default is 151 = 2 digit: F-Route	Assign also in PRG 11-12-20.
11-12-20	Service Code Setup (for Service Access) – External Paging	Default is 703	Change703 Setting based on the K-CCIS network configuration.
44-02-01	Dial Analysis Table for ARS/F-Route Access) – Dial	Analysis Table 1 = 511 Default is No Setting	Assign the access code used to page. Up to eight digits can be assigned.
44-02-02	Dial Analysis Table (ARS/ F-Route Access) – Service Type	0= No Setting (None) 1 = Extension Call (Own) 2 = ARS/F-Route Table (F-Route) 3 = Dial Extension Analyze Table (Option) Default is 0	Assign Service Type.
44-02-03	Dial Analysis Table (ARS/ F-Route Access) – Additional data	0= No Setting 1 = Delete Digits 0~255 (255: Delete All Digits) 2 = 0~500) 3 = Dial Extension Analyze Table Number (0~4) Default is 0	For the Service Type selected in 44-02-02, enter the additional data required.
44-05-01	ARS/F-Route Table – Trunk Group Number	0~ 100, 255 0 = No Setting	Select trunk group number used for outgoing ARS calls. Setting 255 = Internal Extension Call.
44-05-02	ARS/F-Route Table – Delete digits	0 = No setting 0~255 255 = Delete All Digits Default is 0	Enter number of digits to delete from the dialed number.
44-05-03	ARS/F-Route Table – Additional Dial Number Table	0= No Setting 0~1000 Default is 0	Enter Table Number (defined in PRG 44-06) for additional digits to dial.
44-05-09	ARS/F-Route Table – Maximum Digit	0 = No Maximum 0~24 Default is 0	Assign max digits for the Paging Access Code.

Programming Example (External)
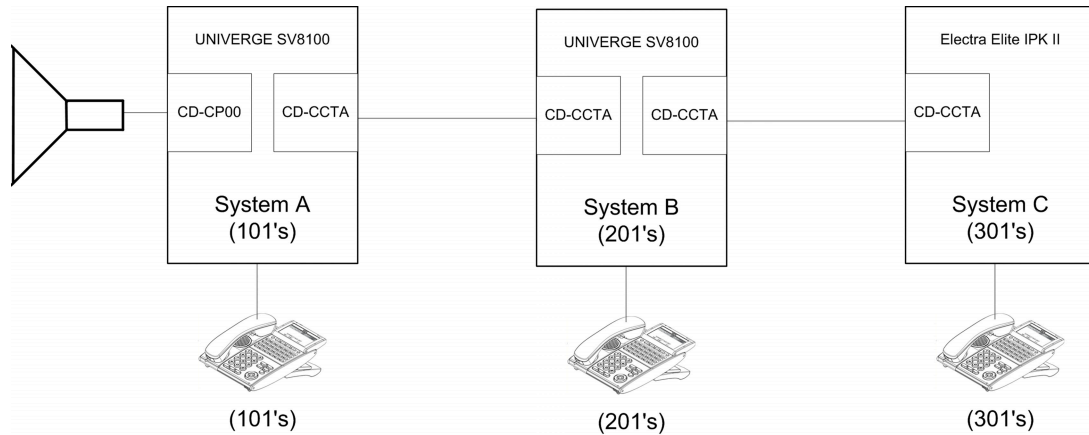
Paging Access for System A = 521(0~9)

Paging Access for System B = 522(0~9)

Paging Access for System C = 523(0~9)

System A (101s)	System B (201s)	System C (301s)
PRG 11-01-01 51 = 2 digits: F-Route	PRG 11-01-01 51 = 2 digits: F-Route	MB 1-1-47 51 = 301
PRG 11-12-20 External Paging = 703 (Page System A)	PRG 11-12-20 External Paging = 703 (Page System B)	MB 1-1-48 Table 01 (3) = 075 (Page System C)
PRG 44-02-01 Table 1 = Dial 511 (Page System A)	PRG 44-02-01 Table 1 = Dial 512 (Page System B)	MB 1-1-48 Table 01 (1) = 604 (ARS Table 4) (Page System A)
PRG 44-02-01 Table 2 = Dial 512 (Page System B)	PRG 44-02-01 Table 2 = Dial 511 (Page System A)	MB 1-1-48 Table 01 (2) = 604 (ARS Table 4) (Page System B)
PRG 44-02-01 Table3 = Dial 513 (Page System C)	PRG 44-02-01 Table3 = Dial 513 (Page System C)	MB 1-14-01 Table 4, No. 02 = 511
PRG 44-02-02 Table 1 = F-Route Table 2 = F-Route Table 3 = F-Route	PRG 44-02-02 Table 1 = F-Route Table 2 = F-Route Table 3 = F-Route	MB 1-14-01 Table 4, No. 02 = 512
PRG 44-02-03 Table 1 = F-Route 1 Table 2 = F-Route 2 Table 3 = F-Route 3	PRG 44-02-03 Table 1 = F-Route 1 Table 2 = F-Route 2 Table 3 = F-Route 3	MB 1-14-02 Table 4, No. 01 = Yes

System A (101s)	System B (201s)	System C (301s)
PRG 44-05-01 Table 1 = TRK GP 255 Table 2 = TRK GP 10 Table 3 = TRK GP 10	PRG 44-05-01 Table 1 = TRK GP 255 Table 2 = TRK GP 10 Table 3 = TRK GP 11	MB 1-14-03 Table 4, No. 01 = RT 01
PRG 44-05-02 Table 1 = Delete Digits 3 Table 2 = Delete Digits 0 Table 3 = Delete Digits 0	PRG 44-05-02 Table 1 = Delete Digits 3 Table 2 = Delete Digits 0 Table 3 = Delete Digits 0	MB 1-14-03 Table 4, No. 02 = RT 01
PRG 44-05-03 Table 1 = Add Dial 1 Table 2 = Add Dial 0 Table 3 = Add Dial 0	PRG 44-05-03 Table 1 = Add Dial 1 Table 2 = Add Dial 0 Table 3 = Add Dial 0	MB 1-14-04 RT No. 01 TG 10
PRG 44-05-09 Table 1 = Max Digit 0 Table 2 = Max Digit 4 Table 3 = Max Digit 4	PRG 44-05-09 Table 1 = Max Digit 0 Table 2 = Max Digit 4 Table 3 = Max Digit 4	MB 1-14-05 RT No. 01 = Delete 00
PRG 44-06-01 Table 1 = Dial 703	PRG 44-06-01 Table 1 = Dial 703	MB 1-14-06 RT No. 01 = Add None MB 1-14-07 RT No. 01 = Max Digit 04

Programming Example (Internal)

System A (101's)	System B (201's)	System C (301's)
Program 11-01-01 51 = 2 digits; F-Route	Program 11-01-01 51 = 2 digits; F-Route	Program 11-01-01 51 = 2 digits; F-Route
Program 11-12-20 External Paging = 703 (Page System A)	Program 11-12-20 External Paging = 703 (Page System B)	Program 11-12-20 External Paging = 703 (Page System C)
Program 44-02-01 Table 1 = Dial 511 (Page System A)	Program 44-02-01 Table 1 =Dial 512 (Page System B)	Program 44-02-01 Table 1 = Dial 513 (Page System C)
Program 44-02-01 Table 2 = Dial 512 (Page System B)	Program 44-02-01 Table 2 = Dial 511 (Page System A)	Program 44-02-01 Table 2 = 511 (Page System A)
Program 44-02-01 Table 3 = Dial 513 (Page System C)	Program 44-02-01 Table 3 = Dial 513 (Page System C)	Program 44-02-01 Table 3 = Dial 512 (Page System B)
Program 44-02-02 Table 1 = F-Route Table 2 = F-Route Table 3 = F-Route	Program 44-02-02 Table 1 = F-Route Table 2 = F-Route Table 3 = F-Route	Program 44-02-02 Table 1 = F-Route Table 2 = F-Route Table 3 = F-Route
Program 44-02-03 Table 1 = F-Route 1 Table 2 = F-Route 2 Table 3 = F-Route 3	Program 44-02-03 Table 1 = F-Route 1 Table 2 = F-Route 2 Table 3 = F-Route 3	Program 44-02-03 Table 1 = F-Route 1 Table 2 = F-Route 2 Table 3 = F-Route 3
Program 44-05-01 Table 1 = TRK GP 255 Table 2 = TRK GP 10 Table 3 = TRK GP 10	Program 44-05-01 Table 1 = TRK GP 255 Table 2 = TRK GP 10 Table 3 = TRK GP 11	Program 44-05-01 Table 1 = TRK GP 255 Table 2 = TRK GP 10 Table 3 = TRK GP 10
Program 44-05-02 Table 1 = Delete Digits 3 Table 2 = Delete Digits 0 Table 3 = Delete Digits 0	Program 44-05-02 Table 1 = Delete Digits 3 Table 2 = Delete Digits 0 Table 3 = Delete Digits 0	Program 44-05-02 Table 1 = Delete Digits 3 Table 2 = Delete Digits 0 Table 3 = Delete Digits 0
Program 44-05-03 Table 1 = Add Dial 1 Table 2 = Add Dial 0 Table 3 = Add Dial 0	Program 44-05-03 Table 1 = Add Dial 1 Table 2 = Add Dial 0 Table 3 = Add Dial 0	Program 44-05-03 Table 1 = Add Dial 1 Table 2 = Add Dial 0 Table 3 = Add Dial 0
Program 44-05-09 Table 1 = Max Digit 0 Table 2 = Max Digit 4 Table 3 = Max Digit 4	Program 44-05-09 Table 1 = Max Digit 0 Table 2 = Max Digit 4 Table 3 = Max Digits 4	Program 44-05-09 Table 1 = Max Digit 0 Table 2 = Max Digit 4 Table 3 = Max Digit 4
Program 44-06-01 Table 1 = Dial 703	Program 44-06-01 Table 1 = Dial 703	Program 44-06-01 Table 1 = Dial 703

Programming Example (Internal)

System A (101's)	System B (201's)	System C (301's)
Program 11-01-01 52 = 2 digits; F-Route	Program 11-01-01 52 = 2 digits; F-Route	Program 11-01-01 52 = 2 digits; F-Route
Program 11-12-19 External Paging = 701 (Page System A)	Program 11-12-19 External Paging = 701 (Page System B)	Program 11-12-19 External Paging = 701 (Page System C)
Program 44-02-01 Table 1 = Dial 521 (Page System A)	Program 44-02-01 Table 1 =Dial 522 (Page System B)	Program 44-02-01 Table 1 =Dial 523 (Page System C)
Program 44-02-01 Table 2 = Dial 522 (Page System B)	Program 44-02-01 Table 2 = Dial 521 (Page System A)	Program 44-02-01 Table 2 = Dial 521 (Page System A)
Program 44-02-01 Table 3 = Dial 523 (Page System C)	Program 44-02-01 Table 3 = Dial 523 (Page System C)	Program 44-02-01 Table 3 = Dial 522 (Page System B)
Program 44-02-02 Table 1 = F-Route Table 2 = F-Route Table 3 = F-Route	Program 44-02-02 Table 1 = F-Route Table 2 = F-Route Table 3 = F-Route	Program 44-02-02 Table 1 = F-Route Table 2 = F-Route Table 3 = F-Route
Program 44-02-03 Table 1 = F-Route 1 Table 2 = F-Route 2 Table 3 = F-Route 3	Program 44-02-03 Table 1 = F-Route 1 Table 2 = F-Route 2 Table 3 = F-Route 3	Program 44-02-03 Table 1 = F-Route 1 Table 2 = F-Route 2 Table 3 = F-Route 3
Program 44-05-01 Table 1 = TRK GP 255 Table 2 = TRK GP 10 Table 3 = TRK GP 10	Program 44-05-01 Table 1 = TRK GP 255 Table 2 = TRK GP 10 Table 3 = TRK GP 11	Program 44-05-01 Table 1 = TRK GP 255 Table 2 = TRK GP 10 Table 3 = TRK GP 10
Program 44-05-02 Table 1 = Delete Digits 3 Table 2 = Delete Digits 0 Table 3 = Delete Digits 0	Program 44-05-02 Table 1 = Delete Digits 3 Table 2 = Delete Digits 0 Table 3 = Delete Digits 0	Program 44-05-02 Table 1 = Delete Digits 3 Table 2 = Delete Digits 0 Table 3 = Delete Digits 0
Program 44-05-03 Table 1 = Add Dial 1 Table 2 = Add Dial 0 Table 3 = Add Dial 0	Program 44-05-03 Table 1 = Add Dial 1 Table 2 = Add Dial 0 Table 3 = Add Dial 0	Program 44-05-03 Table 1 = Add Dial 1 Table 2 = Add Dial 0 Table 3 = Add Dial 0
Program 44-05-09 Table 1 = Max Digit 0 Table 2 = Max Digit 4 Table 3 = Max Digit 4	Program 44-05-09 Table 1 = Max Digit 0 Table 2 = Max Digit 4 Table 3 = Max Digits 4	Program 44-05-09 Table 1 = Max Digit 0 Table 2 = Max Digit 4 Table 3 = Max Digits 4
Program 44-06-01 Table 1 = Dial 701	Program 44-06-01 Table 1 = Dial 701	Program 44-06-01 Table 1 = Dial 701

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Quick Transfer to Voice Mail – K-CCIS

FEATURE DESCRIPTION

A station user transferring a call can force the call to be transferred to the called party voice mail box after the transferred call recalls, after an internal station number is dialed while performing a screened transfer, or during intercom calls.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals allow either operation.

Single line telephones may perform the Quick Transfer only during screened transfer operations. They may not perform Quick Transfer after recall.

Required Components

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

Quick Transfer Across K-CCIS:

To Quick Transfer a call while talking with an outside or internal party:

1. Press Transfer, and wait for an internal dial tone.

2. Enter a station number, and wait for a ring back tone.

3. Dial the Quick Transfer Access Code (default: 7). The outside party is transferred to the station user Voice Mail box.

4. Hang up.

5. The Voice Mail answers.

To leave a message using Quick Transfer to voice mail during an intercom call:

1. Make the intercom call.

2. Dial the Quick Transfer Access Code (default: 8).

3. Leave a voice mail message.

4. Hang up.

SERVICE CONDITIONS

General:

❍ The Quick Transfer to Voice Mail feature is allowed when:

❏ Listening to the ring back tone (RBT)

❏ Listening to the call waiting tone (CWT)

❏ In Handsfree Answerback Mode

❏ In Voice Over Mode

❍ This feature is allowed from a single line telephone (SLT) until the PBR times out (default: 10 sec).

❍ An SLT may perform the Quick Transfer only during screened transfer operations.

❍ InMail voice mail is not supported in a KTS to KTS network for K-CCIS.

RELATED FEATURE LIST

❍ Digital Voice Mail

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-16-09	Single Digit Service Code Setup – Access to Voice Mail	Default is 8	Assign access code for Quick Transfer to Voice Mail.
11-07-01	Department Group Pilot Numbers	Tel Groups 1~64 Dial up to eight digits. Default is No Setting	Assign pilot number to each Department Group set up in PRG 16-02. Extension numbers are assigned in PRGs 11-02, 11-04, 11-06, and 11-08.
15-03-01	Single Line Telephone Basic Data Setup – SLT Signaling Type	0 = DP 1 = DTMF Default is 1	Tell the system which dialing is used by the connected telephone users. For each Voice Mail extension this option must be 0.
15-03-03	Single Line Telephone Basic Data Setup – Terminal Type	0 = Normal 1 = Special Default is 0	Enter 1 to allow a single line port to receive DTMF tones after the initial call setup.
45-01-01	Voice Mail Integration Options – Voice Mail Department Group Number	0 = No Voice Mail 0~64 Groups Default is 0	Assign Department Group Number as the Voice Mail Group.
45-01-14	Voice Mail Integration Options – CCIS Centralized Voice Mail Number	Up to eight digits Default not assigned. Default is No Setting	Assign the CCIS Centralized Voice Mail Pilot Number for Remote Sites.

Programming Example
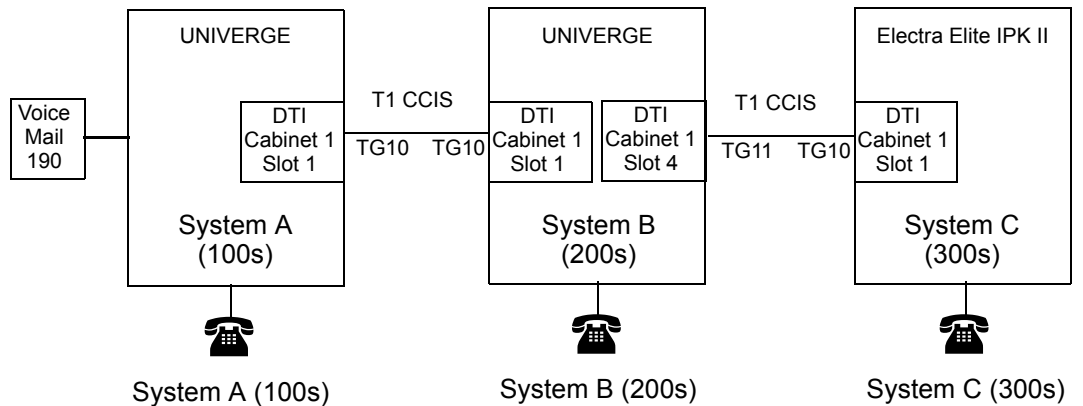
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Station-to-Station Calling – K-CCIS

FEATURE DESCRIPTION

This feature permits any multiline terminal user to dial another multiline terminal directly through a K-CCIS network.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

Normal call handling procedures apply.

SERVICE CONDITIONS

General:

❍   If the called station is off-hook and has Call Queuing disabled, the originating station receives a busy tone. If the called station is idle, the called station rings and the caller hears ringback tone.

❍   If the called station is off-hook on a call and has Call Queuing enabled, the originating station receives ringback tone and the called station receives call alert tone.

❍   Station-to-Station Calling between tenants in the K-CCIS network is not restricted.

❍   The release process is First Party Release.

Restrictions:

❍   The same telephone numbers cannot be duplicated in the same system.

RELATED FEATURE LIST

❍   Call Transfer – All Calls - K-CCIS

❍   Calling Name Display – K-CCIS

❍   Calling Number Display – K-CCIS

❍   Dual Hold – K-CCIS

❍   Elapsed Time Display – K-CCIS

❍   Flexible Numbering of Stations – K-CCIS

❍   Hands-Free Answerback – K-CCIS

❍   Key-Common Channel Interoffice Signaling (K-CCIS)

❍   Uniform Numbering Plan – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Memory Blocks that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-01-01	System Numbering – Extension Number	1X = 3 Digit; Intercom	Assign the system internal (intercom) numbering plan.
11-02-01	Extension Numbering	Default: Port 1~99 = 101~199 Port 100~512 = 3101~3513	Assign the Extension Numbers.

Uniform Numbering Plan – K-CCIS

FEATURE DESCRIPTION

In a K-CCIS network, a Uniform Numbering Plan enables a multiline terminal user to call any other multiline terminal in the network. Two types of numbering plans are provided. In the first plan, the station user dials any telephone number from two to eight digits. The location of the office is identified by the first digit or first two digits of the telephone number. In the second plan, the station user dials a one-, two- or three-digit office code and a telephone number from two to eight digits.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

To call a station at another office using Numbering Plan 1 (Closed Numbering Plan):

1.   Lift the handset or press Speaker.

2.   Dial the remote K-CCIS station number.

To call a station at another office using Numbering Plan 2 (Open Numbering Plan):

1. Lift the handset or press Speaker.

2. Dial the trunk Access Code (normally 8).

3. Dial the Office Code number.

4. Dial the remote K-CCIS station number.

SERVICE CONDITIONS

General:

❍   In a closed numbering plan, the location of the office can be identified by the first digit or first two digits of the telephone number.

❍   In an open numbering plan, each office in the K-CCIS network is assigned a one-, two- or three-digit office code and each station in the office is assigned telephone numbers from two to eight digits.

❍   In the same office, a station-to-station call is made by dialing the telephone number of the desired station.

Restrictions:

❍   For a Closed Numbering Plan network, a maximum of 255 systems can be connected per K-CCIS Network.

❍   When a Closed Numbering plan is used the extensions in the network cannot have the same prefix number.

❍   For an Open Numbering Plan network, the Automatic Route Selection (ARS)/ Flexible Routing (F-Route) feature must be used to place Station-to-Station calls over K-CCIS.

❍   When an Open Numbering plan is used, the extensions in the network can have the same prefix number, however the office location number cannot be the same.

RELATED FEATURE LIST

❍   Call Transfer – All Calls – K-CCIS

❍   Flexible Numbering of Stations – K-CCIS

❍   Key-Common Channel Interoffice Signaling (K-CCIS)

❍   Station-to-Station Calling – K-CCIS

GUIDE TO FEATURE PROGRAMMING

Refer to Chapter 3 System Data Programming.

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Voice Call – K -CCIS

FEATURE DESCRIPTION

This feature provides a voice path, through the K-CCIS network, between a D^{term} in one office and a D^{term} in another office. This path is established from the *calling* party to the *called* party built-in speaker. If the called party MIC is on, the called party can converse hands-free.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

From one D^{term} to another D^{term}:

1. The originating D^{term} user dials the desired station number in a different office and receives ring back tone.

2. Calling party dials 1. A signal tone is transmitted over the K-CCIS network to the called party speaker.

3. The called party can press MIC, or press FEATURE and dial 1 (if the MIC LED is not on) to allow two-way conversation with the calling party.

SERVICE CONDITIONS

General:

❍ The UNIVERGE SV8100 can assign a Feature Access/One Touch Button as a Voice Call key. This performs the same operation as pressing 1.

❍ Any station in the same system can use Directed Call Pick Up to retrieve the Voice Call over K-CCIS.

❍ When a Voice Call is sent to a station that is unable to receive voice announcement, RST is displayed on the originator display.

❍ During Voice Call, the ICM Key is flashing (Red).

Restrictions:

❍ The calling party must wait for at least one ring back before Voice Call is attempted.

❍ After the calling party changes ring back to Voice Call, it cannot be changed back to tone.

❍ Voice Call cannot be set as the initial call across K-CCIS.

❍ Group Call Pick Up is not allowed to retrieve voice calls over K-CCIS.

❍ Single Line terminals can be used to originate a Voice Call Over K-CCIS. However, they are not allowed to receive a voice call.

RELATED FEATURE LIST

❍ Station-to-Station Calling – K-CCIS

❍ Handsfree Answerback – K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Programs that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-12-06	Service Code Setup (for Service Access) – Switching Voice Call and Signal Call	Default is 712	Assigns the access code used to toggle ICM call between Handsfree Answerback and Forced Intercom Ringing for Outgoing Intercom calls.

Voice Mail Integration – K-CCIS

FEATURE DESCRIPTION

This feature allows any station user in the K-CCIS network to use the Voice Mail System (VMS) in another office in the K-CCIS network.

For more details, refer to the UNIVERGE SV8100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All Stations

REQUIRED COMPONENTS:

CD-VM00

CD-CCTA

- OR -

PZ-(X)IPLA

OPERATING PROCEDURES

To access voice mail from a Multiline Telephone in the Main system:

1. Lift the handset or press Speaker.

2. Dial pilot number for voice mail.

3. When voice mail answers use softkeys to navigate.

   - OR -

4. Wait for softkeys to time out and listen to voice prompts to navigate.

5. When finished hang up.

To access voice mail from a Multiline Telephone in the Remote system:

1. Lift the handset or press Speaker.

2. Dial extension number for voice mail.

3. When voice mail answers listen to voice prompts to navigate.

4. When finished hang up.

To program a One-Touch/Feature Access key for easy message access:

1. Press Feature.

2. Dial 751.

3. Press One-Touch/Feature Access key.

4. Dial 1, followed by Voice Mail extension number.

5. Press Hold.

SERVICE CONDITIONS

General:

❍ Any station or Call Arrival (CAR) key can be set for Call Forwarding – Busy/No Answer to voice mail.

❍ The following features **are** supported for voice mail users in remote systems:

   ❏ Message Waiting Indication

   ❏ Automated Attendant

   ❏ Auto Login

   ❏ Call Forward – Busy/No Answer to voice mail

   ❏ Call Forward – All Call to voice mail

❍ A voice mail with at least eight ports should be used in any K-CCIS system with a shared voice mail.

Restrictions:

❍　In the voice mail, only release transfer type is supported for mail boxes of stations in Remote systems.

❍　In a KTS to KTS Network, only digital voice mails are supported for K-CCIS.

❍　In a KTS to KTS network, Centralized Voice Mail is supported only via closed numbering plan and only up to 4-digit station numbers.

❍　In-Mail voice mail is not supported in a KTS to KTS network for K-CCIS.

❍　In a PBX to KTS network, Centralized Voice Mail is supported only via closed numbering plan.

❍　In a PBX to KTS network, Centralized Voice Mail is supported using the PBX voice mail.

❍　Single Line Telephones (SLTs) connected to the AP(A)-R Unit/AP(R)-R Unit or APA-U Unit/APR-U Unit cannot be used to transfer a Trunk call across the K-CCIS Network to another station or Voice Mail.

❍　When a call is forwarded to voice mail by multiple call forwarding, the message is left in the mailbox of the first forwarded station.

❍　Call Forward – Off-Premise must be allowed in Class of Service Feature Selection to set call forwarding to main K-CCIS voice mail.

❍　Trunk-to-Trunk Transfer must be allowed in Program 14-01-13 (Trunk-to-Trunk Transfer Yes/No Selection).

❍　A remote system can have only Message Waiting LED on Line key for extensions in the remote system. Remote system users cannot press a flashing Line key to route to voice mail or the message box of an extension on Message Waiting LED on the Line key.

❍　The following features are **not** supported for voice mail users in remote systems:

　　❏　Live Record

　　❏　Live Monitoring

　　❏　Caller ID Display

　　❏　Softkeys

　　❏　Await Answer transfer from voice mail

　　❏　Call Holding

❏   Constant Message Count Indication

❏   Call Back to VM

❏   Live Transfer (Caller ID Return Call)

❍ The Dial Access Code for Single Line Telephone Hookflash is supported for trunk calls into the main system only.

❍ The voice mail must be installed in the PBX (NEAX system) when 5-, 6-, or 7-digit station numbers are used.

❍ Centralized Voice Mail and Local Voice Mail cannot be mixed in a K-CCIS network.

RELATED FEATURE LIST

❍ Key-Common Channel Interoffice Signaling (K-CCIS)

❍ Call Forwarding – All Calls - K-CCIS

❍ Call Forwarding – Busy/No Answer - K-CCIS

❍ Multiple Call Forwarding – All Calls - K-CCIS

❍ Multiple Call Forwarding – Busy/No Answer - K-CCIS

GUIDE TO FEATURE PROGRAMMING

This guide provides a list of associated Memory Blocks that support this feature.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
10-03-01	ETU Setup – Logical Port Number	VMS Unit Default is 0	The CPUII automatically defines each PCB during installation.
11-01-01	System Numbering	1 = Service Code 2 = Extension Number 3 = Trunk Access 4 = Special Trunk access 5 = Operator Access 6 = Flexible Routing	Defaults for 1X, 2X, and 3X = 2 Extension Number.
11-02-01	Extension Numbering	Assign Station Numbers to Port Numbers	Defaults for Ports 1~512: 101-199 3101-3513
16-01-02	Department Group Basic Data Setup – Department Calling Cycle	0 = Normal Routing (Priority) 1 = Easy – UCD Routing (Circular) Default is 0	Set the call routing for Department Calling.
16-01-03	Department Group Basic Data Setup – Department Routing When busy	0 = Normal (Intercom caller to busy department member hears busy) 1 = Circular (Intercom callers to busy department member route to idle member) Default is 0	Assign how the system routes an Intercom Call to a busy Department Group member.
16-01-04	Department Group Basic Data Setup – Hunting Mode	0 = Last extension is called and hunting is stopped 1 = Circular Default is 0	Assign the action taken when a call reaches the last extension in the Department Group.
16-02-01	Department Group Assignment for Extensions	Groups 1~64 Priority 1~999 Default = 1-XXX	Assign the Department Groups. The initial priority value becomes the numerical port order assigned in PRG 11-02 and 11-04 (Ports 1~256).
11-11-01	Service Code Setup (for Setup/Entry Operation) – Call Forward – All Call	Default is 741	Assign the Call Forward – All Call service Set /Cancel Service Code.
11-11-02	Service Code Setup (for Setup/Entry Operation) – Call Forward – Busy	Default is 742	Assign the Call Forward – Busy Set/ Cancel Service Code.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-11-03	Service Code Setup (for Setup/Entry Operation) – Call Forward – No Answer	Default is 743	Assign the Call Forward – No Answer Set/ Cancel Service Code.
11-11-04	Service Code Setup (for Setup/Entry Operation) – Call Forward – Busy/No Answer	Default is 744	Assign the Call Forward – Busy/No Answer Set/Cancel Service Code.
11-07-01	Department Group Pilot Numbers – Dial	UP to eight digits can be assigned. Default is No Setting.	Assign pilot numbers to each Department Group set up in PGM 16-02.
11-16-09	Single Digit Service Code Setup – Access to Voice Mail	Default is 8	Assign single digit access code for Quick Transfer to Voice Mail.
20-06-01	Class of Service for Extensions	0~15	Default is Extension 101 is in Class 15. All other extensions are in Class 1.
20-09-02	Class of Service Options (Incoming Call Service) – Caller ID Display	0 = Off 1 = On Default is 0	Enable the Caller ID display at an extension.
20-11-01	Class of Service Options (Hold/Transfer Service) – Call Forward All	0 = Off 1 = On Default is 1	Disable Call Forward – All Call at an extension.
20-11-02	Class of Service Options (Hold/Transfer Service) – Call Forward When Busy	0 = Off 1 = On Default is 1	Disable Call Forward – Busy at an extension.
20-11-03	Class of Service Options (Hold/Transfer Service) – Call Forwarding When Unanswered	0 = Off 1 = On Default is 1	Disable Call Forward – No Answer at an extension.
20-11-12	Class of Service Options (Hold/Transfer Service) – Call Forwarding Off Premise (External Call Forwarding)	0 = Off 1 = On Default is 0	Enable Call Forward – Off-Premise at an extension.
20-11-13	Class of Service Options (Hold/Transfer Service) – Operator Transfer After Hold Callback	0 = Off 1 = On Default is 0	Turn Off or On an extension user ability to have a call which recalls from hold transfer to the operator.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
20-11-14	Class of Service Options (Hold/Transfer Service) – Trunk-to-trunk Transfer Restriction	0 = Off 1 = On Default is 0	Enable Trunk-to-Trunk Transfer Restriction at an extension.
15-07-01	Programmable Function Keys	10 = Call Forward – Immediate 11 = Call Forward – Busy 12 = Call Forward – No Answer 13 = Call Forward – Busy/No Answer	Service Codes: 741 742 743 744
14-01-13	Basic Trunk Data Setup – Trunk-to-Trunk Transfer Loop Supervision	0 = Disable (No) 1 = Enable (Yes) Default is 1	Must be enabled for Trunk-to-Trunk Transfer, Call Forward – Off-Premise, and Tandem Trunking.
14-01-22	Basic Trunk Data Setup – Caller ID to Voice Mail	0 = Disable (No) 1 = Enable (Yes Default is 0	Enable the system to send Caller ID Digits to Voice Mail.
22-02-01	Incoming Call Trunk Setup	0 = Normal 1 = VRS (second dial tone if no VRS installed) 2 = DISA 3 = DID 4 = DIL 5 = E&M Tie Line 6 = Delayed VRS 7 = ANI/DNIS 8 = DID(DDI) Mode Switching Default is 0	Set the feature for the trunk being programmed.
22-07-01	DIL Assignment	Extension Number/ Pilot Number (eight digits maximum) Default is No Setting	Assign destination extension or Department Group pilot number programmed in PRG 11-07.
30-03-01	DSS Console Key Assignment	Key Numbers 001~114 00~99 = General Functional Level *00~*99 = Appearance Function Level Default is Extensions. 101~160	Customize Key Assignments for DSS Consoles 1~32.
45-01-01	Voice Mail Integration Options – Voice Mail Department Group Number	0 = No Voice Mail 0~64 Groups Default is 0	Assign an Extension Department Group as the Voice Mail Group.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
45-01-14	Voice Mail Integration Options – CCIS Centralized Voice Mail Number	Up to eight digits Default is No Setting.	Assign the CCIS Centralized Voice Mail Pilot number for remote sites.

SV8100 IP K-CCIS

This chapter describes the system outline, Hardware installation, and programming procedures for providing **IP K-CCIS** using the PZ-32IPLA/ PZ-64IPLA/PZ-128IPLA on the UNIVERGE SV8100 system.

SECTION 1   SYSTEM OUTLINE

1.1   IP K-CCIS Application using the PZ-32IPLA/PZ-64IPLA/ PZ-128IPLA

The system uses the PZ-32IPLA/PZ-64IPLA/PZ-128IPLA to connect multiple systems together over a Data Communication IP Network (Intranet). Key-Common Channel Interoffice Signaling (K-CCIS) is used to provide telephony services between the UNIVERGE SV8100 and another UNIVERGE SV8100 or a NEAX PBX system.

1.1.1   CCIS Networking via IP (Non Peer-to-Peer Connections Basis)

□   IP trunk connections over CCIS Networking via IP is used to provide telephony services between the UNIVERGE SV8100 and UNIVERGE SV8100 and a NEAX IPS, IPX, SV7000 and UNIVERGE SV8300.

□   The SV8100 uses the NEC proprietary CCIS Peer to Peer protocol over IP to communicate between system to system.

□   The PZ-(X)IPLA is required for connections between IP terminals and IP trunks. A maximum of one PZ-(X)IPLA daughter board can be accommodated per system with a maximum of 128 DSP Resources per system.
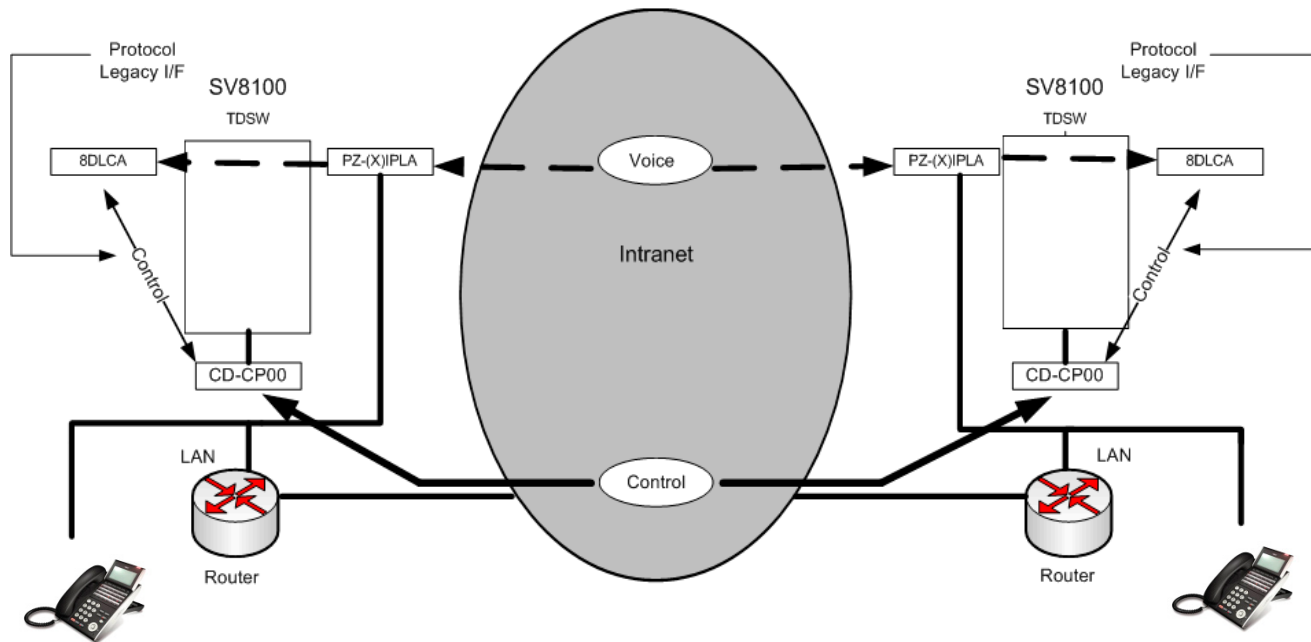
Figure 5-1 Peer to Peer One IP to TDM

1.2   Description

The PZ-(X)IPLA daughter board is an optional interface package for converting the Real Time Transfer Protocol (RTP) packets on the IP network to PCM highway. IP telephones are connected directly to the IP bus. When IP telephones are required to be connected to conventional PCM based digital circuit, the PZ-(X)IPLA daughter board converts IP packet signals. The PZ-(X)IPLA provides the digital signal processors (DSPs) for IP stations and trunks.

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

The following are examples of DSP allocation.

❍   Calling from an IP phone to a TDM phone uses one DSP.

❍   Calling from an IP phone to another IP phone that is registered to the same CPU uses no DSPs.

❍   Calling from a TDM phone to a TDM phone uses no DSPs.

❍   Calling from a TDM phone and out a IP trunk uses one DSP.

❍   Calling from a TDM phone across IP K-CCIS to another TDM phone uses one DSP.

❍   Calling from an IP Phone across IP K-CCIS to another IP Phone uses two DSP resources at each location.

1.3   Systems Requirements

Only voice (RTP/RTCP) processing functions are mounted among VOIP functions on the PZ-32IPLA/PZ-64IPLA/PZ-128IPLA, and all call control functions are handled by the CD-CP00-US.

Three kinds of hardware are offered according to the number of voice channels.

❍   PZ-32IPLA Voice channel 32
(Refer to Figure 5-2 PZ-32IPLA Daughter Board on page 5-4.)

❍   PZ-64IPLA Voice channel 64
(Refer to Figure 5-3 PZ-64IPLA Daughter Board on page 5-4.)

❍   PZ-128IPLA Voice channel 128
(Refer to Figure 5-4 PZ-128IPLA Daughter Board on page 5-5.)

Only one PZ-32IPLA/PZ-64IPLA/PZ-128IPLA blade can be mounted on the CD-CP00-US at any given time.

The PZ-32IPLA/PZ-64IPLA/PZ-128IPLA daughter board has Layer2 Switch ability, along with a Gigabit Ethernet LAN interface and RTP/RTCP packet is transmitted and received directly.

The number of ports supported by the IP K-CCIS (Peer to Peer) application depends on which PZ-32IPLA/PZ-64IPLA/PZ-128IPLA daughter board is installed on the CD-CP00-US and on the number of ports licensed in the CD-CP00-US. For example, if the CD-CP00-US is installed with a PZ-64IPLA, the maximum configuration supported by the CCISoIP Application is 64 CCISoIP channels.
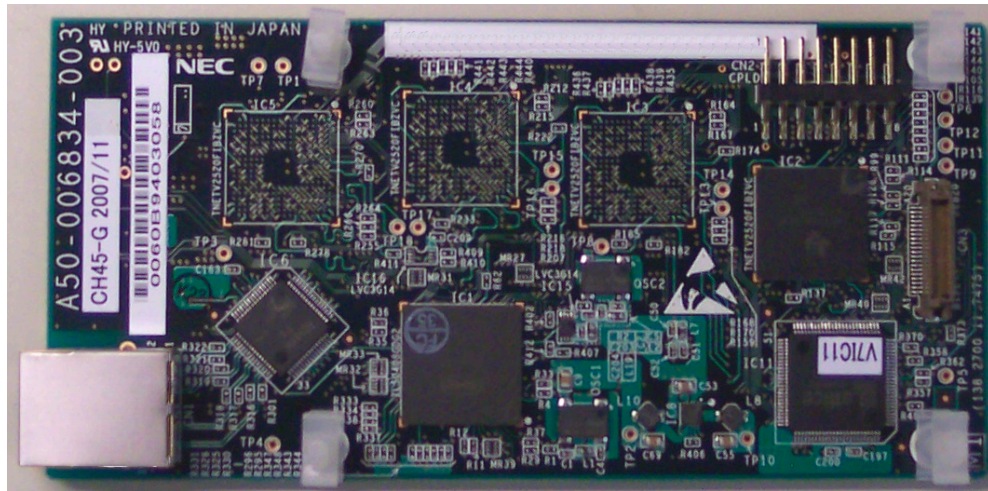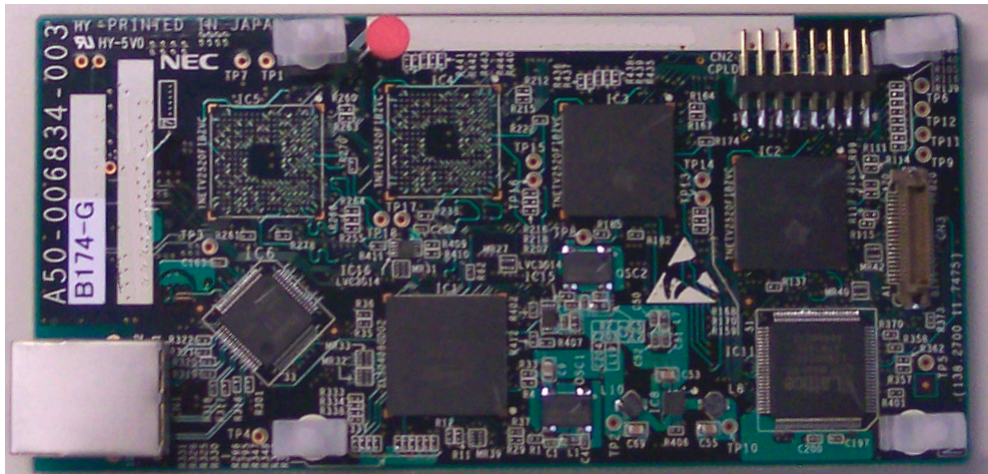
Figure 5-2 PZ-32IPLA Daughter Board

Figure 5-3 PZ-64IPLA Daughter Board

Front View   Components side
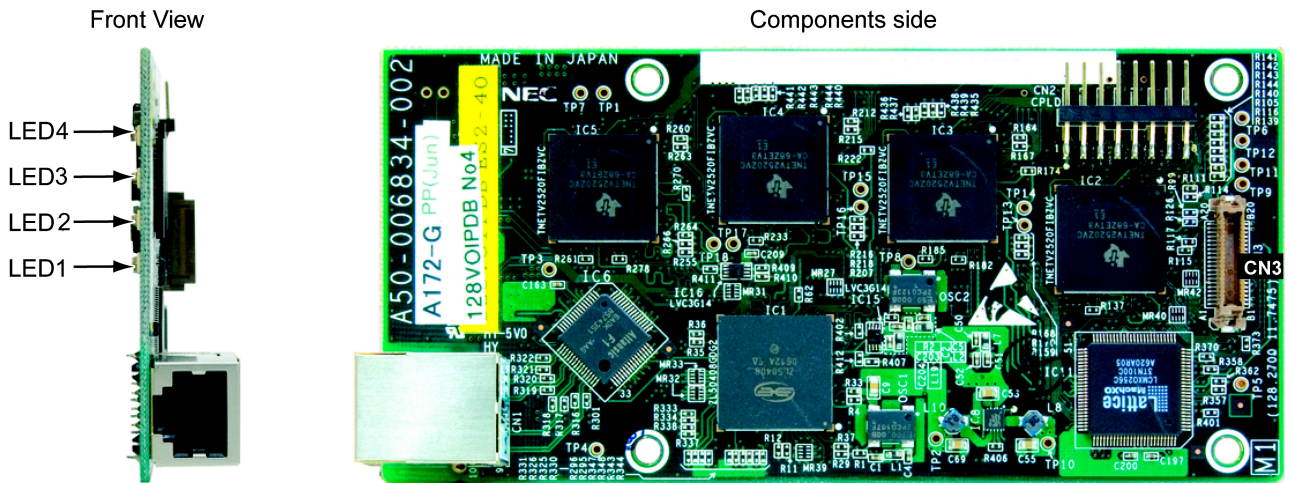
Figure 5-4 PZ-128IPLA Daughter Board

1.4   Installation

The PZ-32IPLA/PZ-64IPLA/PZ-128IPLA **daughter board** can be mounted only on the CD-CP00-US. Only one IPLA daughter board at a time can be mounted on the CD-CP00-US.

1.4.1   Hot Swap

The PZ-32IPLA/PZ-64IPLA/PZ-128IPLA are not hot swappable and cannot be removed from the CD-CP00-US without first powering down the chassis and removing the CD-CP00-US.

1.4.2   Connectors

The PZ-32IPLA/PZ-64IPLA/PZ-128IPLA has the following connectors:

❐   CN1 - RJ-45 Gigabit Ethernet LAN Interface

❐   **CN3 - Connects to the** CD-CP00-US

1.4.3   Ethernet Status

Four LEDs (two red, one green and one yellow) on the PZ-32IPLA/PZ-64IPLA/PZ-128IPLA indicate Ethernet connection status. The yellow LED is on when the Ethernet link is up. The green LED flashes to indicate activity and the two red LEDs are solid to indicate 10 Base-T/100 Base-TX or 1000 Base-T link speed.

1.4.4   Environmental Conditions

Temperature: +32 to 104 degrees Fahrenheit

Humidity: 10% ~ 90% (noncondensing)

Cable length of Ethernet cable: Up to 100m

1.4.5   Installation Procedure

Perform the following steps to install a PZ-32IPLA/PZ-64IPLA/ PZ-128IPLA on the CD-CP00-US:

> *Do not remove or install the CD-CP00-US with the power on.*

1.   Turn off the system power.

2.   Remove the CD-CP00-US, and install the PZ-32IPLA/ PZ-64IPLA/PZ-128IPLA on it.

3.   Insert the CD-CP00-US in slot 1 in the Controlling Chassis.

4.   Connect the PZ-32IPLA/PZ-64IPLA/PZ-128IPLA to the CD-RTB or to an external switching hub using an ethernet cable. Refer to the UNIVERGE SV8100 Programming Manual for detailed programming instructions.

1.4.6   Switch Settings

This daughter boards does not have any switch that needs to be set.

1.4.7   LED Indications

Refer to Figure 5-5 CD-CP00-US Blade with Daughter Boards Installed.

PZ-32IPLA
PZ-64IPLA
PZ-128IPLA   Battery

CompactFlash   PZ-VM21   PZ-ME50-
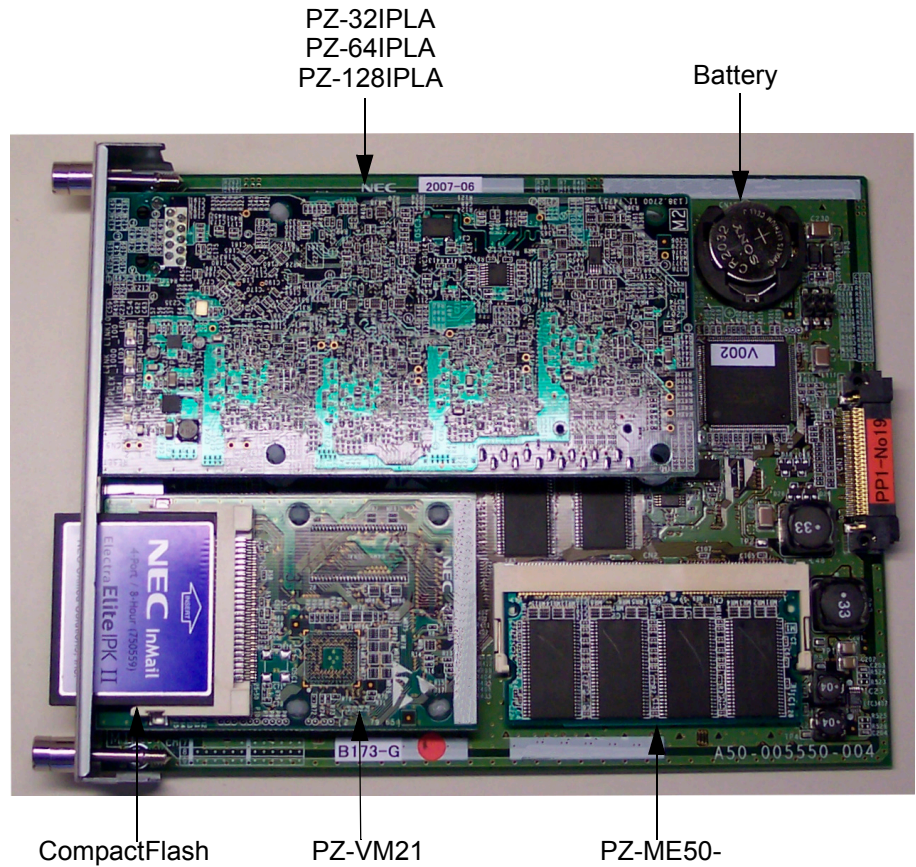
Figure 5-5 CD-CP00-US Blade with Daughter Boards Installed

See Figure 5-6 CD-CP00-US LED Locations for the location of the LEDs on the blade.
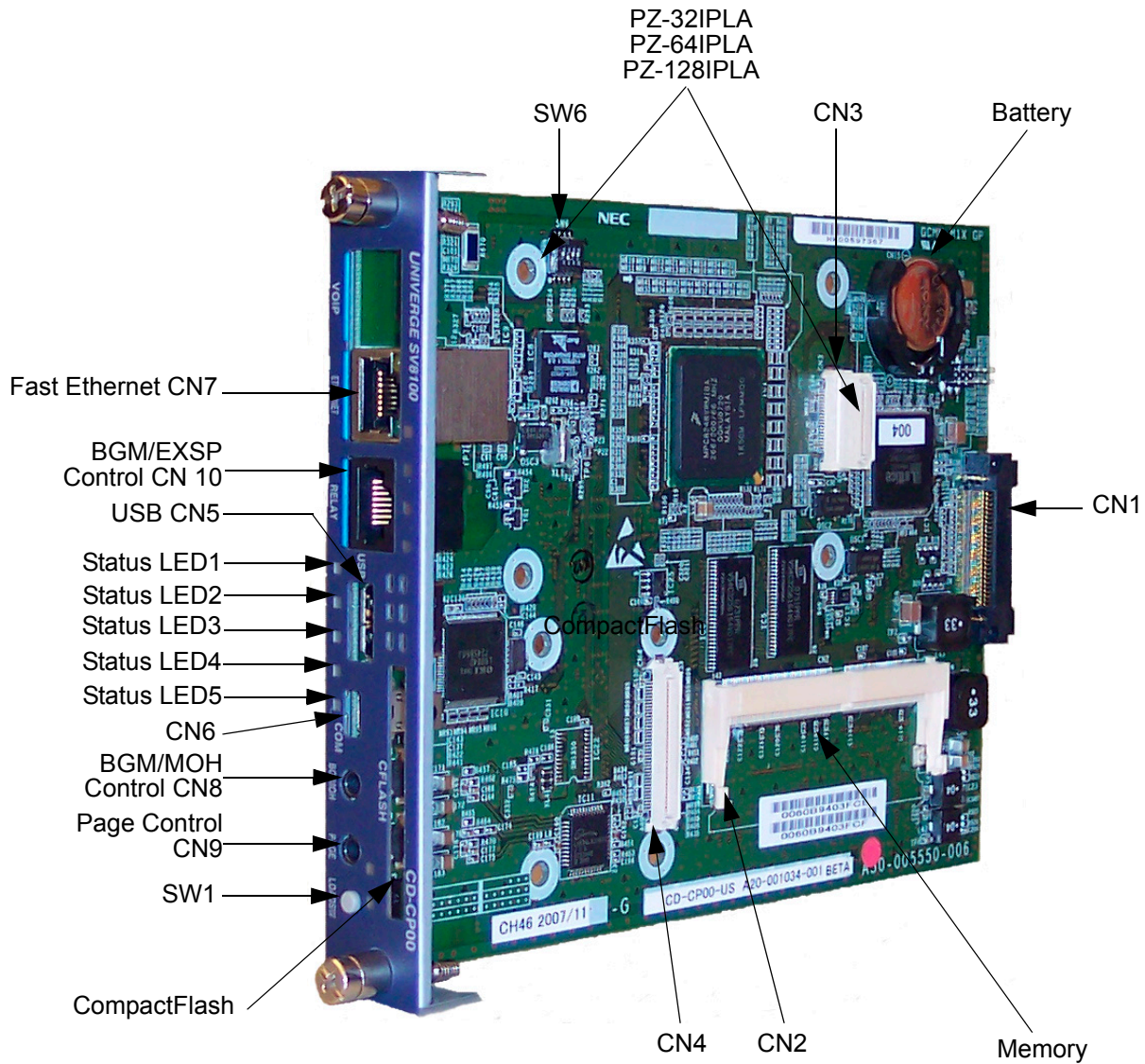
Figure 5-6 CD-CP00-US LED Locations

LED indications for the PZ-32IPLA, PZ-64IPLA or PZ-128IPLA daughter board are shown in Table 5-1 VOIPDB LED Indications. Each LED is listed with its associated function, LED status and Operation status.

Table 5-1 VOIPDB LED Indications

LED	Function	LED Status	Operation Status
Link 10/100 (LED 1)	10 Base-T/100 Base-TX link speed indicator	On Red	10 Base-T/100 Base-TX link up
Link 1000 (LED 2)	1000 Base-T link speed indicator	On Red	1000 Base-T link up
Duplex (LED 3)	Duplex Status	On Yellow	Full duplex operation
ACT (LED 4)	Link activity or data transmission and reception.	On Green	Link up completed

The following table shows the LED indication when transmitting or receiving data on CN1.

Table 5-2 VOIPDB LED CN1 Transmit/Receive Data Indications

LED	Link Up									
	Auto Negotiation Mode					Force Mode				
	1000 Mbps	100Mbps		10Mbps		1000 Mbps	100Mbps		10Mbps	
		Half	Full	Half	Full		Half	Full	Half	Full
ACT (LED4)	ON	ON	ON	ON	ON	ON	ON	ON	ON	ON
DUPLEX (LED3)	ON	OFF	ON	OFF	ON	ON	OFF	ON	OFF	ON
LINK1000 (LED2)	ON	OFF	OFF	OFF	OFF	ON	OFF	OFF	OFF	OFF
LINK10_100 (LED1)	OFF	ON	ON	OFF	OFF	OFF	ON	ON	ON	ON

1.4.8   Connectors

Figure 5-7 VoIP Connections shows a typical connection layout. Figure 5-8 Connecting a VOIPDB to a Network/PC on page 5-11 illustrates how to connect a VoIP Daughter Board to a Network or PC.

Connection Description
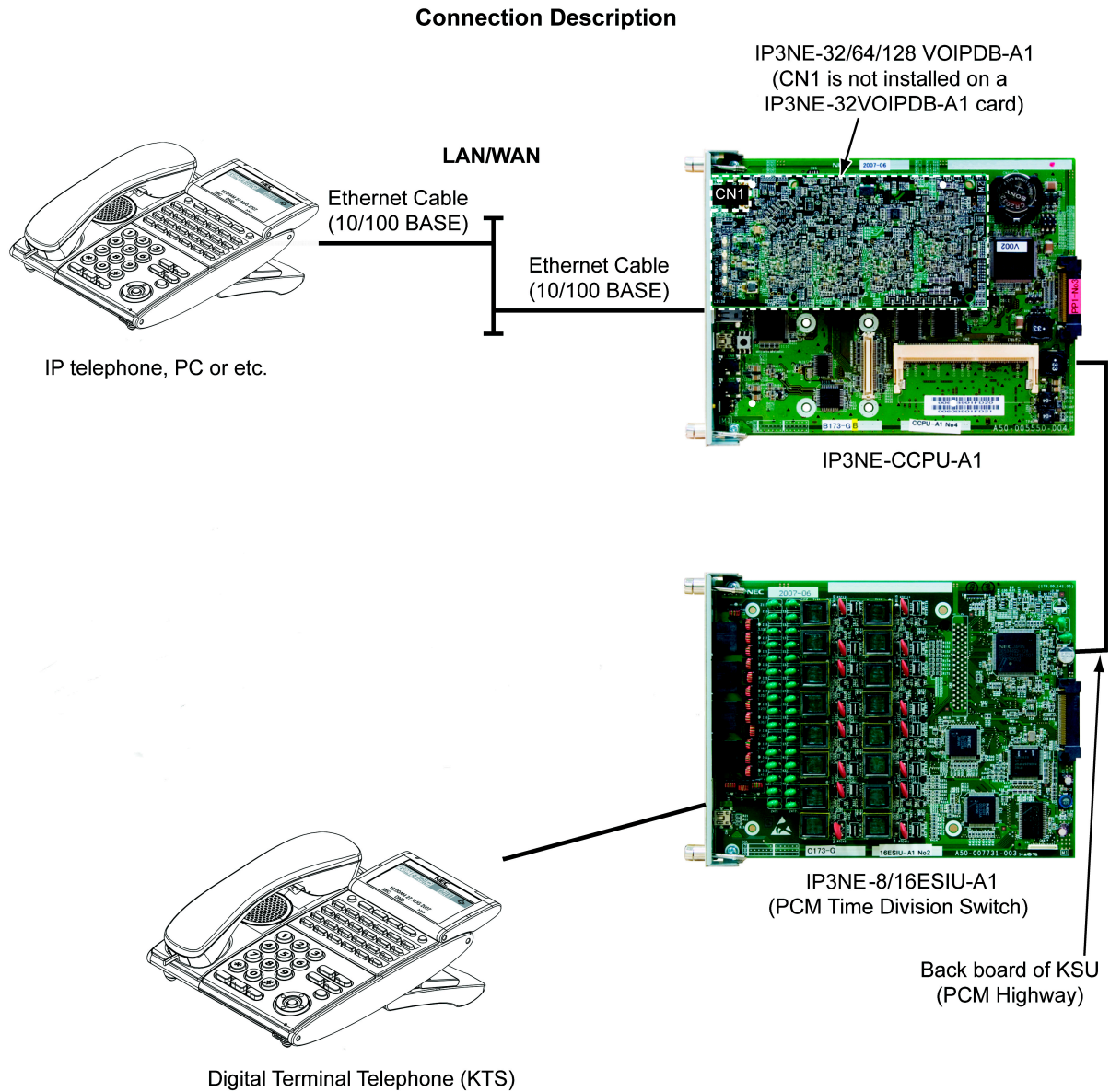
Figure 5-7 VoIP Connections

1.5 IP Addressing

JTAG Connector for CPLD
(8pin PIN HEADER)

To PC:
ispDOWNLOAD Cable
In-System Programming
Interface for the PC

To WAN / LAN network:
Ethernet cable
(CAT5E, 10BASE-T/
100BASE-TX/1000BASE-T)

Pin Number
Location

CN3

RJ61
Connector

LAN Connector
(10BASE-T/100BASE-TX/1000BASE-T)
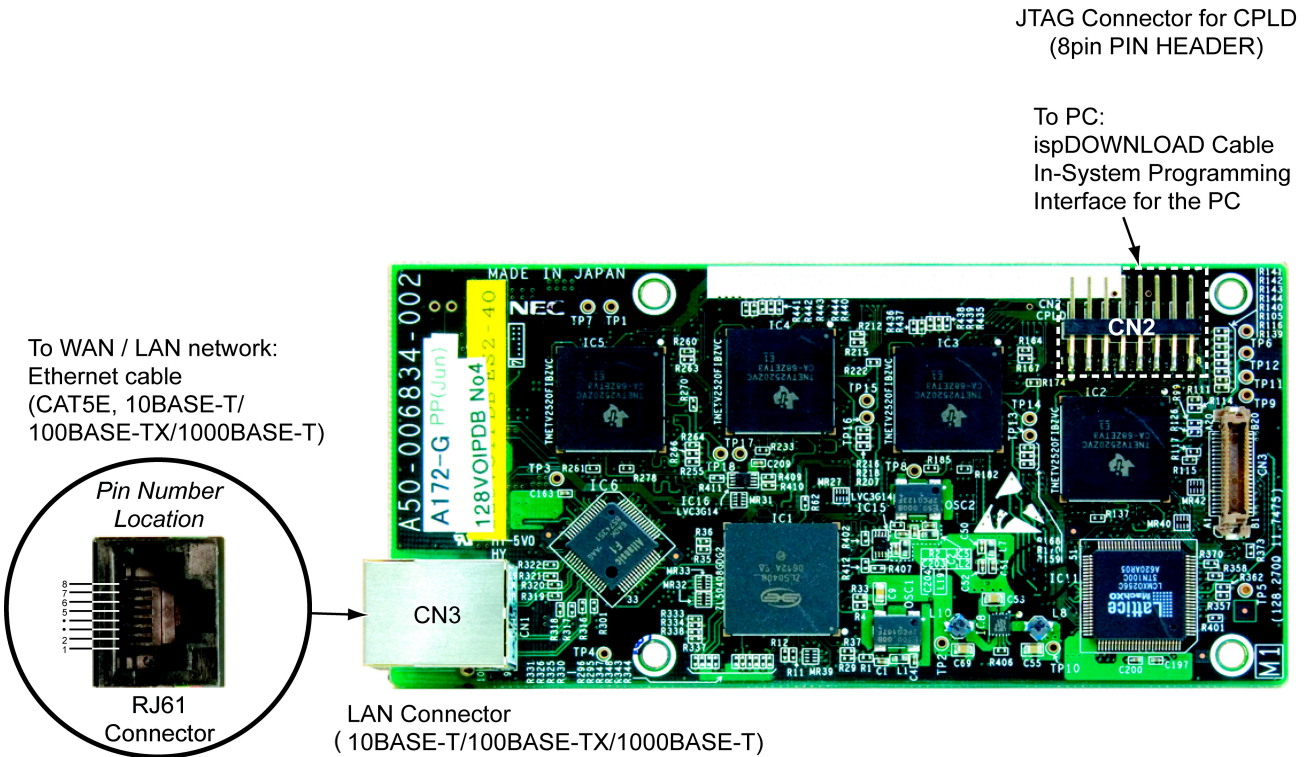
Figure 5-8 Connecting a VOIPDB to a Network/PC

One valid IP address must be assigned for every 16 DSPs that are used. In addition to the DSP IP address, another IP address is required for signaling.

For example, to realize the full functionality of the IPLA 32, three valid IP addresses in the same subnet must be assigned. One address is used for signaling information. The remaining two IP addresses are used for the voice traffic to and from the 32 DSP channels.

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

Card	Minimum IP Addresses	Maximum IP Addresses	Notes
32 IPLA	2	3	One IP address is required per 16 DSP channels. One additional IP Address is needed for signaling.
64 IPLA	2	5	
128 IPLA	2	9	

✎   When assigning the IP addresses to the IPLA card, the addresses must
     be in the same network (subnet). If the CPU is to be connected also to
     the network, it requires a separate IP address in a different network
     (subnet).

     When you have an IPLA card attached to the CPU, the CPU NIC is no
     longer required. All connections that previously terminated to the CPU
     NIC card can now be terminated to the IPLA NIC.

     For example, PC PRO, Web Pro, ACD, etc. terminate to the IPLA NIC
     card, when installed. Both the IPLA and CPU NIC share the same
     gateway assignment. The default gateway command in 10-12-03 is
     used by both NICs, allowing only one device, IPLA or CPU, to route
     outside of its own network.

1.5.1   General IP Configuration

     The voice quality of VoIP depends on variables such as available
     bandwidth, network latency, and quality of service initiatives (QOS),
     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 users IP based voice solution.
     Therefore, NEC recommends connecting the VoIP equipment
     through a local area network using private IP addresses.

     For a network to be suitable for VoIP it must pass specific
     requirements. To make sure that the site meets these requirements
     an IP ready check and a site survey **must** be completed at each site
     before VoIP implementation.

     ❐   One way delay must not exceed 100 ms

     ❐   Round Trip delay must not exceed 200 ms

     ❐   Packet loss must not exceed 1%

     ❐   Data switches must be manageable

     ❐   Routers must provide QOS

     ❐   Adequate bandwidth for estimated VoIP traffic (refer to 1.5.3
          Bandwidth)

     Depending on how QOS policies are built in the network,
     assignments might be needed in the CPU.

1.5.2   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 SV8100 supports two methods for marking the ToS byte.

❐   IP precedence

❐   DSCP (Diffserv)

IP Precedence

IP precedence uses the first three 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, the administrator can assign these precedence values with the lower values having a higher priority.

The following table shows the eight common values for IP precedence.

Table 5-3 IP Precedence Values

TOS Field	Precedence Value		Higher Number Value Correlates to Higher Priority
000	0		Routine or Best Effort
001	1		Priority
002	2		Immediate
003	3		Flash
004	4		Flash Override
005	5		Critical
006	6		Internetwork Control
007	7		Network Control

Working in conjunction with IP precedence, the next four bits in the ToS field influence the delivery of data based on the following:

❐   delay

❐   throughput

❐   reliability

❐   cost

However these fields are usually not used. The following table shows the eight bit ToS field and the associated IP precedence bits.

Table 5-4 ToS Field with 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					

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 table lists the most common DSCP code points, the binary value, and the associated name.

DSCP Code Points	DSCP Value	Name
000000	08	Best Effort (BE)
001000	8	Class Selector 1 (CS1)
001010	10	Assured Forwarding 11 (AF11)
001100	12	Assured Forwarding 12 (AF12)
001110	14	Assured Forwarding 13 (AF13)
010000	16	Class Selector 2 (CS2)
010010	18	Assured Forwarding 21 (AF21)
010100	20	Assured Forwarding 22 (AF22)
010110	22	Assured Forwarding 23 (AF23)
011000	24	Class Selector 3 (CS3)
011010	26	Assured Forwarding 31 (AF31)
011100	28	Assured Forwarding 32 (AF32)
011110	30	Assured Forwarding 33 (AF33)
100000	32	Class Selector 4 (CS4)
100001	34	Assured Forwarding 41 (AF41)
100100	36	Assured Forwarding 42 (AF42)
100110	38	Assured Forwarding 43 (AF 43)
101110	46	Expedited Forwarding (EF)
110000	48	Class Selector 6 (CS6)
111000	56	Class Selector 7 (CS7)

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

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

Assignments for the IP Precedence/Diffserv values in the system are submitted in command 84-10.

RTP/RTCP = voice packets

CCIS= signaling packets
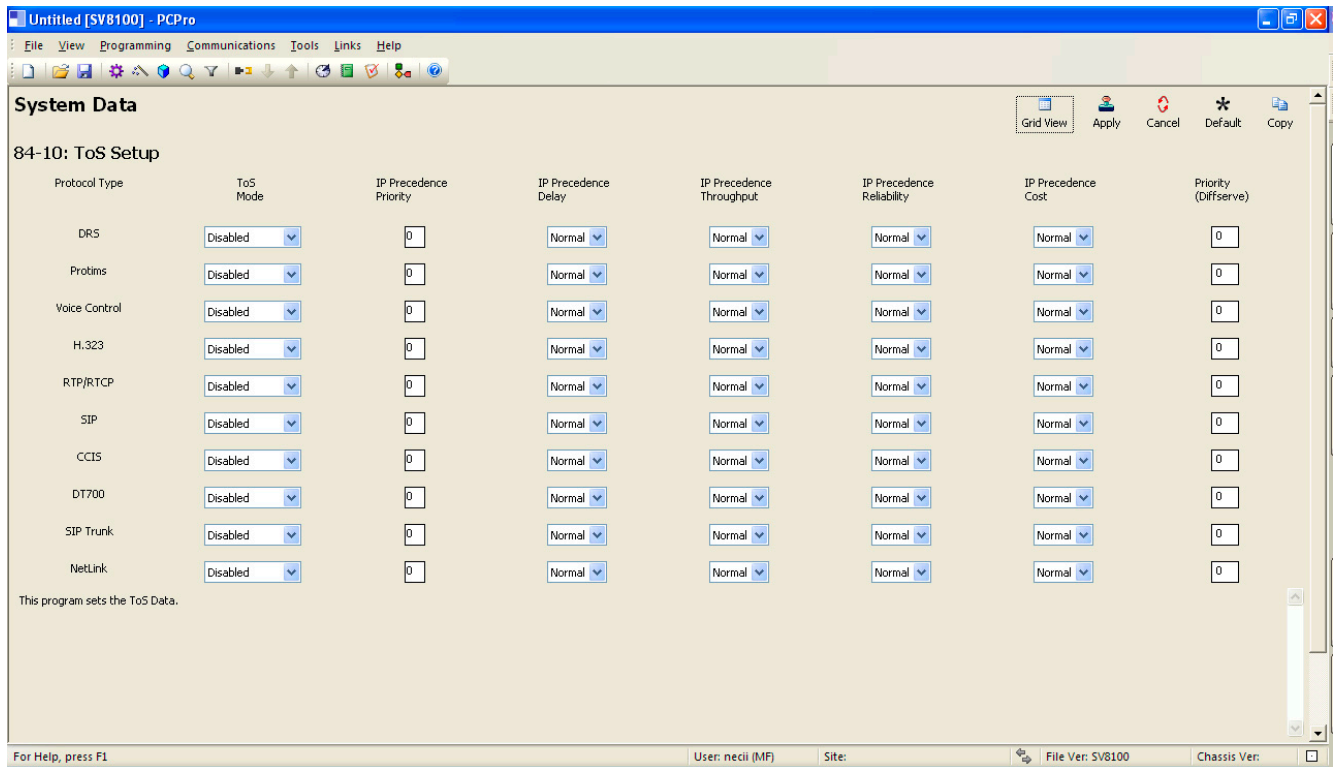
Figure 5-9 System Data 84-10 ToS Setup

1.5.3   Bandwidth

The bandwidth required for VoIP calls depends on several 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 be sent. The downside of this practice is if a packet is dropped/lost a larger piece of voice is missing from the conversation as the system waits the additional delay for the next packet arrival.

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

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

❐   VOIP packet without RTP header compression

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

❐   VOIP packet with RTP header compression

Compressed Header 2 ~ 4-bytes	VOICE-PAYLOAD

Voice Activity Detection (VAD) is suppression of silence packets from being sent across the network. In a VoIP network all conversations are packetized and sent, including silence. On an average a typical conversation contain anywhere from 35% ~ 45% silence. This can be interrupted as 35% ~ 45% transmission of VoIP packets, having no audio, use 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.

1.5.4   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 CCISoIP.

❐   G.711 = 64000 bps

❐   G.722 = 64000 bps

❐   G.729 = 8000 bps

❐   G.726 = 32000 bps

❐   G.723 = 5400 bps

1.5.5   Payload Calculation Voice

(Packet size * Codec bandwidth) / 8 = Voice Payload in Bytes

Example of G.711 with a 20 ms packet size

(.020 * 64000) /8 = 160 Bytes

Example of G.729 with a 30 ms packet size

(.030 * 8000) /8 = 30 Bytes

Now that you have the voice payload in bytes you can now 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 20 ms packet size and not using RTP header compression

(.020 * 64000) / 8 = 160 Bytes for Voice Payload

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

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

Below is a chart showing the supported CODECs, with a few
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
G.726	10	71.2 kbps	78.4 kbps
G.726	20	51.6 kbps	55.2 kbps
G.726	30	45.1 kbps	47.5 kbps
G.726	40	41.8 kbps	43.6 kbps
G.723	30	18.5 kbps	20.8 kbps
G.723	60	11.9 kbps	13.2 kbps

1.5.6   Some Network Considerations

Before adding the UNIVERGE SV8100 to a customer network, a detailed network diagram of the existing network *must* be obtained from the customer. This diagram provides information about any network condition that can prevent or hinder the VoIP equipment from functioning correctly.

Multiple Gateways

Figure 5-10 Multiple Gateways shows the UNIVERGE SV8100 assigned a default Gateway of 192.168.1.1/24. The existing network has two routers in the same broadcast domain connecting different networks. With regular data communications this is not an issue. However, the VOIP cannot function correctly in this type of network environment without specific assignments.
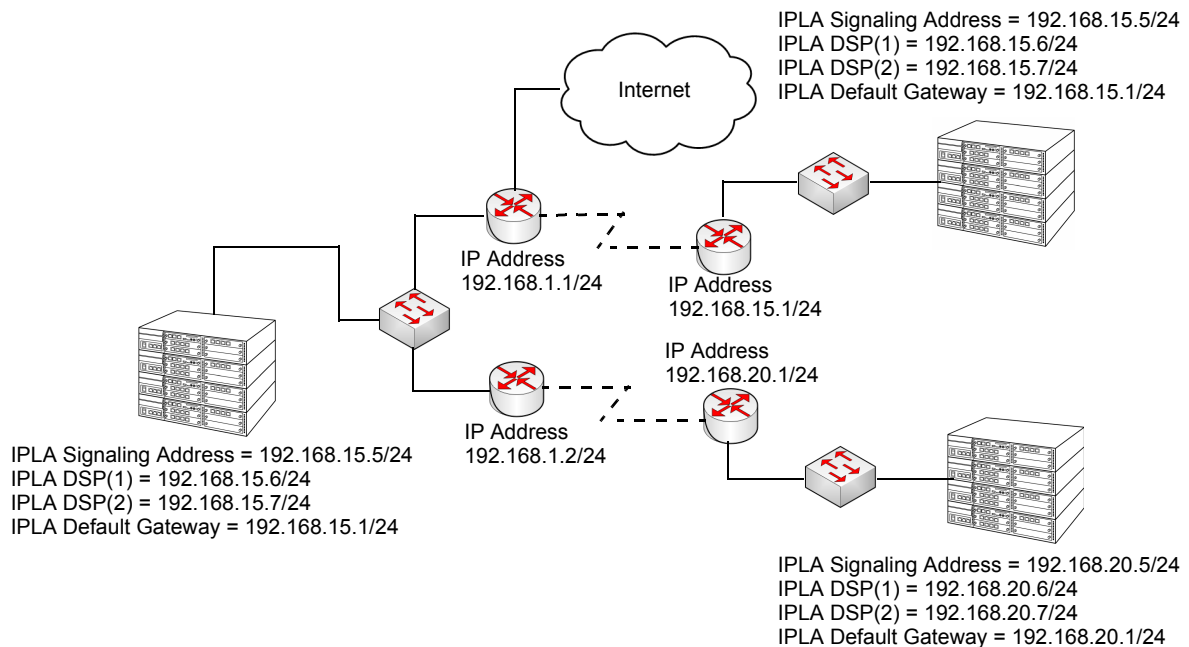
Figure 5-10 Multiple Gateways

Figure 5-10 Multiple Gateways on page 5-20 shows a configuration that allows normal operation of CCISoIP from the **192.168.1.0** network to the **192.168.15.0** network, but the **192.168.20.0** network can have considerable issues. Symptoms can include one way, or no speech path, cutoffs, to name a few. If the default gateway for the SV8100 is changed to 192.168.1.2, then the reverse occurs with the 192.168.15.0 network experiencing issues and the 192.168.20.0 network functioning correctly.

The new VoIP routing table feature (available on the SV8100 only) allows the assignment of tables to route VoIP packets to different gateway addresses. In normal operation, the SV8100 first checks the VoIP routing table for a match. If one is found, that route is selected. If no match is found, the default gateway is then used.

Figure 5-11 Example of VoIP Routing Table is an example of a VoIP routing table allowing operation for all of the IP phones in the previous diagram.
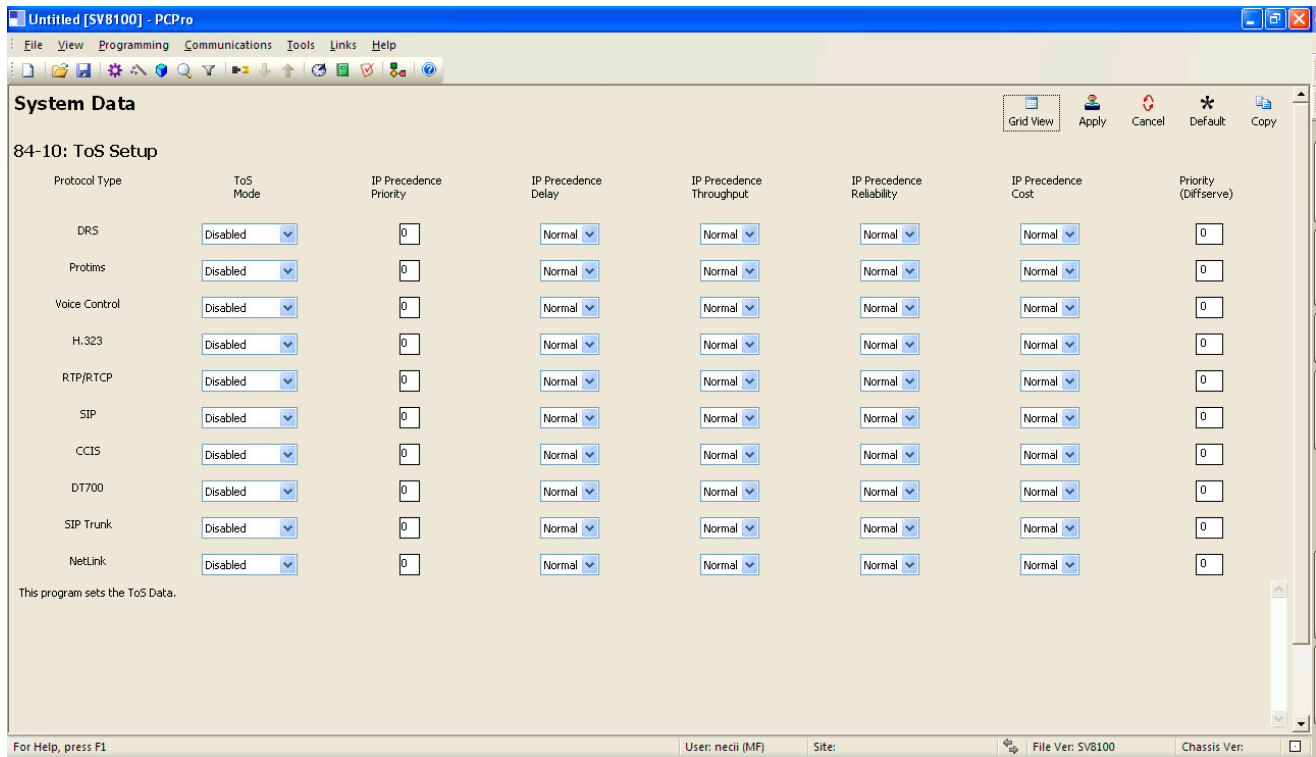
Figure 5-11 Example of VoIP Routing Table

Route table 001 has an IP address of 192.168.20.0 and a subnet mask of 255.255.255.0. This states that any IP address that is destined to 192.168.20.1 ~ 192.168.20.255 will be routed to the IP address of 192.168.1.2. There is no need to add a statement for the 192.168.15.0 network because if there is no match found in this table the packets destined for a distant network are routed to the default gateway.

Firewall

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.

Figure 5-12 Two SV8100 Systems Connected Via the WAN shows two SV8100 systems. One on the corporate local LAN and one on a Remote network connected via the WAN. The remote site cannot call the MAIN site, therefore it is not working.
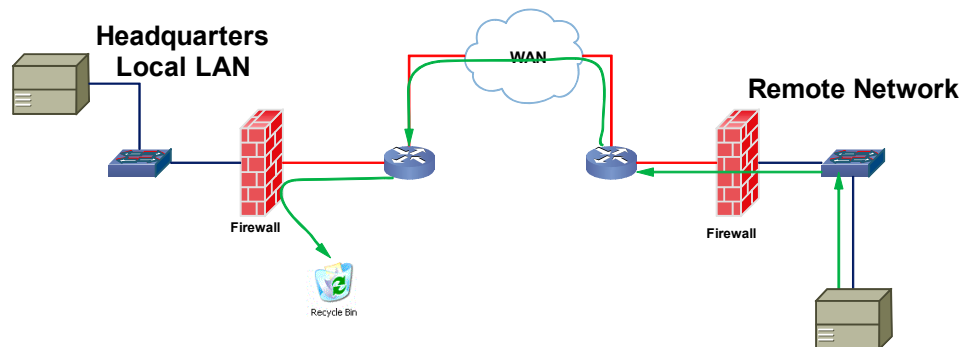
Figure 5-12 Two SV8100 Systems Connected Via the WAN

The green arrow in Figure 5-12 Two SV8100 Systems Connected Via the WAN represents the data packets leaving the REMOTE IPLA card destined for the SV81000 on the Headquarters LAN. The firewall on the Headquarters network is not configured to recognize the TCP/UDP ports utilized by the NEC equipment thus blocking them resulting in registration failure. To solve this issue the ports used by the NEC VoIP equipment have to be opened in the firewall allowing the NEC traffic to pass through to the SV8100.

The ports, **57000** and **59000** (TCP) for signaling and the voice ports, are required to be open at each location. This depends on how many IPLA ports are installed.

❑   IPLA 32 open UDP ports **10020 ~ 10083**

❑   IPLA 64 open UDP ports **10020 ~ 10147**

❑   IPLA 128 open UDP ports **10020 ~ 10275**

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.

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 5-13 VPN Network

Site A can communicate with phones on sites B and C. Phones on sites B and C can not communicate with each other.

To correct this issue, another VPN connection between sites B and C is required.

SECTION 2   UNIVERGE SV8100 CHASSIS PROGRAMMING

The following data programs are used when installing the PZ-32IPLA/PZ-64IPLA/ PZ-128IPLA daughter board for UNIVERGE SV8100 IP (K-CCIS).

✎   If any IPK address or NIC setting is changed, the system must be reset for the changes to take affect.

2.1 Digital Trunk Assignment

Program/ Item No.	Description/ Selection	Assigned Data	Comments
10-03-01	IPLA PKG Setup –Trunk Logical Port Number	0~200 Default is 0	This Program is Read Only.
10-03-02	IPLA PKG Setup – Trunk Type	0 = H.323 1 = SIP Default is 1	
10-12-03	CD-CP00-US Network Setup – Default Gateway	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	Assign the Default Gateway for the CD-CP00-US. When using the PZ-(X)IPLA, the IP Address and Subnet Mask in PRG 10-12-01 and PRG 10-12-02 remain default.
10-12-09	CD-CP00-US Network Setup – IP Address	172.16.0.10 (Slot 1) 172.16.0.13 (Slot 4) Default is 172.16.0.10	Assign the IP Address of the IPLA.
10-12-10	CD-CP00-US Network Setup – Subnet Mask	255.255.0.0 Default is 255.255.0.0	Assign the Subnet Mask of the IPLA.
10-19-01	VoIP DSP Resource Selection – VoIP DSP Resource Selection	Slot Number 1: DSP Resource Number: 01 ~ 128 0 = Common use for both IP extensions and trunks 1 = IP Extension 2 = SIP Trunk 3 = CCIS 4 = Use for NetLink 5 = Blocked Default: Resource 1 = 1 Resources 2~128 = 0	

Program/ Item No.	Description/ Selection	Assigned Data	Comments
10-40-01	IP Trunk Availability – IP Trunk Availability	0 = Disable 1 = Enable Default is 0	
10-40-04	IP Trunk Availability – Number of IP CCIS Ports	0 ~ 128 Default is 0	
14-05-01	Trunk Group	0 ~ 100 Default is 1	Assign Trunks to Trunk Groups.
22-02-01	Incoming Call Trunk Setup – Trunk Type	001 ~ 200 0 = Normal 1 = VRS (second dial tone if no VRS installed) 2 = DISA 3 = DID 4 = DIL 5 = E&M Tie line 6 = Delayed VRS 7 = ANI/DNIS 8 = DID(DDI) Mode Switching Default is 0	Set the feature type for the trunk you are programming.

2.2   VoIP IP Address Assignments

For this feature, the PZ-32IPLA/PZ-64IPLA/PZ-128IPLA daughter board is installed on the CD-CP00-US blade. The PZ-32IPLA/PZ-64IPLA/PZ-128IPLA daughter board reduces the maximum capacity of trunks in the system.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
84-26-01	IPL Basic Setup – IP Address	XXX.XXX.XXX.XXX	Assign the VOIPDB Gateway IP Address. 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

Program/ Item No.	Description/ Selection	Assigned Data	Comments
84-26-02	IPL Basic Setup – RTP Port Number	0~65534	Default: VoIP GW1 = 10020 VoIP GW2 = 10052 VoIP GW3 = 10084 VoIP GW4 = 10116 VoIP GW5 = 10148 VoIP GW6 = 10180 VoIP GW7 = 10212 VoIP GW8 = 10244
84-26-03	IPL Basic Setup – RTCP Port Number (RTP Port Number + 1)	0~65534	Default: VoIP GW1 = 10021 VoIP GW2 = 10053 VoIP GW3 = 10085 VoIP GW4 = 10117 VoIP GW5 = 10149 VoIP GW6 = 10181 VoIP GW7 = 10213 VoIP GW8 = 10245

2.3   Local Numbering Plan Assignment

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-01-01	System Numbering	0 = Not Used 1 = Service Code 2 = Extension Number 3 = Trunk Access 4 = Special Trunk Access 5 = Operator Access 6 = Flexible Routing 9 = Dial Extension Analyze	Defaults for 1X, 2X, 3X = 2 Extension Number.
11-02-01	Extension Numbering	Assign Station Numbers to Port Numbers	Defaults for Ports 1 ~ 512: Ports 1~99 = 101 ~ 199 Ports 100~512 = 3101 ~ 3513
11-04-01	Virtual Extension Numbering	Assign Station Numbers to Port Numbers	Defaults for Ports Ports 1 ~ 99 = 201 ~ 299 Ports 100 ~ 256 = 3601 ~ 3757

2.4   Closed Numbering Plan – Using Closed Number Blocks

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-01-01	System Numbering	2 = Extension Number	Defaults for 1X, 2X and 3X = 2 for Extension Number.
44-02-01	Dial Analysis Table for ARS/F-Route Access – Dial Number of digits to be analyzed by the system	Up to 8 digits can be assigned. Use line key 1 for *Don't Care* digit, @ Default is No Setting	Assign the digits to be dialed across the K-CCIS link. These digits were assigned as F-Route in Program 11-01-01.
44-02-02	Dial Analysis Table for ARS/F-Route Access – Service Type	2 = ARS/F-Route Table (F-Route) Default is No Setting	The service type 2 assigns the digits to be dialed to an F-Route. Program 44-02-03 assigns the F-Route to be used.
44-02-03	Dial Analysis Table for ARS/F-Route Access – Additional Data	2 = 0~500 0 = No Setting Default is 0	When setting data is (2), refer to Program 44-05.
44-05-01	ARS/F-Route Table – Trunk Group Number	0~100, 255 0 = No Setting 255 = Extension Call Default is 0	Select the trunk group used for the outgoing K-CCIS call.

2.5   Open Numbering Plan – Using ARS Table 1, 2, or 3

Program/ Item No.	Description/ Selection	Assigned Data	Comments
11-01-01	System Numbering	2 = Extension Number	Defaults for 1X, 2X and 3X = 2 for Extension Number.
44-02-01	Dial Analysis Table for ARS/F-Route Access – Number of digits to be analyzed by the system	Up to 8 digits can be assigned. Use line key 1 for *Don't Care* digit, @ Default is No Setting	Assign the digits to be dialed across the K-CCIS link. These digits were assigned as F-Route in Program 11-01-01.
44-02-02	Dial Analysis Table for ARS/F-Route Access – Service Type	2 = ARS/F-Route Table (F-Route) Default is 0	The service type (2) assigns the digits to be dialed to an F-Route. Program 44-02-03 assigns the F-Route to be used.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
44-02-03	Dial Analysis Table for ARS/F-Route Access – Additional Data	0~500 (0 = No Setting) Default is 0	When setting data is (2), refer to Program 44-05.
44-05-01	ARS/F-Route Table – Trunk Group Number	Trunk Groups (1~100)	Select the trunk group to be used for the outgoing K-CCIS call.
44-05-02	ARS/F-Route Table – Delete Digits	0 = No Setting 1~255 = Number of digits to delete (255 = Delete All Default is 0	Enter the number of digits to be deleted from the dialed number.
44-05-03	ARS/F-Route Table – Additional Dial Number Table	0 = No Setting 0~1000 Add Digits in Program 44-06-01 Default is 0	
44-05-04	ARS/F-Route Table – Beep Tone	0 = Off 1 = On Default is 0	Select whether or not a beep is heard if a lower priority trunk group is used to dial out.
44-05-05	ARS/F-Route Table – Gain Table Number for Internal Calls	0 = No Setting 0 ~ 500 Default is 0	Select the gain table number to be used for the internal call (defined in Program 44-07).
44-05-06	ARS/F-Route Table – Gain Table Number for Tandem Connections	0 = No Setting 0 ~ 500 Default is 0	Select the gain table number to be used for the tandem call (defined in Program 44-07).
44-05-07	ARS/F-Route Table – ARS Class of Service	0 = No Setting 1 ~ 16 Default is 0	Select the ARS Class of Service to be used for the table. An extensions ARS COS is determined in Program 26-04-01.
44-05-08	ARS/F-Route Table – Dial Treatment	0 = No Setting 0 ~ 15 Default is 0	If a Dial Treatment is selected, Programs 44-05-02 and 44-05-03 are ignored and the Dial Treatment defined in Program 26-03-01 is used instead.
44-05-09	ARS/F-Route Table – Maximum Digit	0 = No Maximum 0 ~ 24 Default is 0	Assign the Max Digits for K-CCIS or ISDN Calls.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
44-05-10	ARS/F-Route Table – CCIS over IP Destination Point Code	0~16367 seconds Default is 0	

2.6   IP K-CCIS Assignment

Program/ Item No.	Description/ Selection	Assigned Data	Comments
50-01-01	CCIS System Setting – CCIS Availability	0 = Disable 1 = Enable Default is 0	
50-02-03	Connecting System Settings – Originating Point Code	0 ~ 16367 Default is 0	Assign the Originating Point Code for Route ID 9 of own side.
50-03-01	CCIS Destination System Settings – Destination Point Code	0 ~ 16367 Default is 0	Assign remote system IP Network information. PRG 50-03 assignments are only used only for CCISoIP.
50-03-03	CCIS Destination System Settings – IP Address (IP only)	xxx.xxx.xxx.xxx (xxx = 0 ~ 255) Default is 0.0.0.0	Assign remote system IP Network information. PRG 50-03 assignments are only used for CCISoIP.
50-03-04	CCIS Destination System Settings - Point Code Availability	0 : Disable 1 : Enable Default is 1	

2.7   IP K-CCIS (Peer to Peer) Assignment

Program/ Item No.	Description/ Selection	Assigned Data	Comments
50-15-01	CCIS over IP Basic Information Setting – Connection Method	0 = Peer to Peer Disable 1 = Peer to Peer Enable Default is 1	Turn Off or On the availability of IP-K-CCIS Peer to Peer.
50-15-02	CCIS over IP Basic Information Setting – TCP Server Port Number	0 ~ 65535 Default is 57000	
50-15-03	CCIS over IP Basic Information Setting – TCP Client Base Port Number	0 ~ 65535 Default is 59000	

2.8   CCIS over IP Codec Information Setup

Program/ Item No.	Description/ Selection	Assigned Data	Comments
84-21-01	CCIS over IP CODEC Information Basic Setup – Number of G.711 Audio Frames	1 ~ 4 1 = 10 ms 2 = 20 ms 3 = 30 ms 4 = 40 ms Default is 3	
84-21-02	CCIS over IP CODEC Information Basic Setup – G.711 Type	0 = A-law 1 = μ-law Default is 1	
84-21-03	CCIS over IP CODEC Information Basic Setup – G.711 Voice Activity Detection Mode	0 = Disable 1 = Enable Default is 0	
84-21-04	CCIS over IP CODEC Information Basic Setup – G.711 Jitter Buffer Min	0 ~ 160 ms Default is 30	

Program/ Item No.	Description/ Selection	Assigned Data	Comments
84-21-05	CCIS over IP CODEC Information Basic Setup – G.711 Jitter Buffer Average	0 ~ 160 ms Default is 60	
84-21-06	CCIS over IP CODEC Information Basic Setup – G.711 Jitter Buffer Max	0 ~ 160 ms Default is 120	
84-21-07	CCIS over IP CODEC – G.729 Audio Frame Number	1 ~ 6 1 = 10 ms 2 = 20 ms 3 = 30 ms 4 = 40 ms 5 = 50 ms 6 = 60 ms Default is 3	
84-21-08	CCIS over IP CODEC Information Basic Setup – G.729 Voice Activity Detection Mode	0 = Disable 1 = Enable Default is 0	
84-21-09	CCIS over IP CODEC Information Basic Setup – G.729 Jitter Buffer Min	0 ~ 270 ms Default is 30	
84-21-10	CCIS over IP CODEC Information Basic Setup – G.729 Jitter Buffer Average	0 ~ 270 ms Default is 60	
84-21-11	CCIS over IP CODEC Information Basic Setup – G.729 Jitter Buffer Max	0 ~ 270 ms Default is 120	
84-21-12	CCIS over IP CODEC Information Basic Setup – G.723 Audio Frame Number	1 = 30 ms 2 = 60 ms Default is 1	
84-21-13	CCIS over IP CODEC Information Basic Setup – G.723 Voice Activity Detection Mode	0 = Disable 1 = Enable Default is 0	

Program/ Item No.	Description/ Selection	Assigned Data	Comments
84-21-14	CCIS over IP CODEC Information Basic Setup – G723 Jitter Buffer Min	0 ~ 270 ms Default is 30	
84-21-15	CCIS over IP CODEC Information Basic Setup – G.723 Jitter Buffer Average	0 ~ 270 ms Default is 60	
84-21-16	CCIS over IP CODEC Information Basic Setup – G.723 Jitter Buffer Max	0 ~ 270 ms Default is 120	
84-21-17	CCIS over IP CODEC Information Basic Setup – TX Gain	0 ~ 40 = (-20dBm ~ +20dBm) 0 = -20 dBm 1 = -19 dBm : 20 = 0 dBm : 39 = +19 dBm 40 = +20 dBm Default is 20	
84-21-18	CCIS over IP CODEC Information Basic Setup – RX Gain	0 ~ 40 = (-20dBm ~ +20dBm) 0 = -20 dBm 1 = -19 dBm : 20 = 0 dBm : 39 = +19 dBm 40 = +20 dBm Default is 20	
84-21-19	CCIS over IP CODEC Information Basic Setup – 1st Priority of Audio Capability	0 = G.711 PT 1 = G.723 PT 2 = G.729 PT 3 = G.722 4 = G.726 5 = Not Used Default is 0	

Program/ Item No.	Description/ Selection	Assigned Data	Comments
84-21-20	CCIS over IP CODEC Information Basic Setup – 2nd Priority of Audio Capability	0 = G.711 PT 1 = G.723 PT 2 = G.729 PT 3 = G.722 PT 4 = G.726 PT 5 =Not Used Default is 1	
84-21-21	CCIS over IP CODEC Information Basic Setup – DTMF Relay Mode	0 = Disable 1 = Inbound (RFC2833) 2 = Outbound (H.245) Default is 0	
84-21-22	CCIS over IP CODEC Information Basic Setup – Jitter Buffer Mode	1 = Static 2 = Silence 3 = Immediate Default is 3	
84-21-23	CCIS over IP CODEC Information Basic Setup – Voice Activity Detection Threshold	0 = Self adjustment 1 = -19dBm (-49dBm) : 20 = 0dbm (-30dBm) : 29 = +9dbm (-21dBm) 30 = +10dbm (-20dBm) Default is 20	
84-21-24	CCIS over IP CODEC Information Basic Setup – Echo Canceller Mode	0 = Disable 1 = Enable Default is 1	
84-21-25	CCIS over IP CODEC Information Basic Setup – NLP Echo Canceller Mode	0 = Disable 1 = Enable Default is 1	
84-21-26	CCIS over IP CODEC Information Basic Setup – UDP Checksum Mode	0 = Disable 1 = Enable Default is 1	

Program/ Item No.	Description/ Selection	Assigned Data	Comments
84-21-27	CCIS over IP CODEC Information Basic Setup – G.722 Audio Frame Number	1 ~ 4 1 = 10 ms 2 = 20 ms 3 = 30 ms 4 = 40 ms Default is 3	
84-21-28	CCIS over IP CODEC Information Basic Setup – G.722 Voice Activity Detection Mode	0 = Disabled 1 = Enabled Default is 0	
84-21-29	CCIS over IP CODEC Information Basic Setup – G.722 Jitter Buffer (min)	0 ~ 160 ms Default is 30	
84-21-30	CCIS over IP CODEC Information Basic Setup – G.722 Jitter Buffer (Average)	0 ~ 160 ms Default is 60	
84-21-31	CCIS over IP CODEC Information Basic Setup – G.722 Jitter Buffer (max)	0 ~ 160 ms Default is 120	
84-21-32	CCIS over IP CODEC Information Basic Setup – G.726 Audio Frame Number	1 ~ 4 1 = 10 ms 2 = 20 ms 3 = 30 ms 4 = 40 ms Default is 3	
84-21-33	CCIS over IP CODEC Information Basic Setup – G.726 Voice Activity Detection Mode	0 = Disabled 1 = Enabled Default is 0	
84-21-34	CCIS over IP CODEC Information Basic Setup – G.726 Jitter Buffer (min)	0 ~ 160 ms Default is 30	

Program/ Item No.	Description/ Selection	Assigned Data	Comments
84-21-35	CCIS over IP CODEC Information Basic Setup – G.726 Jitter Buffer (Average)	0 ~ 160 ms Default is 60	
84-21-36	CCIS over IP CODEC Information Basic Setup – G.726 Jitter Buffer (max)	0 ~ 160 ms Default is 120	
84-21-37	---Not Used---		
84-21-38	---Not Used---		
84-21-39	---Not Used---		
84-21-40	---Not Used---		
84-21-41	---Not Used---		
84-21-42	CCIS over IP CODEC Information Basic Setup – FAX Relay Mode	0 = Disable 1 = Enable 2 = Each Port Mode (each extension) Default is 0	

2.9   SIP-MLT CODEC Information Fixed Mode Setup

Program/ Item No.	Description/ Selection	Assigned Data	Comments
84-29-01	SIP-MLT CODEC Information Fixed Mode Setup – Audio Capability	Type: 1 ~ 5 1 = MultiCast 2 = Reserved 3 = Reserved 4 = Reserved 5 = Reserved Codec: 1 = G.711 A-law 2 = G.711 μ-law 3 = G.729 4 = G.723 5 = G.722 Default is 2	
84-29-02	SIP-MLT CODEC Information Fixed Mode Setup – Number of Audio Frames	Type: 1 ~ 5 1 = MultiCast 2 = Reserved 3 = Reserved 4 = Reserved 5 = Reserved Size 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms 5 = 50ms 6 = 60ms Default is 2	

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SECTION 3   PROGRAMMING EXAMPLE

3.1   UNIVERGE SV8100 IP K-CCIS Programming Example 1

This is an example of UNIVERGE SV8100 program data assignments for a 4-digit Closed Numbering Plan using Closed Number Blocks. The following system configurations are used for all four systems:

❍   Each system is a single chassis with one CHS2U-US.

❍   Each system has 16 multiline terminals.

❍   Each system has eight analog trunks.
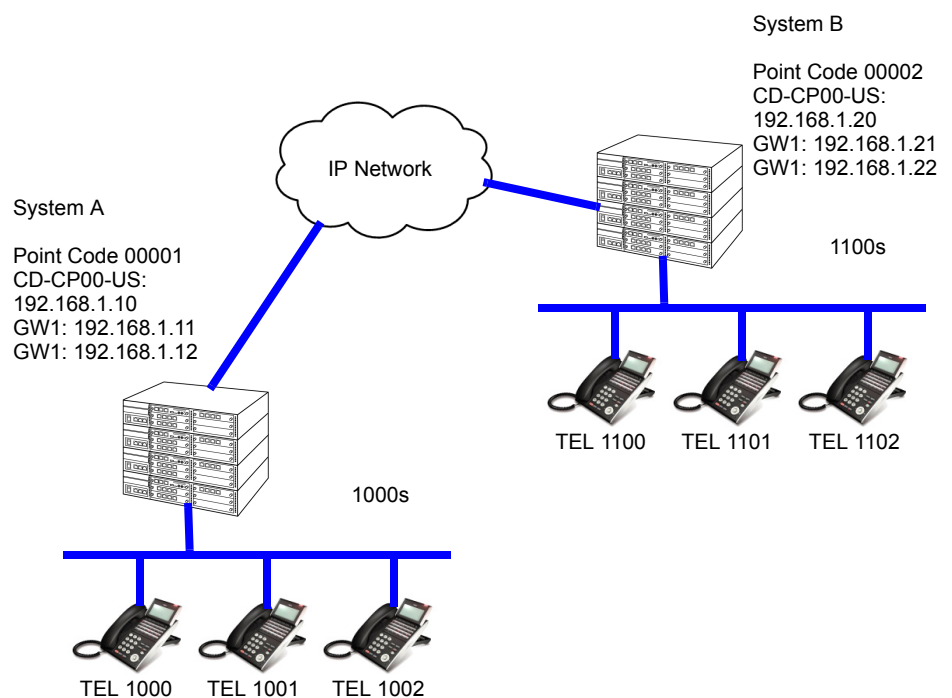
Figure 5-14 Programming Example 1

Assume that the systems are defaulted (first power on) with the following cards installed as described in the following table.

3.1.1   Card Interface Slot Assignment (PRG 10-03)

Slot	Card Type During 1st Power On	CARD TYPE AND PORTS			
		System A		System B	
		Card Type	Ports	Card Type	Ports
1	CD-CP00-US/PZ-32IPLA	CD-CP00-US/PZ-32IPLA	Trunks 5 ~ 28	CD-CP00-US/PZ-32IPLA	Trunks 5 ~ 28
2	CD-8DLCA	CD-8DLCA	Station 1 ~ 8	CD-8DLCA	Station 1 ~ 8
3	CD-8DLCA	CD-8DLCA	Station 9 ~ 16	CD-8DLCA	Station 9 ~ 16
4	None	None	N/A	None	N/A
5	CD-4COTB	CD-4COTB	Trunks 1 ~ 4	CD-4COTB	Trunks 1 ~ 4
6	None	None	N/A	None	N/A

3.1.2   Digital Trunk Assignments

System A	System B
Program 10-03-01 1. Insert PZ-32IPLA on CD-CP00-US in Chassis 1, Slot 1. 2. Verify PRG 10-03-01 (ETU Setup)	Program 10-03-01 1. Insert PZ-32IPLA on CD-CP00-US in Chassis 1, Slot 1. 2. Verify PRG 10-03-01 (ETU Setup).
Program 10-19-01 Assign DSP Resources 001~024 as CCIS (LK4).	Program 10-19-01 Assign DSP Resources 001 ~ 024 as CCIS (LK4).
Program 10-40-01 IP Trunk Availability	
Program 10-40-04 Assign the number of ports for CCISoIP for Slot 1 as 24.	Program 10-40-04 Assign the number of ports for CCISoIP for Slot 1 as 24.
Program 14-05-01 Assign Trunks 5 ~ 28 in Trunk Group 10	Program 14-05-01 Assign Trunks 5 ~ 28 in Trunk Group 10.
Program 22-02-01 Set Trunks 5 ~ 28 as Tie Line (LK6).	Program 22-02-01 Set Trunks 5 ~ 28 as Tie Line (LK6).

3.1.3   VoIP Address Assignments

System A	System B
Program 84-26-01 Slot 1, Assign IP Addresses GW1: 192.168.1.10 GW2: 192.168.1.11	Program 84-26-01 Slot 1, Assign IP Addresses GW1: 192.168.1.20 GW2: 192.168.1.21

3.1.4   IP K-CCIS Availability

System A	System B
Program 50-01-01 Select Option 1 (Enable) Enables the CCIS Feature	Program 50-01-01 Select Option 1 (Enable) Enables the CCIS Feature
Program 50-02-03 Assign Route ID 9 = 01 (Route ID 9 is designated for Host Originating Point Code)	Program 50-02-03 Assign Route ID 9 = 02 (Route ID 9 is designated for Host Originating Point Code)

3.1.5   IP K-CCIS Assignment

System A	System B
Program 50-03-01 System ID 1 ~ 255 – Enter Destination Point Code System ID 1 = DPC 02	Program 50-03-01 System ID 1 ~ 255 – Enter Destination Point Code System ID 1 = DPC 01
Program 50-03-03 System ID 1 ~ 255 – Enter Destination System IP Address. System ID 1 = 192.168.1.20	Program 50-03-03 System ID 1 ~ 255 – Enter Destination System IP Address System ID 1 = 192.168.1.10

3.1.6   IP K-CCIS (Peer to Peer) Assignment

System A	System B
Program 50-15-01 Enable/Disable Connection Method 0 = P2P Disable 1 = P2P Enable (Default)	Program 50-15-01 Enable/Disable Connection Method 0 = P2P Disable 1 = P2P Enable (Default)

System A	System B
Program 50-15-02 Assign the TCP Server Port Number 0 ~ 65535 57000 (Default)	Program 50-15-02 Assign the TCP Server Port Number. 0 ~ 65535 57000 (Default)
Program 50-15-03 Assign the TCP Client Base Port Number 0 ~ 65535 59000 (Default)	Program 50-15-03 Assign the TCP Client Base Port Number. 0 ~ 65535 59000 (Default)

3.1.7   Local Numbering Plan Assignment

System A	System B
Program 11-01-01 1 = Service Code 2 = Extension Number 3 = Trunk Access Code 4 = 2nd Trunk Access Code 5 = Operator Access 6 = ARS/F-Route Access	Program 11-01-01 1 = Service Code 2 = Extension Number 3 = Trunk Access Code 4 = 2nd Trunk Access Code 5 = Operator Access 6 = ARS/F-Route Access
Program 11-02-01 Assign Station Numbers to Port Numbers. Defaults for Ports 1 ~ 512: 101 ~ 198 3101 ~ 3199 3200 ~ 3513	Program 11-02-01 Assign Station Numbers to Port Numbers. Defaults for Ports 1 ~ 512: 101 ~ 198 3101 ~ 3199 3200 ~ 3513

3.1.8   Closed Numbering Plan – Using Closed Number Blocks

System A	System B
Program 11-01-01 Dial 10 = 4 Digit; Extension Dial 11 = 4 Digit; F-Route	Program 11-01-01 Dial 10 = 4 Digit; F-Route Dial 11 = 4 Digit; Extension

System A	System B
Program 11-02-01 Assign Station Numbers to Port Number Defaults for Ports 1 ~ 512 101 ~ 198 3101 ~ 3199 3200 ~ 3513	Program 11-02-01 Assign Station Numbers to Port Number. Defaults for Ports 1 ~ 512 101 ~ 198 3101 ~ 3199 3200 ~ 3513
Program 44-02-01 TBL 1 = 11	Program 44-02-01 TBL 1 = 10
Program 44-02-02 TBL 1 = F-Route	Program 44-02-02 TBL 1 = F-Route
Program 44-02-03 F-Route TBL 1 = 1	Program 44-02-03 F-Route TBL 1 = 1
Program 44-05-01 F-Route TBL 1 = TG10	Program 44-05-01 F-Route TBL 1 = TG10
Program 44-05-02 F-Route TBL 1 Delete Digit = 0	Program 44-05-02 F-Route TBL 1 Delete Digit = 0
Program 44-05-03 F-Route TBL 1 Add Dial = 0	Program 44-05-03 F-Route TBL 1 Add Dial = 0
Program 44-05-09 F-Route TBL 1 Max Digit = 4	Program 44-05-09 F-Route TBL 1 Max Digit = 4
Program 44-05-10 F-Route TBL1 = DPC 2	Program 44-05-10 F-Route TBL1 = DPC 1

3.1.9   Tandem Connections

System A	System B
Program 14-01-13 Trunk to Trunk Transfer 0 = Disable 1 = Enable (Default)	Program 14-01-13 Trunk to Trunk Transfer 0 = Disable 1 = Enable (Default)
Program 14-01-24 Trunk to Trunk Outgoing Caller ID Through Mode 0 = Disable (Off) (Default) 1 = Enable (On)	Program 14-01-24 Trunk to Trunk Outgoing Caller ID Through Mode 0 = Disable (Off) (Default) 1 = Enable (On)

**Figure 5-15 TEL 302 (SIP MLT) makes call to TEL 202 (SIP MLT)
via IP K-CCIS**

3.2 UNIVERGE SV8100 IP K-CCIS and Electra Elite IPK II Programming Example 2

The following example provides programming details for three systems connected by the IP CCH for NEAX application and one system connected by legacy (T1) K-CCIS.

Point Code 00001

Point Code 00002
Address:
1st PVA: 192.168.1.10

2nd PVA: 192.168.1.11

Electra Elite
IPK II

System A
(1000)

T1 for K-CCIS

UNIVERGE
SV8100

System B
(1100)

IP Network
(Intranet)

UNIVERGE
SV8100

System C
(1200)

Point Code 00003
Address: 192.168.1.20
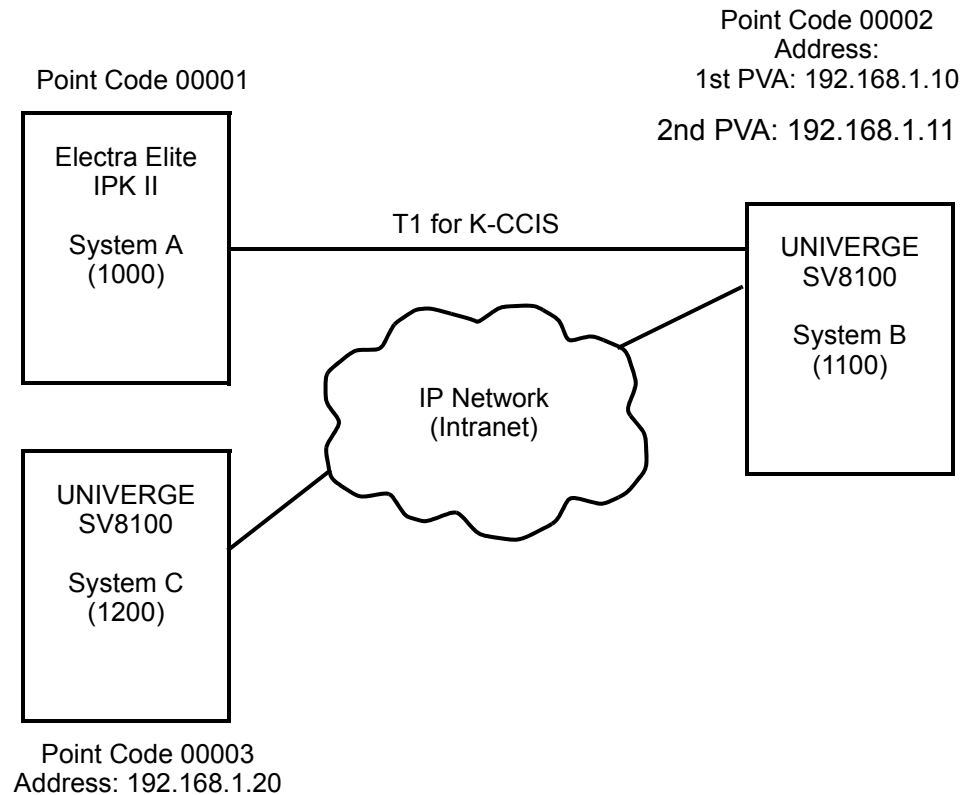
Figure 5-16 Programming Example 2

3.2.1 Chassis Programming Assignments

For this example, the following system configurations are used:

❐ A four-digit Closed Numbering Plan using Closed Number Blocks is used.

❐ System A is a single-cabinet Electra Elite IPK II Expanded port package with one B64-U20 KSU, one DTI-U( ) ETU, and a CCH(4)-U( ) ETU.

❐ System B is a single chassis UNIVERGE SV8100 with one CHS2U-US and one PZ-32IPLA installed on the CD-CP00-US and one CD-CCTA blade installed.

- ❐   System C is a single-cabinet UNIVERGE SV8100 port package with one CHS2U-US and PZ-32IPLA Daughter Board installed on the CD-CP00-US.

- ❐   Each system has 16 multiline terminals.

- ❐   Each system has eight analog trunks.

Assume that the systems were defaulted (first power on) with the following cards installed as described below.

Card Interface Slot Assignment (PRG 10-03)

SYSTEM A			
Slot	Card Type During 1st Power On	Card Type And Ports	
		Card Type	Ports
1	ESI(8)-U10 ETU	ESI(8)-U10 ETU	Station 1 ~ 8
2	ESI(8)-U10 ETU	ESI(8)-U10 ETU	Station 9 ~ 16
3	None	None	N/A
4	COI(8)-U10 ETU	COI(8)-U10 ETU	Trunks 1 ~ 8
5	DTI-U( ) ETU	DTI-U( ) ETU	Trunks 9 ~ 32
6	CCH(4)-U( ) ETU	CCH(4)-U( ) ETU	N/A
7	None	None	N/A
8	None	None	N/A
0A	None	None	N/A

System B			
Slot	**Card Type During 1st Power On**	**Card Type And Ports**	
		Card Type	**Ports**
1	CD-CP00-US with PZ-32IPLA Blade/Daughter Board	CD-CP00-US with PZ-32IPLA Blade/Daughter Board	Trunks 29 ~ 52
2	CD-8DLCA Blade	CD-8DLCA Blade	Station 1 ~ 8
3	CD-8DLCA Blade	CD-8DLCA Blade	Station 9 ~ 16
4	None	None	N/A
5	CD-4COTB Blade	CD-4COTB Blade	Trunks 1 ~ 4
6	CD-CCTA Blade	CD-CCTA Blade	Trunks 5 ~ 28

System C			
Slot	**Card Type During 1st Power On**	**Card Type And Ports**	
		Card Type	**Ports**
1	CD-CP00-US with PZ-32IPLA Blade/Daughter Board	CD-CP00-US with PZ-32IPLA Blade/Daughter Board	Trunks 5 ~ 28
2	CD-8DLCA Blade	CD-8DLCA Blade	Station 1 ~ 8
3	CD-8DLCA Blade	CD-8DLCA Blade	Station 9 ~ 16
4	None	None	N/A
5	CD-4COTB Blade	CD-4COTB Blade	Trunks 1 ~ 4
6	None	None	None

3.2.2   Digital Trunk Assignments

System A	System B	System C
Program 10-03-01 1. Notice the Logical Trunk Number	Program 10-03-01 1. Notice the Logical Trunk Number	Program 10-03-01 1. Select Slot 1. 2. Notice the Logical Trunk Number.
Program 10-03-02 0 = D4 (Multiframe) (Default) 1 = ESF (24 Multiframe)	Program 10-03-02 0 = D4 (Multiframe) (Default) 1 = ESF (24 Multiframe)	*Program 10-19-01 1. Assign DSP Resources 001 ~ 024 as CCIS (LK 4).
Program 10-03-03 0 = B8ZS (Default) 1 = ANI/ZCS	Program 10-03-03 0 = B8ZS (Default) 1 = ANI/ZCS	Program 10-40-01 0 = Disable (Default) 1 = Enable
Program 10-03-05 Assign Clocking 0 = Internal 1 = External (Default)	Program 10-03-05 Assign Clocking 0 = Internal 1 = External (Default)	*Program 10-40-04 1. Assign Number of Ports for CCIS for Slot 1 = 24.
Program 14-05-01 1. Trunks 9 ~ 32 2. Assign Trunk Group 10	*Program 10-19-01 1. Assign DSP Resources 001 ~ 024 as CCIS (LK 4).	Program 14-05-01 1. Trunks 5 ~ 28 2. Assign Trunk Group 10
Program 22-02-01 1. Trunks 9 ~ 32 2. Assign LK 6 (Tie Line)	*Program 10-40-04 1. Assign Number of Ports for CCIS for Slot 1 = 24	Program 22-02-01 1. Trunks 5 ~ 28 2. Assign LK 6 (Tie Line)
Program 34-01-01 1. Trunks 9 ~ 32 2. Assign LK 2 (Wink)	Program 14-05-01 1. Trunks 5 ~ 28 2. Assign Trunk Group 11 3. Trunks 29 ~ 52 4. Assign Trunk Group 10	Program 34-01-01 1. Trunks 5 ~ 28 2. Assign LK 2 (Wink)
	Program 22-02-01 1. Trunks 5 ~ 52 2. Assign LK 6 (Tie Line)	
	Program 34-01-01 1. Trunks 5 ~ 52 2. Assign LK 2 (Wink)	

* For CCISoIP only

3.2.3   CCH Assignment

System A	System B	System C
Program 14-13-01 1. Select Trunk Group 10 2. Assign CCIS Route ID 1	*Program 44-05-10 F-Route Table 1 : 3 F-Route Table 2 : 4	*Program 44-05-10 F-Route Table 1 : 2 F-Route Table 2 : 4
Program 14-14-01 1. Trunks 9 ~ 32 2. Assign CIC 1 ~ 24	Program 14-13-01 1. Select Trunk Group 10 2. Assign CCIS Route ID 1	Program 50-01-01 1. Enable CCIS
Program 50-01-01 1. Enable CCIS	Program 14-14-01 1. Trunks 5 ~ 28 2. Assign CIC 1 ~ 24	Program 50-02-03 1. Route ID 9 : 3
Program 50-02-01 1. Select CCIS Route ID 1 2. Assign Channel Port 28	Program 50-01-01 1. Enable CCIS	*Program 50-03-01 1. System ID 1 : 2 2. System ID 2 : 4
Program 50-02-02 1. Select CCIS Route ID 1 2. Assign Data Speed LK 2 (56k)	Program 50-02-01 1. Select CCIS Route ID 1 2. Assign Channel Port 28	
	Program 50-02-02 1. Select CCIS Route ID 1 2. Assign Data Speed LK 2 (56k)	
	*Program 50-02-03 Route ID 9 : 2	
	*Program 50-03-01 1. System ID 1 : 3 2. System ID 2 : 4	

* For CCISoIP only

SECTION 4   PORT DESIGNATIONS

IP Application	IP Port Numbers	Comments
SIP Trunk		
SIP Trunk Signaling	5060	UDP
SIP Trunk Voice w/32 IPLA	10020~10083	UDP
SIP Trunk Voice w/64 IPLA	10020~10147	UDP
SIP Trunk Voice w/128 IPLA	10020~10275	UDP
3rd Party SIP		
SIP SLT Signaling	5070	UDP
SIP SLT Voice w/32 IPLA	10020~10083	UDP
SIP SLT Voice w/64 IPLA	10020~10147	UDP
SIP SLT Voice w/128 IPLA	10020~10275	UDP
H.323 Trunk		
H.323 Signaling	1718/1719/1720	TCP
H.323 Trunk Voice w/32 IPLA	10020~10083	UDP
H.323 Trunk Voice w/64 IPLA	10020~10147	UDP
H.323 Trunk Voice w/128 IPLA	10020~10275	UDP
SV8100 IP - K-CCIS		
IP K-CCIS Signaling	57000/59000	TCP
IP Trunk Voice w/32 IPLA	10020~10083	UDP
IP Trunk Voice w/64 IPLA	10020~10147	UDP
IP Trunk Voice w/128 IPLA	10020~10275	UDP

IP Application	IP Port Numbers	Comments
SV8100 NetLink		
Primary System Signaling	58000/58001	TCP
Secondary System Signaling	58002	TCP
IP DSP Resource w/32 IPLA	10020~10083	UDP
IP DSP Resource w/64 IPLA	10020~10147	UDP
IP DSP Resource w/128 IPLA	10020~10275	UDP
NEC Proprietary SIP (SIP MLT)		
SIP MLT Signaling	5080	UDP
SIP Trunk Voice w/32 IPLA	10020~10083	UDP
SIP Trunk Voice w/64 IPLA	10020~10147	UDP
SIP Trunk Voice w/128 IPLA	10020~10275	UDP

SECTION 5   SERVICE CONDITIONS

For this feature, the PZ-(X)IPLA is installed on the CD-CP00-US. Only one PZ-(X)IPLA can be installed per system for a maximum of 128 VoIP resources.

The audio quality of speech connections depends on the available bandwidth between the PZ-(X)IPLA daughter boards in the data network. As internet is an uncontrolled data network compared to an intranet, using this application in intranet WAN environment with known, or controlled and assured, Quality of Service (QoS), is highly recommended.

The CD-CCTA blade is not required to support this feature. It can be installed and used in a system using traditional K-CCIS with point to point T1 lines allowing both IP K-CCIS and traditional K-CCIS to be used with the same system.

The LAN connection is provided by a 10 Base-T/100 Base-TX/1000 Base-T, Auto sensing, full duplex Ethernet.

5.1   Restrictions

❍   The UNIVERGE SV8100 can send billing information to a billing center office (NEAX2000/2400/SV8300), but it cannot receive the billing information as the billing center office.

❍   Voice Calls – K-CCIS, voice announce is not supported for a forwarded call across IP K-CCIS to NEAX.

❍   Not all data networks are suitable to support Voice over Internet Protocol (VoIP). A good VoIP solution requires a low-latency, low jitter and low packet loss network. Accordingly, a network must be evaluated for latency, packet loss, and jitter to qualify and determine if it can provide toll-quality speech paths.

❍   The PZ-(X)IPLA can support 32, 64 or 128 VoIP resources dependent on the PZ-(X)IPLA daughter board installed on the CD-CP00-US which may be configured to support SIP MLT, 3rd Party SIP Stations and Trunks or IP K-CCIS Applications.

❍   A CD-CP00-US license is not required for IP K-CCIS trunks.

❍   A CD-CP00-US license is required for SIP Clients to include SIP stations and SIP trunks.

❍   The PZ-(X)IPLA daughter board may configure VoIP DSP usage criteria with SV8100 Program 10-19-01 DSP Resources. Each PZ-(X)IPLA can flag individual DSP resources as:

■   IP Ext – IP Extensions (includes SIP MLT Station or 3rd Party SIP Stations)

■   SIP Trk – SIP Trunks

■   IP K-CCIS – CCIS Networking

■   Common – Common usage for CCIS Networking, 3rd Party SIP Station, SIP MLT Stations, SIP Trunks.

■   Pressing Feature + 4 from any MLT Terminal shows the following:

❍   What type of PZ-(X)IPLA daughter board is installed on the CD-CP00-US.

❍   How many VoIP resources are Active (call in progress) and Reserved (call setup in progress).

❍   The MAC Address of the PZ-(X)IPLA.

❍   One way delay must not exceed 100 ms.

❍   Round trip delay must not exceed 200 ms.

❍   Packet loss must not exceed 1%.

❍   When an SIP MLT phone establishes a call via IP K-CCIS to another SIP MLT phone in a remote system, a VoIP Resource is used from both systems.

❍   When a TDM phone establishes a call via IP K-CCIS and calls another
     TDM phone in a remote system,   a VoIP resource is used from both
     systems.

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KCCIS-IP with PVA

SECTION 1   INTRODUCTION

This feature provides the benefits and additional feature compatibility of Key-Common Channel Interoffice Signaling (K-CCIS) between multiple systems including NEAX PBX systems connected together over a Data Communication IP Network (Intranet).

Voice Signals and common signaling data from/to the distant offices are converted into IP Packets and transmitted through the Data IP Network. When using this feature, both Voice and Data Communication Lines are integrated into one Network and communication costs can be reduced.

This feature is available between UNIVERGE SV8100, Electra Elite IPK II, Electra Elite IPK and NEAX PBX systems.

The following features are provided:

❍   Automatic Recall ***

❍   Brokerage Hotline

❍   Call Forwarding – All Calls

❍   Call Forwarding – Busy/No Answer

❍   Call Park Retrieve

❍   Call Transfer – All Calls

❍   Calling Name Display

❍   Calling Number Display

❍   Calling Party Number (CPN) Presentation from Station

❍   Centralized Billing

❍   Centralized BLF (K-CCIS) **

❍   Centralized Day/Night Mode Change

❍   Centralized E911

❍ Dial Access to Attendant

❍ Direct Inward Dialing

❍ Dual Hold

❍ Elapsed Time Display

❍ Flexible Numbering of Stations

❍ Hands-free Answerback

❍ Hot Line

❍ Link Reconnect

❍ Multiple Call Forwarding – All Calls

❍ Multiple Call Forwarding – Busy/No Answer

❍ Paging Access

❍ Quick Transfer to Voice Mail

❍ Station-to-Station Calling

❍ Uniform Numbering Plan

❍ Voice Call

❍ Voice Mail Integration *

*Not supported with VM8000 InMail

** Not supported with NEAX PBX

*** Not supported with IPK

SECTION 2   FEATURE OVERVIEW

This section lists feature details of the KCCIS_IP with PVA.

2.1   System Availability

❍ All Terminals

2.2   Required Components:

❍ This feature is available with Version 3.01 or higher.

❍ CD-PVAA w/K-CCIS-IP Compact Flash [PVA CCISoIP APP CF (670918)]

❍ LK-PVA-CCISoIP 4-Port – LIC license increments

❍ LK-PVA-CCISoIP 24-Port – LIC license

2.3   Operating Procedures

❍   Normal call handling procedures apply.

❍   For programming details, refer to Section 10 Univerge SV8100
      PROGRAMMING on page 6-28.

2.4   Service Conditions

❍   The CD-PVAA blade requires the K-CCIS-IP with PVA Compact Flash
      card.

❍   The K-CCIS-IP with PVA application can be licensed in increments of 4,
      or a single 24-ports.

❍   Each CD-PVAA blade reduces the maximum capacity of trunks in the
      system.

❍   The CD-PVAA is required for K-CCIS-IP with PVA connection to the IPK,
      IPK II, IPS and NEAX.

❍   When connecting the CD-PVAA KCCIS-IP with PVA application to the
      IPK and IPKII, the PVA(X)-U( ) ETU installed in the IPK and IPKII must
      be upgraded to the latest Service Pack and Firmware.

❍   One CD-PVAA is required to support up to 24 channels for KCCIS-IP
      with PVA.

❍   Up to (8) CD-PVAA blades can be installed in the Univerge SV8100.

❍   Port assignment of the CD-PVAA package depends on the KCCIS – IP
      with PVA license. You must input the license code of "6200" in
      PRG 10-54 for each slot the PVA-CCIS package is installed.

❍   When multiple CD-PVAA blades are installed in the same Univerge
      SV8100, it is necessary to input PRG 10-54 for each slot the KCCIS-IP
      with PVA is installed. The total PVA-CCIS call count is limited by the
      license quantity installed in the system.

The following table illustrates examples of KCCIS-IP with PVA licensing and
slot assignments:

CD-PVAA blade is installed in slot# 6:

Example #1	
License Quantity: 8 PRG 10-54 = 8	Can make or receive a total of 8 calls. Can assign 8 ports and CD-PVAA blade starts.
Example #2	

License Quantity: 8 PRG 10-54 = 0	Cannot assign ports and CD-PVAA blade does not start.
Example #3	
License Quantity: 0 PRG 10-54 = 8	Cannot make or receive calls. Can assign 8 ports and CD-PVAA blade starts.
Example #4	
License Quantity: 8 PRG 10-54 = 24	Can make or receive a total of 8 calls. Can assign 24 ports and CD-PVAA blade starts.

CD-PVAA blade is installed in slot# 4 and slot# 6:

Example #1	
License Quantity: 16 PRG 10-54(slot#4) = 8 PRG 10-54(slot#6) = 8	Can make or receive a total of 16 calls. Can assign 8 ports and CD-PVAA blade starts. Can assign 8 ports and CD-PVAA blade starts.
Example #2	
License Quantity: 16 PRG 10-54(slot#4) = 16 PRG 10-54(slot#4) = 16	Can make or receive a total of 16 calls. Can assign 16 ports and CD-PVAA blade starts. Can assign 16 ports and CD-PVAA blade starts.

❍   The KCCIS-IP with PVA feature shares the CO/PBX/Tie/DID trunks
    available for the system.

❍   The CD-PVAA blade supports only those CODECs that are approved to
    provide toll-quality speech path. The following voice compression
    methods are supported for the KCCIS-IP with PVA application:

    ❐   G.711 u-Law – Highest Bandwidth

    ❐   G.729 – Mid-Range Bandwidth

    ❐   G.723 – Lowest Bandwidth

❍   Each voice call requires at least the bandwidth listed in the following
    table:

CODEC	Transmit Data Rate	Receive Data Rate	Time Between Packets	Packetization Delay	Default Jitter Buffer Delay	Theoretical Maximum MOS
G711u-law	90 Kbps	90 Kbps	20 ms	1.5 ms	2 datagrams (40 ms)	4.4
G729	34 Kbps	34 Kbps	20 ms	15.0 ms	2 datagrams (40 ms)	4.07

CODEC	Transmit Data Rate	Receive Data Rate	Time Between Packets	Packetization Delay	Default Jitter Buffer Delay	Theoretical Maximum MOS
G.723	25 Kbps	25 Kbps	30 ms	37.5 ms	2 datagrams (60 ms)	3.87

✎ This includes the overheads of VoIP communication including signaling.

✎ In voice communications, particularly Internet telephony, the mean opinion score (MOS) provides a numerical measure of the quality of human speech at the destination end of the circuit. The scheme uses subjective tests (opinionated scores) that are mathematically averaged to obtain a quantitative indicator of the system performance. A score of 5.0 is the maximum for the Mean Opinion Score.

○ Data calls (Modem Data) across the VoIP connection are only supported when the G.711 u-Law CODEC is used.

○ The CD-PVAA blade contains a regular TCP/RTP/IP stack that can handle real time media. The blade, from the network administration perspective, is an endpoint on the IP network.

○ The CD-PVAA with KCCIS – IP with PVA application loaded uses Common Channel Interoffice Signaling over IP networks. Currently this protocol does not allow for communications across networks where Network Address Translation (NAT) is performed.

○ The audio quality of speech connections depends greatly on the available bandwidth between the CD-PVAA blade in the data network. As Internet is an uncontrolled data network compared to an Intranet, using this application in Intranet WAN environment, with known (or controlled and assured) Quality of Service (QoS), is highly recommended.

○ If any network where the CD-PVAA blade is connected uses NAT for connecting the voice calls (including firewall), consulting your network provider/administrator and specifically requesting service for VoIP or real-time media support on your networks is highly recommended.

○ A static IP Address is required for each CD-PVAA with KCCIS – IP with PVA.

○ This feature allows Point-to-Multipoint peer-to-peer connections for calls through the IP K-CCIS Network. If a call is transferred or forwarded to a different system in the Network, the trunks in the transferring system are released and a new Point to Point connection is established.

   * When a call terminates back to the originating system because of call forwarding or transfer, the intermediate trunks are not released until the call is completed by an answer at the called party or until the call is forwarded to a different system (similar to the linkreconnect function in TDM CCIS.

❍   The CD-CCTA blade is not required to support this feature. It can be installed and used in a system using traditional K-CCIS with point to point T1 lines allowing both KCCIS – IP with PVA and traditional K-CCIS to be used with the same system.

❍   The LAN connection is provided by a 10/100/1000 Base T, Auto sensing, full-duplex Ethernet.

❍   When assigning a Closed Numbering Plan and DID conversion across KCCIS-IP with PVA is required, the UNIVERGE SV8100 uses the ARS/ F-Route Tables.

❍   The UNIVERGE SV8100 uses the F-Route Tables to assign an Open Numbering Plan.

❍   When all KCCIS – IP with PVA voice channels are busy, the UNIVERGE SV8100 originator of a KCCIS-IP with PVA call hears a busy tone from the system.

❍   Outgoing CO calls in a K-CCIS network can be routed over the KCCIS – IP with PVA link and use the distant system CO lines.

❍   Distant system extension numbers in the KCCIS – IP with PVA network can be assigned to One Touch keys and Speed Dial buffers.

❍   When a KCCIS – IP with PVA trunk is on hold, the Specified Line Seizure access codes can be used to retrieve the call from its held state.

❍   FoIP (Fax over Internet Protocol) with T.38 protocol is not supported.

❍   Each Univerge SV8100 can support both KCCIS-IP, KCCIS-IP with PVA and SV8100 NetLink simultaneously.

❍   Each Univerge SV8100 can support up to (8) CD-PVAA blades – (3) per chassis. Refer to the SV8100 SystemHardware Manual for more information regarding maximum capacities.

❍   KCCIS-IP with PVA does not require the use of the PZ-(X)IPLA (VoIPDB).

❍   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.

❍   Adequate bandwidth for estimated VoIP traffic.

❍   Centralized Voice Mail is supported from the Univerge SV8100 or
     Electra Elite IPKII when using 5-, 6-, 7- and 8-digits station numbers,
     when the Voice Mail is installed in the SV8100.

2.5   Restrictions:

❍   The Univerge SV8100 can send billing information to a billing center
     office (Univerge SV8300/SV8500) but cannot receive the billing
     information as the billing center office.

❍   Voice Calls – (K-CCIS) voice announce is not supported for a forwarded
     call across KCCIS – IP with PVA to an IPK.

❍   Not all data networks are suitable to support Voice over Internet Protocol
     (VoIP). A good VoIP solution requires a low-latency, low jitter and low
     packet loss network. Accordingly, a network must be evaluated for
     latency, packet loss, and jitter to qualify and determine if it can provide
     toll-quality speech paths.

❍   KCCIS - IP with PVA Application will support trunks configured in
     increments of two contiguous DSP resources.

❍   Blade Configuration – The KCCIS – IP with PVA Application Package is
     identified as an CD-PVA blade type.

❍   When the KCCIS – IP with PVA application is installed in the Univerge
     SV8100, the system assigns the first available logical port numbers for
     CCISoIP trunks.

❍   If the Univerge SV8100 system does not have registered CD-PVAA
     KCCIS-IP with PVA licenses, the Logical Trunk will not be assigned.

❍   The UNIVERGE SV8100 can support only 1~8-digit station numbers.

❍   Station Numbers are assigned by the 10s group for 4-digit station
     numbers, 100s group for 5-digit station numbers, 1000s group for 6-digit
     station numbers, 10000s group for 7-digit station numbers.

❍   When Voice Mail Message Waiting status must be sent across the
     K-CCIS to a remote system, F-Routes must be used.

❍   For a Closed Numbering Plan network using F-Routes, a maximum of
     120 F-Route Tables are available allowing a maximum of 121 connected
     systems per K-CCIS network.

❍   When a Closed Numbering Plan Network is used, a user can call
     another station by dialing the distant extension number, but extensions in
     the network cannot have the same prefix.

❍ For an Open Numbering Plan network, a user can dial another station by dialing the office location number plus an extension number and the extension number can have the same prefix, but the office location cannot be the same.

❍ An UNIVERGE SV8100 K-CCIS network should never have more than five hops (tandem connections) because of the message delay through each tandem system.

❍ The maximum number of systems depends on the Numbering Plan used and the maximum number of hops (tandem connections).

❍ K-CCIS requires assigning a point code for each office. Point codes differentiate between an originating office and a destination office in the K-CCIS Network. Assigning point codes requires the following considerations:

❍ The point code must be unique in the network.

❍ The UNIVERGE SV8100 can have a maximum of 255 codes assigned to distant systems.

❍ Using a Key System-to-Key System network, centralized voice mail is not supported when an Open Numbering Plan is used.

❍ Centralized E911 – K-CCIS is supported.

❍ When Centralized E911 – K-CCIS is not used, each UNIVERGE SV8100 system in a K-CCIS network must have at least one trunk for Emergency 911 calls.

❍ Using a NEAX-to-UNIVERGE SV8100 network, the PBX must supply centralized voice mail.

❍ Multiline terminals must have an available Call Appearance (CAP) key to originate or answer a KCCIS-IP with PVA trunk call.

❍ Direct access of KCCIS-IP with PVA voice or data channels using Line keys or Specified Line Seizure access codes is prohibited.

❍ The Recall key or Drop key does not function on KCCIS-IP with PVA calls. When either key is pressed, operation is ignored, and the call continues.

❍ Trunk queuing is prohibited on a KCCIS-IP with PVA trunk route.

❍ The ability to route an incoming DID call directly across a KCCIS – IP with PVA link (Direct Inward Dialing – K-CCIS) is supported only when a Closed Numbering Plan using F-Routes is used.

❍ This feature is not supported by the CD-4ODTA Analog Line interface.

❍   Extension numbers cannot start with 0 or 9.

❍   Internal Calls, transferred calls, and KCCIS-IP with PVA calls do not provide Caller ID to single line telephones.

❍   Caller ID Call Return feature is not supported with KCCIS-IP with PVA calls.

❍   Any IPK/IPKII in a KCCIS - IP with PVA network must use the PVA(X)-U10 ETU/with SPXX and Application software XXX or higher.

❍   The Automatic Recall feature is not supported when the call is originated in the Electra Elite IPK System.

❍   Centralized Voice Mail is not supported from the Electra Elite IPK, when using 5-, 6- or 7-digit station numbers. Refer to the IPK K-CCIS Manual for more information.

❍   In a KCCIS - IP with PVA network, the Electra Elite IPK must be programmed internally to match the same extension number length of other systems in the network.

❍   UCB is not supported as the Centralized Voice Mail

2.6   Related Feature List

❍   K-CCIS - IP

2.7   Guide To Feature Programming

Refer to Section 10 Univerge SV8100 PROGRAMMING for programming details.

2.8   Operation

Refer to the UNIVERGE SV8100 Networking Manual for detailed feature information.

SECTION 3   SYSTEM OUTLINE

3.1   KCCIS-IP with PVA Application using the CD-PVAA Blade

The system uses the CD-PVAA blade to connect multiple systems together over a Data Communication IP Network (Intranet). Key-Common Channel Interoffice Signaling (K-CCIS) is used to provide telephony services between the Univerge SV8100 and another Electra Elite IPK or Electra Elite IPK II.

The illustration below shows a system outline of KCCIS-IP with PVA Network via IP.
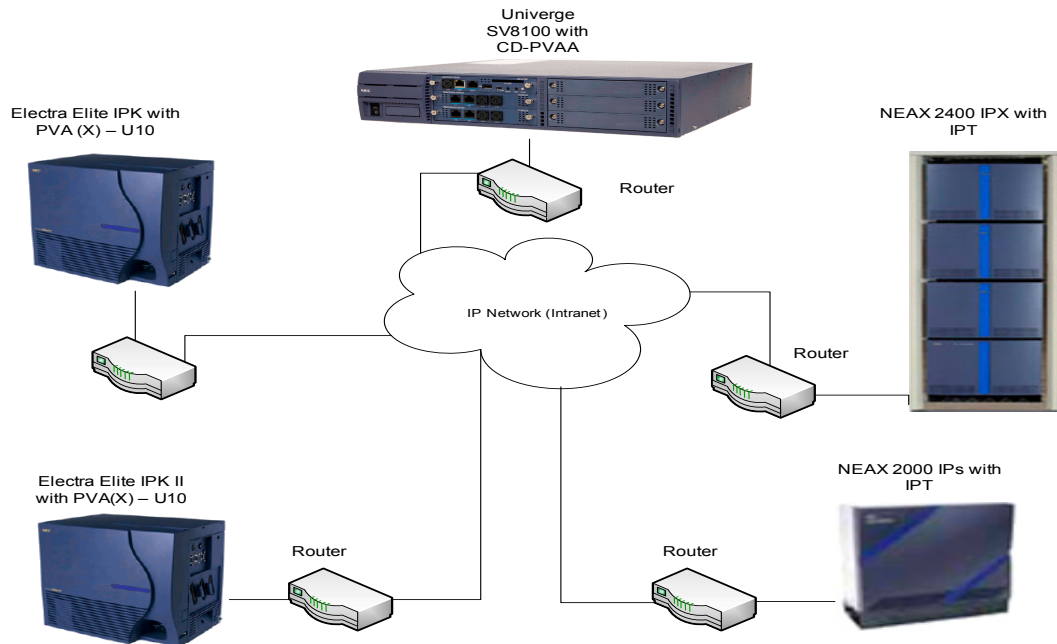
Figure 6-1 KCCIS-IP with PVA Network via IP

3.2   Point-to-Multipoint Connection

The CD-PVAA blade allows a point-to-multipoint connection. A point-to-multipoint connection designates multiple destination IP Addresses per CD-PVAA as shown below.
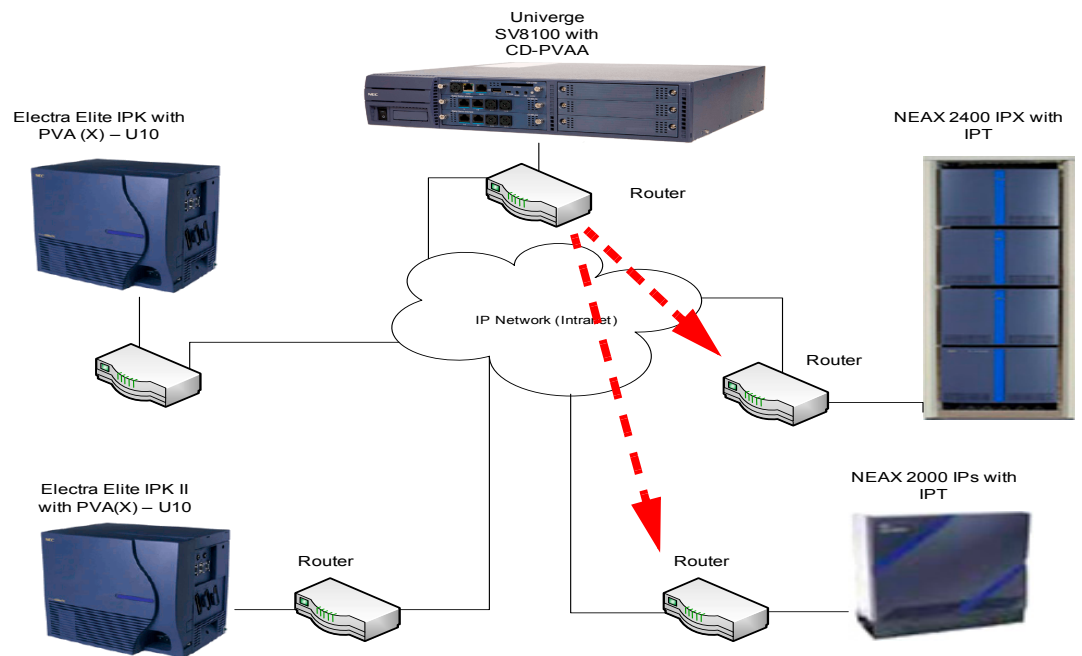
Figure 6-2 Example of Point-to-Multipoint Connection

SECTION 4   CD-PVAA SPECIFICATIONS

4.1   Indicators, Switches and Connectors
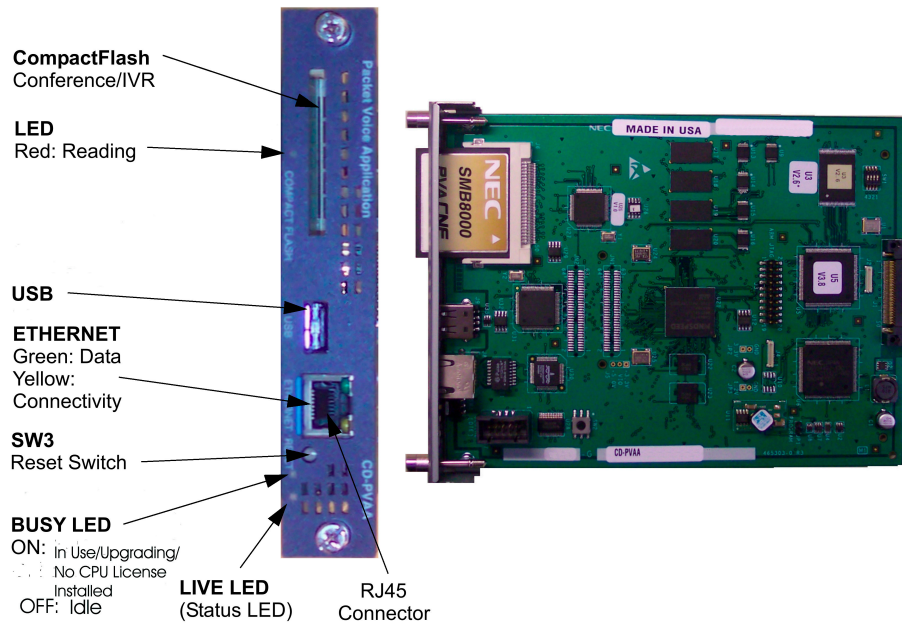
Figure 6-3 CD-PVAA Blade

4.2   Switches

The Multimedia Conference Bridge Application has the following switches.

○   DIP Switch SW1

Reserved for future use.

○   RESET Switch SW3

4.3   Connectors

❍   Ethernet Connector

The Multimedia Conference Bridge Application has a single 10/100 MPS Ethernet connector. The port has Auto-MDIX (Auto-Medium Dependent Interface Crossover) giving users the ability use either a straight-through Ethernet cable or a crossover Ethernet cable. A crossover cable is not required for direct connection to a PC.

The Ethernet Interface setting allows for manual configuration of the Ethernet port mode/speed from the Web Interface. This can be set to operate in the following port speed and duplex mode: 10MB/Full Duplex,10MB/Half Duplex, 100MB/Half Duplex, 100MB Full Duplex, Auto-Negotiate.

❍   Serial Connector

A 10-pin header (J2) provides access to a serial terminal on the Multimedia Conference Bridge Application. The serial connector is used for debug operations only.

4.4   Environmental Conditions

The following environmental conditions apply for the CNF module operation:

❍   Operating Temperature: +32°F ~ +104°F (0°C ~ 40°C)

❍   Operating Humidity: 10% to 90% (non-condensing)

SECTION 5   INSTALLING THE **KCCIS-IP** WITH **PVA** APPLICATION

5.1   System Requirements

The KCCIS-IP with PVA Application is a hot-swappable blade that can be installed in any slot in an SV8100. It is recognized by the system as a PVA-CCIS blade.

The Ethernet connector can use a Category 5 (CAT5) unshielded, twisted pair (UTP) cable to connect to an Ethernet hub or switch within the customer premises.

Use the following steps to install the KCCIS-IP with PVA application.

1.   License CD-CP00-US for PVA-CCIS ports.

2.   Using telephone programming only, set the following programs. This must be done via telephone programming:

❒   PRG 10-54-01: For the slot the PVA-CCIS blade will be installed in set ID01 license code 6200.

❒   PRG 10-54-02: For the slot the PVA-CCIS blade will be installed in set ID01 license num to the number of PVA-CCIS ports licensed in step 1.

❒   PRG 10-55-01: For the slot the PVA-CCIS blade will be installed, and set the IP Address.

❒   PRG 10-55-04: For the slot the PVA-CCIS blade will be installed, and set the subnet mask.

3.   Insert the PVA-CCIS Application (Compact Flash) into position J6 of the CD-PVAA blade.

4.   Place the PVA-CCIS Application in the desired slot in the chassis and wait for it to finish booting. This could take several minutes.

5.   Connect the Ethernet cable coming from a hub or switch from the customer premises to the connector on the PVA-CCIS Application. Verify that the link light shows green.

At this point, the PVA-CCIS Application is physically accessible. Refer to Section 8 KCCIS-IP with PVA Application Configuration on page 6-17 for configuration instructions.

5.2   Installation Precautions

Observe the following precautions when installing the blades to avoid static electricity damage to hardware or exposure to hazardous voltages.

The blades used in this system make extensive use of CMOS technology that is very susceptible to static electricity. Static discharge must be avoided when handling blades. Always use the following precautions.

❍   Wear a grounded wrist strip any time you handle a blade.

❍   Make all blade DIP switch and jumper setting changes before inserting the blade in the chassis.

❍   Carry blade in a conductive polyethylene bag to prevent static electricity damage.

SECTION 6   PROGRAMMING USING THE MULTILINE TELEPHONE

The CD-PVAA blade with the PVA-CCIS Application can be programmed utilizing the three methods of programming on the SV8100 system: PC Pro, Web Pro and Handset Programming.

License assignment in program 10-54-01 must be done before the system recognizes the CD-PVAA blade.

1.   To program the system, the CD-PVAA blade should be installed in the desired slot.

2.   The CD-PVAA with the PVA-CCIS Application package installed will consume the number of ports licensed on the CD-CP00, and programmed in PRG 10-40-04 and PRG 10-55-02.

3.   Access the SV8100 System Programming mode.

✎   Consult the System Administrator for the system password.

4.   Access Data Program 10-03-01 (Blade setup), and enter the slot number of the CD-PVAA blade to verify which ports are used. In the following example, the CD-PVAA has been seated in slot 6:

Example: 10-03-02 Slot No 6
PVA Logical Port 1
Port 1 (Trunk 1) is the first port assigned to the CD-PVAA.

SECTION 7   LOADING THE COMPACTFLASH APPLICATION IN THE CD-PVAA

In this section, the procedure for loading a CompactFlash (CF) into the CD-PVAA is provided. The same procedure applies for any other application (e.g., IVR, CNF).

❍   Verify Service Package prior to installing the application.

❍   Access to this website requires login privileges.

7.1   Loading The CompactFlash Application On The CD-PVAA

A successful application installation resets all system configurations to the default settings. Make note of the card configuration prior to performing a firmware upgrade.

1. Unzip the installation files to a USB drive. The USB drive should be 512MB or larger, and not have any other files on it.

    There will be two files once unzipped. For example, if your USB is drive F: the files will be extracted to the following location on your USB drive:

    F:/ install/ccis_verA3.pkg
    F:/install/ ccis_verA3.pkg.md5

2. Remove the CD-PVAA from the chassis and confirm the CompactFlash is located in the CompactFlash slot.

3. Insert the the USB stick in the CD-PVAA USB slot.

4. Re-seat the PVA Blade in the Chassis.

5. Upon boot up, the installation process will begin updating the application on the CompactFlash.

    It will take approximately 10 minutes for the upgrade to complete. Do not remove the USB drive, unseat the CD-PVAA, or turn off the phone system during this process.

6. Once the card has completed the installation process, the CD-PVAA will automatically reboot.

7. Log back into the PVA to confirm the application has updated. Login Credentials - admin/password.

8. Select the System button in far left corner to confirm the version. If the installation process does not complete successfully, repeat steps 2~7 above.

9. Remove USB drive from CD-PVAA.

SECTION 8   **KCCIS-IP WITH PVA APPLICATION CONFIGURATION**

When installed with a PVA-CCIS CF drive, the PVA-CCIS Application comes up with the IP Address parameters assigned in programming, PRG 10-55-01 for the slot the blade is installed.

When installed without a PVA-CCIS CF drive, the PVA-CCIS Application comes up with the factory default parameters shown in Table 6-1 Network Settings for TCP/IP Addressing.

Table 6-1 Network Settings for TCP/IP Addressing

IP Address	192.168.1.100
Subnet Mask	255.255.255.0
Default Gateway	0.0.0.0

In this configuration, the blade cannot run in every environment. Therefore, its parameters must be defined again before it can work in your environment.

❍   Internet Explorer 6.0 or higher is recommended.

❍   The network setting example may not match your environment setting. This information totally depends on the default IP setting in Data Program 10-55-01. Be sure to reference this Data Program prior to configuring your network settings.

8.1   Configuring the PC

After the CD-PVAA with the PVA-CCIS Application is installed in the chassis, the PC that is communicating with the PVA-CCIS Application must be configured to recognize the PVA-CCIS blade. The PC and PVA-CCIS blade must be connected using a straight Ethernet cable, crossover cable, and/or a small hub. To ensure proper communication between the PC and the PVA-CCIS blade, they must be attached to the same network.

To configure the PC:

1.   Press **Start** and select **Control Panel** from the menu.

✎   *Windows XP Professional operating system is used in this example. When using a different Windows operating system, the screens may look slightly different.*
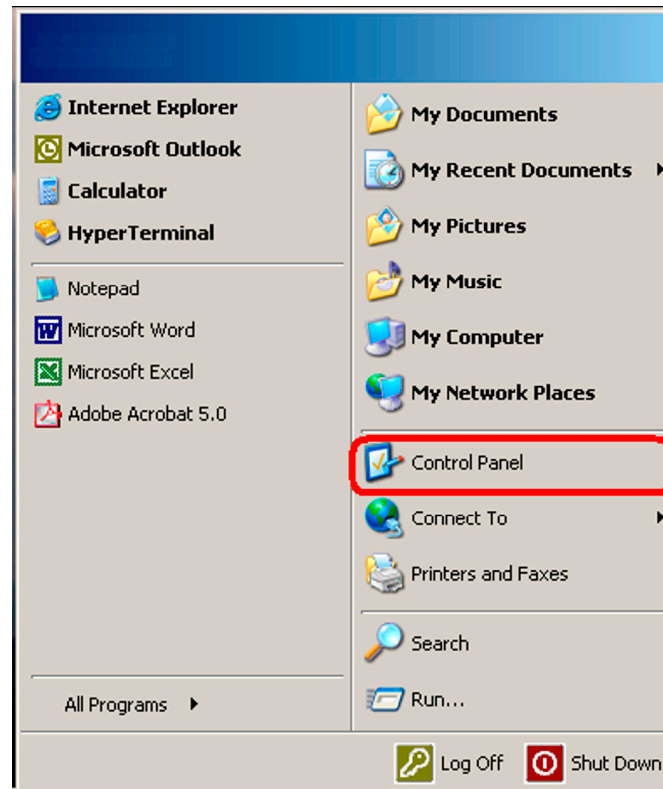
Figure 6-4 Windows Start Screen

2.   On the Control Panel menu, double click on **Network Connections**.
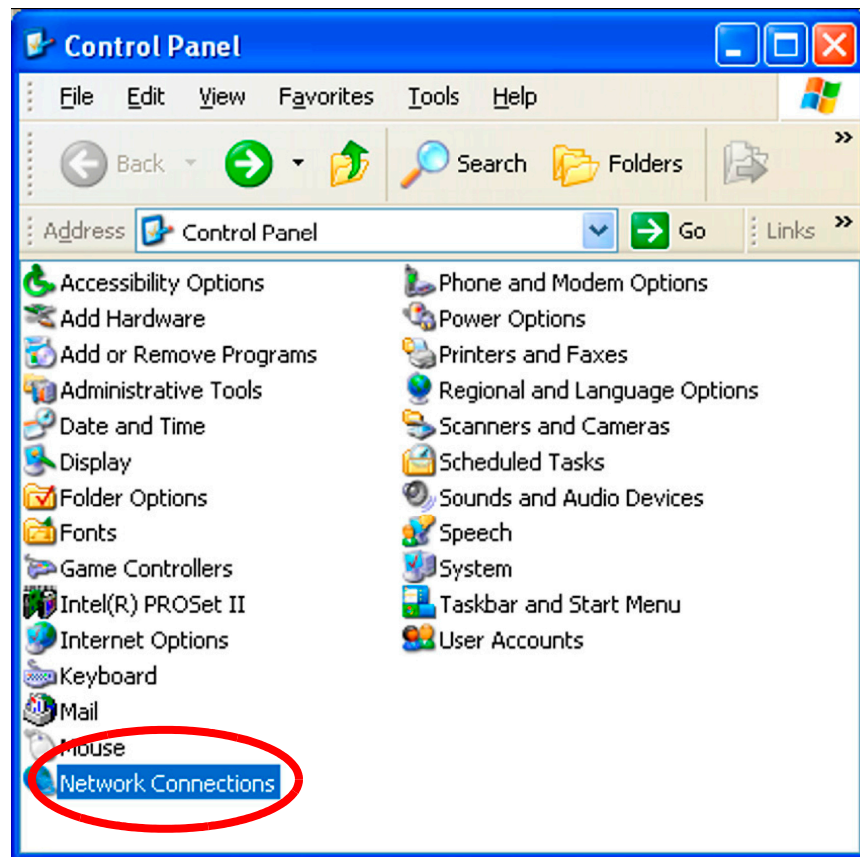
Figure 6-5 Select Network Connections

3.   From the Network Connections screen, double click on **Local Area Connection**.
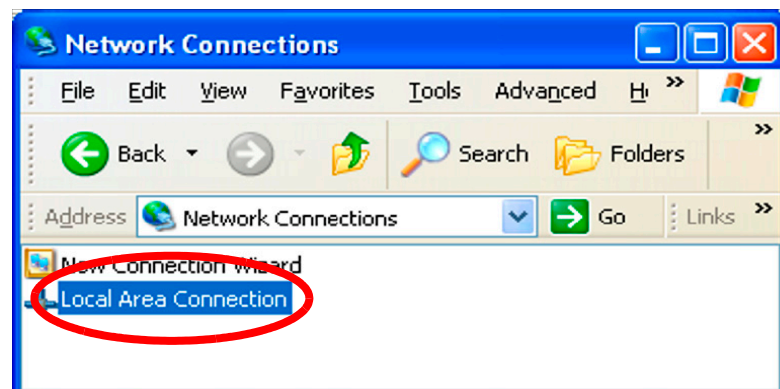
Figure 6-6 Click on Local Area Connection

4.   When the Local Area Connection Status screen is displayed, click the
      Properties button.
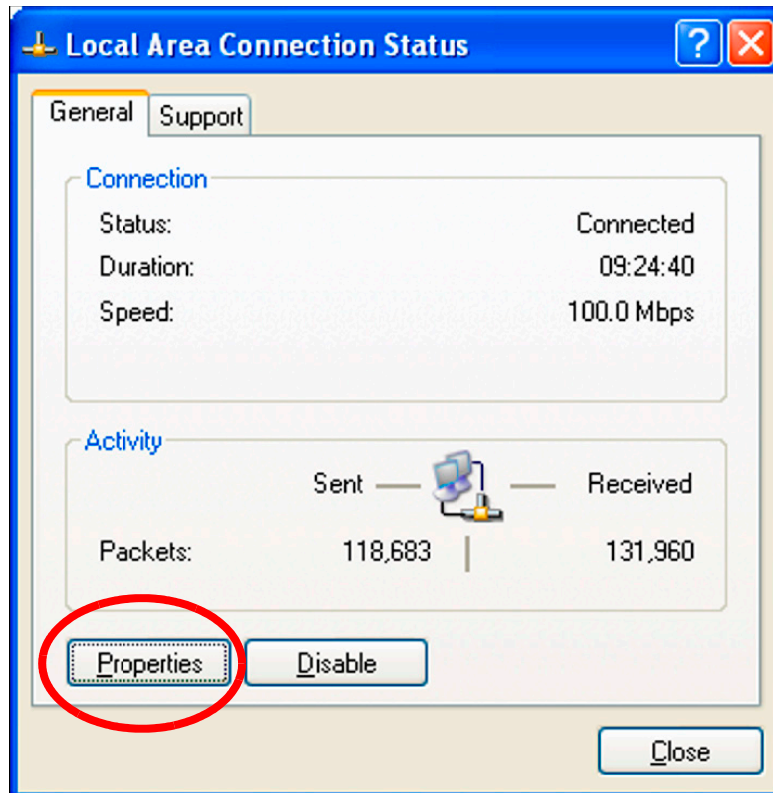
Figure 6-7 LAN Connection Status Screen

5.   From the Local Area Connection Properties screen, select **Internet Protocol (TCP/IP)**, and then **Properties**.
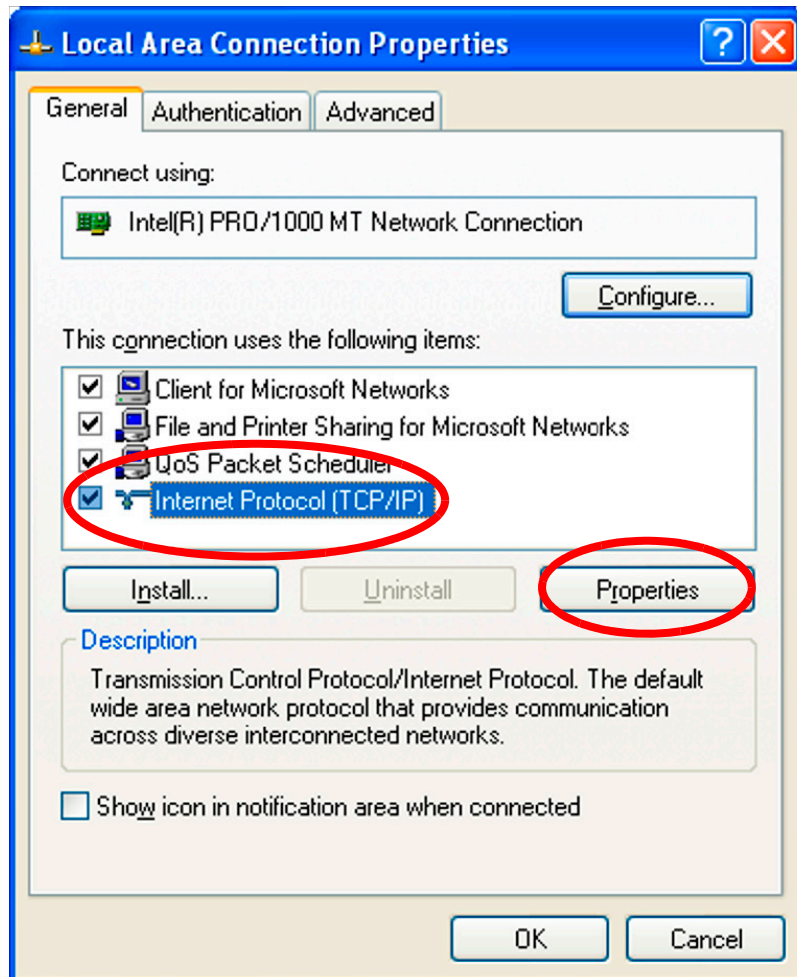
Figure 6-8 Selecting Internet Protocol Properties

6.   Click on **Use the following IP address**. For this example, enter
      192.168.1.99 for the IP address, and 255.255.255.0 for the Subnet
      mask. Click **OK** to accept the changes.
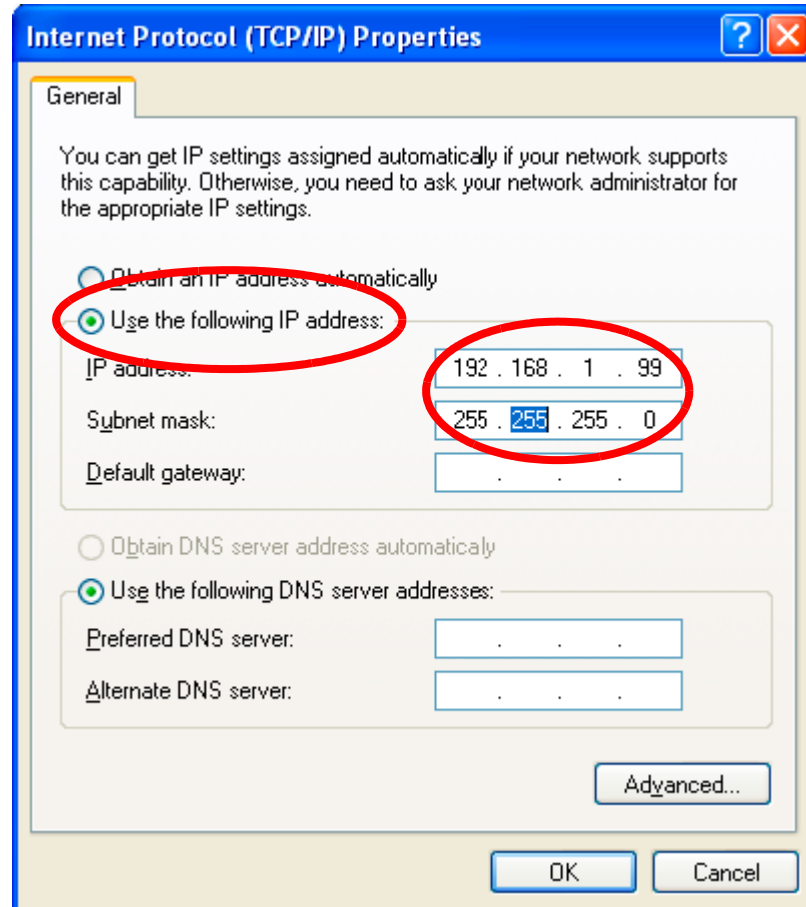
Figure 6-9 Enter IP Address and Subnet Mask

7.   On the Local Area Connection Status screen, click **Close**, then close the
      Network Connections screen by clicking ⊠ .

SECTION 9   **KCCIS-IP WITH PVA APPLICATION WEB PROGRAMMING**

The SV8100 CD-PVAA blade Web Programming is used for viewing firmware versions, password change, resetting the blade and Debugging only. All network configurations are made using Univerge SV8100 System programming.

✎   After the PVA-CCIS CF Application is installed the default IP Address 192.168.1.100 is no longer effective. The IP address, configured in SV8100 PRG 10-55-01 for the slot the PVA-CCIS blade is installed in, goes into effect. Refer to the Univerge SV8100 Programming Manual for more details.

9.1   PVA-CCIS Application Interface

The PVA-CCIS Web Interface application allows the PVA-CCIS user to interact with the application. The Web interface can be used for viewing system information, turning on PVA-CCIS traces, changing the password of the PVA-CCIS application and performing a forced or safe reset.

9.2   Logging Into PVA-CCIS Application

The PVA-CCIS Web Interface application is configured using an Internet Browser. Microsoft Internet Explorer 6.0 or higher is recommended. The Login screen allows a user to enter their user name and password for network access.
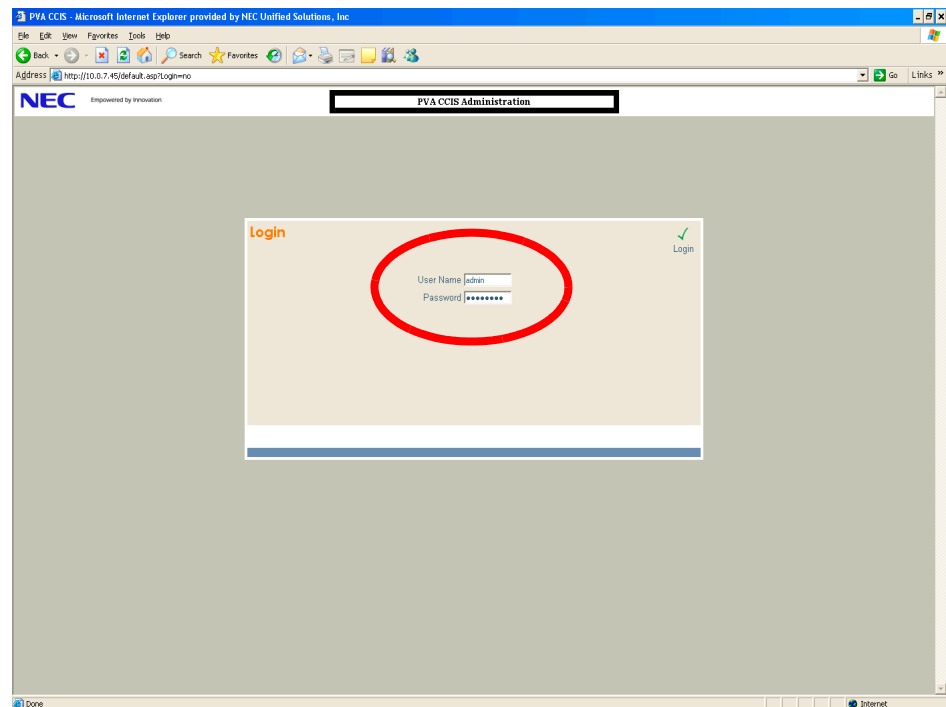
Figure 6-10 PVA-CCIS Application Admin Screen

9.2.1   To log in:

1.   Start your Internet Explorer from a PC connected to the same network as the PVA-CCIS Application.

2.   Enter the PVA-CCIS Application default IP Address (192.168.1.100) in the address link of your browser.

3.   When the Login screen is displayed, enter the default User Name (admin) and the default Password (password).

4.   After entering the User Name and Password, click **Login** to access the PVA-CCIS Application.

9.3   System

This screen provides information about the Firmware Revision and Card Type.
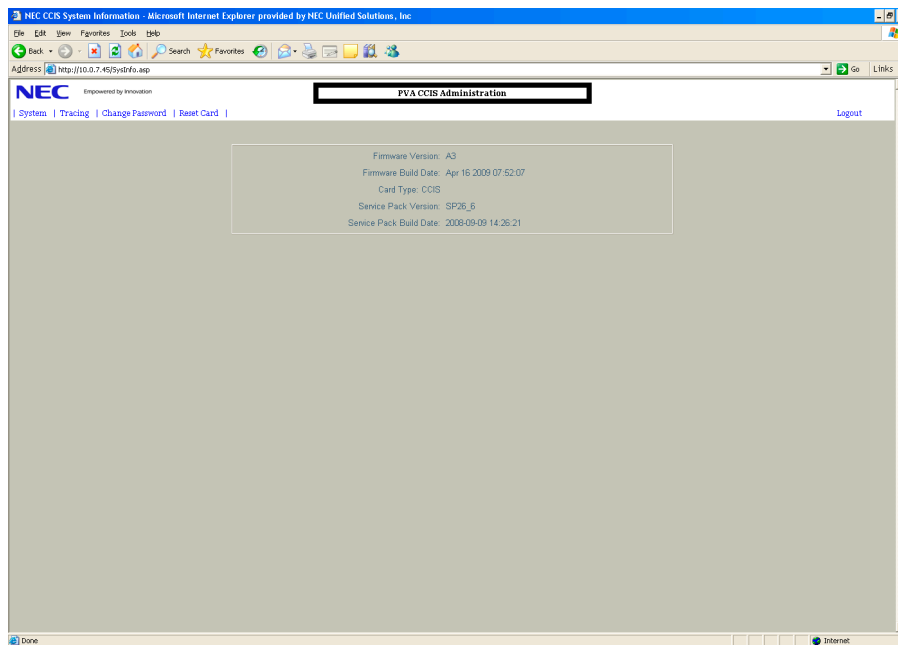
Figure 6-11 System Screen

9.4   Tracing

This screen is used to analyze communication errors and is intended to be used by technical support personnel. Keep all boxes unchecked during normal operation, unless otherwise specified.
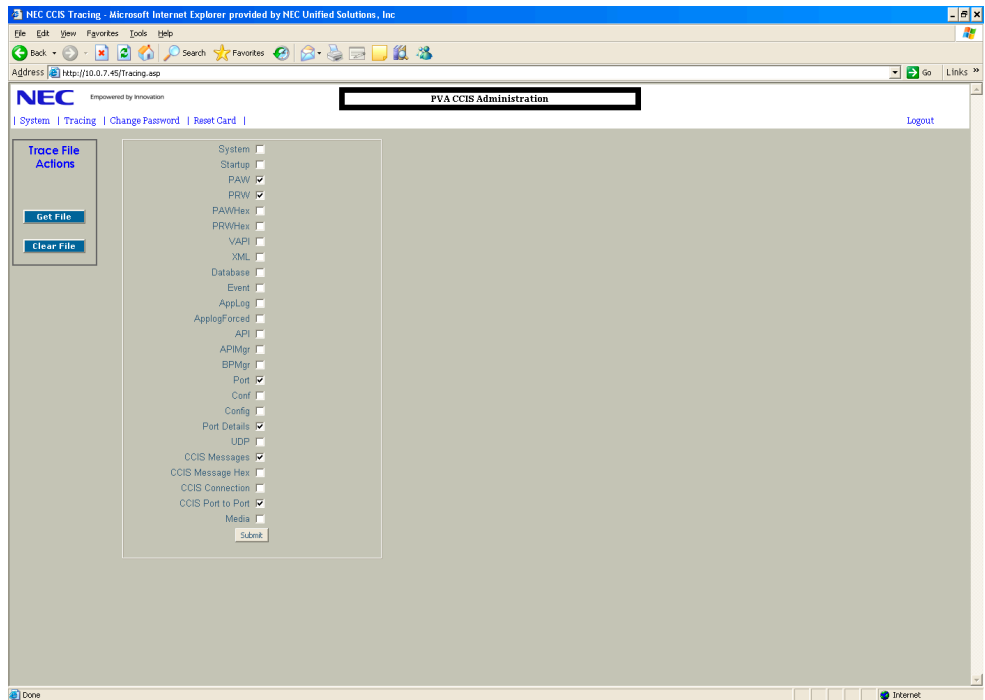
Figure 6-12 Tracing Screen

9.5   Reset Password

Reset password enables the **admin** user to change the current password. You cannot change the user name from **admin**.
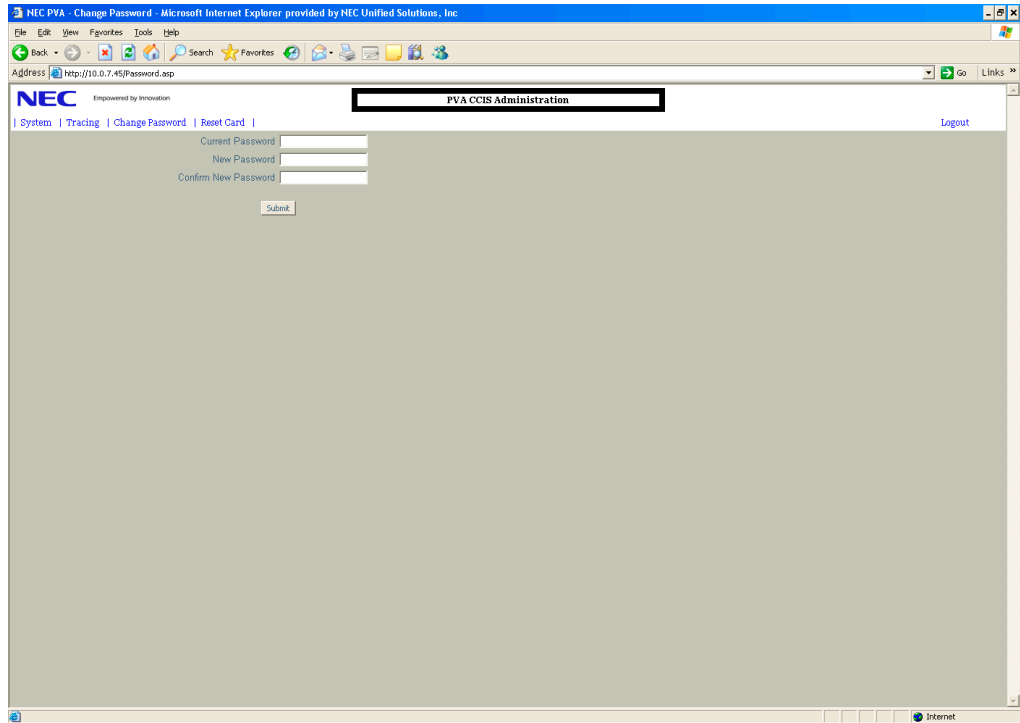
Figure 6-13 Reset Password Screen

9.6   Reset Card

Reboot is necessary when changing the operating modes and network participation parameters.

Safe Reset occurs when it is selected. The PVA blade resets only when no calls are in progress.
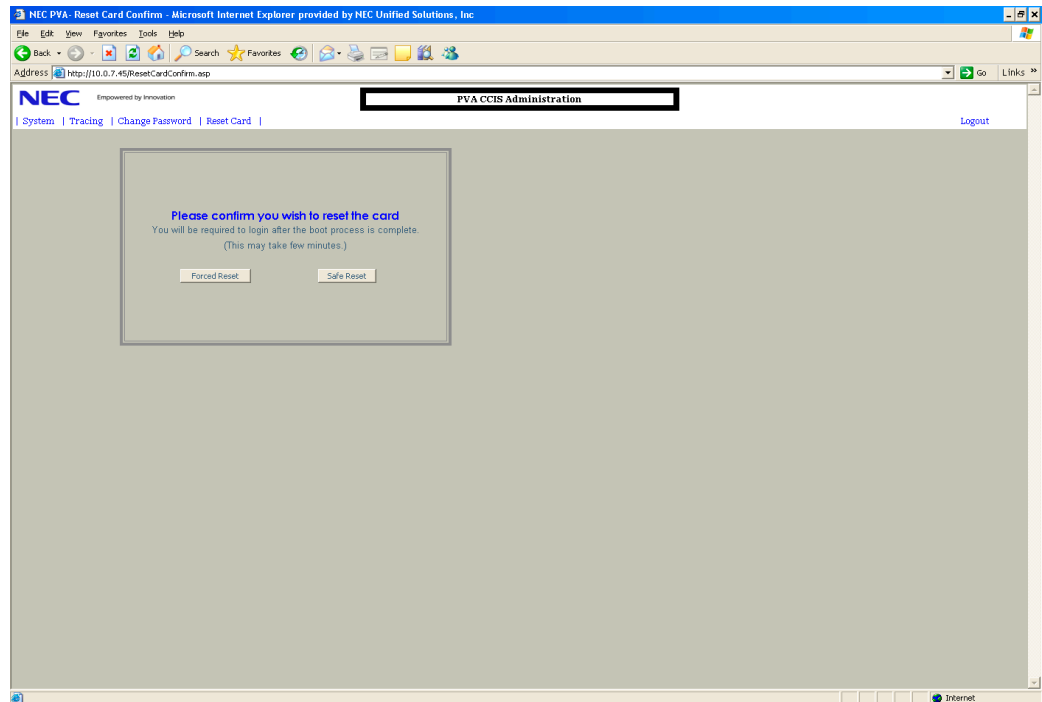
Figure 6-14 Reset Card Screen

9.7   Logout

Logout disconnects from the PVA blade.

SECTION 10   UNIVERGE SV8100 PROGRAMMING

The following Data Programs are used when installing the CD-PVAA blade for PVA-CCIS.

CD-PVA Setup

Program /Item No.	Description/Selection	Assigned Data	Comments
10-03-01	PCB Setup – Logical Port Number	0~200	The CD-CP00 automatically assigns each blade during installation.
10-54-01	License Configuration for Each Package - License Code	0000~9999	The PVA-CCIS requires license code 6200.
10-54-02	License Configuration for Each Package - License Quantity	0~255	Set the license information for each unit.
10-55-01	Package Network Setup – IP Address	0.0.0.0~126.255.255.254 128.0.0.1~191.255255.254 192.0.0.1223.255.255.254 (default = 172.16.1.100)	Define the IP Address for the CD-PVA.When the blade is deleted from the system using Program 90-05, the programming for the slot in 10-55 is set back to default.
10-55-03	Package Network Setup – Main/Add-on	Use Main setting to distribute an IP Address to the blade.	0 = Main 1 = Add-on (default = 1)

Program /Item No.	Description/Selection	Assigned Data	Comments
10-55-04	Package Network Setup – Sub Net Mask	Define the subnet mask for the CD-PVA.	128.0.0.0 192.0.0.0 224.0.0.0 240.0.0.0 248.0.0.0 252.0.0.0 254.0.0.0 255.0.0.0 255.128.0.0 255.192.0.0 255.224.0.0 255.240.0.0 255.248.0.0 255.252.0.0 255.254.0.0 255.255.0.0 255.255.128.0 255.255.192.0 255.255.224.0 255.255.240.0 255.255.248.0 255.255.252.0 255.255.254.0 255.255.255.0 255.255.255.128 255.255.255.192 255.255.255.224 255.255.255.240 255.255.255.248 255.255.255.252 255.255.255.254 255.255.255.255 (default = 255.255.0.0)
10-55-05	Package Network Setup – Default Gateway	Define the default gateway for the CD-PVA.	0.0.0.0~126.255.255.254 128.0.0.1~191.255255.254 192.0.0.1~223.255.255.254 (default = 0.0.0.0)

System Numbering

Program /Item No.	Description/Selection	Assigned Data	Comments
11-01-01	System Numbering – Service Code	Customize the system internal (Intercom) numbering plan.	Refer to Univerge SV8100 System Program Manual.
11-02-01	Extension Numbering	Up to eight digits 1 - 101 2 - 102 3 - 103 ~ ~ 99 - 199 100 - 3101 ~ ~ 512 - 3513	Assign extension numbers to extension ports. The telephone programming identity follows the port number – not the extension number.

Trunk Programming

Program /Item No.	Description/Selection	Assigned Data	Comments
14-05-01	Assign Trunks to Trunk Groups/Outbound Priority	Trunk Group = 1◄ Priority = 1 ~ 200 ◄ = Default	Default priorities for trunks 1~200 is 1~200.
22-02-01	Incoming Call Trunk Setup	0=Normal◄ 1=VRS 2=DISA 3=DID 4=DIL 5=E&M Tie Line 6=Delayed DID 7=ANI/DNIS ◄ = Default	Set the feature type for the trunk you are programming. Second dial tone for option 1 if no VRS is installed.
34-01-01	DID/E&M Start Signaling	0=2nd Dial Tone 1=Wink◄ 2=Immediate 3=Delay ◄ = Default	Set the signaling mode for DID and TIE trunks. DID and TIE trunks can use either Immediate start or Wink start signaling.

F-Route Programming

Program /Item No.	Description/Selection	Assigned Data	Comments
44-02-01	Dial Analysis Table for ARS/ F-Route – Number of digits to be analyzed by the system.	Up to "8" digits can be assigned. Default is not assigned	Assign the digits to be dialed across the K-CCIS link. These digits were assigned as F-Route in Program 11-01-01. (Use line key "1" for "Don't Care" digit, @).
44-02-02	Dial Analysis Table for ARS/ F-Route – Service Type.	2=ARS/F-Route Table. Default is not assigned.	The service type (2) assigns the digits to be dialed to an F-Route. Program 44-02-03 assigns the F-Route to be used.
44-02-03	Dial Analysis Table for ARS/ F-Route – Additional Data.	2=0~500 (0=No Setting) Default is not assigned.	When setting data is (2), refer to Program 44-05.
44-05-01	ARS/F-Route Table – Trunk Group Number	Trunk Groups (1 ~ 100)	Select the trunk group to be used for the outgoing K-CCIS call.
44-05-02	ARS/F-Route Table – Delete Digits	0=No Setting◄ 1 ~ 255 = Number of digits to delete. (255 = Delete All) ◄ = Default	Enter the number of digits to be deleted from the dialed number.
44-05-03	ARS/F-Route Table – Additional Dial Number Table	0=No Setting◄ 1 ~ 1000 = Add Table in Program 44-06-01. ◄ = Default	Enter the table number (defined in Program 44-06) for additional digits to be dialed.
44-05-04	ARS/F-Route Table – Beep Tone	0 = Off◄ 1 = On ◄ = Default	Select whether or not a beep is heard, if a lower priority trunk group is used to dial out.
44-05-05	ARS/F-Route Table – Gain Table Number for Internal Calls	0 = No Setting◄ 1 ~ 500 ◄ = Default	Select the gain table number to be used for the internal call (defined in Program 44-07).
44-05-06	ARS/F-Route Table – Gain Table Number for Tandem Connections	0 = No Setting◄ 1 ~ 500 ◄ = Default	Select the gain table number to be used for the tandem call (defined in Program 44-07).

Program /Item No.	Description/Selection	Assigned Data	Comments
44-05-07	ARS/F-Route Table – ARS Class of Service	0 = No Setting◄ 1 ~ 16 ◄ = Default	Select the ARS Class of Service to be used for the table. An extension ARS COS is determined in Program 26-04-01.
44-05-08	ARS/F-Route Table – Dial Treatment	0 = No Setting◄ 1 ~ 15 ◄ = Default	If a Dial Treatment is selected, Programs 44-05-02 and 44-05-03 are ignored and the Dial Treatment defined in Program 26-03-01 is used instead.
44-05-09	ARS/F-Route Table – Max Digit Assignment	0 = No Maximum◄ 1 ~ 24 ◄ = Default	Assign the Max Digits for K-CCIS or ISDN Calls.
44-05-10	ARS/F-Route Table – CCISoIP Destination Point Code	0 = No Maximum 1~24	

PVA-CCIS Setup

Program /Item No.	Description/Selection	Assigned Data	Comments
50-01-01	CCIS System Setting – CCIS Availability	1 = Enable 2 = Disable◄ ◄ = Default	Set the availability of CCIS in the SV8100. No other CCIS settings function if this program is disabled.
50-02-03	Originating Point Code – Route ID 9	Assign Originating Point Code Default Not Assigned.	Route ID 1~8: CCIS via CD-CCTA Route ID 9: CCIS via PZ-32/64/128IPLA (IP-CCIS) and CD-PVA KCCIS – IP Point to Multipoint.
50-03-01	CCIS Destination System Settings – Destination System Point Code	Assign the IP Address to a CCIS System ID. Default Not Assigned.	Assign remote system IP Network information. PRG 50-03 assignments are only used for CCISoIP.
50-03-03	CCIS Destination System Settings – Destination System IP Address	xxx.xxx.xxx.xxx (xxx = 0~255)	Assign remote system IP Network information. PRG 50-03 assignments are only used for CCISoIP.
50-03-04	CCIS Destination System Settings – Point Code Availability	0:Disable 1:Enable◄ ◄ = Default	

PVA-CCIS Data Setup

Program /Item No.	Description/Selection	Assigned Data	Comments
84-06-01	PVA Data Setting – RTP Port Number	0~65534 Default is 10020	
84-06-02	PVA Data Setting – RTCP Port Number	RTP Port Number + 1 Default is 10021	It has to be RTP Port Number + 1.
84-06-04	PVA Data Setting – Fract Lost Threshold	0~100% Default is 0	The data is sent to the CD-CP00-US if the value exceeds the defined value.
84-06-05	PVA Data Setting – Packets Lost Threshold	0~16777215 Default is 0	The data is sent to the CD-CP00-US if the value exceeds the defined value.

Program /Item No.	Description/Selection	Assigned Data	Comments
84-06-07	PVA Data Setting – Jitter Threshold	0~4294967295 (sec) Default is 0	The data is sent to the CD-CP00-US if the value exceeds the defined value.
84-06-09	PVA Data Setting – Delay LSR Threshold	0~4294967295 (sec) Default is 0	The data is sent to the CD-CP00-US if the value exceeds the defined value.
84-06-16	PVA Data Setting – IMCP Redirect	0=No 1=Yes Default is 0	The VOIP blade supports sending the Internet Message Control Protocol (IMCP) redirect message.

PVA-CCIS VLAN Setup

Program /Item No.	Description/Selection	Assigned Data	Comments
84-09-01	VLAN Setup – VLAN (Future Use)	0 = Disable (Off) ◄ 1 = Enable (On) ◄ = Default	
84-09-02	VLAN Setup – VLAN ID (Future Use)	1~4094 Default is 0	
84-09-03	VLAN Setup - Priority (Future Use)	0~7 Default is 0	

PVA-CCIS ToS Setup

Program /Item No.	Description/Selection	Assigned Data	Comments
84-10-01	ToS Setup – ToS Mode	0 = Disable (Invalid) ◄ 1 = IP Precedence 2 = Diffserv ◄ = Default	
84-10-02	ToS Setup – Priority, IP Precedence	0~7 0 = Low◄ 7 = High ◄ = Default	
84-10-03	ToS Setup – Low Delay	0~1 0 = Normal Delay, Low Delay Default is 0	
84-10-04	ToS Setup – Wideband (Throughout)	0~1 0 = Normal Throughput◄ 1 = High Throughput ◄ = Default	
84-10-05	ToS Setup – High Reliability 0~1 0 = Normal Reliability◄ 1 = Low Reliability ◄ = Default		
84-10-07	ToS Setup - Priority (D.S.C.P. - Differentiated Services Code Point)	0~63 Default is 0	

PVA-CCIS CODEC Setup

Program /Item No.	Description/Selection	Assigned Data	Comments
84-30-01	PVA-CCIS CODEC Information Basic Setup – Number of audio frames (G.711)	1: 20ms 2: 30ms◄ ◄ = Default	
84-30-02	PVA-CCIS CODEC Information Basic Setup – Type of G.711	0: A-law 1: u-law◄ ◄ = Default	
84-30-03	PVA-CCIS CODEC Information Basic Setup – Mode of G.711	0: Disable◄ 1: Enable ◄ = Default	
84-30-04	PVA-CCIS CODEC Information Basic Setup – Minimum Jitter Buffer (G711)	0~200ms Default is 30	
84-30-05	PVA-CCIS CODEC Information Basic Setup – Average Jitter Buffer (G.711)	0~200ms Default is 60	
84-30-06	PVA-CCIS CODEC Information Basic Setup – Maximum Jitter Buffer (G.711)	0~200ms Default is 120	
84-30-07	PVA-CCIS CODEC Information Basic Setup – Number of G.729 audio frames	1: 20ms 2: 30ms◄ 3: 40ms ◄ = Default	
84-30-08	PVA-CCIS CODEC Information Basic Setup – Mode of G.729 VAD	0: Disable◄ 1: Enable ◄ = Default	
84-30-09	PVA-CCIS CODEC Information Basic Setup – Minimum G.729 Jitter Buffer	0~500ms Default is 30ms	
84-30-10	PVA-CCIS CODEC Information Basic Setup – Average G.729 Jitter Buffer	0~500ms Default is 60ms	
84-30-11	PVA-CCIS CODEC Information Basic Setup – Maximum G.729 Jitter Buffer	0~500ms Default is 120ms	
84-30-12	PVA-CCIS CODEC Information Basic Setup – Number of G.723 audio frames	1: 30ms◄ 2: 60ms ◄ = Default	

Program /Item No.	Description/Selection	Assigned Data	Comments
84-30-13	PVA-CCIS CODEC Information Basic Setup – Mode of G.723 VAD	0: Disable◄ 1: Enable ◄ = Default	
84-30-14	PVA-CCIS CODEC Information Basic Setup – Minimum G.723 Jitter Buffer	0~500ms Default is 30ms	
84-30-15	PVA-CCIS Codec Information Basic Setup – Average G.723 Jitter Buffer	0~500ms Default is 60ms	
84-30-16	PVA-CCIS Codec Information Basic Setup – Maximum G.723 Jitter Buffer	0~500ms Default is 120ms	
84-30-17	PVA-CCIS CODEC Information Basic Setup – TX Gain	0~40 (-20dBm to +20dBm) 0: -20dBm 1: -19dBm : 20: 0dBm : 39: 10dBm 40: 20dBm Default is 20	
84-30-18	PVA-CCIS CODEC Information Basic Setup – RX Gain	0~40 (-20dBm to +20dBm) 0: -20dBm 1: -19dBm : 20: 0dBm : 39: 10dBm 40: 20dBm Default is 20	
84-30-19	PVA-CCIS CODEC Information Basic Setup – 1st Priority of Audio Capability	0: G.711_PT◄ 1: G.723_PT 2: G.729_PT ◄ = Default	
84-30-20	PVA-CCIS CODEC Information Basic Setup – 2nd Priority of Audio Capability	0: G.711_PT 1: G.723_PT◄ 2: G.729_PT ◄ = Default	

Program /Item No.	Description/Selection	Assigned Data	Comments
84-30-21	PVA-CCIS CODEC Information Basic Setup – DTMF Relay Mode	0: Disable◄ 1: Inbound 2: Outbound ◄ = Default	
84-30-22	PVA-CCIS CODEC Information Basic Setup – Jitter Buffer Mode	1: Static 2: Immediate (no tone) 3: Immediate◄ ◄ = Default	
84-30-23	PVA-CCIS CODEC Information Basic Setup – VAD Threshold	0~30 (19db to + 10db) 0: Auto Default is 20db	
84-30-24	PVA-CCIS CODEC Information Basic Setup – Echo Canceller Mode	0: Disable 1: Enable◄ ◄ = Default	
84-30-25	PVA-CCIS CODEC Information Basic Setup – Echo Canceller NLP Mode	0: Disable 1: Enable ◄ = Default	
84-30-26	PVA-CCIS CODEC Information Basic Setup – UDP check Sum Mode	0: Disable 1: Enable◄ ◄ = Default	

SECTION 11   UNIVERGE SV8100 KCCIS-IP WITH PVA PROGRAMMING EXAMPLE 1

This is an example of UNIVERGE SV8100 program data assignments for a 4-digit Closed Numbering Plan. The following system configurations are used for all four systems:

❍   Each system is a single-cabinet, expanded port package with one CHS2U-US.

❍   Each system has 16 Multiline Terminals.

❍   Each system has eight analog Trunks.
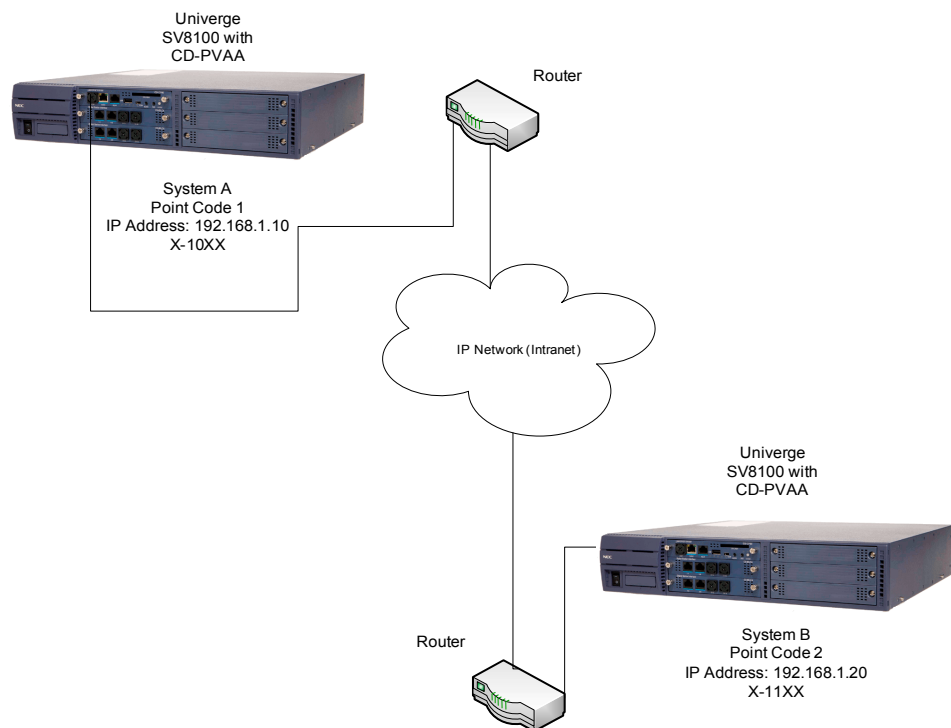
Figure 6-15 PVA Programming Example 1

It is assumed that the systems were defaulted (first power on) with the following blades installed as described below.

11.1   Card Interface Slot Assignment (PRG 10-03)

Slot	Card Type During 1st Power On	CARD TYPE AND PORTS			
		System A		System B	
		Card Type	Ports	Card Type	Ports
1	CD-CP00-US	CD-CP00-US		CD-CP00-US	
2	CD-8DLCA	CD-8DLCA	Station 1 ~ 8	CD-8DLCA	Station 1 ~ 8
3	CD-8DLCA	CD-8DLCA	Station 9 ~ 16	CD-8DLCA	Station 9 ~ 16
4	CD-PVAA	CD-PVAA	Trunks 1 ~ 24	CD-PVAA	Trunks 1 ~ 24
5	CD-4COTB	CD-4COTB	Trunks 25 ~ 32	CD-4COTB	Trunks 25 ~ 32
6	None	None	N/A	None	N/A

11.2   PVA-CCIS Trunk Assignments

Use the table below to make the appropriate assignments for digital trunks.

System A	System B
Program 10-03-01 1. Insert CD-CP00-US in Chassis 1, Slot 1. 2. Insert CD-PVA with PVA-CCIS CF in Slot 4. 3. Verify PRG 10-03-01 (Blade Setup).	Program 10-03-01 1. Insert CD-CP00-US in Chassis 1, Slot 1. 2. Insert CD-PVA with PVA-CCIS CF in Slot 4. 3. Verify PRG 10-03-01 (Blade Setup).
Program 10-54-01 1. Slot 4 – ID01 License Code = 6200.	Program 10-54-01 1. Slot 4 – ID01 License Code = 6200.
Program 10-54-02 1. ID01 License Code = 24.	Program 10-54-02 1. ID01 License Code = 24.
Program 14-05-01 Assign Trunks 1 ~ 24 in Trunk Group 10	Program 14-05-01 Assign Trunks 1 ~ 24 in Trunk Group 10
Program 22-02-02 Set Trunks 1 ~ 24 as Tie Lines (LK6).	Program 22-02-02 Set Trunks 1 ~ 24 as Tie Lines (LK6).

11.3   Tandem Connection Assignments

Use the following table to assign Tandem connections.

System A	System B
Program 14-01-13 Trunk to Trunk Transfer 0 = Disable 1 = Enable (Default)	Program 14-01-13 Trunk to Trunk Transfer 0 = Disable 1 = Enable (Default)
Program 14-01-24 Trunk to Trunk Outgoing Caller ID Through Mode 0 = Disable (Off) (Default) 1 = Enable (On)	Program 14-01-24 Trunk to Trunk Outgoing Caller ID Through Mode 0 = Disable (Off) (Default) 1 = Enable (On)

11.4   VoIP Address Assignments

Use the following table to assign VoIP Addresses.

System A	System B
Program 10-55-01 Slot 4, Assign IP Addresses 192.168.1.10	Program 10-55-01 Slot 4, Assign IP Addresses 192.168.1.20
Program 10-55-03 Slot 4, Main	Program 10-55-03 Slot 4, Main
Program 10-55-04 Slot 4, Subnet Mask 255.255.255.0	Program 10-55-04 Slot 4, Subnet Mask 255.255.255.0
Program 10-55-05 Slot 4, Default Gateway = 0.0.0.0	Program 10-55-05 Slot 4, Default Gateway = 0.0.0.0

11.5   CCIS Availability

Use the table below to make the appropriate assignments for CCIS availability.

System A	System B
Program 50-01-01 Select Option 1 (Enable) Enables the CCIS Feature	Program 50-01-01 Select Option 1 (Enable) Enables the CCIS Feature
Program 50-02-03 Assign Route ID 9 = 01 (Route ID 9 is designated for Host Originating Point Code)	Program 50-02-03 Assign Route ID 9 = 02 (Route ID 9 is designated for Host Originating Point Code)

11.6   CCISoIP Assignment

Use the table below to make the appropriate assignments for CCIS availability.

System A	System B
Program 50-03-01 System ID 1 ~ 255 – Enter Destination Point Code System ID 1 = DPC 02	Program 50-03-01 System ID 1 ~ 255 – Enter Destination Point Code System ID 1 = DPC 01
Program 50-03-03 System ID 1 ~ 255 – Enter Destination System IP Address. System ID 1 = 192.168.1.20	Program 50-03-03 System ID 1 ~ 255 – Enter Destination System IP Address System ID 1 = 192.168.1.10

11.7   Local Numbering Plan Assignment

System A	System B
Program 11-01-01 1 = Service Code 2 = Extension Number 3 = Trunk Access Code 4 = 2nd Trunk Access Code 5 = Operator Access 6 = ARS/F-Route Access	Program 11-01-01 1 = Service Code 2 = Extension Number 3 = Trunk Access Code 4 = 2nd Trunk Access Code 5 = Operator Access 6 = ARS/F-Route Access

System A	System B
Program 11-02-01 Assign Station Numbers to Port Numbers. Defaults for Ports 1 ~ 512: 101 ~ 198 3101 ~ 3199 3200 ~ 3513	Program 11-02-01 Assign Station Numbers to Port Numbers. Defaults for Ports 1 ~ 512: 101 ~ 198 3101 ~ 3199 3200 ~ 3513

11.8   Closed Numbering Plan – Using Closed Number Blocks

System A	System B
Program 11-01-01 Dial 10 = 4 Digit; Extension Dial 11 = 4 Digit; F-Route	Program 11-01-01 Dial 10 = 4 Digit; F-Route Dial 11 = 4 Digit; Extension
Program 11-02-01 Assign Station Numbers to Port Number Defaults for Ports 1 ~ 512 101 ~ 198 3101 ~ 3199 3200 ~ 3513	Program 11-02-01 Assign Station Numbers to Port Number. Defaults for Ports 1 ~ 512 101 ~ 198 3101 ~ 3199 3200 ~ 3513
Program 44-02-01 TBL 1 = 11	Program 44-02-01 TBL 1 = 10
Program 44-02-02 TBL 1 = F-Route	Program 44-02-02 TBL 1 = F-Route
Program 44-02-03 F-Route TBL 1 = 1	Program 44-02-03 F-Route TBL 1 = 1
Program 44-05-01 F-Route TBL 1 = TG10	Program 44-05-01 F-Route TBL 1 = TG10
Program 44-05-02 F-Route TBL 1 Delete Digit = 0	Program 44-05-02 F-Route TBL 1 Delete Digit = 0
Program 44-05-03 F-Route TBL 1 Add Dial = 0	Program 44-05-03 F-Route TBL 1 Add Dial = 0
Program 44-05-09 F-Route TBL 1 Max Digit = 4	Program 44-05-09 F-Route TBL 1 Max Digit = 4
Program 44-05-10 F-Route TBL1 = DPC 2	Program 44-05-10 F-Route TBL1 = DPC 1

SECTION 12   UNIVERGE SV8100 KCCIS-IP WITH PVA AND ELECTRA ELITE IPK/ IPKII PROGRAMMING EXAMPLE 2

The following example provides programming details for three systems connected by the PVA-CCIS application.
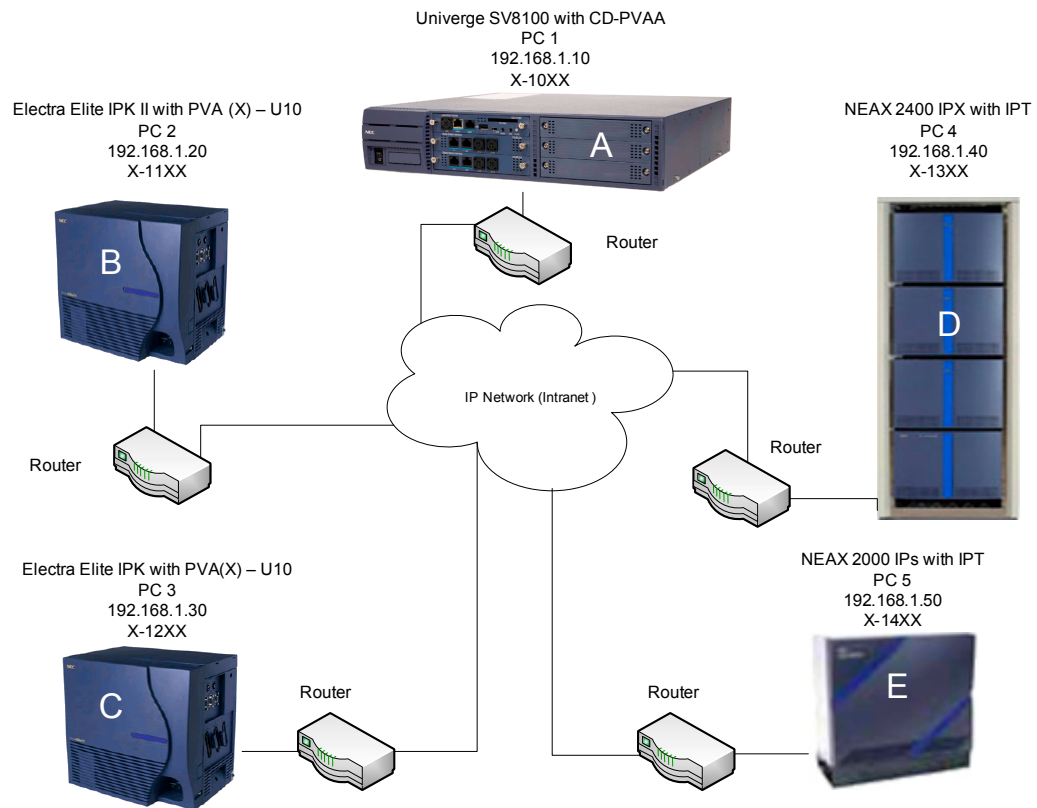
Figure 6-16 PVA Programming Example 2

12.1   Programming Assignments

For this example, the following system configurations are used:

❍   A 4-digit, Closed Numbering Plan using Closed Number Blocks is used.

❍   System A is a single-cabinet UNIVERGE SV8100 with one CHS2U-US and one CD-PVAA blade with the PVA-CCIS CF Application.

❍   System B is a single-cabinet Electra Elite IPKII Expanded port package with one B64-U20 KSU and one PVA(X)-U10 ETU.

❍   System C is a single-cabinet Electra Elite IPK Expanded portpackage with one B64-U20 KSU and one PVA(X)-U( ) ETU.

❍   System D and E are NEAX systems. Refer to the applicable system manuals for programming details.

❍   Each system has eight multiline terminals.

❍   Each system has eight analog trunks.

It is assumed that the systems were defaulted (first power on) with the following cards installed as described below.

12.2   Card Interface Slot Assignment (PRG 10-03 Blade Setup)

Slot	Card Type During 1st Power On	System A (SV8100)		
		Card Type and Ports		
		Card Type	Ports	
1	CD-CP00-US	CD-CP00-US	
2	CD-8DLCA	CD-8DLCA	Station 1~8
3	None	None	N/A
4	COI(8)-U10 ETU	COI(8)-U10 ETU	Trunks 1~8
5	CD-PVAA	CD-PVAA	Trunks 9~32
6	None	None	N/A

Slot	Card Type During 1st Power On	System B (IPK II)		
		Card Type and Ports		
		Card Type	Ports	
1	ESI(8)-U10 ETU	ESI(8)-U10 ETU	Station 1~8
2	COI(8)-U10 ETU	COI(8)-U10 ETU	Trunks 9~32
3	None	None	N/A
4	PVA(X)-U( ) ETU	PVA(X)-U( ) ETU	Trunks 9~32
5	None	None	N/A
6	None	None	N/A
7	None	None	N/A
8	None	None	N/A
0A	MIFM-U10 ETU	MIFM-U10 ETU	N/A

Slot	Card Type During 1st Power On	System C (IPK)	
		Card Type and Ports	
		Card Type	Ports
1	ESI(8)-U10 ETU	ESI(8)-U10 ETU	Station 1~8
2	COI(8)-U10 ETU	COI(8)-U10 ETU	Trunks 9~32
3	None	None	N/A
4	PVA(X)-U( ) ETU	PVA(X)-U( ) ETU	Trunks 9~32
5	None	None	N/A
6	None	None	N/A
7	None	None	N/A
8	None	None	N/A
0A	MIFM-U10 ETU	MIFM-U10 ETU	N/A

12.3   Digital Trunk Assignments

Use the table below to make the appropriate assignments for digital trunks.

System A (SV8100)	System B (IPK II)	System C (IPK)
Program 10-03-01 1. Insert CD-CP00-US in Chassis 1, Slot 1. 2. Verify PRG 10-03-01 (Blade Setup).	**Program 10-03-01** 1. Insert PVA(X)-U( ) in Cabinet 1, Slot 4. 2. Verify PRG 10-03-01 (ETU Setup).	**Memory Block 7-1** 1. Insert PVA(X)-U( ) in Cabinet 1, Slot 4. 2. Select Cabinet 1, Slot 4. 3. Assign Page 6, LK 6 (DTI24).
Program 10-54-01 Slot 4 – ID01 License Code = 6200.	**Program 10-03-02** 0 = D4 (12 Multiframe) (**default**) 1 = ESF (24 Multiframe)	**M.B. 1-8-33** No assignment required.
Program 10-54-02 Slot 4 – ID01 License Code = 24.	**Program 10-03-03** 0 = B8ZS (**default**) 1 = ANI/ZCS	**M.B. 1-11-05** 1. Select DTI 1, blocks 1~6. 2. Page 1, LK 1~24=On.
Program 14-05-01 Assign Trunks 9~32 in Trunk Group 10.	**Program 10-19-01** Assign DSP Resources 009~032 as CCIS (LK4).	**M.B. 1-11-07** 1. Select DTI 1, blocks 1~6. 2. Assign LK 2 (E&M).
Program 22-02-02 Set Trunks 9~32 as Tie Line (LK6).	**Program 10-40-01** Slot 4, SIP Trunk =Enable.	**M.B. 3-03** 1. Trunks 9~32. 2. Assign Trunk Group 10.

System A (SV8100)	System B (IPK II)	System C (IPK)
Program 34-01-01 1. Trunks 9~32. 2. Assign LK 2 (Wink).	**Program 10-40-03** Slot 4, CCISoIP=Fixed.	**M.B. 3-14** 1. Trunks 9~32. 2. Assign LK 2 (Wink).
	Program 10-40-04 Slot 4 = 24.	**M.B. 3-91** 1. Trunks 9~32. 2. Assign LK 3 (TIE).
	Program 14-05-01 Assign Trunks 9~32 in Trunk Group 10.	
	Program 22-02-02 Set Trunks 9~32 as Tie Line (LK6).	
	Program 34-01-01 1. Trunks 9~32. 2. Assign LK 2 (Wink).	

12.4 Tandem Connection Assignment

Use the table below to make assignments for tandem connections.

System A (SV8100)	System B (IPK II)	System C (IPK)
Program 14-01-13 Trunk to Trunk Transfer 0 = Disable 1 = Enable (Default)	**Program 14-01-13** Trunk to Trunk Transfer 0 = Disable 1 = Enable (Default)	**M.B. 1-8-08** Page 3, LK 5 On = Allow
Program 14-01-24 Trunk to Trunk Outgoing Caller ID Through Mode 0 = Disable (Off) (default) 1 = Enable (On)	**Program 14-01-24** Trunk to Trunk Outgoing Caller ID Through Mode 0 = Disable (Off) (default) 1 = Enable (On)	**M.B. 3-04** All Trunks = LK 2: Yes

12.5   VoIP Address Assignments

Use the table below to make the appropriate assignments for VoIP daughter board (PZ-32IPLA/PZ-64IPLA/PZ-128IPLA) addresses.

System A (SV8100)	System B (IPK II)	System C (IPK)
Program 10-55-01 Slot 4, Assign IP Addresses 192.168.1.10	**Program 84-05-01** Slot 4, Assign IP Addresses 192.168.1.20	Programmed locally in the PVA(X)-U( ) ETU.
Program 10-55-03 Slot 4, Main	**Program 84-05-02** Slot 4, Auto	
Program 10-55-04 Slot 4, Subnet Mask = 255.255.255.0	**Program 84-05-03** Slot 4, Master	
Program 10-55-05 Slot 4, Default Gateway = 0.0.0.0	**Program 84-05-04** Slot 4, Subnet Mask = 255.255.255.0	
	Program 84-05-05 Slot 4, Default Gateway = 0.0.0.0	

12.6   CCIS Availability

Use the table below to make the appropriate assignments for CCIS availability.

System A (SV8100)	System B (IPK II)	System C (IPK)
Program 50-01-01 Select Option 1 (Enable)	**Program 50-01-01** Select Option 1 (Enable)	**M.B. 1-11-03** 1. Select DTI1. 2. Assign LK 2 (Yes).
Program 50-02-03 Assign Route ID 9 = 01 (Route ID 9 is designated for Host Originating Point Code).	**Program 50-02-03** Assign Route ID 9 = 02 (Route ID 9 is designated for Host Originating Point Code).	**M.B. 3-70** No assignment required.
		M.B. 1-15-01 No assignment required.

System A (SV8100)	System B (IPK II)	System C (IPK)
		M.B. 1-15-02 No assignment required.
		M.B. 1-15-03 No assignment required.
		M.B. 1-15-04 No assignment required.
		M.B. 1-15-05 No assignment required.
		M.B. 5-05 No assignment required.
		M.B. 5-06 1. Select Trunk Group 10. 2. Assign H>L (Default).
		M.B. 1-15-00 No assignment required.

12.7   CCISoIP Assignment

Use the table below to make the appropriate assignments for CCIS over IP.

System A (SV8100)	System B (IPK II)	System C (IPK)
Program 50-03-01 System ID 1 = DPC 02 System ID 2 = DPC 03 System ID 3 = DPC 04 System ID 4 = DPC 05	**Program 50-03-01** System ID 1 = DPC 01 System ID 2 = DPC 03 System ID 3 = DPC 04 System ID 4 = DPC 05	Programmed locally in the PVA(X)-U( ) ETU.
Program 50-03-01 System ID 1 = 192.168.1.10 System ID 2 = 192.168.1.30 System ID 3 = 192.168.1.40 System ID 4 = 192.168.1.50	**Program 50-03-01** System ID 1 = 192.168.1.10 System ID 2 = 192.168.1.30 System ID 3 = 192.168.1.40 System ID 4 = 192.168.1.50	
Program 50-03-04 0 = Disable 1 = Enable (Default)	**Program 50-03-04** 0 = Disable 1 = Enable (Default)	

12.8 Closed Numbering Plan - using Closed Number Blocks

Use the table below to make closed numbering plan assignments.

System A (SV8100)	System B (IPK II)	System C (IPK)
Program 11-01-01 Dial 10 = 4 Digit; Extension Dial 11 = 4 Digit; F-Route Dial 12 = 4 Digit ; F-Route Dial 13 = 4 Digit ; F-Route Dial 14 = 4 Digit ; F-Route	**Program 11-01-01** Dial 11 = 4 Digit; Extension Dial 10 = 4 Digit; F-Route Dial 12 = 4 Digit ; F-Route Dial 13 = 4 Digit ; F-Route Dial 14 = 4 Digit ; F-Route	**M.B. 1-2-03** LK 3 = 4 digit
Program 11-02-01 Assign Station Numbers to Port Number Defaults for Ports 1~512 101~198 3101~3199 3200~3513	**Program 11-02-01** Assign Station Numbers to Port Number Defaults for Ports 1~512 101~198 3101~3199 3200~3513	**M.B. 1-1-46** Access Code. 001 = 000
Program 44-02-01 TBL 1 = 11 TBL 2 = 12 TBL 3 = 13 TBL 4 = 14	**Program 44-02-01** TBL 1 = 10 TBL 2 = 12 TBL 3 = 13 TBL 4 = 14	**M.B. 1-1-47** Access Code 12 = 001 Access Code 10 = 401 Access Code 11 = 402 Access Code 13 = 403 Access Code 14 = 404
		M.B 4-10 All Ports 12XX
Program 44-02-02 TBL 1 = F-Route TBL 2 = F-Route TBL 3 = F-Route TBL 4 = F-Route	**Program 44-02-02** TBL 1 = F-Route TBL 2 = F-Route TBL 3 = F-Route TBL 4 = F-Route	**M.B. 1-1-49** Block 01 = 110 Block 02 = 110 Block 03 = 110 Block 04 = 110
Program 44-02-03 F-Route TBL 1 = 11 F-Route TBL 2 = 12 F-Route TBL 3 = 13 F-Route TBL 4 = 14	**Program 44-02-03** F-Route TBL 1 = 10 F-Route TBL 2 = 12 F-Route TBL 3 = 13 F-Route TBL 4 = 14	**M.B. 1-1-50** Block 01 = 10 Block 02 = 11 Block 03 = 13 Block 04 = 14
Program 44-05-01 F-Route TBL 1 = TG10 F-Route TBL 2 = TG10 F-Route TBL 3 = TG10 F-Route TBL 4 = TG10	**Program 44-05-01** F-Route TBL 1 = TG10 F-Route TBL 2 = TG10 F-Route TBL 3 = TG10 F-Route TBL 4 = TG10	
Program 44-05-02 F-Route TBL 1 Delete Digit = 0	**Program 44-05-02** F-Route TBL 1 Delete Digit = 0	
Program 44-05-03 F-Route TBL 1 Add Dial = 0 F-Route TBL 2 Add Dial = 0 F-Route TBL 3 Add Dial = 0 F-Route TBL 4 Add Dial = 0	**Program 44-05-03** F-Route TBL 1 Add Dial = 0 F-Route TBL 2 Add Dial = 0 F-Route TBL 3 Add Dial = 0 F-Route TBL 4 Add Dial = 0	

System A (SV8100)	System B (IPK II)	System C (IPK)
Program 44-05-09 F-Route TBL 1 Max Digit = 4 F-Route TBL 2 Max Digit = 4 F-Route TBL 3 Max Digit = 4 F-Route TBL 4 Max Digit = 4	**Program 44-05-09** F-Route TBL 1 Max Digit = 4 F-Route TBL 2 Max Digit = 4 F-Route TBL 3 Max Digit = 4 F-Route TBL 4 Max Digit = 4	
Program 44-05-10 F-Route TBL1 = DPC 2 F-Route TBL2 = DPC 3 F-Route TBL3 = DPC 4 F-Route TBL4 = DPC 5	**Program 44-05-10** F-Route TBL1 = DPC 1 F-Route TBL2 = DPC 3 F-Route TBL3 = DPC 4 F-Route TBL4 = DPC 5	

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Book 2 – SV8100 IP Networking

General Information

SECTION 1   UNIVERGE SV8100 IP NETWORKING

UNIVERGE SV8100 is an enterprise IP Telephony solution that allows businesses and organizations to converge their voice and data network to secure the many advantages of IP telephony, while enjoying the hundreds of features that they have come to expect from the telephone systems.

SECTION 2   VOICE OVER IP

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

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

UNIVERGE SV8100 supports the use of IP Phones. These telephones provide the same functionality as a multiline telephone but use the data network rather then the traditional telecoms infrastructure. This can reduce costs and allow the use of UNIVERGE SV8100 telephones in locations that would not normally be supported by multiline telephones.

UNIVERGE SV8100 can also use VoIP technologies to connect two or more telephone systems together. This can eliminate inter-site call charges, and can also simplify calling between sites (as desk-to-desk dialing is possible).

Table 1-1 VoIP Specifications lists the specifications for various aspects of UNIVERGE SV8100 VoIP system.

General Information

Table 1-1 VoIP Specifications

Category	Feature	Notes
IP Address	DHCP Server	CD-CP00-US
	DHCP Client	IP Phone
QoS	802.1p/1q	CD-CP00-US
	L3 QoS (ToS)	Diffserv/IP Precedence
Maintenance	HTTP Server	CD-CP00-US
VLAN	Tag and port-based VLAN	
VoCoder	G.711 µ-law/A-law	
	G.729a	
Jitter Buffer Size	Set by system programming	
RTP Length	Set by system programming	
Echo Canceller Tail Size	Set by system programming	
Level Adjustment	Set by system programming	
IP Phone	SIP Phone	SIP Phone
SIP Trunk	SIP Trunk	Maximum 200 Trunks
IP CCIS	IP CCIS Trunks	Maximum 200 Trunks

IP Networking

SECTION 1   INTRODUCTION[1]

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

Two types of IP Networking are available on the UNIVERGE SV8100 CCIS Network and SIP trunks. These methods are explained below.

SECTION 2   UNIVERGE SV8100 IP K-CCIS NETWORK

The SV8100 IP K-CCIS package provides a seamless connection of multiple systems into a single virtual communications system using VoIP lines with a unified numbering plan.

The SV8100 IP K-CCIS Network allows many offices to connect their UNIVERGE SV8100 systems so they appear as one. This gives them the ability to have only one operator to manage the system and share one voice mail system in the network.

An extension user in the network can easily dial another extension or transfer a call in the SV8100 IP K-CCIS Network system. Calls are passed from network node to network node using a protocol that contains information about the source of the call, the type of call and the destination of the call.

SV8100 IP K-CCIS is explained in detail in the UNIVERGE SV8100 Key-Common Channel Interoffice Signaling (K-CCIS) section. Please refer to this for a complete description and installation instructions.

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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 3   IP TRUNKS

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

Refer to SIP Trunking for a a detailed description of SIP trunking and for set up instructions.

To set up IP trunks:

1.   Connect the system to the Data Network. (Refer to General IP Configuration for detailed instructions.)

2.   Configure the IP trunks.

3.   Configure the IP K-CCIS or SIP information.

4.   Configure the F-Route.

3.1   Configure IP Trunks

When installing a PZ-32IPLA/PZ-64IPLA/PZ-128IPLA daughter board in the UNIVERGE SV8100 system, external line ports are allotted in accordance with the number of Licensed ports for the particular IP Application.

The UNIVERGE SV8100 system now has the required information about the remote destinations and the SIP/IP K-CCIS configuration is complete. The only remaining task is to configure F-Route to route calls to remote destinations via the IP trunks. F-Route configuration is discussed in detail in the Automatic Route Selection (ARS) feature in the UNIVERGE SV8100 Programming Manual. A basic list of the programming commands required for F-Route is shown in the example below.

SECTION 4   EXAMPLE CONFIGURATIONS

4.1   Network Configurations

Figure 2-1 Example IP Network Configuration shows four sites networked via IP trunks. Each site has a Point Code and an IP address.

The programing for Office A and C is shown below. This would be sufficient programming to make a call from Office A to Office C.

UNIVERGE SV8100

UNIVERGE SV8100

Ext. 1000   **Office A**

Office B   Ext 1100

IP Address (CD-CP00-US):
   192.168.1.10
Point Code: 00001

IP Address (CD-CP00-US):
   192.168.1.20
Point Code: 00002

IP Network
(Intranet)

PBX

PBX

Office C

Office D

Ext. 1200

Ext. 1300

IP Address: 192.168.1.30
Point Code: 00003

IP Address: 192.168.1.40
Point Code: 00004
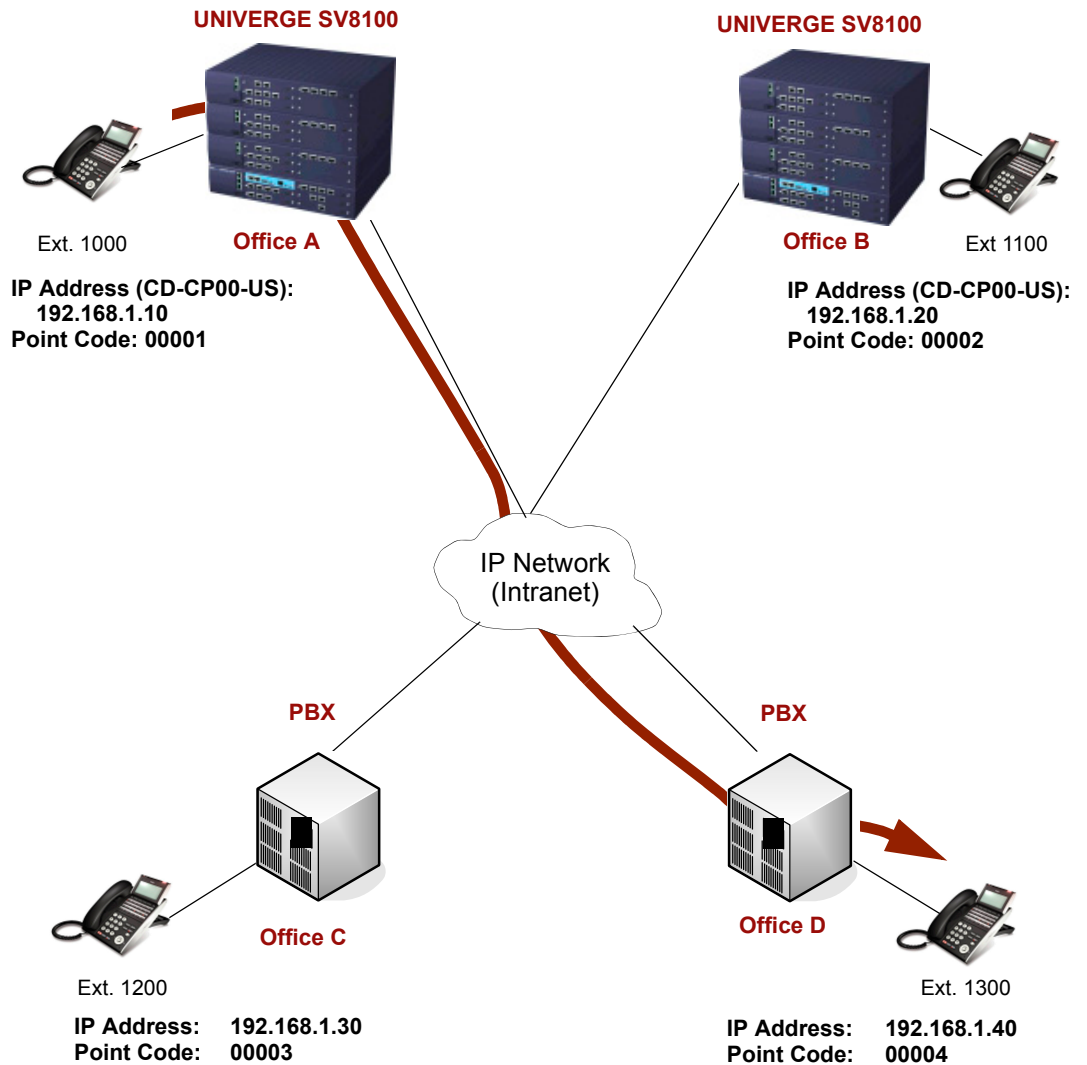
Figure 2-1 Example IP Network Configuration

4.2   UNIVERGE SV8100 IP K-CCIS Programming Example 1

This is an example of UNIVERGE SV8100 program data assignments for a 4-digit Closed Numbering Plan using Closed Number Blocks. The following system configurations are used for all four systems:

❍   Each system is a single cabinet expanded port package with one CHS2U-US.

❍   Each system has 16 multiline terminals.

❍   Each system has eight analog Trunks.

Point Code: 00001   Point Code: 00002
Address: CD-CP00-US: 192.168.1.10   Address: 1st CD-CP00-US Resource: 192.168.1.20

Point Code: 00003
Address: 192.168.1.30

Point Code: 00004
Address: 192.168.1.40
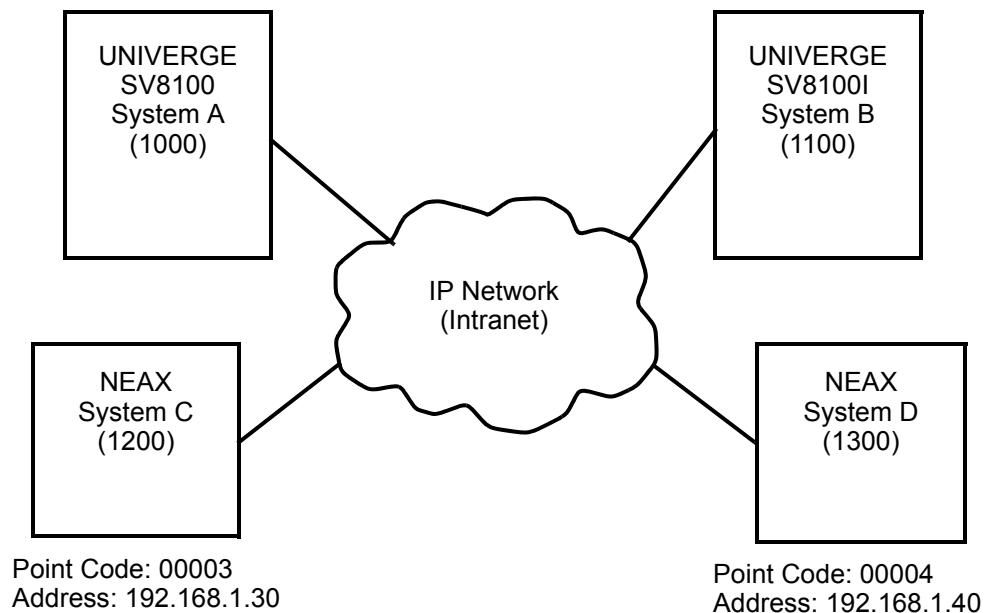
✎   Systems C and D are NEAX systems. Refer to the applicable system manuals for programming details.

Figure 2-2 Programming Example 1

Assume that the systems are defaulted (first power on) with the following cards installed as described in the following table.

4.2.1   Card Interface Slot Assignment (PRG 10-03 ETU Setup)

Slot	Card Type During 1st Power On	Card Type and Ports			
		System A		System B	
		Card Type	Ports	Card Type	Ports
1	CD-CP00/PZ-32IPLA	CD-CP00/PZ-32IPLA	Trunks 5 ~28	CD-CP00/PZ-32IPLA	Trunks 5~28
2	CD-8DLCA	CD-8DLCA	Station 1~8	CD-8DLCA	Station 1~8
3	CD-8DLCA	CD-8DLCA	Station 9~16	CD-8DLCA	Station 9~16
4	None	None	N/A	None	N/A
5	CD-4COTB	CD-4COTB	Trunks 1~4	CD-4COTB	Trunks 1~4
6	None	None	N/A	None	N/A

4.2.2   Digital Trunk Assignments

Use the table below to make the appropriate assignments for digital trunks.

System A	System B
Program 10-03-01 1. Insert PZ-32IPLA on CD-CP00 in Chassis 1, Slot 1. 2. Verify PRG 10-03-01 (ETU Setup).	**Program 10-03-01** 1. Insert PZ-32IPLA on CD-CP00 in Chassis 1, Slot 1. 2. Verify PRG 10-03-01 (ETU Setup).
Program 10-19-01 Assign DSP Resources 001~024 as CCIS (LK4).	**Program 10-19-01** Assign DSP Resources 001~024 as CCIS (LK4).
Program 10-40-04 Assign the number of ports for CCISoIP for Slot 1 as 24.	**Program 10-40-04** Assign the number of ports for CCISoIP for Slot 1 as 24.
Program 14-05-01 Assign Trunks 5~28 in Trunk Group 10.	**Program 14-05-01** Assign Trunks 5~28 in Trunk Group 10.
Program 22-02-01 Set Trunks 5~28 as Tie Line (LK6).	**Program 22-02-01** Set Trunks 5~28 as Tie Line (LK6).

4.2.3   VoIP Address Assignments

Use the table below to make the appropriate assignments for VoIPDB (PZ-32IPLA/PZ-64IPLA/PZ-128IPLA) addresses.

Program/Item No.	Description/ Selection	Assigned Data	Comments
84-26-01	IPLA Basic Setup – IP Address	XXX.XXX.XXX.XXX	Assign the IPLA IP Address Default: 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
84-26-02	IPLA Basic Setup – RTP Port Number	0~65534	Default: VoIP GW1 = 10020 VoIP GW2 = 10052 VoIP GW3 = 10084 VoIP GW4 = 10116 VoIP GW5 = 10148 VoIP GW6 = 10180 VoIP GW7 = 10212 VoIP GW8 = 10244
84-26-03	IPLA Basic Setup – RTCP Port Number (RTP Port Number + 1)	0~65534	Default: VoIP GW1 = 10021 VoIP GW2 = 10053 VoIP GW3 = 10085 VoIP GW4 = 10117 VoIP GW5 = 10149 VoIP GW6 = 10181 VoIP GW7 = 10213 VoIP GW8 = 10245

4.2.4   CCIS Availability

Use the table below to make the appropriate assignments for CCIS availability.

System A	System B
Program 50-01-01 Select Option 1 (Enable) Enables the CCIS Feature	**Program 50-01-01** Select Option 1 (Enable) Enables the CCIS Feature
Program 50-02-03 Assign Route ID 9 = 01 (Route ID 9 is designated for Host Originating Point Code)	**Program 50-02-03** Assign Route ID 9 = 02 (Route ID 9 is designated for Host Originating Point Code)

4.2.5   IP CCISoIP Assignment

Use the table below to make the appropriate assignments for CCIS over IP.

Program/Item No.	Description/ Selection	Assigned Data	Comments
50-03-01	CCIS Destination System Settings – Destination Point Code	0~16367 seconds Default is 0	PRG 50-03 assignments are only used for CCISoIP.
50-03-03	CCIS Destination System Settings – IP Address (IP only)	xxx.xxx.xxx.xxx (xxx = 0~255) Default is 0.0.0.0	PRG 50-03 assignments are only used for CCISoIP.
50-03-04	CCIS Destination System Settings – Point Code Availability	0:Disable 1:Enable Default is 1	

4.2.6   CCIS Assignment

Use the table below to make the appropriate system-wide CCIS assignments.

System A	System B
Program 50-03-01 1. System ID 1~255 – Enter Destination Point Code 2. System ID 1 = DPC 02	**Program 50-03-01** 1. System ID 1~255 – Enter Destination Point Code 2. System ID 1 = DPC 01
Program 50-03-03 1. System ID 1~255 – Enter Destination System IP Address. 2. System ID 1 = 192.168.1.20 3. System ID 2 = 192.168.1.21	**Program 50-03-03** 1. System ID 1~255 – Enter Destination System IP Address 2. System ID 1 = 192.168.1.10 3. System ID 2 = 192.168.1.11
Program 50-04-01 CCIS Office Code Assignment Assign up to four digits. Default not assigned.	**Program 50-04-01** CCIS Office Code Assignment Assign up to four digits. Default not assigned.
Program 50-13-01 CCIS Centralized Response Timeout Assignment - IAI Response Timer Enter 00~99.	**Program 50-13-01** CCIS Centralized Response Timeout Assignment - IAI Response Timer Enter 00~99.
Program 50-06-01 CCIS Feature Availability - Link Reconnect 0 = Not Available 1 = Available (default)	**Program 50-06-01** CCIS Feature Availability - Link Reconnect 0 = Not Available 1 = Available (default)

System A	System B
Program 50-05-01 CCIS Maximum Call Forwarding Hop Counter 1~7 7 = Default	**Program 50-05-01** CCIS Maximum Call Forwarding Hop Counter 1~7 7 = Default

4.2.7   Centralized Day Night Switching Assignments

Use the table below to make the appropriate assignments for centralized day and night switching.

System A	System B
PRG 50-06-02 CCIS Feature Availability - Centralized Day/Night Switching 0 = Disable (default) 1 = Enable	**MB 1-15-13 Centralized Day/Night Switching for Remote Office Assignment** Select Data: LK 1 = Deny (default) LK 2 = Allow
PRG 50-11-02 CCIS Centralized Day/Night Switching Sending Group Assignment - CCIS Route ID Send Group (1~8) CCIS Route ID 0 = No Setting (default) 1~4	**MB 1-15-14 Centralized Day/Night Switching for Main Office Assignment** Select Table 01~16. Enter Point Code (00001~16367). Enter CCH Channel (1~4) or i.
PRG 50-12-01 CCIS Centralized Day/Night Mode to System Mode Assignment - Day Mode Day Mode = Mode 1~8 1 = default Night Mode = Mode 1~8 2 = default	**Not Applicable**

4.2.8   Centralized BLF Assignments

Use the table below to make the appropriate assignments for centralized BLF.

System A	System B
PRG 50-08-01 CCIS Centralized BLF Sending Group Assignment – Destination Point Code Select Group ID (1~8) + Destination Point Code. Default not assigned.	**MB 1-15-15 Centralized BLF Send Point Code Assignment** 1. Select Table (1~8). 2. Enter Point Code (00001~16367).
PRG 50-08-02 CCIS Centralized BLF Sending Group Assignment – CCIS Route ID Select Group ID 1~8 + CCIS Route ID. 1~16367 0 = Not Assigned (default) 1~16367	**MB 1-15-16 Centralized BLF Send Extension Number Assignment** Select Table (001~120). Enter Extension Number. Select Data: LK 1: Send Group 1 LK 2: Send Group 2 LK 3: Send Group 3 LK 4: Send Group 4 LK 5: Send Group 5 LK 6: Send Group 6 LK 7: Send Group 7 LK 8: Send Group 8
PRG 50-09-01 CCIS Centralized BLF Sending Extension Number Assignment – Extension Number Select Table Number 1 ~120 and Enter Extension Number. Default not assigned.	**MB 1-15-17 CCIS Centralized BLF Send Time Assignment** Select Data: LK 1: 4 seconds (default) LK 2: 8 seconds LK 3: 12 seconds LK 4: 16 seconds
PRG 50-09-02 Centralized BLF Sending Extension Number Assignment – Send to Sending Group 1 Select Table Number 1 ~120. 0 = Disable (default) 1 = Enable	**MB 1-15-18 Centralized BLF Receive Extension Number Assignment** Select Table (001~120). Enter Extension Number: 3-digit (100~899) 4-digit (1000~8999)
PRG 50-10-01 CCIS Centralized BLF Interval Time Assignment– Type of Interval Time 0 = 4 seconds (default) 1 = 8 seconds 2 = 12 seconds 3 = 16 seconds	**Not Applicable**

4.2.9   Local Numbering Plan Assignment

Use the table below to make local numbering plan assignments.

Program/Item No.	Description/ Selection	Assigned Data	Comments
11-01-01	System Numbering	0 = Not Used 1 = Service Code 2 = Extension Number 3 = Trunk Access 4 = Special Trunk Access 5 = Operator Access 6 = Flexible Routing 9 = Dial Extension Analyze	Defaults for 1X, 2X, 3X = 2 Extension Number
11-02-01	Extension Numbering	Assign Station Numbers to Port Numbers	Defaults for Ports 1~512: Ports 1~99 = 101~199 Ports 100~512 =3101~3513
11-04-01	Virtual Extension Numbering	Assign Station Numbers to Port Numbers	Defaults for Ports Ports 1~99 = 201~299 Ports 100~256 = 3601~3757

4.2.10   Closed Numbering Plan - using Closed Number Blocks

Use the table below to make closed numbering plan assignments.

Program/Item No.	Description/ Selection	Assigned Data	Comments
11-01-01	System Numbering	2 = Extension Number	Defaults for 1X, 2X and 3X = (2) for Extension Number
44-02-01	Dial Analysis Table for ARS/F-Route Access – Number of digits to be analyzed by the system	Up to eight digits can be assigned. Default is No Setting	Assign the digits to be dialed across the K-CCIS link. These digits were assigned as F-Route in Program 11-01-01 (Use line key 1 for "Don't Care" digit, @)
44-02-02	Dial Analysis Table for ARS/F-Route Access – Service Type	0 = No Setting (None) 2 = ARS/F-Route Table (F-Route) Default is 0	The service type (2) assigns the digits to be dialed to an F-Route. Program 44-02-03 assigns the F-Route to be used.
44-02-03	Dial Analysis Table for ARS/F-Route Access – Additional Data	2 = 0~500 (0=No Setting) Default is 0	When setting data is (2), refer to Program 44-05.

Program/Item No.	Description/ Selection	Assigned Data	Comments
44-05-01	ARS/F-Route Table – Trunk Group Number	0~100, 255 0 = No Setting 255 = Extension Call Default is 0	Select the trunk group to be used for the outgoing K-CCIS call.

4.2.11 Tandem Connections

Use the table below to make assignments for tandem connections.

System A	System B
Program 14-01-13 Trunk to Trunk Transfer 0 = Disable 1 = Enable (Default)	**Program 14-01-13** Trunk to Trunk Transfer 0 = Disable 1 = Enable (Default)
Program 14-01-24 Trunk to Trunk Outgoing Caller ID Through Mode 0 = Disable (Off) (default) 1 = Enable (On)	**Program 14-01-24** Trunk to Trunk Outgoing Caller ID Through Mode 0 = Disable (Off) (default) 1 = Enable (On)

SECTION 5   DTMF RELAY

When sending DTMF over an SV8100 CCISoIP network it may be necessary to configure the item below.   This allows the sending UNIVERGE SV8100 to detect DTMF tones and convert them to data, then regenerate the DTMF tones at the receiving side.

➲   **84-21-21   CCIS over IP CODEC - DTMF Relay Mode**

When using G.729, NEC recommends setting this item to (1) = Inbound (RFC2833).

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General IP Configuration

SECTION 1   INTRODUCTION

The UNIVERGE SV8100 system uses IP for various applications, including:

❍   System Programming

❍   Voice Over IP

❍   Switching

This section describes the procedure for connecting the UNIVERGE SV8100 system to an existing data network and configuring TCP/IP. This is the first step in implementing VoIP and other IP applications.

SECTION 2   NETWORK ADDRESSING OVERVIEW

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

2.1   IP Address

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

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

General IP Configuration

❍   Class A 10.0.0.0 ~ 10.22.255.255

❍   Class B 172.16.0.0. ~ 172.31.255.255

❍   Class C 192.168.0.0 ~ 192.168.255.255

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

2.2   Subnet Mask

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

❍   Class A 255.0.0.0

❍   Class B 255.255.0.0

❍   Class C 255.255.255.0

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

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

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

2.3   DHCP

Dynamic Host Configuration Protocol (DHCP) assigns a dynamic IP address. Network control may be easier with DHCP as there is no need to assign and program individual IP addresses for the LAN equipment. To use a dynamic IP address, a DHCP server must be provided. The UNIVERGE SV8100 system CD-CP00-US blade provides an internal DCHP server enabling the ability to use DHCP.

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

When the request for an address is recognized, the DHCP server assigns an IP address, Subnet mask, and the IP address of the router, based on UNIVERGE SV8100 system programming.

Note that the CD-CP00-US blade must always have a static IP address. This address is set in Program 10-12-01 : CD-CP00-US Network Setup – IP Address (default: 192.168.0.10).

SECTION 3   CONFIGURATION EXAMPLES

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

3.1   Example Configuration 1 - Existing Network with Static Addressing

Figure 3-1 Example Configuration 1 - Existing Network with Static IP Address on page 3-4 shows a typical network configuration that uses Static IP Addressing.

Each client device has a manually assigned IP address in the 192.168.1.0/24 network (i.e., 192.168.1.1 to 192.168.1.254 with a subnet mask of 255.255.255.0). They also have a default gateway address configured (192.168.1.254) this allows the device to route packets to destinations that exist outside of their own LAN.
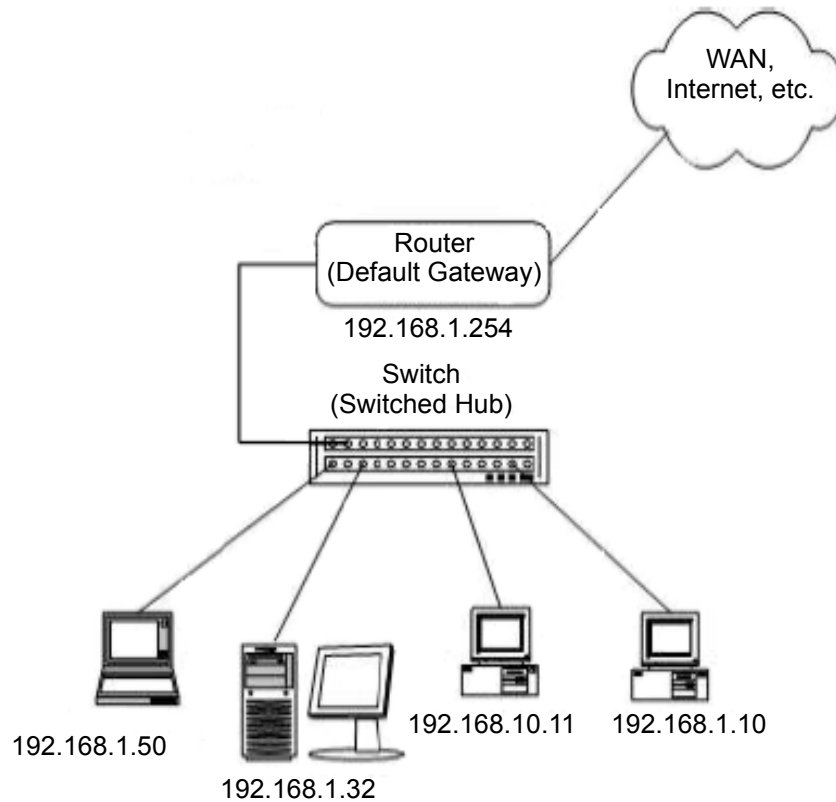
Figure 3-1 Example Configuration 1 - Existing Network with Static IP Address

Assume that a UNIVERGE SV8100 is added to the existing data network. The Network Administrator (or IT Manager) should provide the following:

❍   IP Address (for the CD-CP00-US blade)

❍   IP Addresses (for the PZ-(X)IPLA daughter board)

❍   Subnet Mask

❍   Default Gateway

❍   A spare switch/hub port

First, program the UNIVERGE SV8100:

❍   192.168.1.200

❍   255.255.255.0

❍   PRG10-12-03: 192.168.1.254

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

Now connect the CD-CP00-US blade Ethernet Port to the switch/hub port, using a standard Cat-5 patch cable. The UNIVERGE SV8100 is now configured on the network and should be accessible by other devices on the network. Refer to Figure 3-2 Example Configuration 1 - Adding the UNIVERGE SV8100 Chassis to the Network.

WAN,
Internet, etc.

UNIVERGE SV8100
Chassis with CCPU
Installed

Router
(Default Gateway)

192.168.1.254

Switch
(Switched Hub)

192.168.1.200

192.168.1.50

192.168.10.11   192.168.1.10

192.168.1.32
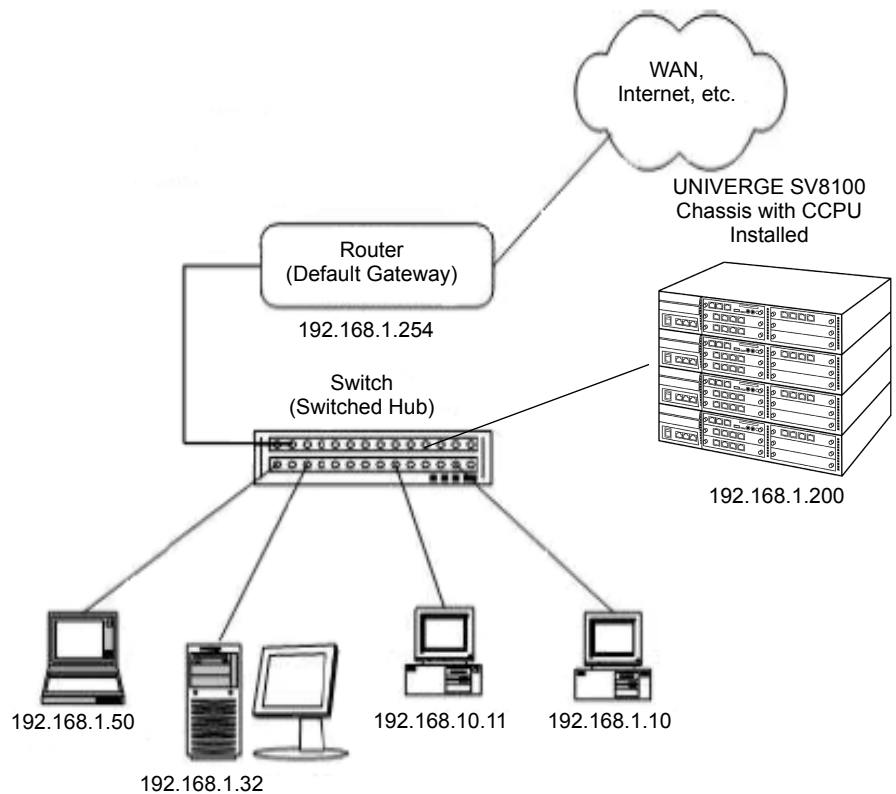
**Figure 3-2   Example Configuration 1 - Adding the UNIVERGE SV8100 Chassis
to the Network**

3.2   Example Configuration 2 - New Network with Dynamic Addressing

Figure 3-3 Example Configuration 2 - New Network with Dynamic Addressing on page 3-6 shows a typical network configuration using Dynamic IP Addressing, and the UNIVERGE SV8100 Internal DHCP server. In most cases, the customer would use an external DHCP server (for example on a Windows 2003 Server) or static addressing (as illustrated in Figure 3-2 Example Configuration 1 - Adding the UNIVERGE SV8100 Chassis to the Network on page 3-5). However, if the UNIVERGE SV8100 is to be installed on a new network the Network Administrator may want to use the UNIVERGE SV8100 internal server (this is called inDHCP).

In this example, the client PCs get an IP address, subnet mask, and default gateway from the inDHCP server. The server also uses DHCP, but should always be given the same IP address (192.168.1.32).

The Network Administrator (or IT Manager) should provide the following:

❍   IP Address (for the CD-CP00-US blade)

❍   IP Addresses (for the PZ-(X)IPLA)

❍   Subnet Mask

❍   Default Gateway

❍   Range(s) of IP Addresses to assign

❍   List of permanent IP addresses, with corresponding MAC Addresses

❍   A spare switch/hub port
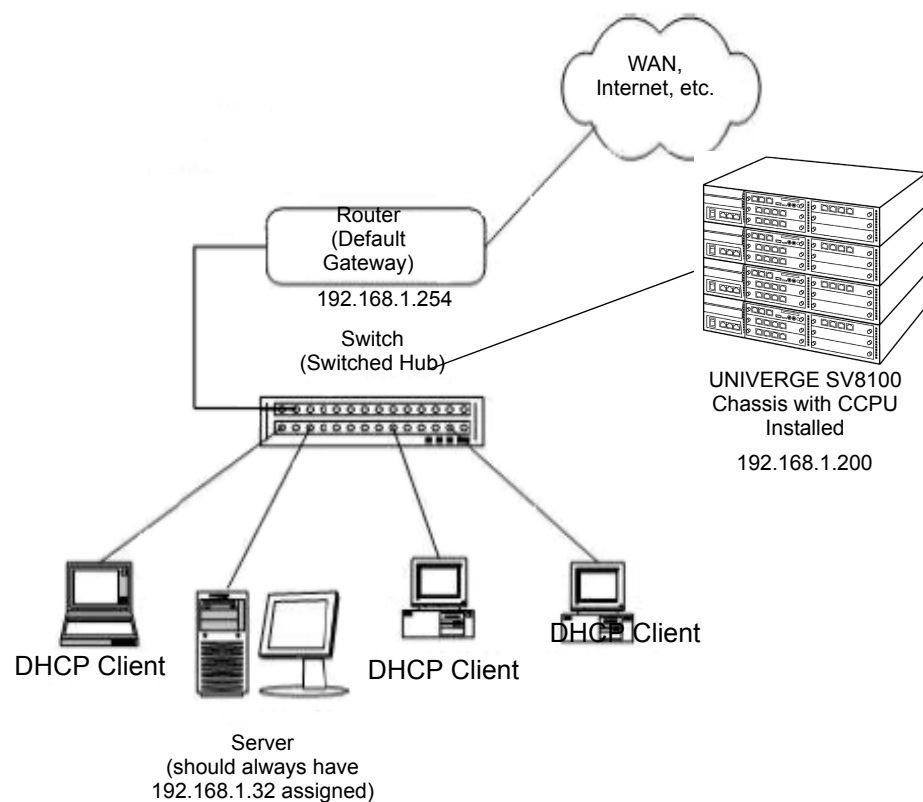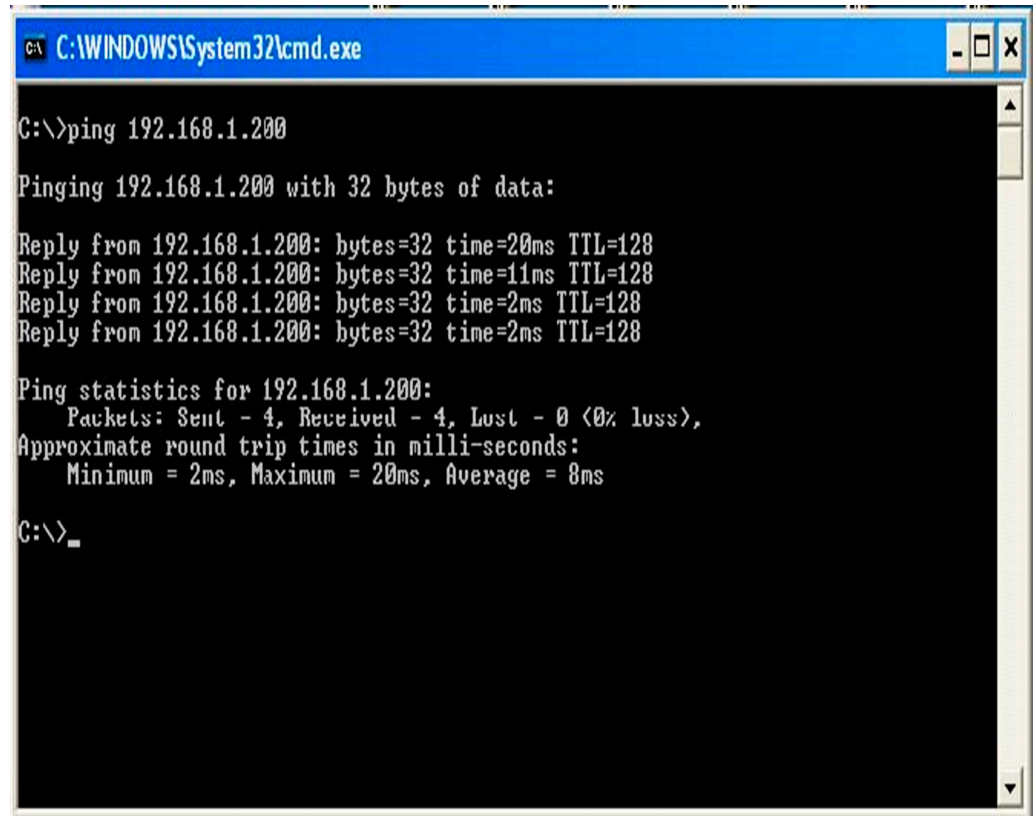
Figure 3-3 Example Configuration 2 - New Network with Dynamic Addressing

Now connect the UNIVERGE SV8100 PZ-(X)IPLA Ethernet Port to the switch/hub port, using a standard CAT-5 patch cable. The UNIVERGE SV8100 is now configured on the network and its DHCP server is ready to allocate IP addresses. The client PCs should be set to *Obtain IP Address Automatically*. Refer to Figure 3-4 TCP/IP Properties Screen on page 3-7.


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

C:\>ping 192.168.1.200

Pinging 192.168.1.200 with 32 bytes of data:

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

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

C:\>_
```

Figure 3-4 TCP/IP Properties Screen
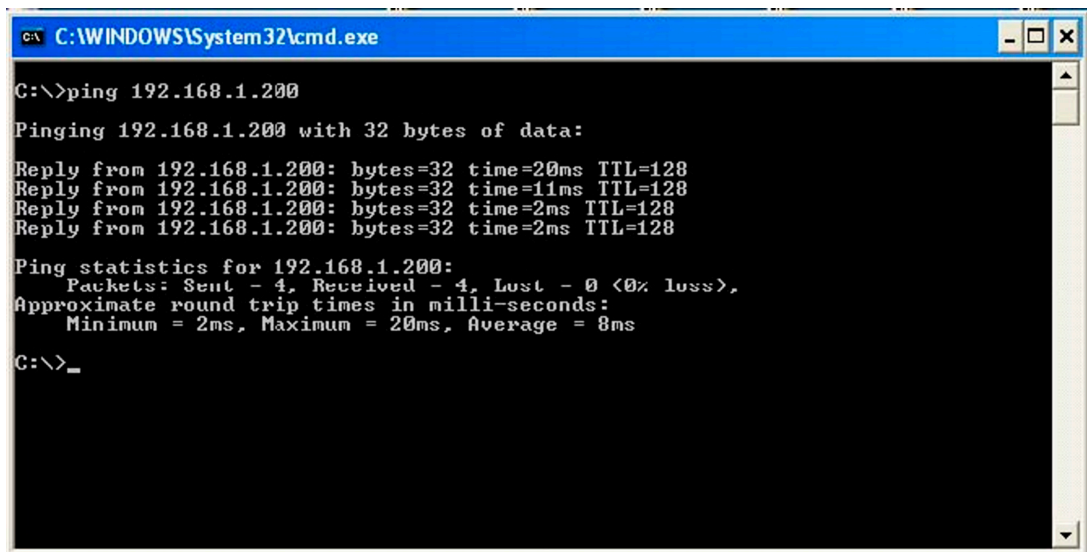
If the client PCs are now connected to the network (and restarted), they should be assigned an IP address in the range 192.168.1.50 to 192.168.1.150, a subnet mask of 255.255.255.0 and a default gateway of 192.168.1.254. When the server tries to obtain an IP address, the inDHCP server allocates IP address 192.168.1.32, as it is statically assigned to the server MAC address.

SECTION 4   TESTING THE **UNIVERGE SV8100** NETWORK CONNECTION

To test the UNIVERGE SV8100 network connection, it is possible to use the ICMP (Internet Control Message Protocol) Ping command. This basically sends a small amount of data from one device to another and then waits for a reply. This test confirms that the IP addressing and physical connection are good. To perform this test, from a Windows PC:

1.   Click **Start**.

2.   Click **Run...** .

3.   In the Open dialogue box, enter **command**.

4.   Click **OK**. A Command prompt window opens.

5.   Type **ping 192.168.1.200**.

Figure 3-5 Testing the Network Connection shows that the UNIVERGE SV8100 system has replied to the Ping request – this indicates that the UNIVERGE SV8100 system is correctly connected to the network.

Figure 3-5 Testing the Network Connection

Programming 4

SECTION 1   BEFORE YOU START PROGRAMMING

This chapter provides you with detailed information about the UNIVERGE SV8100 program blocks that may be required to connect the SV8100 to a data network and to configure the VoIP function. The configuration and programming examples, found in the earlier chapters, can be a useful reference when programming the data.

When using this chapter, note that the information on each program is subdivided into the following headings:

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

The reverse type (white on black) just beneath the Description heading is the program access level. You can use only the program if your access level meets or exceeds the level the program requires. Refer to Section 2 How to Enter Programming Mode on page 4-2 for a list of the system access levels and passwords.

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

Programming

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

To enter the programming mode:

1.   15-07-01

> **15-07-01   TEL**
> **KY01 = *01**
> ←   →

Dial 150701 from the telephone dial pad. See the message 15-07-01 TEL on the first line of the telephone display. This indicates the program number (15-07), item number (01), and that the options are being set for the extension.

The second row of the display KY01 = *01 indicates that Key 01 is being programmed with the entry of *01. The third row allows you to move the cursor to the left or right, depending on which arrow is pressed.

To learn how to enter the programming mode, refer to Section 2 How to Enter Programming Mode below.

SECTION 2   **HOW TO ENTER PROGRAMMING MODE**

To enter programming mode:

1.   Go to any working display telephone.

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

2.   *Do not* lift the handset.

3.   Press **Speaker**.

4.   **# ✱ # ✱** .

> **Password**

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

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

Password	User Name	Level	Programs at this Level
47544	**necii**	1 (MF)	Manufacturer (MF): All programs
12345678	**tech**	2 (IN)	Installation (IN): All programs in this section not listed below for SA and SB
0000	**ADMIN1**	3 (SA)	System Administrator - Level 1 (SA): 10-01, 10-02, 10-12, 10-13, 10-14, 10-15, 10-16, 10-17, 10-18, 10-22, 12-02, 12-03, 12-04, 15-01, 15-07, 15-09, 15-10, 15-11, 20-16, 21-07, 21-14, 22-04, 22-11, 25-08, 30-03, 32-02, 40-02, 41-02, 41-03, 41-04, 41-05, 41-06, 41-07, 41-08, 41-09, 41-10, 41-11, 41-12, 41-13, 41-14, 41-15, 41-16, 41-17, 41-18, 90-03, 90-04, 90-06, 90-07, 90-18, 90-19
9999	**ADMIN2**	4 (SB)	System Administrator - Level 2 (SB): 13-04, 13-05, 13-06

SECTION 3   HOW TO EXIT PROGRAMMING MODE

To exit the programming mode:

To exit programming mode, first exit the programming options mode.

1.   Press **Answer** to exit program options, if needed.

```
              Program Mode
        Base  Service  OP1   OP2
```

2.   Press **Speaker**. If changes were made to the system programming, Saving System Data is displayed.

3.   When completed, the display shows Complete Data Save and exits the telephone to idle.

✎   To save a customer database, a blank Compact Flash (CF) is required. Insert the CF into the CD-CP00-US and, using Program 90-03, save the software to the Compact Flash. (Program 90-04 is used to reload the customer data if necessary.) Note that a Compact Flash can hold only one customer database. Each database to be saved requires a separate card.

SECTION 4   USING KEYS TO MOVE AROUND IN THE PROGRAMS

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

Table 4-1 Keys for Entering Data

Keys for Entering Data	
Use this key...	**When you want to...**
0~9 and ✱	Enter data into a program.
Transfer	Complete the programming step you just made (e.g., pressing **Enter** on a PC keyboard). When a program entry displays, press **Transfer** to bypass the entry without changing it.
Conf	Delete the entry to the left (e.g., pressing **Backspace** on a PC keyboard).
Hold	Delete or clear all characters to the right of the cursor.
Answer	Exit one step at a time from the program window currently being viewed. For example, if programming item 5 in 15-03, pressing **Answer** allows you to enter a new option in program 15-03. Pressing **Answer** again allows you to select a new program in the 15-XX series. Pressing **Answer** a third time allows you to enter a new program beginning with **1**. Pressing **Answer** one last time brings you to the beginning program display, allowing you to enter any program number.
Redial	Switch between the different input data fields by pressing **Redial**. The cursor moves up to the top row of the display. Pressing **Redial** again moves the cursor back to the middle row.
Line Keys	Use pre-programmed settings to help with the program entry. These settings vary between programs from LINE 1 = 0 (off) and LINE 2 = 1 (on) to preset values for timers where LINE 1 = 5, LINE 2 = 10, LINE 3 = 15, etc. For programs with this option, the line key, which currently matches the programmed setting, lights steady. The display can also indicate Softkey, which will allow you to select the values as well (-1 and +1 will step through these pre-programmed settings.)
Line Key 1	Program a pause into an Speed Dialing bin.
Line Key 2	Program a recall/flash into an Speed Dialing bin.
Line Key 3	Program an @ into an Speed Dialing bin.

Table 4-1 Keys for Entering Data (Continued)

Keys for Entering Data	
Use this key...	**When you want to...**
VOL ▲	Scroll backward through a list of entry numbers (e.g., from extension etc.) or through entries in a table (e.g., Common Permit Table). If you enter data and then press this key, the system accepts the data before scrolling forward.
VOL ▼	Scroll forward through a list of entry numbers (e.g., from extension etc.) or through entries in a table (e.g., Common Permit Table). If you enter data and then press this key, the system accepts the data before scrolling backward.

SECTION 5   PROGRAMMING NAMES AND TEXT MESSAGES

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

Table 4-2 Keys for Entering Names

Use this keypad digit . . .	When you want to. . .
1	Enter characters: 1 @ [¥] ^ _ ` { | } Æ ¨ Á À Â Ã Ç É Ê ì ó
2	Enter characters: A-C, a-c, 2.
3	Enter characters: D-F, d-f, 3.
4	Enter characters: G-I, g-i, 4.
5	Enter characters: J-L, j-l, 5.
6	Enter characters: M-O, m-o, 6.
7	Enter characters: P-S, p-s, 7.
8	Enter characters: T-V, t-v, 8.
9	Enter characters: W-Z, w-z, 9.
0	Enter characters: 0 ! " # $ % & ' ( ) ô Õ ú ä ö ü α ϵ θ
∗	Enter characters: ∗ + , - . / : ; < = > ? π Σ σ Ω □ ¢ £

Table 4-2 Keys for Entering Names

Use this keypad digit . . .	When you want to. . .
#	# = Accepts an entry (only required if two letters on the same key are needed - ex: TOM). Pressing # again = Space. (In system programming mode, use the right arrow Softkey instead to accept and/or add a space.)
Conf	Clear the character entry one character at a time.
Hold	Clear all the entries from the point of the flashing cursor and to the right.

SECTION 6   USING SOFTKEYS FOR PROGRAMMING

Each UNIVERGE SV8100 display telephone provides interactive Softkeys for intuitive feature access. The options for these keys will automatically change depending on where you are in the system programming. Simply press the Softkey located below the option you wish and the display will change accordingly.

```
 _
              Program Mode
Base      Service    OP1      OP2
  ◯         ◯        ◯        ◯
```

Pressing the VOLUME ▲ or VOLUME ▼ will scroll between the menus.

SECTION 7   WHAT THE SOFTKEY DISPLAY PROMPTS MEAN

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

Table 4-3 Softkey Display Prompts

<table>
<tr><td colspan="2" align="center">Softkey Display Prompts</td></tr>
<tr><td align="center">If you press this Softkey . . .</td><td align="center">The system will. . .</td></tr>
<tr><td align="center">back</td><td>Go back one step in the program display.
You can press VOLUME ▲ or VOLUME ▼ to scroll forward or backward through a list of programs.</td></tr>
<tr><td align="center">↑</td><td>Scroll down through the available programs.</td></tr>
<tr><td align="center">↓</td><td>Scroll up through the available programs.</td></tr>
<tr><td align="center">select</td><td>Select the currently displayed program.</td></tr>
<tr><td align="center">←</td><td>Move the cursor to the left.</td></tr>
<tr><td align="center">→</td><td>Move the cursor to the right.</td></tr>
<tr><td align="center">-1</td><td>Move back through the available program options.</td></tr>
<tr><td align="center">+1</td><td>Move forward through the available program options.</td></tr>
</table>

SECTION 8   PROGRAMS

This sections describes the programs used to connect the SV8100 to a data network and to configure the VoIP functions.

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Program 10 : System Configuration Setup

10-03 : ETU Setup

Level:
IN

Description

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

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

Input Data

For CNF PKG Setup

Physical Port Number	01~16

Item No.	Item	Input Data	Default
01	**Logical Port Number**	0~256	0

For DLCA PKG Setup

Physical Port Number	01~16

Item No.	Item	Input Data	Default
01	**Terminal Type (B1)**	0 = Not set 1 = Multiline Terminal 2 = SLT Adapter 3 = Bluetooth Cordless Handset 6 = PGD(2)-U10 ADP (Paging) 7 = PGD(2)-U10 ADP (Tone Ringer) 8 = PGD(2)-U10 ADP (Door Box) 9 = PGD(2)-U10 ADP (ACI) 10 = DSS Console 11 = -- Not Used ---	0

Item No.	Item	Input Data	Default
02	**Logical Port Number (B1)**	0 = Not set 1 = Multiline Terminal (1~256) 2 SLT Adapter (1~256) 6 = PGD(2)-U10 ADP (Paging) (1~8) 7 = PGD(2)-U10 ADP (for Tone Ringer) (1~8) 8 = PGD(2)-U10 ADP (for Door Box) (1~8) 9 = PGD(2)-U10 ADP (for ACI) (1~96) 10 = DSS (1~32) 11 = --- Not Used ---	0
03	**--- Not Used ---**		
04	**Optional Installed Unit 1 (Only applies to DTH/ DTR style telephones)**	0 = None 1 = APR Module 2 = APA Module 3 = ADA Module 4 = CTA/CTU Module	0
05	**Optional Installed Unit 2**	0 = None 1 = APR Module 2 = APA Module 3 = ADA Module 4 = CTA/CTU Module	0

B-Channel 2			
Item No.	Item	Input Data	Default
06	**Terminal Type (B2)**	0 = Not set 6 = PGD(2)-U10 ADP (Paging) 7 = PGD(2)-U10 ADP (Tone Ringer) 8 = PGD(2)-U10 ADP (Door Box) 9 = PGD(2)-U10 ADP (ACI) 12 = APR(B2 Mode)	0
07	**Logical Port Number (B2)**	0 = Not set 6 = PGD(2)-U10 ADP (Ext. Speaker) 7 = PGD(2)-U10 ADP (Paging/Tone Ringer) (1~8) 8 = PGD(2)-U10 ADP (for Door Box) = (1~8) 9 = PGD(2)-U10 ADP (ACI) = (1~96) 12 = APR (for B2 mode) (193~512)	0

08	**Multiline Telephone Type**	0 = DT3** 1 = D^{term} 8 2 = D^{term} 7	0
09	**Side Option Information**	0 = No option 1 = 8LK Unit 2 = 16LK Unit 3 = 24ADM	0
10	**Bottom Option Information (Only applies to DTL style telephones)**	0 = No option 1 = APR 2 = ADA 3 = BHA	0
11	**Handset Option Information**	0 = No option 1 = PSA/PSD 2 = Bluetooth Cordless Handset	0

For LCA PKG Setup

Physical Port Number	01~16

Item No.	Item	Input Data	Default
01	**Logical Port Number**	0~256	0
03	**Transmit Gain Level (S-Level)**	1~63 (-15.5 +15.5dB)	32 (0dB)
04	**Receive Gain Level (R-Level)**	1~63 (-15.5 +15.5dB)	32 (0dB)

For COTA Unit Setup

Physical Port Number	1~8

Item No.	Item	Input Data	Default
01	**Logical Port Number**	0~200	0

For ODTA PKG Setup

Physical Port Number	01~04

Item No.	Item	Input Data	Default
01	**Logical Port Number**	0~200	0
02	**2/4 Wire**	0 = 2 Wire 1 = 4 Wire	1
03	**E&M Line Control Method**	0 = TYPE I 1 = TYPE V	0

For DIOP PKG Setup

Physical Port Number	01~04

Item No.	Item	Input Data	Default
01	**LD/OPX Specification**	0 = LD Trunk 1 = OPX	0
02	**Logical Port Number**	0 = 1~200 (LD Trunk) 1 = 1~256 (OPX)	0

For BRIA PKG Setup

ISDN Line Number	01~04

Item No	Item	Input Data	Default
01	**ISDN Line Mode**	0 = Not Used 1 = T-Point	1
02	**Logical Port Number** ✎ *The starting port number of a BRI line is displayed. Two logic ports are automatically assigned to a BRI line.*	0 = Not Used 1 = For T-Bus (1~200)	0
03	**Connection Type**	0 = Point-to-Multipoint 1 = Point-to-Point	0
04	**Layer 3 Timer Type** ✎ *Each timer value of Layer 3 is set up for every type using Program 81-06 (T-Bus)*	1~5	1
05	**CLIP Information Announcement** Based on this setting, the system includes a Presentation Allowed (1) or Presentation Restricted (0) in the Setup message to allow or deny the Calling Party Number. Program 15-01-04 must also be set to 1 if this option is enabled.	0 = Disable 1 = Enable	1
06	**--- Not Used ---**		
07	**--- Not Used ---**		
08	**Dial Sending Mode** ISDN Protocol definition	0 = Enblock Sending 1 = Overlap Sending	1
09	**Dial Information Element** ISDN Protocol definition [Only when Dialing Sending Mode (10-03-08) is set for 1 (Overlap Sending]	0 = Keypad Facility 1 = Called Party Number	0
10	**Master/Slave System** If set to 0, system is synchronized to network clock. If set to 1, system is not synchronized to the network clock.	0 = Slave System 1 = Master System	0
11	**--- Not Used ---**		
14	**--- Not Used ---**		
15	**--- Not Used ---**		
17	**ISDN Line Ringback Tone** If Telco does not provide ringback tone, SV8100 can if set to 1:Enable.	0 = Disable 1 = Enable	0

Item No	Item	Input Data	Default
18	**Type of Number** ISDN Protocol definition	0 = Unknown 1 = International number 2 = National number 3 = Network specific number 4 = Subscriber number 5 = Abbreviated number	2
19	**Numbering Plan Identification** ISDN Protocol definition.	0 = Unknown 1 = ISDN numbering plan 2 = Data numbering plan 3 = Telex numbering plan 4 = National standard numbering plan 5 = Private numbering plan	1
22	**--- Not Used ---**		
23	**Straight/Cross Wiring**	0 = Auto 1 = Manual (Cross) 2 = Manual (Straight)	0
24	**--- Not Used ---**		

For PRTA PKG Setup

ISDN Line Number	01~24

Item No.	Item	Input Data	Default
01	**--- Not Used ---**		
02	**Logical Port Number** ✎ *The start port number of a PRI line is displayed.*	1 = for T-Bus 1~200	1
03	**--- Not Used ---**		
04	**Layer 3 Timer Type** ✎ *Each timer value of Layer 3 is set up for each type in Program 81-06 (T-Bus)*	1~5	1
05	**CLIP Information** Based on this setting, the system includes a Presentation Allowed (1) or Presentation Restricted (0) in the Setup message to allow or deny the Calling Party Number. Program 15-01-04 must also be set to 1 if this option is enabled.	0 = Disable 1 = Enable	1
06	**Length of Cable**	0 = Level 1 1 = Level 2 2 = Level 3 3 = Level 4 4 = Level 5	2
07	**--- Not Used ---**		
08	**Dial Sending Mode** ISDN Protocol definition	0 = Enbloc Sending 1 = Overlap Sending	0
09	**Dial Information Element** ISDN Protocol definition (Only when Dialing Sending Mode (10-03-08) is set for 1 (Overlap Sending)	0 = Keypad Facility 1 = Called Party Number	0
10	**--- Not Used ---**		
11	**--- Not Used ---**		
12	**--- Not Used ---**		

Item No.	Item	Input Data	Default
13	**Loss-Of-Signal Detection Limit** If the transmit/receive voltage is less than the setting in 10-03-13, the system considers this as Loss-Of-Signal and the PRTA does not come up. Note that there are different values based on the setting in 10-03-12 for the PRI.	0 = Level 0 (lowest sensitivity) 1 = Level 1 2 = Level 2 3 = Level 3 4 = Level 4 5 = Level 5 6 = Level 6 7 = Level 7 (highest sensitivity)	2
14	**--- Not Used ---**		
15	**--- Not Used ---**		
16	**--- Not Used ---**		
17	**ISDN Ringback Tone** If Telco does not provide ringback tone, SV8100 can if 10-03-17 is set to 1:Enable.	0 = Disable 1 = Enable	1
18	**Type of Number** ISDN Protocol definition. Select the number type for the ISDN circuit.	0 = Unknown 1 = International number 2 = National number 3 = Network Specific number 4 = Subscriber number 5 = Abbreviated number	2
19	**Numbering Plan Identification** ISDN Protocol definition. Select the Numbering Plan used for the ISDN circuit.	0 = Unknown 1 = SDN numbering plan 2 = Data numbering plan 3 = Telex numbering plan 4 = National standard numbering plan 5 = Private numbering plan	1
20	**Network Exchange Selection** Select the ISDN protocol for the ISDN circuit.	0 = Standard (same as NI-2) 1 = reserved 2 = reserved 3 = DMS (A211) 4 = 5ESS 5 = DMS (A233) 6 = 4ESS 7 = NI-2	0
21	**Number of Ports**	0 = Auto 1 = 4 Ports 2 = 8 Ports 3 = 12 Ports 4 = 16 Ports 5 = 20 Ports	0
22	**--- Not Used ---**		

Item No.	Item	Input Data	Default
23	**Straight/Cross Wiring**	0 = Auto 1 = Manual (Cross) 2 = Manual (Straight)	0

For DTI (T1) PKG Setup

Physical Port Number	01~24

Item No.	Item	Input Data	Default
01	**Logical Port Number** The start port number of a T1 line is displayed, and 24 logic ports are automatically assigned to a DTI (T1) line.	0~200	0
02	**T1 Signal Format Selection**	0 = D4 (12 Multi Frame) 1 = ESF (24 Multi Frame)	1
03	**Zero Code Suppression**	0 = B8ZS 1 = AMI/ZCS	0
04	**Line Length Selection**	0 = 0 feet ~ 133 feet 1 = 133 feet ~ 266 feet 2 = 266 feet ~ 399 feet 3 = 399 feet ~ 533 feet 4 = 533 feet ~ 655 feet	0
05	**T1 Clock Source**	0 = Internal 1 = External	1
06	**Number of Ports**	0 = Auto 1 = 4 Ports 2 = 8 Ports 3 = 12 Ports 4 = 16 Ports 5 = 20 Ports	0
07	**Straight/Cross Wiring**	0 = Auto 1 = Manual (Cross) 2 = Manual (Straight)	0

For IPLA PKG Setup

Physical Port Number	001~128

Item No.	Item	Input Data	Default
01	**Trunk Logical Port Number**	0~200	0
02	**Trunk Type**	0 = H.323 1 = SIP	1
03	**CCIS Trunk**	0 = not CCIS 1 = CCIS	0

For VM00 PKG Setup

Physical Port Number	01~16

Item No.	Item	Input Data	Default
01	**Logical Port Number**	0~256	0

For CCTA PKG Setup

Physical Port Number	01~24

Item No.	Item	Input Data	Default
01	**Logical Port Number** The start port number of a T1 line is displayed, and 24 logic ports are automatically assigned to a DTI (T1) line.	0~200	0
02	**T1 Signal Format Selection**	0 = D4 (12 Multi Frame) 1 = ESF (24 Multi Frame)	1
03	**Zero Code Suppression**	0 = B8ZS 1 = AMI/ZCS	0
04	**Line Length Selection**	0 = 0 feet ~ 133 feet 1 = 133 feet ~ 266 feet 2 = 266 feet ~ 399 feet 3 = 399 feet ~ 533 feet 4 = 533 feet ~ 655 feet	0
05	**T1 Clock Source**	0 = Internal 1 = External	1

Item No.	Item	Input Data	Default
06	**Number of Ports**	0 = Auto 1 = 4 Ports 2 = 8 Ports 3 = 12 Ports 4 = 16 Ports 5 = 20 Ports	0
07	**Straight/Cross Wiring**	0 = Auto 1 = Manual (Cross) 2 = Manual (Straight)	0

Conditions

❍ When changing a defined terminal type, first set the type to 0 and then plug the new device in to have the system automatically define it, or redefine the type manually.

❍ The system must have a blade installed to view/change the options for that type of blade.

Feature Cross Reference

❑ Universal Slots

Program 10 : System Configuration Setup

10-04 : Music On Hold Setup

Level:
IN

Description

Use **Program 10-04 : Music on Hold Setup** to set the Music on Hold (MOH) source. For internal Music on Hold, the system can provide a service tone callers on hold or one of eleven synthesized selections.

Input Data

Item No.	Item	Input Data	Default	Description
01	**Music on Hold Source Selection**	0 = No Tone 1 = External MOH 2 = Service Tone 3 = Internal MOH	2	The Music on Hold (MOH) source can be internal (synthesized) or from a customer-provided music source. The customer-provided source can connect to a PGD(2)-U10 ADP or the connector on the side of the Base Cabinet MOH/IN connection. Trunk MOH and Extension MOH music source use the same Music on Hold source.
02	**Music on Hold Tone Selection**	[In case Item 1 is 0] 1 = Download File1 2 = Download File2 3 = Download File3 [In case Item 1 is 1, 2, or 3] 1~100 = VRS Message Number		
03	**Audio Gain Setup**	1~63 (-15.5 ~ +15.5dB)	32 (0dB)	

Conditions
None

Feature Cross Reference

❏ Analog Communications Interface (ACI)

❏ Background Music

❏ Music on Hold

Program 10 : System Configuration Setup
10-12 : CD-CP00-US Network Setup

Level:
SA

Description

Use **Program 10-12 :** CD-CP00-US **Network Setup** to setup the IP Address, Subnet-Mask, and Default Gateway addresses.

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

Input Data

Item No.	Item	Input Data			Default	Description
01	IP Address	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.254.255.254 192.0.0.1 ~ 223.255.255.254			192.168.0.10	Set for CD-CP00-US.
02	Subnet Mask	128.0.0.0 240.0.0.0 254.0.0.0 255.192.0.0 255.248.0.0 255.255.0.0 255.255.224.0 255.255.252.0 255.255.255.128 255.255.255.240 255.255.255.254	192.0.0.0 248.0.0.0 255.0.0.0 255.224.0.0 255.252.0.0 255.255.128.0 255.255.240.0 255.255.254.0 255.255.255.192 255.255.255.248 255.255.255.255	224.0.0.0 252.0.0.0 255.128.0.0 255.240.0.0 255.254.0.0 255.255.192.0 255.255.248.0 255.255.255.0 255.255.255.224 255.255.255.252	255.255.255.0	The setting of Subnet Mask is invalid when all Host Addresses are 0. If the network section is: 0, 127, 128.0, 191.255, 192.0.0, 223.255.255 The setting of Subnet Mask is invalid.
03	Default Gateway	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.254.255.254 192.0.0.1 ~ 223.255.255.254			0.0.0.0	IP Address for Router.

Input Data (Continued)

Item No.	Item	Input Data	Default	Description
04	Time Zone	0~24 (0 = -12 Hours and 24 = +12 Hours)	+7 (-5 hours)	Determine the offset from Greenwich Mean Time (GMT) time. Then enter its respective value. For example, Eastern Time (US and Canada) has a GMT offset of -5. The program data would then be 7 (0= -12, 1= -11, 2= -10, 3= -9, 4= -8, 5= -7, 6= -6, 7= -5, ……24= +12)
05	**NIC Interface**	0 = Auto Detect 1 = 100Mbps, Full Duplex 2 = 100Mbps, Half Duplex 3 = 10Mbps, Full Duplex 4 = 10Mbps, Half Duplex	0	NIC Auto Negotiate (CD-CP00-US)
06	**NAT Router Setup**	0 = No (Disable) 1 = Yes (Enable)	0	If using an external NAT Router or not.
07	**NAPT Router IP Address (Default Gateway [WAN])**	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0	Set the IP address on the WAN side of router.
08	**ICMP Redirect**	0= (Enable) 1= (Disable)	0	When receiving ICMP redirect message, this determines if the IP Routing Table updates automatically or not.
09	**IP Address**	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	172.16.0.10	Set for IPLA.

Input Data (Continued)

Item No.	Item	Input Data			Default	Description
10	**Subnet Mask**	128.0.0.0	192.0.0.0	224.0.0.0	255.255.0.0	Set for IPLA.
		240.0.0.0	248.0.0.0	252.0.0.0		
		254.0.0.0	255.0.0.0	255.128.0.0		
		255.192.0.0	255.224.0.0	255.240.0.0		
		255.248.0.0	255.252.0.0	255.254.0.0		
		255.255.0.0	255.255.128.0	255.255.192.0		
		255.255.224.0	255.255.240.0	255.255.248.0		
		255.255.252.0	255.255.254.0	255.255.255.0		
		255.255.255.128	255.255.255.192	255.255.255.224		
		255.255.255.240	255.255.255.248	255.255.255.252		
		255.255.255.254	255.255.255.255			
11	**NIC Setup**	0 = Auto Detect			0	Set for IPLA.
		1 = 100Mbps, Full Duplex				
		2 = 100Mbps, Half Duplex				
		3 = 10Mbps, Full Duplex				
		4 = 10Mbps, Half Duplex				
		5 = 1 Gbps, Full Duplex				
		6 = 1 Gbps, Half Duplex				

Conditions

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

Feature Cross Reference

❑   Voice Over Internet Protocol (VoIP)

Program 10 : System Configuration Setup
10-13 : In-DHCP Server Setup

Level:
SA

Description

Use **Program 10-13 : In-DHCP Server Setup** to setup the DHCP Server built into the CD-CP00-US blade.

Input Data

Item No.	Item	Input Data	Default	Description
01	**DHCP Server Mode**	0 = Disable 1 = Enable	0	Enable/Disable the the built-in DHCP Server.
02	**Lease Time**	Days 0~255	0 day	Lease Time of the IP address to a client.
		Hour 0~23	0 hour	✎ *Press Transfer to increment to the next setting data.*
		Minutes 1~59	30 minutes	
05	**Last DHCP Data**	0 = Disable 1 = Enable	1	If 10-13-01 is enabled, Enable/Disable DHCP resource.

Conditions
None

Feature Cross Reference

❑   Voice Over Internet Protocol (VoIP)

Program 10 : System Configuration Setup

10-14 : Managed Network Setup

**Level:
SA**

Description

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

Item No.	Item	Input Data	Default	Related Program
01	**The range of the IP address to lease.** When Maximum has not been entered, the maximum value equals the minimum value. When **Single** is selected in 10-13-04, only 1 scope range can be entered. When **Divide Same Network** is selected in 10-13-04, a maximum of 10 scope ranges can be entered.	Minimum: 1.0.0.1 ~ 126.255.255.254 128.1.0.1 ~ 191.254.255.254 192.0.1.1 ~ 223.255.254.254	172.16.0.100	
		Maximum: 1.0.0.1 ~ 126.255.255.254 128.1.0.1 ~ 191.254.255.254 192.0.1.1 ~ 223.255.254.254	172.16.5.254	

Conditions
None

Feature Cross Reference

❑   Voice Over Internet Protocol (VoIP)

Program 10 : System Configuration Setup

10-15 : Client Information Setup

Level:
SA

Description

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

Input Data

Client Number	1~512

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

Conditions
None

Feature Cross Reference

❑   Voice Over Internet Protocol (VoIP)

Program 10 : System Configuration Setup

10-16 : Option Information Setup

Description

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

Input Data

Item No.	Item	Input Data	Default
01	**Router** Set the Router IP address.	Code number 0~255	3 (Fixed)
		IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0
02	**DNS Server** Set IP address of DNS Server.	Code number 0~255	6 (Fixed)
		IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0
03	**TFTP Server** Set the name for the TFTP Server.	Code number 0~255	66 (Fixed)
		Maximum 64 character strings	No setting
05	**MGC**	Code number 0~255	129 (Fixed)
		IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	172.16.0.10
06	**Client Host Name** Set the Client Host Name.	Code number 0~255	12 (Fixed)
		Maximum 64 character strings	No setting
07	**DNS Domain Name** Set the DNS Domain Name.	Code number 0~255	15 (Fixed)
		Maximum 20 character strings	No setting

Input Data (Continued)

Item No.	Item	Input Data	Default
08	**Download Protocol** Set Download Protocol used for AutoConfig (for DT700 Series).	Code number 0~255	43 (Fixed)
		Sub code number	163
		1 = FTP 2 = HTTP	1
09	**Encryption Information** Set an Encryption Information used for AutoConfig (for DT700 series).	Code number 0~255	43 (Fixed)
		Sub code number	164
		Maximum 128 character strings	No setting
10	**FTP Server Address** Set a FTP Server Address used for AutoConfig.	Code number 0~255	43 (Fixed)
		Sub code number	141
		IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0
11	**Config File Name** Set a File Name used for AutoConfig.	Code number 0~255	43 (Fixed)
		Sub code number	151
		Maximum 15 character strings	No setting
12	**Vender Class ID**	Code number 0~255	60 (Fixed)
		Maximum 256 character strings	NECDT700
13	**SNMP Server**	Code number 0~255	69 (Fixed)
		IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0
14	**POP3 Server**	Code number 0~255	70 (Fixed)
		IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0
16	**SIP Server (IP Address)**	Code number 0~255	120 (Fixed)
		IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	172.16.0.10

Input Data (Continued)

Item No.	Item	Input Data	Default
17	**SIP Server (Domain Name)**	Code number 0~255	120 (Fixed)
		Maximum 20 character strings	No setting
18	**FTP Server**	Code number 0~255	141 (Fixed)
		IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0
19	**Config File Name**	Code number 0~255	151 (Fixed)
		Maximum 15 character strings	No setting
20	**LDS Server 1**	Code number 0~255	162 (Fixed)
		IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0
21	**LDS Server 2**	Code number 0~255	162 (Fixed)
		IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0
22	**LDS Server 3**	Code number 0~255	162 (Fixed)
		IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0
23	**LDS Server 4**	Code number 0~255	162 (Fixed)
		IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0
24	**Next Server IP Address**	IP address 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0
27	**SIP Server Receive Port**	Code number 0~255	168 (Fixed)
		Port: 1~65535	5080

Conditions
None

Feature Cross Reference

❑   Voice Over Internet Protocol (VoIP)

Program 10 : System Configuration Setup
10-19 : VoIP DSP Resource Selection

Level:
SA

Description

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

Input Data

Slot Number	1

Input Data

DSP Resource Number	01~128

Input Data

Item No.	Item	Input Data	Default
01	**VoIP DSP Resource Selection**	0 = Common use for both IP extensions and trunks 1 = IP Extension 2 = SIP Trunk 3 = CCIS 4 = Use for NetLink 5 = Blocked	Resource 1 = 1 Resource 2~128 = 0

Conditions
None

Feature Cross Reference

❑   None

Program 20 : System Option Setup
15-05 : IP Telephone Terminal Basic Data Setup

Level:

IN

Description

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

Input Data

Extension Number	Maximum eight digits

Item No.	Item	Input Data	Default	Description	Related Program
01	**Terminal Type**	0 = NGT 1 = H.323 2 = SIP 3 = MEGACO 4 = SIP-MLT	0	Viewing Only – No changes permitted	
02	**IP Phone Fixed Port Assignment**	MAC address 00-00-00-00-00-00 to FF-FF-FF-FF-FF-FF	00-00-00-00-00-00	MAC Address of registered MLT SIP phone is stored and/or can input the MAC address of an MLT SIP phone so when it comes online it will be provided with the extension in which the MAC address matches.	15-05-01
04	**Nickname**	Up to 48 characters	No setting	Nickname section on Invite message. Example: Extension 100 has a Nickname set to PAUL. Extension 101 has command 15-05-17 set to Nickname. The inbound call to extension 101, from 100, will show PAUL.	15-05-17
07	**Using IP Address**	0.0.0.0~255.255.255.255	0.0.0.0	Informational Only registered IP Phones	15-05-01
15	**CODEC Type**	1-Type 1 2-Type 2 3-Type 3 4-Type 4 5-Type 5	1	Assign the CODEC Type of the MLT SIP.	84-24-XX

Item No.	Item	Input Data	Default	Description	Related Program
16	**Authentication Password**	Up to 24 characters	None	Assign the authentication password for SIP single line telephones.	15-05-01
17	**Calling Party Display Info**	0 = Nickname 1 = Display Name 2 = User Part 3 = Extension	0	The part of the Invite message the calling party information is taken from. There are four choices: **Nickname**: Displays the nickname programmed in 15-05-04. **Display Name**: Some SIP phones have a Field called Display Name. If configured, in the SIP phone, this displays on a call from that station. **User Part**: Some SIP phones have a field called User Part. If configured in the SIP phone, this displays on a call from that station. **Extension**: Display shows extension of the SIP phone.	
18	**IP Duplication Allowed Group**	0 = Not Used 1 = Group 1 2 = Group 2 3 = Group 3 4 = Group 4 5 = Group 5 6 = Group 6 7 = Group 7 8 = Group 8 9 = Group 9 10 = Group 10	0	For an adapter with one IP address coming into it but multiple extensions off of it. Assign all the extensions to a group so that way the CPU knows that the one IP address is assigned to multiple extensions.	15-05-01
19	**Side Option Information**	0 = No Option 1 = 8LK Unit 2 = 16LK Unit 3 = 24ADM	0	This is a read only program that shows what type of Line Key unit is installed on the ITH-style telephone.	10-03-09 15-05-22
20	**Bottom Option Information**	0 = No Option 1 = ADA 2 = BHA	0	This is a read only program that shows what type of adapter is installed on the ITH-style telephone.	10-03-10
21	**Handset Option Information**	0 = Normal Handset 1 = Handset for power failure (PSA/PSD) 2 = BCH	0	This is a read only program that shows what type of Handset is installed on the ITH-style telephone.	10-03-11 15-05-23

Item No.	Item	Input Data	Default	Description	Related Program
22	**Side Option Additional Data**	0 = No Setting 1~32 = DSS Console number	0	This is a read only program that shows the DSS console number when one is installed on a ITH-style telephone.	30-01 30-02 30-03 30-04 30-05 30-06
23	**Handset Option Additional Information** Determine to use TEN or not.	0 = No Setting 1~16 = Terminal equipment number (TEN) of Bluetooth Cordless Handset (BCH)	0		
24	**Protection Service**	0 = Not Used 1 = Used	0	Enable this to allow the MLT SIP telephones to use the security key. If disabled, and the key is pressed, nothing happens.	90-40-01 90-40-02
26	**DT700 Terminal Type**	0 = Not Set 1 = ITL-**E-1D/IP-*E-1 2 = ITL-**D-1D/ITL-24BT1D/ITL- 4PA-1D [without 8LKI(LCD)-L] 3 = ITL-**D-1D/ITL-24BT1D/ITL-24PA-1D [with 8LKI(LCD)-L] 4 = ITL-320C-1 5 = Softphone 6 = CTI 7 = AGW	0		
27	**Personal ID Index**	0~512	0	Use 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-XX
28	**Addition Information Setup** Select whether to inform of additional information or not.	0 = Disable 1 = Enable	0		
29	**Terminal WAN-side IP Address**	0.0.0.0~255.255.255.255	0.0.0.0		

Item No.	Item	Input Data	Default	Description	Related Program
30	**DTMF play during conversation at Receive Extension**	0 = Do Not Play 1 = Play	0		
31	**Alarm Tone during conversation** (RTP packet loss alarm)	0 = Not Ringing 1 = Ringing	1		
32	**Dial Key Talkie**	0 = Not Used 1 = Used	0		
33	**LAN Side IP Address of Terminal**	0.0.0.0~255.255.255.255	0.0.0.0.	Read-only	
34	**Terminal Touch Panel On/Off**	0 = Off 1 = On	1	Set whether the touch screen used on ITL-320C-1 (BK) TEL can be used (On) or cannot be used (Off).	

Conditions

❍ 15-05-04 – Nickname must be unique in the system.

Feature Cross Reference

None

❍ Voice Over Internet Protocol (VoIP)

Conditions

❍ The system programming must be exited before these program options take affect.

Feature Cross Reference

❍ Voice Over Internet Protocol (VoIP)

Program 84 : Hardware Setup for VoIP
84-03 : IP Phone Information Basic Setup

Level:
IN

Description

Use **Program 84-03 : IP Phone Information Basic Setup** to define the details of IP Terminals and Adapters.

Input Data

Terminal Type	1 = D^{term} IP / UNIVERGE SV8100 Multiline Terminal with IP Adapter 2 = Soft Phone 3 = Bundle IP Phone

Item No.	Item	Input Data	Default
06	**Area Number**	0 = Japan 1 = USA 2 = Australia 3 = EU 4 = Asia 5 = Other Country 6 = Germany 7 = Italy 8 = Netherlands 9 = Austria 10 = Belgium 11 = Spain 12 = Sweden 13 = UK 14 = Denmark 15 = Greece 16 = Switzerland 17 = RSA 18 = New Zealand	1
10	**Signaling Server Port**	0~65535	5029
11	**Negotiation Timer**	0~255 sec	5

Conditions
None

Feature Cross Reference

Voice Over Internet Protocol (VoIP)

Program 84 : Hardware Setup for VoIP

84-06 : PVA Data Setting

Level:
IN

Description

Use **Program 84-06 : PVA Data Setting** to set up threshold levels of every CD-PVAA slot. (This Program is not available with V1000).

Input Data

Slot Number	01~24

Item No.	Item	Input Data	Default	Description
01	**RTP Port Number**	0~65534	10020	
02	**RTCP Port Number**	RTP Port Number + 1	10021	It has to be RTP Port Number + 1.
04	**Fract Lost Threshold**	0~100%	0	The data is sent to the CD-CP00-US if the value exceeds the defined value.
05	**Packets Lost Threshold**	0~16777215	0	The data is sent to the CD-CP00-US if the value exceeds the defined value.
07	**Jitter Threshold**	0~4294967295 (sec)	0	The data is sent to the CD-CP00-US if the value exceeds the defined value.
09	**Delay LSR Threshold**	0~4294967295 (sec)	0	The data is sent to the CD-CP00-US if the value exceeds the defined value.
16	**IMCP Redirect**	0=No 1=Yes	0	The VOIP blade supports sending the Internet Message Control Protocol (IMCP) redirect message.

Conditions

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

Feature Cross Reference

❑   Voice Over Internet Protocol (VoIP)

Program 84 : Hardware Setup for VoIP

84-09 : VLAN Setup

Level:
IN

Description

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

Input Data

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

Conditions

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

Feature Cross Reference

❑ Voice Over Internet Protocol (VoIP)

Program 84 : Hardware Setup for VoIP
84-10 : ToS Setup

Description

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

Input Data

Protocol Type	1 = Not Used
	2 = Not Used
	3 = Voice Control
	4 = H.323
	5 = RTP/RTCP
	6 = SIP
	7 = CCISoIP
	8 = DT700 MLT
	9 = SIP Trunk
	10 = Net-Link

Item No.	Item	Input Data	Default	Description
01	**ToS Mode**	0 = Disable (Invalid) 1 = IP Precedence 2 = Diffserv	0	When Input Data is set to 1, Item No. 07 is invalid. When Data is set to 2, Item No. 02 ~ 06 are invalid.
02	**Priority, IP Precedence**	0~7 0 = Low 7 = High	0	1 = Router queuing priority
03	**Low Delay**	0~1 0 = Normal Delay, Low Delay	0	1 = Optimize for low delay routing
04	**Wideband (Throughout)**	0~1 0 = Normal Throughput 1 = High Throughput	0	1 = Optimize for high bandwidth routing
05	**High Reliability**	0~1 0 = Normal Reliability 1 = Low Reliability	0	1 = Optimize for reliability routing
07	**Priority (D.S.C.P. - Differentiated Services Code Point)**	0~63	0	DSCP (Differentiated Services Code Point)

Conditions

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

Feature Cross Reference

❑   Voice Over Internet Protocol (VoIP)

Program 84 : Hardware Setup for VoIP
84-11 : D^{term} IP CODEC Information Basic Setup

Level:
IN

Description

Use **Program 84-11 : D^{term} IP CODEC Information Basic Setup** to set voice (RTP packet) encoding parameters.

Input Data

Type	1~5

Item No.	Item	Input Data	Default
01	**Number of G711 Audio Frames**	2~3	3
02	**G711 VAD mode**	0 = Disable 1 = Enable	0
03	**G711 Type**	0 = A-law 1 = μ-law	1
04	**G.711 Jitter Buffer Min**	0~160 ms	30
05	**G.711 Jitter Buffer Average**	0~160 ms	60
06	**G.711 Jitter Buffer Max**	0~160 ms	120
07	**Number of G.729 Audio Frames**	1~4	3
08	**G.729 VAD mode**	0 = Disable 1 = Enable	0
09	**G.729 Jitter Buffer Min**	0~200 ms	30
10	**G.729 Jitter Buffer Average**	0~200 ms	60
11	**G.729 Jitter Buffer Max**	0~200 ms	120
17	**Jitter Buffer Mode**	1 = Static 2 = Immediate	2

Item No.	Item	Input Data	Default
18	**VAD Threshold**	0~30 = -19dB~+10dB 0 = Adaptec Threshold 1 = -19dB(-49dBm) : 20 = 0dB (-30dBm) : 29 = 9dBm(-21dBm) 30 = 10dBm(-20dBm)	20
26	**TX Gain**	0~40 (-20dBm~ +20dBm) 0 = -14dBm 1 = -13dBm : 14 = 0dBm : 19 = 5dBm 20 = 6dBm	14
27	**RX Gain**	0~40 (-20dBm~ +20dBm) 0 = -14dBm 1 = -13dBm : 14 = 0dBm : 19 = 5dBm 20 = 6dBm	14
28	**Audio Capability Priority**	0 = G.711_PT 2 = G.729_PT	0

Conditions
None

Feature Cross Reference

❏   Voice Over Internet Protocol (VoIP)

Program 84 : Hardware Setup for VoIP
84-17 : IPL Echo Canceller Control Setup (IN Level)

Level:
IN

Description

Use **Program 84-17 : IPL Echo Canceller Control Setup** to configure echo cancellation.

Input Data

Type	4~5 (Type 1, 2 and 3 are not used) Type 4 = Analog trunks Type 5 = Digital/MEGACO trunks

Input Data

Item No.	Item	Input Data	Default
01	**Echo Canceller Mode**	0 = Disable 1 = Enable	Type 4 = 1 Type 5 = 1
02	**Echo Canceller Tail Size**	1 = 8 ms 2 = 16 ms 3 = 32 ms 4 = 64 ms 5 = 128 ms	Type 4 = 5 Type 5 = 1
03	**Echo Canceller NLP Mode**	0 = Disable 1 = Enable	Type 4 = 1 Type 5 = 0
04	**Echo Canceller NLP Noise Setting**	0 = Adaptive 1 = Fixed	0
05	**Echo Canceller NLP Noise Level**	40~70 (-40dbm~ -70dbm)	60

Conditions
None

Feature Cross Reference

None

❍   Voice Over Internet Protocol (VoIP)

Program 90 : Maintenance Program
90-23 : Deleting Registration of IP Telephones

Level:
IN

Description

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

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

Input Data

Extension Number	Up to 8 digits

Item No.	Item	Input Data
01	**Delete IP Telephone** This assignment removes the station number association with the MAC address of the IP station.	[Delete?] : Dial **1** + press **Transfer** (Press **Transfer** to cancel.)

Conditions
None

Feature Cross Reference

❏   Voice Over Internet Protocol (VoIP)

Program 90 : Maintenance Program

90-34 : Firmware Information

Level:
IN

Description

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

Input Data

Slot No.	1~24

Item No.	Item	Display Data
01	**Pkg Name**	PKG Name
02	**Firmware Version Number**	00.00~0F.FF

Conditions

❍ These Programs are 'Read Only.'

Feature Cross Reference

None

Network Design Considerations

SECTION 1   INTRODUCTION

This chapter explains some issues that should be considered when planning a UNIVERGE SV8100 VoIP installation. This is a generalized explanation and therefore does not discuss vendor-specific issues and solutions. Typically, different solutions are implemented by different manufacturers.

SECTION 2   QOS

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

This chapter also describes the problems that can occur and some possible solutions. Each network equipment manufacturer (NEC, 3Com, Cisco, etc.) has slightly different methods of implementing QoS and these are not discussed in this document. This chapter provides an overview to classify voice traffic on the UNIVERGE SV8100 so that the network equipment can impose QoS.

2.1   QoS Definitions

This section lists common definitions used with QoS for VoIP.

Latency (Delay):
If at any point the usage on the network exceeds the available bandwidth, the user experiences delay, also called latency. In more traditional uses of an IP data network, the applications can deal with this latency. If a person is waiting for a web page to download, they can accept a certain amount of wait time. This is not so for voice traffic. Voice is a real time application, which is sensitive to latency. If the end-to-end voice latency becomes too long (250ms, for example),

the call quality is usually considered poor. It is also important to remember that packets can get lost. IP is a best effort networking protocol. This means the network tries to get the information there, but there is no guarantee.

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

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

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

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

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

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

All transport systems exhibit some jitter. As long as jitter falls within defined tolerances, it does not impact service quality. Excessive jitter can be overcome by buffering, but this increases delay, which can cause other problems. With intelligent discard mechanisms, IP telephony/VoIP systems try to synchronize a communication flow by selective packet discard, in an effort to avoid the

walkie-talkie phenomenon caused when two sides of a conversation have significant latency. UNIVERGE SV8100 incorporates a Jitter Buffer to avoid these problems.

Packet Loss

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

2.2   Voice Quality Improvements

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

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

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

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

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

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

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

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

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

2.3   Types of Classifications for Traffic for QoS

Classification uses information from a packet (or frame) to define the type of
data and therefore how the data should be handled for QoS on the network.
Using packet classification, you can partition network traffic into multiple
priority levels or Types of Service (ToS). UNIVERGE SV8100 supports
methods of marking a packet with its classification information in the Layer 2
or 3 headers.

❍   VLAN (802.1Q):
    Virtual LANs work at Layer 2 of the OSI model and can be equated to a
    broadcast domain. More specifically, VLANs can be seen as a group of
    end stations, perhaps on multiple physical LAN segments that are not
    constrained by their physical location and therefore, communicate as if
    they were on a common LAN. Packets can be marked as important by

using layer 2 classes of service (CoS) settings in the User Priority bits of the 802.1Pq header. Refer to Program 84-09 : VLAN Setup on page 5-30 for information for VLAN configuration.

Layer 2
802.1Q/p

| PREAM | SFD | DA | SA | Type | TAG 4 Bytes | PT | Data | FCS |

| PRI | CFI | VLAN ID |

Three bits used for CoS
(user priority)
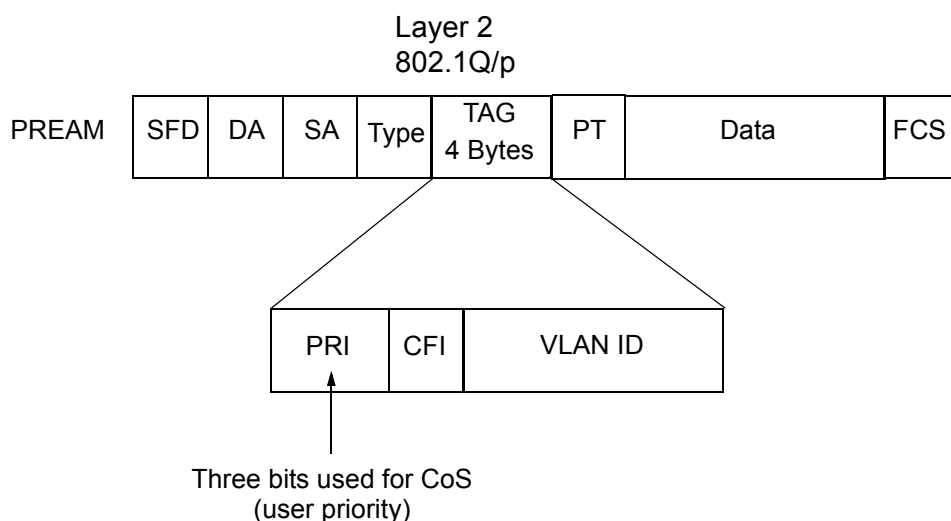
Figure 5-1 Layer 2 Diagram (802.1Q)

❍   IP Precedence - Layer 3 QoS:
Allows you to specify the class of service for a packet. You use the three precedence bits in the Ipv4 header type of service (ToS) field for this purpose. Using the ToS bits, you can define up to six classes of service. Other devices configured throughout the network can then use these bits to determine how to treat the packet in regard to the type of service to grant it. These other QoS features can assign appropriate traffic-handling policies including congestion management and bandwidth allocation. By setting IP Precedence levels on incoming traffic and using them in combination with QoS queuing features, you can create differentiated service. (Refer to Program 84-10 : ToS Setup on page 5-31 for detailed programming information.)

❍   Differentiated service (Diffserv) - Layer 3 QoS:
Provides services differentiated on performance using weighted priority queuing. DiffServ requires that edge routers classify traffic flows into a member from a set of categories based on the TCP/IP header fields in what is called a micro flow. Because the Diffserv is present in every packet header, each node can provide differentiated services per-hop. Refer to Program 84-10 : ToS Setup on page 5-31 for detailed programming information.

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

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

Asymmetric Data Rates

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

Contention

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

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

Network Address Translation (NAT)

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

VPN

Due to the use of NAT, and non-routable IP addressing, it is necessary to implement a VPN solution. This is outlined in VPN Tunneling below. (Refer to 4.3 Virtual Private Network (VPN) Tunnelling on page 5-9.)

QoS

As discussed earlier, it is essential to have some form of Quality of Service implemented. With internet based connections, we are not in control of the many routers, switches and other network hardware that reside between our two VoIP endpoints. This means that we cannot specify any QoS parameter on these devices.

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

When implementing UNIVERGE SV8100 IP over internet based connections it is very important that these factors are considered, and that the customer is made aware that neither the installer nor NEC are held responsible for any quality issues experienced.

SECTION 4   FIREWALLS AND NAT

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

4.1   Understanding the Infrastructure

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

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

❍   QoS devices to support precedence settings of voice packets.

❍   Elimination of hubs in place of switches to support 100Mbps full-duplex transmission.

❍   Firewall integration to protect the internal network from external attack.

❍   Network Address Translation (NAT) devices are widely deployed to support the addressing issues.

❍   Virtual Private Network (VPN) Servers were added to Enterprise networks to support the security and connectivity issues for remote users.

Some solutions, such as the hub replacement and integration of QoS, are done behind the scenes and should have no effect on the voice application. Other solutions such as NAT and Firewall cause major disturbance to VoIP. Implementing a VPN is the only way to resolve these issues.

4.2   Firewall Integration

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

❍   Firewalls
Firewalls can be implemented in both hardware and software, or a combination of both. Firewalls are frequently used to prevent unauthorized Internet users from accessing private networks connected to the Internet, especially intranets. All messages entering or leaving the intranet pass through the firewall, which examines each message and blocks those that do not meet the specified security criteria.

❍   Proxy Server
Proxy server intercepts all messages entering and leaving the network. The proxy server effectively hides the true network address.

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

The ports that need to be open on the firewall/proxy vary depending on the particular application being used. A list of these ports is shown below, however it should be noted that the preferred solution would be to allow all ports on the UNIVERGE SV8100 device to be open, or to place the SV8100 outside of the firewall.

Applications	Rx Port	UNIVERGE SV8100 Programming
Pc Programming	7	
DHCP Server	67	
SIP MLT Listening Port	5080	10-46-06
SIP Trunk Listening Port	5060	10-29-04
SIP Single Line Stations	5070	84-20-01
Realtime Transport Protocol	10020	84-06-01

Applications	Rx Port	UNIVERGE SV8100 Programming
Realtime Transport Control Protocol (RTCP)	10021	84-06-02

4.3   Virtual Private Network (VPN) Tunnelling

A Virtual Private Network is a private data network that maintains privacy through using a tunneling protocol and security procedures. Allowing for remote networks (including VoIP devices), which reside behind NATs and/or Firewalls to communicate freely with each other. In UNIVERGE SV8100 VoIP networks, implementation of VPNs can resolve the issues with NAT that are described in the previous section.

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

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

❍   Remote User to Corporate Site VPN
       Allows employees to use their local ISP fastest connection such as cable modems, DSL, and ISDN. For traveling users, all they need to do is dial into their ISP local phone number.

❍   Site-to-site VPN
       Allows companies to make use of the Internet for the branch-to-branch connections, cutting the cost of the expensive point to point leased line service.

❍   Extranet
       Extranet describes one application using VPN technology. The concept allows a company and a vendor/supplier to access network resources at each site. For example, a customer may have access to a suppliers intranet for access to product information.

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

The diagram below is example of how a VPN tunnel may be implemented. The red lines in the diagram show the tunnels that are created through the internet. Each network can connect to the others as though they are connected with private connections (kilostream, etc.), without the issues relating to NAT.
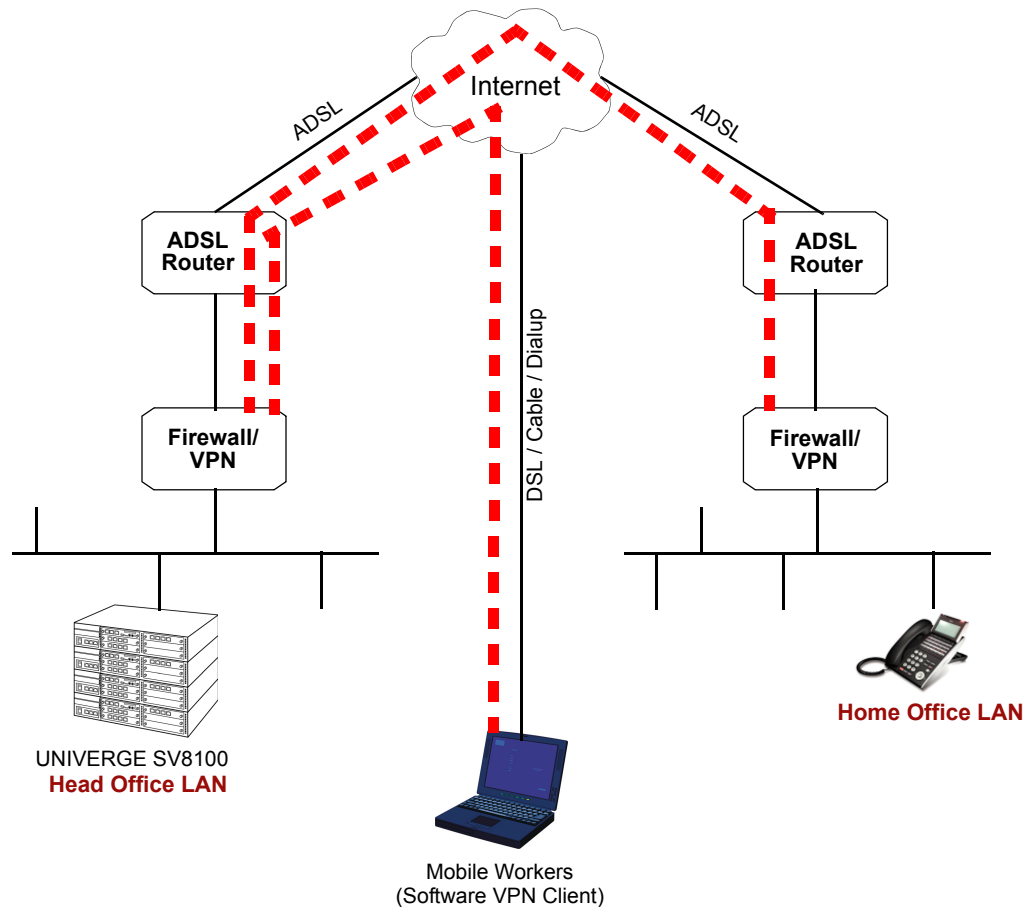
Figure 5-2 Virtual Private Network (VPN) Example

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

The common scenario for remote IP deployment is:

❍   Implementation of an IP Phone with a public IP address talking with an
       UNIVERGE SV8100 behind NAT. An example would be a telecommuter.

       Implementation of an IP Phone behind a NAT, which connects to the
       internet, terminates in a UNIVERGE SV8100 behind a different NAT.
       When selecting VPN equipment it is important to consider Quality of
       Service. Generally, VPN hardware is connected to internet connections
       which are unreliable and out of the control of the customer. However, it is
       possible to set prioritization on some VPN units for voice traffic. This does
       not solve the unreliability of the internet, but helps to ensure that the data
       traffic to and from the LAN do not impair the quality of the voice traffic.
       (Refer to Section 2 QoS on page 5-1).

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

SECTION 5   CODEC AND BANDWIDTH

This section describes CODEC and bandwidth and their application with the
UNIVERGE SV8100 system.

5.1   CODECs

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

The UNIVERGE SV8100 system supports three different CODECs:

❍   G.711
       This is the ITU-T recommendation for coding of speech at 64kbps using
       PCM (pulse code modulation). This CODEC is often described as
       uncompressed as it uses the same sampling rate as Time-Division
       Multiplexing (TDM). G.711 has a MOS[1] score of 4.2 but uses a large
       bandwidth for transmission. This CODEC is not commonly used due to
       the bandwidth required, although it can be acceptable in LAN
       environment (i.e., IP Phones connected over a 100Mbps LAN).

❍   G.722
       G.722 is an ITU standard CODEC that provides 7kHz wideband audio at
       data rates from 48 to 64kbps. This is useful in a fixed network Voice
       Over IP applications, where the required bandwidth is typically not

---

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

prohibitive, and offers a significant improvement in speech quality over older narrowband codecs such as G.711, without an excessive increase in implementation complexity.

❍   G.726
G.726 is an ITU-T ADPCM speech CODEC standard covering voice transmission at rates of 16, 24, 32, and 40kbit/s. It was introduced to supersede both G.721, which covered ADPCM at 32kbit/s, and G.723, which described ADPCM for 24 and 40kbit/s. G.726 also introduced a new 16kbit/s rate. The four bit rates associated with G.726 are often referred to by the bit size of a sample as 2-bits, 3-bits, 4-bits, and 5-bits respectively.

❍   G.729A
This ITU-T recommendation describes the algorithm for coding of speech signals at 8kbps using CS-ACELP (conjugate-structure algebraic code-excited linear prediction). This CODEC samples the analog signal at 8000Hz and uses a frame size of 10ms. This CODEC has a MOS score of 4.0.

G.729 is the most commonly used CODEC for UNIVERGE SV8100 VoIP installations. This is due to the fact that it offers high compression (and therefore low bandwidth) while maintaining good speech quality.

❍   G.723
This ITU-T recommendation describes a very low bit-rate compression algorithm. The standard describes two versions 5.3Kbps and 6.4Kbps. UNIVERGE SV8100 uses the higher bit rate. This CODEC offers low bandwidth speech transmission, but has a lower MOS score of 3.9. This CODEC is not commonly used on the UNIVERGE SV8100, but is particularly suited to low bandwidth WAN connections.

❍   iLBC
The iLBC CODEC is an algorithm that compresses each basic frame (20ms or 30ms) of 8000 Hz, 16-bit sampled input speech, into output frames with rate of 400 bits for 30ms basic frame size and 304 bits for 20ms basic frame size. This CODEC is suitable for real-time communications such as, telephony and video conferencing, streaming audio, archival and messaging.

Packet Size:

Each CODEC has a set frame length. This is the time that the frame encapsulates. For G.729 and G.711 the frame length is 10ms and for G.723 the frame length is 30ms. It is possible to configure the packet size in the UNIVERGE SV8100 programming. To do this, we tell the UNIVERGE SV8100 how many frames to encapsulate into each packet for transmission.

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

5.2   Bandwidth

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

❍   Number of simultaneous calls

❍   CODEC used

❍   Frame Size

❍   Data Networking Protocol used

❍   UNIVERGE SV8100 CCIS Network parameters

The more frames encapsulated into each packet, the less bandwidth is required. This is because each packet transmitted has the same header size. Therefore, if numerous very small packets are sent then bandwidth is also being used for a large amount of header information. If we add several frames to the packet, less packets are transmitted and therefore have less header information sent.

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

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

Examples:

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

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

SECTION 6   DSP RESOURCE CALCULATION

Voice over IP (NECi SIP, SIP stations, SIP trunks) requires DSP resources to be able to convert from TDM[1] to IP technologies. DSPs (Digital Signal Processors) take a TDM signal and convert to Realtime Transport Protocol (RTP) for transmission as VoIP, and vice versa. Each IP to TDM conversion requires a DSP resource.

---

1.   TDM = Time Division Multiplexing - traditional circuit-based telephony

DSP resources are provided by VoIP blades. For example, an MG16 VoIP blade has 64 DSP resources. It can be difficult to work out how many DSP resources are required in an UNIVERGE SV8100 system, because:

❍   not all IP Extensions/trunks are used at the same time

❍   peer-to-peer calls do not use a DSP resource

PZ-(X)IPLA IP Addressing

For every 16 DSPs used, one valid IP address must be assigned. Along with the DSP IP address, another IP address is required for registration and signaling. For example, for full functionality of the IPLA 32, three valid IP addresses must be assigned in the same subnet. One address is used for the IP phones to register and pass signaling information. The remaining two IP addresses are used for the voice traffic to and from the 32 DSP channels.

✎   When assigning the IP addresses to the IPLA card, they must be in the same network (subnet). If the CPU will be connected to the network it requires a separate IP address in a different network (subnet). When an IPLA 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 NIC. For example, PC PRO, Web Pro, and ACD all terminate to the IPLA NIC card when installed. Both the IPLA and CPU NIC share the same gateway assignment. The default gateway command in Program 10-12-03 is used by both NICs, allowing only one device, IPLA or CPU, to route outside of its own network.

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

Card	Minimum IP Addresses	Maximum IP Addresses	Notes
32 IPLA	2	3	Remember that one IP address is required per 16 DSP channels. One additional IP Address is needed for registration and signaling.
64 IPLA	2	5	
128 IPLA	2	9	

To calculate the maximum DSPs required:

The manual calculations listed below are used in the UNIVERGE SV8100.

Non-Peer-to-Peer Mode (Program10-26-01=0):

It is easy to calculate the maximum number of DSPs for a system that is not peer-to-peer. This is a simple addition of:

```
VoIP extensions (VoIPE) + VoIP trunks (VoIPT)
```

Combine the resource figures:

Combine the (extension resource figure x DSPs required for extensions) + (trunk resource figure x DSPs required for trunks) equals the total card resource required.

```
nTotalCardResourceRequired = (nExtCardResourceFactor x
nDSPsForExt)+(nTrkCardResourceFactor x nDSPsForTrk) 1
```

SECTION 7   QUALITY OF SERVICE (QOS) IMPLEMENTATION

Section 2.2 Voice Quality Improvements on page 5-3 discusses some of the problems associated with voice quality. This section describes how QoS can be implemented on data networks to provide the "best case" for VoIP traffic.

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

Quality of Service is commonly used to describe the actual implementation of prioritization on network hardware. This prioritization (at Layer 2 and Layer 3 of the OSI model) is described in Figure 5-1 Layer 2 Diagram (802.1Q) on page 5-5.

7.1   Prioritization

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

Consider data communication between the two computers shown in the diagram Figure 5-1 Layer 2 Diagram (802.1Q). The Hosts can transmit data at 100 Mbps. When a packet from Host A, destined for Host B, reaches the router, the available bandwidth is reduced to 256Kbps and the packet flow must be reduced. Figure 5-3 Network Bottleneck Example shows a diagram of this condition.

---

1.   This figure is different only to the number of required DSPs if the CODECS used are the faster ones. All other CODECS are a multiplication factor of 1 thus not effecting the calculation.
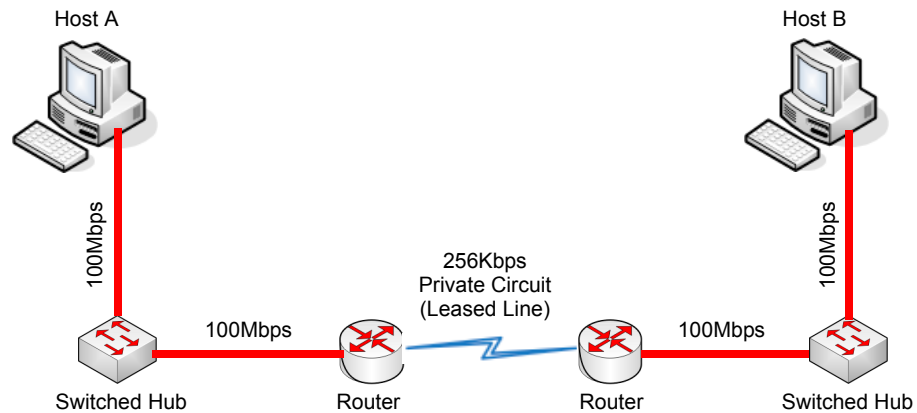
Figure 5-3 Network Bottleneck Example

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

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

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

Figure 5-4 Voice and Data Network Implementation shows how a voice and data network can be implemented.
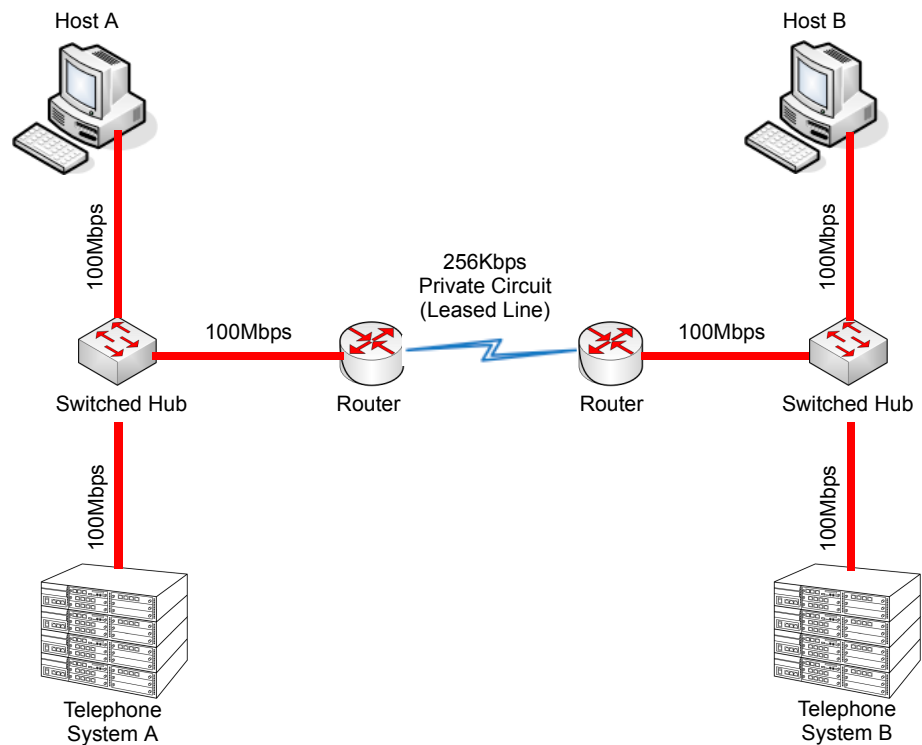
Figure 5-4 Voice and Data Network Implementation

---

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

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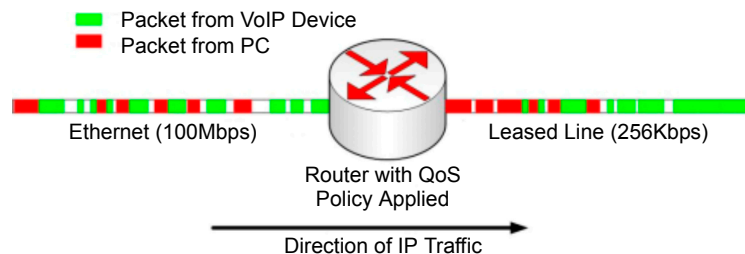
Figure 5-5 Priority Queuing on Voice and Data Networks

To enable this type of queuing it is necessary to:

a   Configure the VoIP equipment to mark its packets with a specific value so that the switches/routers can identify that it is voice – Called **Marking**.

b   Configure the network equipment to recognize the difference between the different Marked packets – Called **Classification.** (i.e., informs the router what a voice packet looks like.

c   Configure the network equipment to give priority to the packets that have been classified as voice – Called **Priority Queuing**.

7.2   Layer 2 QoS (802.1pq)

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

Layer 2 devices work with Ethernet frames (encapsulated IP packets) rather than IP addresses. These devices are usually Switched Hubs (Switches). As the IP header information is encapsulated, the Switched Hubs cannot reference the Type of Service (Layer 3 QoS) field in the IP header to determine the priority of a frame.

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

Protocol Structure - IEEE 802.1p: LAN Layer 2 QoS

Figure 5-6 Protocol Structure for Layer 2 QoS illustrates the format of an Ethernet frame and the User Priority field that is used for Layer 2 QoS.

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

Expanded View of TCI Field		
3 Bits	1 Bit	12 Bits
User Priority	CFI	Bits of VLAN ID (VID) to identify possible VLANs
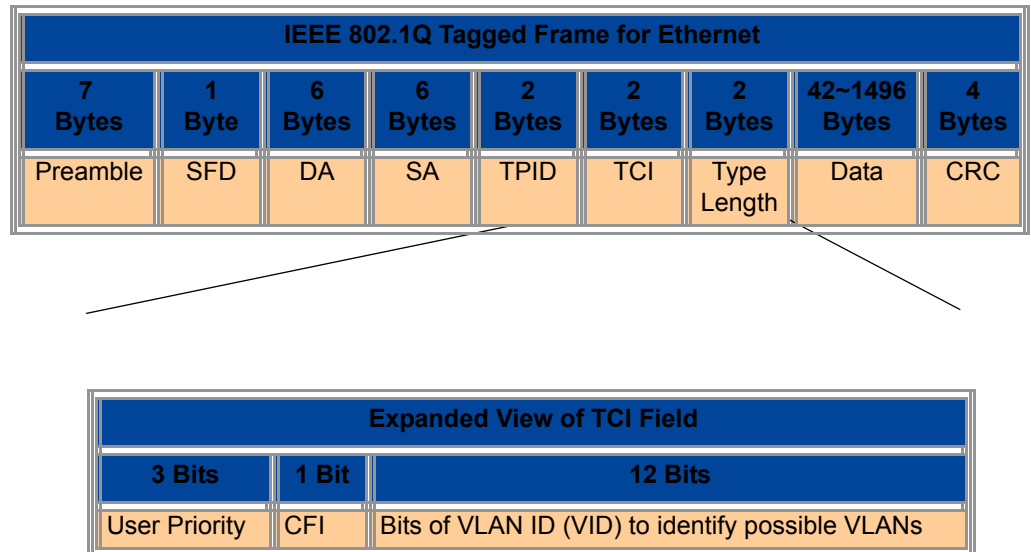
Figure 5-6 Protocol Structure for Layer 2 QoS

The following define the fields used for the protocol structure:

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

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

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

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

Tag Protocol Identifier (TPID) - The defined value of SV8100 in hex. When a frame has the EtherType equal to SV8100, this frame carries the tag IEEE 802.1Q / 802.1P.

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

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

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

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

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

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

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

Example Ethernet Frame with Layer 2 QoS Enabled

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

Ethernet Frame Example - Layer 2 QoS Enabled

```
Source          Destination      Protocol
172.16.0.101    172.16.0.21      RTP
Info
Payload type=ITU-T G.729, SSRC=701655963, Seq=28165, Time=21520

Frame 160 (78 bytes on wire, 78 bytes captured)
    Arrival Time: Jan 18, 2005 13:55:44.842738000
    Time delta from previous packet: 0.008241000 seconds
    Time since reference or first frame: 2.910072000 seconds
    Frame Number: 160
    Packet Length: 78 bytes
    Capture Length: 78 bytes
Ethernet II, Src: 00:60:b9:c6:6e:45, Dst: 00:60:b9:c1:ab:a3
    Destination: 00:60:b9:c1:ab:a3 (Nitsuko_c1:ab:a3)
    Source: 00:60:b9:c6:6e:45 (Nitsuko_c6:6e:45)
    Type: 802.1Q Virtual LAN (0xSV8100)
```

Ethernet Frame Example - Layer 2 QoS Enabled (Continued)

```
802.1q Virtual LAN
    101. .... .... .... = Priority: 5        (Layer 2 Priority = 5)
    ...0 .... .... .... = CFI: 0
    .... 0000 0110 0011 = ID: 99
    Type: IP (0x0800)
Internet Protocol, Src Addr: 172.16.0.101 (172.16.0.101), Dst Addr:
172.16.0.21 (172.16.0.21)
    Version: 4
    Header length: 20 bytes
    Differentiated Services Field: 0x00 (DSCP 0x00: Default; ECN: 0x00)
        0000 00.. = Differentiated Services Codepoint: Default
(0x00)
        .... ..0. = ECN-Capable Transport (ECT): 0
        .... ...0 = ECN-CE: 0
    Total Length: 60
    Identification: 0x0086 (134)
    Flags: 0x00
        0... = Reserved bit: Not set
        .0.. = Don't fragment: Not set
        ..0. = More fragments: Not set
    Fragment offset: 0
    Time to live: 30
    Protocol: UDP (0x11)
    Header checksum: 0x4391 (correct)
    Source: 172.16.0.101 (172.16.0.101)
    Destination: 172.16.0.21 (172.16.0.21)
User Datagram Protocol, Src Port: 10022 (10022), Dst Port: 10020
(10020)
    Source port: 10022 (10022)
    Destination port: 10020 (10020)
    Length: 40
    Checksum: 0x0581 (correct)
Real-Time Transport Protocol
    Stream setup by SDP (frame 1)
        Setup frame: 1
        Setup Method: SDP
    10.. .... = Version: RFC 1889 Version (2)
    ..0. .... = Padding: False
    ...0 .... = Extension: False
    .... 0000 = Contributing source identifiers count: 0
    0... .... = Marker: False
    .001 0010 = Payload type: ITU-T G.729 (18)
    Sequence number: 28165
    Timestamp: 21520
    Synchronization Source identifier: 701655963
    Payload: 76AC9D7AB6ACE2510B3A3338646DA738...
```

7.3   Layer 3 QoS

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

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

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

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

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

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

Type of Service Field (Diffserv)

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

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

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

Figure 5-7 Layer 3 QoS Example

Listed below are the fields used in Figure 5-7 Layer 3 QoS Example.

Version – the version of IP currently used.

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

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

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

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

Flags – Consists of a 3-bit field of which the two low-order (least-significant) bits control fragmentation. The low-order bit specifies whether the packet can be fragmented. The middle bit specifies whether the packet is the last fragment in a series of fragmented packets. The third or high-order bit is not used.

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

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

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

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

Source Address – Specifies the sending node.

Destination Address – Specifies the receiving node.

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

Data – Contains upper-layer information.

7.4   IP Precedence

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

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

Table 5-1 Type of Service Field (IP Precedence - i Ref. REC 1349)

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

IP Precedence Value

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

Throughput

Value	Description
0	Normal Throughput
1	High Throughput

Reliability

Value	Description
0	Normal Reliability
1	High Reliability

Delay

Value	Description
0	Normal Delay
1	Low Delay

Cost

Value	Description

7.5   Diffserv (Differentiated Service)

Differentiated Services (Diffserv) uses the ToS field in an IP header. Diffserv is now commonly used instead of IP Precedence (refer to 7.4 IP Precedence on page 5-24) as it provides greater flexibility. This method uses six bits of the ToS field to determine the priority – which provides up to 64 possible values. The combination of binary digits is known as the Diffserv Codepoint (DSCP).

Table 5-2 Diffserv Parameters

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

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

Ethernet Frame Example - Containing One RTP (Speech) Packet

```
Source           Destination        Protocol
172.16.0.21      172.16.0.101       RTP
Info
Payload type=ITU-T G.729, SSRC=732771006, Seq=30885, Time=20560

Frame 159 (65 bytes on wire, 65 bytes captured)
    Arrival Time: Jan 18, 2005 13:55:44.834497000
    Time delta from previous packet: 0.000445000 seconds
    Time since reference or first frame: 2.901831000 seconds
    Frame Number: 159
    Packet Length: 65 bytes
    Capture Length: 65 bytes
Ethernet II, Src: 00:60:b9:c1:ab:a3, Dst: 00:60:b9:c6:6e:45
    Destination: 00:60:b9:c6:6e:45 (Nitsuko_c6:6e:45)
    Source: 00:60:b9:c1:ab:a3 (Nitsuko_c1:ab:a3)
    Type: IP (0x0800)
Internet Protocol, Src Addr: 172.16.0.21 (172.16.0.21), Dst Addr:
172.16.0.101 (172.16.0.101)
    Version: 4
    Header length: 20 bytes
    Diff Services Field: 0xa0 (DSCP 0x28: Class Selector 5; ECN: 0x00)
        1010 00.. = Diff Services Codepoint: Class Selector 5 (0x28)
        .... ..0. = ECN-Capable Transport (ECT): 0
        .... ...0 = ECN-CE: 0
```

Ethernet Frame Example - Containing one RTP (Speech) Packet (Continued)

```
   Total Length: 44
     Identification: 0x0069 (105)
     Flags: 0x00
         0... = Reserved bit: Not set
         .0.. = Don't fragment: Not set
         ..0. = More fragments: Not set
     Fragment offset: 0
     Time to live: 30
     Protocol: UDP (0x11)
     Header checksum: 0x431e (correct)
     Source: 172.16.0.21 (172.16.0.21)
     Destination: 172.16.0.101 (172.16.0.101)
User Datagram Protocol, Src Port: 10020 (10020), Dst Port: 10022
(10022)
     Source port: 10020 (10020)
     Destination port: 10022 (10022)
     Length: 24
     Checksum: 0x5293 (correct)
Real-Time Transport Protocol
     Stream setup by SDP (frame 112)
         Setup frame: 112
         Setup Method: SDP
     10.. .... = Version: RFC 1889 Version (2)
     ..1. .... = Padding: True
     ...0 .... = Extension: False
     .... 0000 = Contributing source identifiers count: 0
     0... .... = Marker: False
     .001 0010 = Payload type: ITU-T G.729 (18)
     Sequence number: 30885
     Timestamp: 20560
     Synchronization Source identifier: 732771006
     Payload: 3ED0
     Padding data: 00
     Padding count: 2
```

7.6   Comparison of IP Precedence and Diffserv Values

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

For example, if the VoIP equipment supports IP Precedence and the router can prioritize only using the DSCP they can be set to the same value. Refer to Table 5-3 IP Precedence and Diffserv Values Comparison for the values.

Table 5-3 IP Precedence and Diffserv Values Comparison

DSCP Decimal	DSCP Binary	IP Precedence	Description
0	000000	0	Class Selector 0
1	000001		
2	000010		
3	000011		
4	000100		
5	000101		
6	000110		
7	000111		
8	001000	1	Class Selector 1
9	001001		
10	001010		AF11 (Assured Forwarding)
11	001011		
12	001100		AF12 (Assured Forwarding)
13	001101		
14	001110		AF13 (Assured Forwarding)
15	001111		
16	010000	2	Class Selector 2
17	010001		
18	010010		AF21 (Assured Forwarding)
19	010011		
20	010100		AF22 (Assured Forwarding)
21	010101		
22	010110		AF23 (Assured Forwarding)
23	010111		
24	011000	3	Class Selector 3
25	011001		
26	011010		AF31 (Assured Forwarding)
27	011011		

Table 5-3 IP Precedence and Diffserv Values Comparison (Continued)

DSCP Decimal	DSCP Binary	IP Precedence	Description
28	011100		AF32 (Assured Forwarding)
29	011101		
30	011110		AF33 (Assured Forwarding)
31	011111		
32	100000	4	Class Selector 4
33	100001		
34	100010		AF41 (Assured Forwarding)
35	100011		
36	100100		AF42 (Assured Forwarding)
37	100101		
38	100110		AF43 (Assured Forwarding)
39	100111		
40	101000	5	Class Selector 5
41	101001		
42	101010		
43	101011		
44	101100		
45	101101		
46	101110		EF (Expedited Forwarding)
47	101111		
48	110000	6	Class Selector 6
49	110001		
50	110010		
51	110011		
52	110100		
53	110101		
54	110110		
55	110111		

Table 5-3 IP Precedence and Diffserv Values Comparison (Continued)

DSCP Decimal	DSCP Binary	IP Precedence	Description
56	111000	7	Class Selector 7
57	111001		
58	111010		
59	111011		
60	111100		
61	111101		
62	111110		
63	111111		

7.7   Programming QoS in the UNIVERGE SV8100 System

7.7.1   Marking Voice Traffic - Program 84-10-XX

Before programming the UNIVERGE SV8100 system, discuss the requirements with the network engineering staff or the managed network provider. If the ToS markings that are used are not specifically configured into the network equipment, the voice traffic is handled by the default queue and is given lowest priority.

7.7.2   UNIVERGE SV8100 Voice Protocols

The UNIVERGE SV8100 system supports the following types of VoIP traffic (refer to the Input Data section of Program 84-10 : ToS Setup on page 5-33).

Number	Protocol Type	Description
1	DRS	Not Used
2	Protims	Not Used
3	Voice Control (H.245)	Communication from PZ-(X)IPLA to PZ-(X)IPLA
4	H.323	Communication from PZ-(X)IPLA to PZ-(X)IPLA (TIE Lines) Or Communication from PZ-(X)IPLA to Network
5	RTP/RTCP	Voice (RTP) and Call Quality Data (RTCP)
6	SIP	Communication from PZ-(X)IPLA to Standard SIP device.

Number	Protocol Type	Description
7	CCISoIP	CCIS Signaling Messages
8	SIP MLT	Communication from PZ-(X)IPLA to SIP Multiline Terminal.
9	SIP Trunks	Communication from PZ-(X)IPLA to Network
10	NetLink	Communication from PZ-(X)IPLA to PZ-(X)IPLA

7.7.3   Configuring Diffserv

Use Program 84-10-10 to select the logic for marking the ToS field (refer to Program 84-10 : ToS Setup on page 5-33). The choices are:

Number	ToS Mode	Programs Enabled
0	None	None – ToS bits are: 00000000
1	IP Precedence	84-10-02 Priority – 0=Lowest ~ 7=Highest (ToS bits: 0~2) 84-10-03 Delay – 0=Normal, 1=Low (Tos Bit: 3) 84-10-04 Throughput – 0=Normal, 1=High (Tos Bit: 4) 84-10-05 Reliability – 0=Normal, 1=Low (ToS Bit: 5) 84-10-06 Cost – 0=Normal, 1=Low (ToS Bit: 6) ToS Bit 7: Always 0 ✎ Typically, only one of bits 3~6 is set to 1 and the other three bits are set to 0. For example, to maximize route reliability, set 84-10-05 to 1 and leave 84-10-03, 84-10-04 and 84-10-06 at 0 (default).
2	Diffserv	84-10-07 DSCP Value in Decimals: 0~63 (ToS bits: 0~5) ToS Bits 6 & 7 are not evaluated

7.7.4   Configuration Examples for Classification and Queuing

Figure 5-8 Common Network with Cisco Router shows a typical network scenario and an example of a Cisco router configuration.

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

NEC does not endorse or provide support on any third party equipment unless it is supplied by NEC.
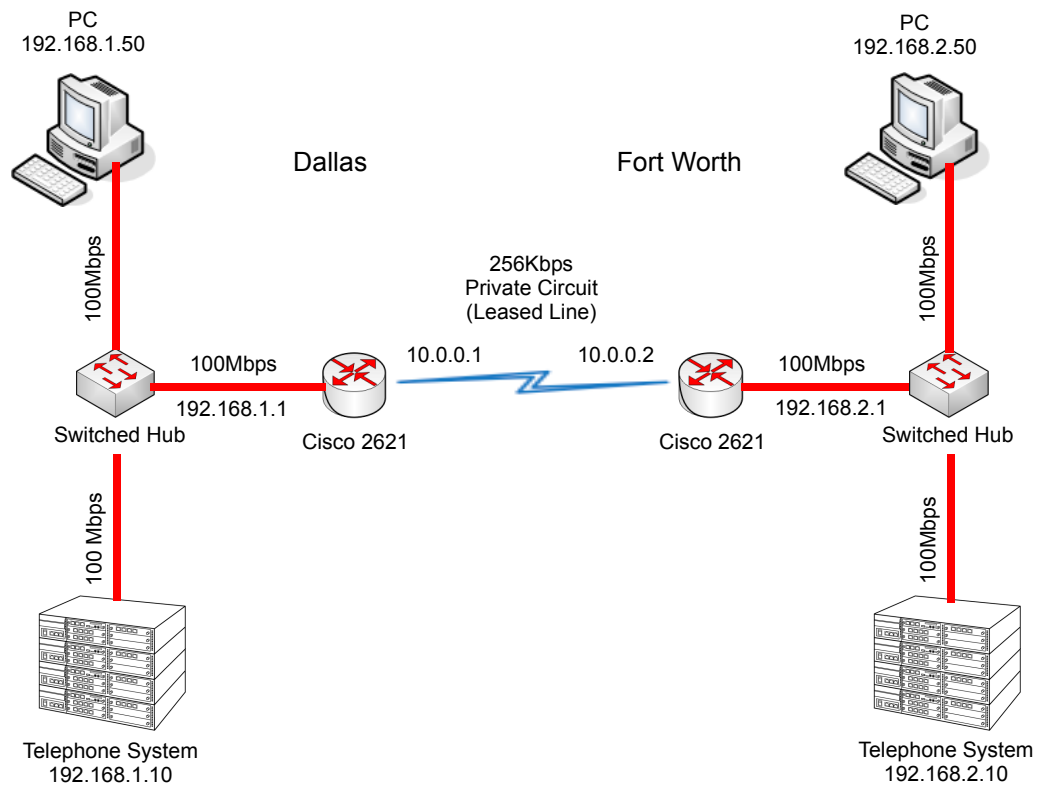
Figure 5-8 Common Network with Cisco Router

See Table 5-4 Cisco Router Configuration Example for configuration information about the Cisco 2621 router. A description of key commands follows.

Table 5-4 Cisco Router Configuration Example

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

Configuration Example Explanation:

1.   Defines a Class Map called VoIPClass.

2.   Matches any packets that have the ToS field set to IP Precedence 5 / DSCP 40 and assigns them to VoIPClass.

3.   Defines a Policy Map called VoIPPolicy.

4.   Creates a Class called VoIPClass and assigns this to the VoIPPolicy.

5.   Allocates 50Kbps of bandwidth to the VoIPClass.

6 & 7.Determines that any data that does not match VoIPClass should be processed using the "fair-queue" method (i.e., No Prioritization).

8.   Determines the amount of bandwidth available on the Serial interface – essential for the QoS calculations.

9.   Applies the VoIP Policy to any packets that exit the serial interface. This means that data being received (input) does not use this policy.

Program 84 : Hardware Setup for VoIP
84-10 : ToS Setup

Level:
IN

Description

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

Input Data

Protocol Type	1 = Not Used
	2 = Not Used
	3 = Voice Control
	4 = H.323
	5 = RTP/RTCP
	6 = SIP
	7 = CCISoIP
	8 = DT700 MLT
	9 = SIP Trunk
	10 = NetLink

Item No.	Item	Input Data	Default	Description
01	**ToS Mode**	0 = Disable (Invalid) 1 = IP Precedence 2 = Diffserv	0	When Input Data is set to 1, Item No. 07 is invalid. When Data is set to 2, Item No. 02 ~ 06 are invalid.
02	**Priority, IP Precedence**	0~7 0 = Low 7 = High	0	1 = Router queuing priority
03	**Low Delay**	0~1 0 = Normal Delay, Low Delay	0	1 = Optimize for low delay routing
04	**Wideband (Throughout)**	0~1 0 = Normal Throughput 1 = High Throughput	0	1 = Optimize for high bandwidth routing

Item No.	Item	Input Data	Default	Description
05	**High Reliability**	0~1 0 = Normal Reliability 1 = Low Reliability	0	1 = Optimize for reliability routing
07	**Priority (D.S.C.P. - Differentiated Services Code Point)**	0~63	0	DSCP (Differentiated Services Code Point)

Conditions

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

Feature Cross Reference

❑ Voice Over Internet Protocol (VoIP)

SECTION 8   PORT DESIGNATIONS

IP Application	IP Port Numbers	Comments
SIP Trunk		
SIP Trunk Signaling	5060	UDP
SIP Trunk Voice w/32 IPLA	10020~10083	UDP
SIP Trunk Voice w/64 IPLA	10020~10147	UDP
SIP Trunk Voice w/128 IPLA	10020~10275	UDP
3rd Party SIP		
SIP SLT Signaling	5070	UDP
SIP SLT Voice w/32 IPLA	10020~10083	UDP
SIP SLT Voice w/64 IPLA	10020~10147	UDP
SIP SLT Voice w/128 IPLA	10020~10275	UDP
H.323 Trunk		
H.323 Signaling	1718/1719/1720	TCP
H.323 Trunk Voice w/32 IPLA	10020~10083	UDP
H.323 Trunk Voice w/64 IPLA	10020~10147	UDP
H.323 Trunk Voice w/128 IPLA	10020~10275	UDP
SV8100 IP - K-CCIS		
IP K-CCIS Signaling	57000	TCP
IP Trunk Voice w/32 IPLA	10020~10083	UDP
IP Trunk Voice w/64 IPLA	10020~10147	UDP
IP Trunk Voice w/128 IPLA	10020~10275	UDP

IP Application	IP Port Numbers	Comments
SV8100 NetLink		
Primary System Signaling	58000/58001	TCP
Secondary System Signaling	58002	TCP
IP DSP Resource w/32 IPLA	10020~10083	UDP
IP DSP Resource w/64 IPLA	10020~10147	UDP
IP DSP Resource w/128 IPLA	10020~10275	UDP
NEC Proprietary SIP (SIP MLT)		
SIP MLT Signaling	5080	UDP
SIP Trunk Voice w/32 IPLA	10020~10083	UDP
SIP Trunk Voice w/64 IPLA	10020~10147	UDP
SIP Trunk Voice w/128 IPLA	10020~10275	UDP

SIP Trunking

SECTION 1   VoIP

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

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

SECTION 2   IP NETWORKING

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

The following Networking modes are available on the UNIVERGE SV8100 system:

❍   CCISoIP

❍   SIP Trunks

---

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

SECTION 3   SIP TRUNKING

3.1   INTRODUCTION

SIP (Session Initiation Protocol) is used for Voice over IP. It is defined by the IETF (Internet Engineering Task Force) in RFC2543 and RFC3261. SIP trunking is the term used for linking a PBX, such as the UNIVERGE SV8100 system, to the public telephone network by means of VoIP. This provides the ability for users to place and receive communications and services from any location and for networks to identify the users wherever they are located.

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

The UNIVERGE SV8100 system implementation and programming for SIP are very similar. The call routing, call features and speech handling (RTP) are the same. Only the signaling protocol is different.

With the UNIVERGE SV8100 system, SIP trunks can receive incoming calls with Caller ID, place outgoing calls, and transfer SIP trunks to IP, SIP, analog and digital stations, and across a network.

Currently, however, SIP Centrex Transfer and E.164 Numbering Plan are not supported.

If a common carrier supports SIP, the UNIVERGE SV8100 can connect the SIP Carrier and outgoing calls to the PSTN (Public Switched Telephone Network) network and the common IP network using an UNIVERGE SV8100 SIP trunk.
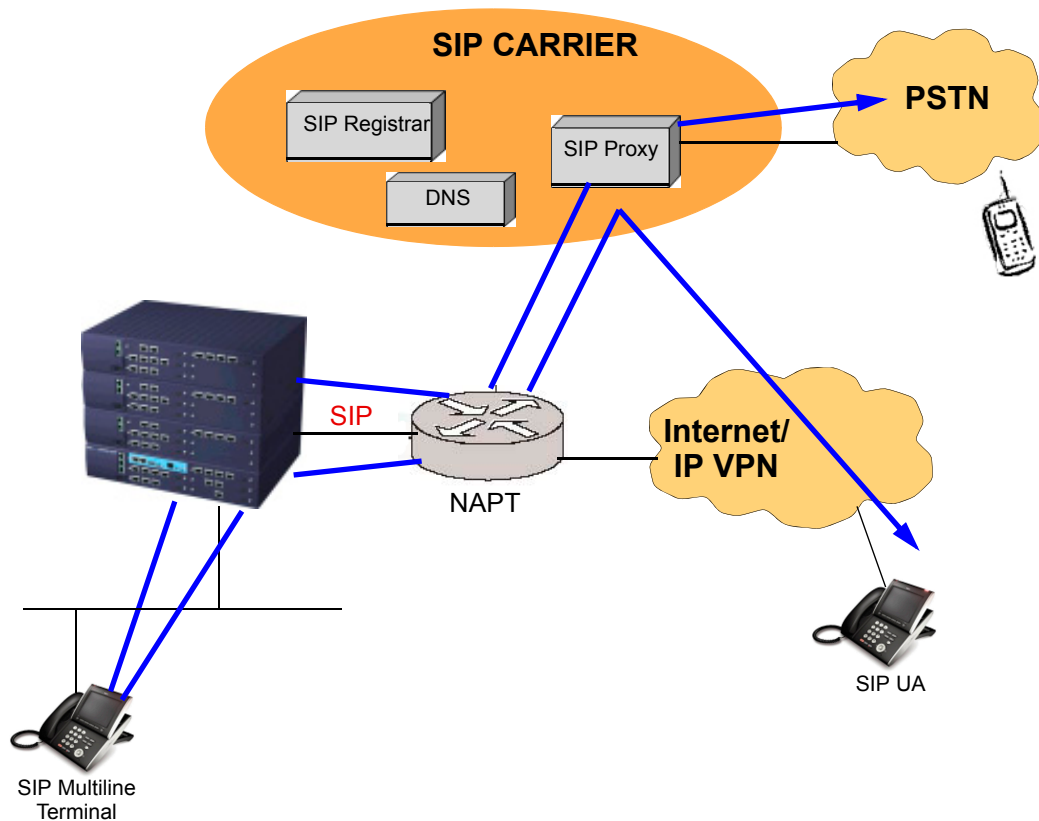
Figure 6-1 Common IP Network using UNIVERGE SV8100 SIP Trunk

3.1.1   SIP Trunking Requirements

The following are required when using the SIP trunk on the UNIVERGE SV8100 system:

❐   CD-CP00-US software

❐   PZ-32IPLA/PZ-64IPLA/PZ-128IPLA (VoIP Daughter Board) Programming Conditions

The following conditions apply when programming the UNIVERGE SV8100 system for SIP Trunking:

❐   If entries are made in Program 84-20-xx for a SIP Server and the SIP Server is then removed or not used, the entries in Program 84-20-xx must be set back to their default settings. Even if **Program 84-20-01 : SIP Proxy Setup – Outbound Proxy** is set to *0*, the UNIVERGE SV8100 system checks the settings in the remaining 84-20 programs.

❐   The UNIVERGE SV8100 system does not support the simultaneous use of a SIP trunk interconnection and a SIP trunk carrier connection.

❐   The UNIVERGE SV8100 system restricts an outgoing call under the following conditions:

  ❍   SIP configuration failed

  ❍   SIP registration failed

  ❍   CD-CP00-US / PZ-32IPLA/PZ-64IPLA/PZ-128IPLA daughter board link down

  ❍   Lack of PZ-32IPLA/PZ-64IPLA/PZ-128IPLA DSP resource

  ❍   Lack of bandwidth

❐   UNIVERGE SV8100 system SIP does not support T.38 FAX.

3.1.2   SIP Trunking Setup

Use the following steps to initially set up SIP Trunking for the UNIVERGE SV8100 system:

1.   By default, the UNIVERGE SV8100 is assigned a static IP address and runs behind a NAT router.

✎   *When using an SV8100 on a LAN behind an NAPT router, forward port 5060 to the IP address of the UNIVERGE SV8100 CD-CP00-US (commonly called the SV8100 CPU) since the signaling is handled by the CD-CP00-US. Then, since the media stream (the speech) uses a large range of ports for the RTP packages, forward the ports (10020~10083) to the IP address of the PZ-32IPLA/PZ-64IPLA/ PZ-128IPLA. Or, use the DMZ option for the PZ-32IPLA/PZ-64IPLA/ PZ-128IPLA. This means that the PZ-32IPLA/PZ-64IPLA/PZ-128IPLA is not actually behind the firewall. This is achieved by connecting the PZ-32IPLA/PZ-64IPLA/PZ-128IPLA to a physical or virtual DMZ port. You can also achieve the same result by port forwarding 10020 to 10083.*

2.   Define the SIP Carrier account information (user name, password, domain name/IP address to the provider).

3.   Define the trunk ports as SIP.

4.   Set the Expire Time.

SECTION 4   SIP TRUNK OVERVIEW

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

4.1   GENERAL INFORMATION

The UNIVERGE SV8100 SIP Trunks behave like some SIP User Agents (UA). UA is the client application used with a particular network protocol.

The UNIVERGE SV8100 provides a maximum of 32 register IDs and can register these IDs with any SIP server.

A maximum number of 200 SIP trunks can be used with the UNIVERGE SV8100 system, with a maximum number of 200 simultaneous calls.

The UNIVERGE SV8100 can connect any SIP server over a NAPT router by one static global IP address. The UNIVERGE SV8100 system supports a DNS resolution access and a IP address direct access for SIP servers and supports the sub-address feature with SIP trunk interconnection.

4.2   Supported Protocols

The following protocols are supported:

❍   SIP (RFC2543 bis04)

❍   SIP RFC3261 Supported [updated version of RFC2543]

❍   The SIP stack has been updated from RFC2543 Base to RFC3261 Base

❍   SDP

❍   RTP/RTCP

❍   UDP

❍   IPv4

4.3   Supported SIP Methods

The following SIP methods are supported with the UNIVERGE SV8100 system:

❍   REGISTER

❍   INVITE

❍   BYE

❍   CANCEL

❍   ACK

❍   PRACK

❍   Response 1xx 2xx, 3xx, 4xx, 5xx, 6xx

The following features are available:

❍   ***Support the 401 response for the Initial Invite***
If 401 message is sent for the Initial Invite, with previous software, the system cannot respond to the message correctly.

❍   ***Support the 401/407 response for the Invite of Session Timer***
If 401/407 response is sent for the invite of Session Timer, the system can send the Invite message with Authentication header.

❍   ***Support the 128 byte size of Nonce max value sent by the 401/407 message***
The Nonce maximum size sent by the 401/407 message is expanded to 128 byte. With previous software, 64 byte size can be received.

4.4   Supported SIP Trunking Options

The options listed below are supported with SIP Trunking.

❍   100rel

❍   Session Timer

❍   Early Media

4.5   Supported CODEC

SIP Trunking can use the following CODECs.

❍   G.729. Low bandwidth requirement, and is used on most Wide Area
     Network links

    ❑   VAD

    ❑   VIF size 20ms~30ms

❍   G.711 – High bandwidth requirement – usually used on Local Area
     Networks.

❍   G.722 – This CODEC is useful in fixed network, Voice over IP
     applications, where the required bandwidth is typically not prohibitive.

❍   G.726 – is an ITU-T ADPCM speech coded standard covering the
     transmission of voice at rates of 16, 24, 32, and 40Kbps.

❍   ILBC (Internet Low Bitrate CODEC) – Narrow band speech. ILBC
     enables graceful speech quality degradation for lost frames, which
     occurs in connection with lost or delayed IP packets

SECTION 5   SUPPORTED SIP TRUNKING FUNCTIONS

The functions in this section are supported by UNIVERGE SV8100 SIP Trunking.

5.1   ADDRESS RESOLUTION

If using any SIP Proxy Server, the UNIVERGE SV8100 always sends
messages to an external SIP server.

If not using any SIP Proxy Server, the UNIVERGE SV8100 uses the internal
address table (Program 10-23 : SIP System Interconnection Setup).

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

5.2   Authentication Process

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

UNIVERGE SV8100 SIP trunks support HTTP digest authentication process
(MD5). This process is done on a Register process and Initial INVITE
process.

5.3 Caller ID

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

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

Programs follow program priority as follows: 21-19 > 21-17 > 10-28-04

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

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

5.4 Carrier Support

If a common carrier supports SIP, the UNIVERGE SV8100 can connect:

❍   the SIP Carrier

❍   the outgoing call to the PSTN network

❍   the common IP network via an UNIVERGE SV8100 SIP trunk.

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

5.5 Early Media

When the UNIVERGE SV8100 receives the response *18x w/SDP* and the CODEC negotiation is a success, the UNIVERGE SV8100 starts to send/receive RTP packets.

5.6 Fault Tolerance

When the UNIVERGE SV8100 uses an external SIP Server and if the registration process fails, the UNIVERGE SV8100 blocks outgoing calls with SIP trunks. All SIP trunk ports are placed into a busy status.

If the UNIVERGE SV8100 has trunk groups that include both SIP trunks and ISDN trunks, and all SIP trunks are busy, a user can make an outgoing call using an ISDN trunk as a bypass.

5.7   Network Address Port Translation (NAPT)

UNIVERGE SV8100 SIP trunk can pass through a NAPT router. The related system data is **Program 10-12-06 :** CD-CP00-US **Network Setup** – **NAPT Router (On/Off)** and **Program 10-12-07 :** CD-CP00-US **Network Setup** – **NAPT Router IP Address** (do not set with a dynamic router IP address).

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

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

5.8   Quality of Service (QoS)

UNIVERGE SV8100 SIP trunks support Type of Service (ToS) (**Program 84-10 : ToS Setup**, protocol type = 6 (SIP Trunk).

5.9   Registration

5.9.1   Registration Process

When the UNIVERGE SV8100 system registers its own IDs with an external SIP server, the following system data are sent:

Register ID #	User ID	Authentication ID	Authentication Password
Register ID 0	10-28-04	10-30-02	10-30-03
Register ID 1	10-36-02	10-36-03	10-36-04
⋮	10-36-02	10-36-03	10-36-04
Register ID 31	10-36-02	10-36-03	10-36-04

The UNIVERGE SV8100 sends the REGISTER Message when the system starts up, register timer expires, CCPU LAN links and recover timer expires.

5.9.2   Registration Recover Process

The UNIVERGE SV8100 has a registration recovery process for registration failure. When a registration fails, the UNIVERGE SV8100 sets an internal recover timer. When the timer expires, the UNIVERGE SV8100 sends a REGISTER message per register ID again.

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

SECTION 6   SIP TRUNK PROGRAMMING

6.1   SIP TRUNK BASIC SETUP

Program/ Item No.	Description/ Selection	Assigned Data	Comments
10-28-01	SIP System Information Setup – Domain Name	64 characters maximum Default not assigned	Define the domain name. This information is generally provided by the SIP carrier.
10-28-02	SIP System Information Setup – Host Name	48 characters maximum Default not assigned	Define the Host name. This information is generally provided by the SIP carrier.
10-28-03	SIP System Information Setup – Transport Protocol	0 = UDP 1 = TCP Default = 0	Define the Transport type. This option will always be set to UDP.
10-28-04	SIP System Information Setup – User ID	32 characters maximum Default not assigned	Define the User ID. ✎ *Program 10-23-04 for system interconnection. This entry must be numeric as 10-23-04 does not allow text entry - only numeric. This information is generally provided by the SIP carrier.*
10-28-05	SIP System Information Setup – Domain Assignment	0 = IP Address 1 = Domain Name Default = 0	Define the Domain Assignment. This entry is determined by what information the SIP carrier provides. If the SIP carrier provides a server name: **SIPconnect-sca@L0.cbeyond.net** the domain would be: **@L0.cbeyond.net** and the host name would be: **SIPconnect-sca**.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
10-29-14	SIP Server Information Setup – SIP Carrier Choice	0 = Standard 1 = Carrier A 2 = Carrier B 3 = Carrier C 4 = Carrier D 5 = Carrier E 6 = Carrier F 7 = Carrier G Default = 0	Define the SIP Carrier Choice. This should currently stay at an entry of *0* (default).
10-40-01	IP Trunk Availability – IP Trunk Availability	0 = Disable 1 = Enable Default = 0	Determine SIP Trunk availability.
10-40-02	IP Trunk Availability – Number of Ports	0~128 Default = 0	Determine the IP trunk type setup. With SIP trunking, some ports must be defined for SIP.
22-02-01	Incoming Call Trunk Setup	0 = Normal 1 = VRS (second dial tone if no VRS installed) 2 = DISA 3 = DID 4 = DIL 5 = E&M Tie line 6 = Delayed VRS 7 = ANI/DNIS 8 = DID(DDI) Mode Switching Default = 0	Define the SIP trunks as type 3 (DID). In addition to the SIP trunk programming, refer to the Direct Inward Dialing (DID) feature in the UNIVERGE SV8100 Features and Specifications Manuals and the UNIVERGE SV8100 Programming Manual for detailed information.

6.2   IP DSP Resource

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

Program/Item No.	Description/ Selection	Assigned Data	Comments
10-03-01	ETU Setup – Logical Port Number	0 ~ 200	The CD-CP00-US automatically assigns each blade during installation. **READ ONLY**
10-03-02	ETU Setup – Trunk Type	0 = H.323 1 = SIP Default = 1	Assign each trunk type for each DSP Resource.
10-19-01	VoIP DSP Resource Selection – VoIP DSP Resource Selection	0 = Common (use for both IP extensions and trunks) 1 = IP Extension 2 = SIP Trunk 3 = CCIS 4 = Use for NetLink 5 = Blocked Default: Resource 1 = 1 Resources 2~128 = 0	Define the criteria for each DSP resource on the VoIP blade.
14-05-01	Trunk Group	Trunk Group = 0~100 Priority = 1 ~ 200 Default = Group1 for all trunk ports and priorities	Default priorities for trunks 1~200 is 1~200. Assign Trunks to Trunk Groups/ Outbound Priority
22-02-01	Incoming Call Trunk Setup	0 = Normal 1 = VRS (second dial tone if no VRS installed) 2 = DISA 3 = DID 4 = DIL 5 = E&M Tie line 6 = Delayed VRS 7 = ANI/DNIS 8 = DID(DDI) Mode Switching Default = 0	Set the feature type for the trunk you are programming. (Second dial tone for option 1 if no VRS is installed)

Program/Item No.	Description/ Selection	Assigned Data	Comments
84-26-01	IPL Basic Setup – IP Address	XXX.XXX.XXX.XXX	Assign the PZ-32IPLA/ PZ-64IPLA/PZ-128IPLA Gateway IP Address. Slot 1 = **:** Slot 4 = 172.16.0.44 VoIP GW Number 1~8: 172.16.0.20~172.16.0.27
84-26-02	IPL Basic Setup – RTP Port Number	0 ~ 65534	Default: VoIP GW1 = 10020 VoIP GW2 = 10052 VoIP GW3 = 10084 VoIP GW4 = 10116 VoIP GW5 = 10148 VoIP GW6 = 10180 VoIP GW7 = 10212 VoIP GW8 = 10244
84-26-03	IPL Basic Setup – RTCP Port Number (RTP Port Number + 1)	0 ~ 65534	Default: VoIP GW1 = 10021 VoIP GW2 = 10053 VoIP GW3 = 10085 VoIP GW4 = 10117 VoIP GW5 = 10149 VoIP GW6 = 10181 VoIP GW7 = 10213 VoIP GW8 = 10245

6.3   SIP Authentication Information

Program/Item No.	Description/ Selection	Assigned Data	Comments
10-30-02	SIP Authentication Information Setup – User Name	64 characters maximum Default not assigned	Define the authentication User name provided by the SIP carrier.

Program/Item No.	Description/ Selection	Assigned Data	Comments
10-30-03	SIP Authentication Information Setup – Password	32 characters maximum Default not assigned	Enter the authentication password provided by the SIP carrier. When the UNIVERGE SV8100 registers its own ID with the carrier SIP server or makes an outgoing call via the carrier SIP server, the SIP server requests the authentication. This data is used as Register ID 0.
10-30-04	SIP Authentication Information Setup – Authentication Trial	0 ~ 9 Default = 1	Define the Authorization Trial. When a call tries to register with the SIP carrier and they refuse, this entry determines how many times the UNIVERGE SV8100 sends authorization.

6.4   SIP Caller ID

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

Program/Item No.	Description/ Selection	Assigned Data	Comments
21-19-01	IP Trunk (SIP) Calling Party Number Setup for Extension	1 ~ 0, *, # (to equal up to 16 digits) Default not assigned	Assign the Calling Party Number for each extension. The assigned number is sent to the central office when the caller places an outgoing call. If the Calling Party Number is assigned by both Program 21-17 and 21-18/21-19, the system uses the data in Program 21-18/21-19.

6.5   SIP CODEC Trunk

Program/Item No.	Description/ Selection	Assigned Data	Comments
84-13-01	SIP Trunk CODEC Information Basic Setup – G.711 Audio Frame Number	1 ~ 4 Default is 2	Set the G.711 Audio Frame Number.
84-13-02	SIP Trunk CODEC Information Basic Setup – G.711 Voice Activity Detection Mode	0 = Disable 1 = Enable Default is 0	Enable/Disable the G.711 VAD Detection Mode.
84-13-03	SIP Trunk CODEC Information Basic Setup – G.711 Type	0 = A-law 1 = µ-law Default is 1	Define the G.711 type.
84-13-04	SIP Trunk CODEC Information Basic Setup - G.711 Jitter Buffer (min)	0 ~ 160ms Default is 20	Set the minimum G.711 Jitter Buffer.
84-13-05	SIP Trunk CODEC Information Basic Setup – G.711 Jitter Buffer (average)	0 ~ 160ms 40ms = Default is 40	Set the average G.711 Jitter Buffer.
84-13-06	SIP Trunk CODEC Information Basic Setup – G.711 Jitter Buffer (max)	0 ~ 160ms Default is 60	Set the maximum G.711 Jitter Buffer.
84-13-07	SIP Trunk CODEC Information Basic Setup – G.729 Audio Frame Number	1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms 5 = 50ms 6 = 60ms Default is 2	Set the G.729 Audio Frame Number.

Program/Item No.	Description/ Selection	Assigned Data	Comments
84-13-08	SIP Trunk CODEC Information Basic Setup – G.729 Voice Activity Detection Mode	0 = Disable 1 = Enable Default is 0	Enable/Disable the G.729 VAD Detection Mode.
84-13-09	SIP Trunk CODEC Information Basic Setup – G.729 Jitter Buffer (min)	0 ~ 270ms Default is 20	Set the minimum G.729 Jitter Buffer.
84-13-10	SIP Trunk CODEC Information Basic Setup – G.729 Jitter Buffer (average)	0 ~ 270ms Default is 40	Set the average G.729 Jitter Buffer.
84-13-11	SIP Trunk CODEC Information Basic Setup – G.729 Jitter Buffer (max)	0 ~ 270ms Default is 80	Set the maximum G.729 Jitter Buffer.
84-13-17	SIP Trunk CODEC Information Basic Setup – Jitter Buffer Mode	1 = Static 2 = Adaptive During Silence 3 = Adaptive Immediately Default is 3	Set the Jitter Buffer Mode.
84-13-18	SIP Trunk CODEC Information Basic Setup – VAD Threshold	Entries 0 ~ 30 (-19dBm~10dBm) 0 = Self adjustment 1 = -19dB (-49dBm) : 20 = 0dB (-30dBm) : 29 = 9dB (-21dBm) 30 =10dB (-20dBm) Default is 20	Set the VAD (voice activity detection) threshold.
84-13-26	SIP Trunk CODEC Information Basic Setup – TX Gain	Entries 0~40 (-20dBm~+20 dBm) 0 = -20dBm 1 = -19dBm : 20 = 0dBm : 39 = 19dBm 40 = 20dBm Default is 20	Set the transmit gain.

Program/Item No.	Description/ Selection	Assigned Data	Comments
84-13-27	SIP Trunk CODEC Information Basic Setup – RX Gain	Entries 0~40 (-20dBm~+20 dBm) 0 = -20dBm 1 = -19 dBm : 20 = 0dBm : 39 = 19dBm 40 = 20dBm Default is 20	Set the receive gain.
84-13-28	SIP Trunk CODEC Information Basic Setup – Audio Capability Priority	0 = G.711_PT 1 = G.723_PT 2 = G.729_PT 3 = G.722_PT 4 = G.726_PT 5 = Not Used Default is 0	Define the CODEC Priority.
84-13-31	SIP Trunk CODEC Information Basic Setup – DTMF Payload Number	96 ~ 127 Default is 110	Define the DTMF Payload Number.
84-13-32	SIP Trunk CODEC Information Basic Setup – DTMF Relay Mode	0 = Disable 1 = RFC2833 Default is 0	Determine the DTMF setup.
84-13-33	SIP Trunk CODEC Information Basic Setup – Number of G.722 Audio Frames	1~4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms Default is 3	
84-13-34	SIP Trunk CODEC Information Basic Setup – G.722 VAD Mode	0 = Disable 1 = Enable Default is 0	
84-13-35	SIP Trunk CODEC Information Basic Setup – G.722 Jitter Buffer (min)	0~160ms Default is 30	

6.6   SIP DNS Setup

Program/Item No.	Description/ Selection	Assigned Data	Comments
10-29-08	SIP Server Information Setup – DNS Server Mode	0 = Off 1 = On Default is 0	Define the DNS Mode. If the SIP carrier provides a domain name, turn this option on.
10-29-09	SIP Server Information Setup – DNS Server IP Address	0.0.0.1 - 126.255.255.254 128.0.0.1 - 191.255.255.254 192.0.0.1 - 223.225.255.254 Default is 0.0.0.0	Define the DNS IP Address (normally provided by the SIP carrier). Enter the carrier-provided information or enter a valid DNS server IP address.
10-29-10	SIP Server Information Setup – DNS Port Number	0 ~ 65535 Default is 53	Define the DNS Transport port.

6.7   SIP NAPT Router Setup

Program/Item No.	Description/ Selection	Assigned Data	Comments
10-12-06	CD-CP00 Network Setup – NAPT Router Setup	0 = Disable 1 = Enable Default is 0	Enable/Disable the NAPT Router Setup. With SIP trunking behind a NAPT router, this must be set to enable (1).
10-12-07	CD-CP00 Network Setup – NAPT Router IP Address	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.254.255.254 192.0.0.1 ~ 223.255.255.254 Default is 0.0.0.0	Set the NAPT Router IP address. With SIP trunking, the IP address of the WAN side of the router must be entered.
10-12-08	CD-CP00 Network Setup – ICMP Redirect	0 = Disable 1 = Enable Default is 0	Enable/Disable ICMP (Internet Control Message Protocol) Redirect.

6.8   SIP System Interconnection Setup

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

6.9   SIP Protocol

Program/Item No.	Description/ Selection	Assigned Data	Comments
84-14-06	SIP Trunk Basic Information Setup – SIP Trunk Port Number	1 ~ 65535 Default is 5060	Set the SIP UA (User Authorized) Trunk port number (Receiving Transport for UNIVERGE SV8100 SIP).
84-14-07	SIP Trunk Basic Information Setup – Session Timer Value	0 ~ 65535 seconds Default is 0	Set the Session Timer Value. 0 means session timer is Off
84-14-08	SIP Trunk Basic Information Setup – Minimum Session Timer Value	0 ~ 65535 seconds Default is 1800	Set the Minimum Session Timer Value.
84-14-09	SIP Trunk Basic Information Setup – Called Party Information	0 = Request URI 1 = To Header Default is 0	Set the Called Party Information.

6.10   SIP Server Information Setup

Program/Item No.	Description/ Selection	Assigned Data	Comments
10-29-01	SIP Server Information Setup – Default Proxy (Outbound)	0 = Off 1 = On Default is 0	Define the SIP Proxy setup, Default Proxy (Outbound). When SIP trunking is used, this must be on. If entries are made in Program 10-29-xx for a SIP Server and the SIP Server is then removed or not used, the entries in Program 10-29-xx must be set back to their default settings. Even if 10-29-01 is set to 0, the UNIVERGE SV8100 still checks the settings in the remaining 10-29 programs.
10-29-02	SIP Server Information Setup – Default Proxy (Inbound)	0 = Off 1 = On Default is 0	Define the Default Proxy (inbound).
10-29-03	SIP Server Information Setup – Default Proxy IP Address	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.225.255.254 Default is 0.0.0.0	Enter the default Proxy IP Address if the SIP carrier is using an IP Address for the Proxy. In most cases, this is left at the default entry as the domain name is used.
10-29-04	SIP Server Information Setup – Default Proxy Port Number	0 ~ 65535 Default is 5060	Define the Proxy Port Number.
10-29-12	SIP Server Information Setup – Proxy Domain Name	64 characters maximum Default not assigned	Define the Proxy Domain Name (UNIVERGE SV8100 domain name).
10-29-13	SIP Server Information Setup – Proxy Host Name	48 characters maximum Default not assigned	Define the Proxy Host name (UNIVERGE SV8100 proxy name).

6.11   SIP Registrar Setup

Program/Item No.	Description/ Selection	Assigned Data	Comments
10-29-05	SIP Server Information Setup – Registrar Mode	0 = None 1 = Manual Default is 0	Define the Registrar Mode. This should always be set to manual when using SIP trunking.
10-29-06	SIP Server Information Setup – Registrar IP Address	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 Default is 0.0.0.0	Define the Registrar IP Address. The carrier may provide an IP address. In most cases, a domain name will be used so this entry will be left at the default.
10-29-07	SIP Server Information Setup – Registrar Port Number	0 ~ 65535 Default is 5060	Define the Registrar Port Numbers.
10-29-11	SIP Server Information Setup – Registrar Domain Name	128 characters maximum Default not assigned	Define the Registrar Domain Name (normally provided by the SIP carrier). Example: **SIPconnect-sca@L0.cbeyond.net**
10-29-15	SIP Server Information Setup – Registration Expiry (Expire) Time	120 ~ 65535 seconds Default is 3600	Define the Registration Expire time – the time allowed to register with the SIP carrier. This should stay at the default entry.

6.12   SIP Server Status

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

6.13   SIP Trunk Registration Information Setup

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

6.14   SIP UPnP

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

SECTION 7   SIP TRUNK FLOWCHART

The SIP Station Flowchart provides the process for setting up SIP trunks.

SECTION 8   SIP PACKET SEQUENCE

The SIP Packet Sequence illustrates the sequence of events when establishing a SIP trunk call.

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SIP Station Flowchart

```
Start
  │
  ▼
┌──────────────────────┐      Go to the following system data and set the correct
│ 1.Set "IP address"   │──────values.
│   of CPU1.           │         10-12-01 CPU IP address
└──────────────────────┘         10-12-02 CPU Sub net mask
  │                              10-12-03 CPU Default G/W IP address
  │                           If you change the value, the system must be
  ▼                           rebooted to reflect this change after exiting
┌──────────────────────┐      program mode.
│ 2.Set "IP address"   │
│   of MG.             │──────Go to the following system data and set the correct
└──────────────────────┘      values.
  │                              84-05-01 Slot X: MG IP address
  │                              84-05-04 Slot X: MG Sub net mask
  ▼                              84-05-05 Slot X: MG Default G/W IP address
┌──────────────────────┐      If you changed this value, MG is rebooted
│ 3. Register SIP      │      automatically when the MG status is idle.
│    licenses on the   │
│    CPU.              │
└──────────────────────┘
  │
  ▼
┌──────────────────────┐
│ SIP Server on CPU is │
│ ready to receive     │
│ REGISTER message     │
│ from SIP phones at   │
│ 5070 port.           │
└──────────────────────┘
  │
  ▼
  ⊗
```

SIP Server on CPU Advanced Setting

```
SIP Server on CPU
Advanced Setting
        │
        ▼
┌───────────────────────┐
│ You want to change    │───Yes──▶  Go to Program 84-20-01.
│ listening port?       │
└───────────────────────┘
        │ No
        ▼
┌───────────────────────┐
│ You want to enable    │───Yes──▶  Go to Program 10-33-02 and enable.
│ "Authentication       │
│ mode" for SIP station?│           Go to Program 15-05-16
└───────────────────────┘           and set password for each SIP station.
        │ No
        ▼
┌───────────────────────┐
│ You want to register  │───Yes──▶  Go to Program 15-05-18
│ several SIP phones    │           and set group# for each SIP station.
│ with the same IP      │
│ address.              │                    ▼
└───────────────────────┘                   ⊗
        │ No
        ▼
```

301@172.16.0.10
302@172.16.0.10

<SIP Phone>

UA1   UA2

If the SIP phone has multiple UAs capability, and each UA (User Agent) has extension number of SV8100, you can put these extensions to the same group to allow using the same IP address.

SIP Station ToS

Go to Program 84-10
Select "SIP".

⊗

SIP Station CODEC

Program 84-19 and 84-20

⊗

Nickname

You want to send "Nickname" to SIP Phone instead of extension number?

Yes →

Go to Program 15-05-04
and
Set "Nickname" for each TDM phone or IP phone.

→

Go to Program 15-05-17
and you can choose the data which you want to send to other SIP phones.
TDM Phone: Name or Extension #
SIP Phones: Name / Display Field / User ID field / Extension #

CPU puts this data on "Display Field" and sends it to destination SIP Phone.

→ ⊗

No

SIP Packet Sequence

SIP Server

UA (User Agent)

Register

401 — If authentication required.

Register + Auth

200 OK

INVITE

401 or 407 — If authentication required.

ACK

INVITE + Auth

100 Trying

180 Ringing

1,180 Ringing
-> Local RBT
2,180 Ringing + SDP
-> Far side in-band RBT
3,183 Session Progress + SDP
-> Far side in-band RBT

PRACK

200 OK — If 100rel option supported.

200 OK — **Answer**

ACK

RTP

BYE

200 OK

H.323 Trunking

SECTION 1   INTRODUCTION

H.323 is an International Telecommunication Union (ITU) standard for Packet Based Multimedia Communication Systems. The UNIVERGE SV8100 can use H.323 to connect to another UNIVERGE SV8100 system or a third party product.

The feature set is limited. When using H.323, it is not possible to use the advanced networking features. If these features are required, use IP K-CCIS.

The UNIVERGE SV8100 Voice over IP Trunk Daughter Board H.323 package sends the real-time voice over the corporate LAN or WAN. The voice from the telephone is digitized and then put into frames to be sent over a network using Internet Protocol.

The UNIVERGE SV8100 Voice over IP Trunk – H.323 daughter board package allows communication using standard H.323 (Normal and Fast Start) Protocol and allows connectivity to any H.323 standards compliant voice gateway and gatekeeper. This VoIP Trunk daughter board also allows Registration and Authentication Server (RAS) support to register with an RAS Server and use Gatekeeper for dynamic call routing.

The PZ-(X)IPLA daughter board – H.323 is an optional interface that can provide IP trunks and Tie Lines. It can operate in the following modes:

❍   COI

❍   DID

❍   TLI

Depending on the requirements and resource allocation in the LAN/WAN/ Internet, the PZ-(X)IPLA – H.323 can be configured to use any of the following voice compressions:

❍   G.729 Low bandwidth requirement used on most Wide Area Network links.

❍   G.711 High bandwidth requirement usually used on Local Area Networks.

❍   G.722 This CODEC is useful in fixed network, Voice over IP applications, where the required bandwidth is typically not prohibitive.

❍   G.723 This CODEC is a ITU-T standard wideband speech CODEC. This is an extension of Recommendation G.721 adaptive differential pulse code modulation to 24 and 40Kbps for digital circuit multiplication equipment application.

SECTION 2   H.323 TRUNK IMPLEMENTATION

The UNIVERGE SV8100 implementation and programming for SIP and H.323 are very similar. The call routing, call features and speech handling (RTP) are the same – only the signaling protocol is different. For this reason, the description and examples below can be used for either implementation (the differences are clearly defined). IP Trunk is used to describe SIP or H.323 trunks.

The information below relates to basic networking, without the use of external H.323 Gatekeeper. The following steps are required to configure IP Trunks:

2.1   Determine Numbering Plan

When planning the IP Trunk implementation, it is important to determine the numbering plan for the sites. A properly planned VoIP network allows flexibility and simplifies additions or changes to the network.

There are two approaches to the numbering scheme;

❍   Open Numbering

In this case the sites are identified by dialing codes that are appended to the dialed digits. This allows the extension numbers to be the same at multiple sites.

Example:

	System A	System B
Extension Number	200-299	200-299
Site Code	1	2

Extension 200 at System A can dial Extension 200 at System B by dialing the extension 2200.

This numbering scheme is ideal when there are a large number of sites in the VoIP network or when many extension numbers are in use at each site.

❍ Closed Numbering

In this case, the first digit of the extension number is used for routing calls. This alleviates the need for dialing a site code and makes a more unified numbering plan. This is feasible only if there is a relatively small number of sites or the sites have very few extensions.

2.2   Configure VoIPDB Trunk Ports

When installing a PZ-(X)IPLA (VoIP Daughter Board) in the UNIVERGE SV8100 system, external line ports are automatically assigned once Program10-40-01 and Program10-40-02 are assigned.

For the slot assigned as a VoIPDB slot, view the starting trunk number.

By default, the VoIPDB trunk ports are allocated to SIP in Program10-03-02. When using H.323 it is necessary to change the trunk ports to H.323 in Program10-03-02.

2.2.1   H.323 Setup

Program/Item No.	Description/ Selection	Assigned Data	Comments
10-03-01	ETU Setup – Logical Port Number	0~200	The CD-CP00-US automatically assigns each PCB during installation. READ ONLY
10-03-02	ETU Setup – Trunk Type	0 = H.323 1 = SIP Default is 1	Assign each trunk type for each DSP Resource.
14-05-01	Trunk Group – Trunk Group Number	0~100 Priority = 1 ~ 200 Default is Group 1	Default priorities for trunks 1~200 is 1~200. Assign Trunks to Trunk Groups/Outbound Priority
22-02-01	Incoming Call Trunk Setup	0 = Normal 1 = VRS (second dial tone if no VRS installed) 2 = DISA 3 = DID 4 = DIL 5 = E&M Tie Line 6 = Delayed VRS 7 = ANI/DNIS 8 =DID(DDI) Mode Switching Default is 0	Set the feature type for the trunk you are programming. (Second dial tone for option 1 if no VRS is installed)

2.2.2 Configure VoIPDB Networking Information

Program/Item No.	Description/ Selection	Assigned Data	Comments
10-40-01	IP Trunk Availability – IP Trunk Availability	0 = Disable 1 = Enable Default is 0	Assign the H.323 Trunk availability.
10-40-02	IP Trunk Availability – Number of Ports	0~128 Default is 0	Assign the number of H.323 trunks.

2.2.3 VoIPDB (DSP) Basic Setup

Program/Item No.	Description/ Selection	Assigned Data	Comments
84-26-01	IPL Basic Setup – IP Address	Slot 1 = 172.16.0.20 : Slot 4 = 172.16.0.44 Default is Slot 1	Assign IP Addresses for each DSP resource to be used. IP Address must be assigned for every 16 DSP Resources used. Note: The VoIPDB IP Addresses must be statically assigned.
84-26-02	IPL Basic Setup – RTP Port Number	0~65534 Default for DSP1 = 10020	Assign the RTP Port Number.
84-26-03	IPL Basic Setup – RTCP Port Number (RTCP Port Number + 1)	0~65534 Default for DSP1 = 10021	Assign the RTCP Port Number.

2.3   Configure Signaling Information (H.323)

H.323 requires a gatekeeper to be configured to manage registration of H.323 endpoints. The UNIVERGE SV8100 has a built-in H.323 gatekeeper and also supports connection to an external gatekeeper. In most cases, the SV8100 internal Gatekeeper is sufficient. External gatekeepers are used usually only in very large H.323 networks.

To configure the UNIVERGE SV8100 to use its internal Gatekeeper, set Program 10-17-01 to Manual, and then set the CD-CP00-US IP address (Program10-12-09) as the gatekeeper address in Program 10-17-02.

The UNIVERGE SV8100 has to register an Alias address with the gatekeeper. This is used to route incoming calls to the correct destination.

2.3.1   H.323 GateKeeper Setup

Program/Item No.	Description/ Selection	Assigned Data	Comments
10-17-01	H.323 Gatekeeper Setup – Gatekeeper Mode	0 = No Gatekeeper 1 = Automatic 2 = Manual Default is 0	Set IP Address either automatically or manually if using an external Gatekeeper. If Automatic Gatekeeper is selected, the system searches for the gatekeeper. If Manual Gatekeeper is selected, the IP address defined in Program 10-17-02 is used.
10-17-02	H.323 Gatekeeper Setup – Gatekeeper IP Address	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 0.0.0.0 Default is 0.0.0.0	If Program 10-17-01 is set to 2, enter the IP address of the Gatekeeper. This should match the entry made in Program 10-12-09.

Program/Item No.	Description/ Selection	Assigned Data	Comments
10-17-04	H.323 Gatekeeper Setup – Preferred Gatekeeper	Maximum 124 characters Default not assigned	If Program 10-17-01 is set to 1, enter the Gatekeeper ID. When registering with an external Gatekeeper using Gatekeeper search, two or more GRQs (Gate Keeper Request) may be assigned. In this case, if this ID is set up, it registers with a Gatekeeper using the ID set up in this program (124 characters max).

2.3.2   H.323 Alias Address Setup

Program/Item No.	Description/ Selection	Assigned Data	Comments
10-18-01	H.323 Alias Address Setup – Alias Address	Dial up to 12 digits (0~9, *, #) Default not assigned	Enter the Alias Address of the UNIVERGE SV8100 system registered into the external Gatekeeper. (12 digits max). This is the System Code of Local System.
10-18-02	H.323 Alias Address Setup – Alias Address Type	0 = E164 Default = 0	Define the type of Alias Address registered to the external Gatekeeper. At this point, only E.164 can be entered.

2.4   Configure Call Routing for H.323

Enter the remote destination information in Program 10-23. This allows up to 1000 SIP or H.323 destinations (remote systems) to be entered. If more than 1000 destinations are required, it is necessary to use an external H.323 Gatekeeper or SIP Server.

It may be advantageous to configure F-route to route calls to remote destinations via the IP trunks (this can simplify dialing for the users). F-route configuration is discussed fully in the UNIVERGE SV8100 Features and Specifications Manual.

2.4.1   H.323 Call Routing

Program/Item No.	Description/ Selection	Assigned Data	Comments
10-23-01	SIP System Interconnection Setup – System Interconnect	0 = Disable (No) 1 = Enable (Yes) Default is 0	Enable/Disable the remote destination.
10-23-02	SIP System Interconnection Setup – IP Address	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 Default = 0.0.0.0	Enter the IP address for the remote system.
10-23-04	SIP System Interconnection Setup – Dial Number	Up to 12 digits (0~9) Default not assigned	Enter the alias number for the remote system. This is the E.164 number assigned to the remote destinations.

2.4.2   Caller ID (H.323)

Program/Item No.	Description/ Selection	Assigned Data	Comments
21-17-01	IP Trunk (SIP) Calling Party Number Setup for Trunk	Up to 16 digits (1~0, *, #) Default not assigned	Enter the CPN to be used for calls made via the selected trunk.
21-18-01	IP Trunk (H.323) Calling Party Number Setup for Extension	Up to 16 digits (1~0, *, #) Default not assigned	Enter the CPN to be used for calls made via the selected H.323 Extension. This Program overrides the programming in Program 21-17
20-08-13	Class of Service Options (Outgoing Call Service) – ISDN CLIP	0 = Off 1 = On Default is 0	Determine if the Calling Line Identity presentation and screening indicators are allowed.
20-09-02	Class of Service Options (Incoming Call Service) – Caller ID Display	0 = Off 1 = On Default is 0	Turn Off or On the Caller ID display at an extension.

The CLIP that is actually sent is based on the following priority:
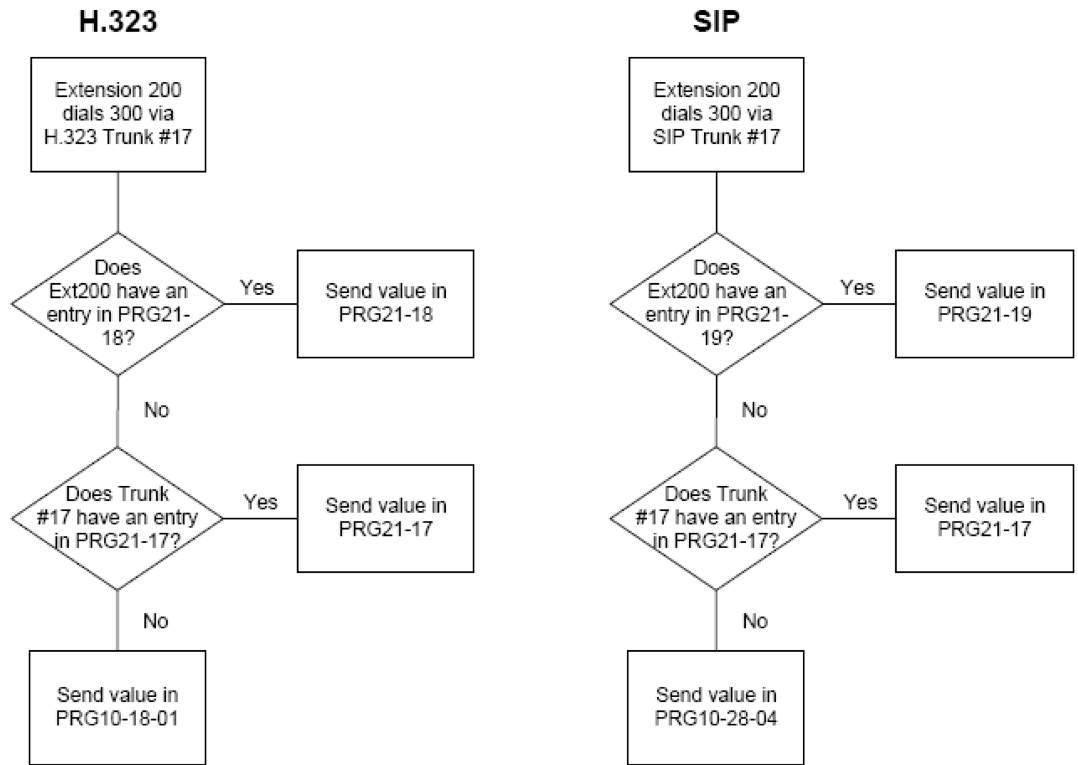
2.5   CODEC Selection for IP Networking

IP Networking can use various CODECs. A CODEC is a standard for converting an analog signal to digital. This conversion process is handled by the DSP (Digital Signal Processors) on VoIP Daughter Board.

Each CODEC has different voice quality and compression properties. The correct choice of CODEC is based on the bandwidth available, the number of calls required, and the voice quality required.

2.5.1   CODEC Selection for IP Networking (H.323)

Program/Item No.	Description/ Selection	Assigned Data	Comments
84-01-02	H.323 Trunk Basic Information Setup – Number of G.711 audio frames	1~4 (SV8100) Default =3	

Program/Item No.	Description/ Selection	Assigned Data	Comments
84-01-03	H.323 Trunk Basic Information Setup – G.711 VAD mode	0 = Disable 1 = Enable Default is 0	
84-01-04	H.323 Trunk Basic Information Setup – G.711 Type	0 = A-law 1 = μ-law Default is 1	
84-01-05	H.323 Trunk Basic Information Setup – Number of G.729 audio frames	1~6 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms 5 = 50ms 6 = 60ms Default is 3	
84-01-06	H.323 Trunk Basic Information Setup – G.729 VAD mode	0 = Disable 1 = Enable Default = 0	
84-01-07	H.323 Trunk Basic Information Setup – G.729 Jitter Buffer (min)	0~270ms Default is 30	
84-01-08	H.323 Trunk Basic Information Setup – G.729 Jitter Buffer (average)	0~270ms Default is 60	
84-01-09	H.323 Trunk Basic Information Setup – G.729 Jitter Buffer (max)	0~270ms Default is 120	
84-01-11	H.323 Trunk Basic Information Setup – Number of G.723 audio frames	1~2 Default is 1	
84-01-12	H.323 Trunk Basic Information Setup – G.711 VAD mode	0 = Disable 1 = Enable Default is 0	
84-01-15	H.323 Trunk Basic Information Setup – Jitter Buffer Mode	1 = Fixed 2 = Self adjusting (silence inside) 3 = Self adjusting Default is 3	

Program/Item No.	Description/ Selection	Assigned Data	Comments
84-01-16	H.323 Trunk Basic Information Setup – G.711 Jitter Buffer(min)	0~160ms Default is 30	
84-01-17	H.323 Trunk Basic Information Setup – G.711 Jitter Buffer (average)	0~160ms Default is 60	
84-01-18	H.323 Trunk Basic Information Setup – G.711 Jitter Buffer (max)	0~160ms Default is 120	
84-01-19	H.323 Trunk Basic Information Setup – G.723 Jitter Buffer (min)	0~270ms Default is 30	
84-01-20	H.323 Trunk Basic Information Setup – G.723 Jitter Buffer (average)	0~270ms Default is 60	
84-01-21	H.323 Trunk Basic Information Setup – G.723 Jitter Buffer (max)	0~270ms Default is 120	
84-01-22	H.323 Trunk Basic Information Setup – Silent Compression Threshold	0~30 (-19dB~ +10dB and self adjustment) 0 = Self adjustment 1 = -19dB (-49dBm) : 20 = 0dB (-30dBm) : 29 = 9dB (-21dBm) 30 = 10dB (-20dBm) Default is 20	
84-01-23	H.323 Trunk Basic Information Setup – Idle Noise Level	-5000dBm ~ -7000dBm 7000dBm Default is 7000	
84-01-24	H.323 Trunk Basic Information Setup – Echo Canceller Mode	0 = Disable 1 = Enable Default is 1	

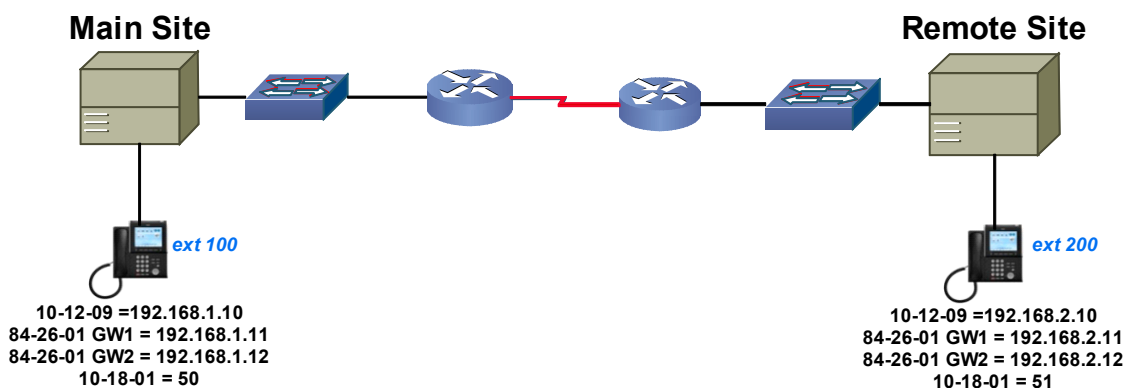| Program/Item No.	Description/ Selection	Assigned Data	Comments
84-01-25	H.323 Trunk Basic Information Setup – Echo Canceller Tail Size	1 = 4ms 2 = 8ms 3 = 16ms 4 = 32ms 5 = 64ms 6 = 128ms Default is 6	
84-01-26	H.323 Trunk Basic Information Setup – Echo Canceller NLP Mode	0 = Disable 1 = Enable Default is 1	
84-01-28	H.323 Trunk Basic Information Setup – Echo Canceller NLP Noise Setting	0 = Automatic level adjustment 1 = Fixed level Default is 0	
84-01-30	H.323 Trunk Basic Information Setup – TX Gain	0~40 (-20dBm~ +20dBm) 20 Default is 20	
84-01-31	H.323 Trunk Basic Information Setup – RX Gain	0~40 (-20dBm~ +20dBm) 20 Default is 20	
84-01-33	H.323 Trunk Basic Information Setup – Priority CODEC setting Priority of voice encoding method.	0~3 0 = G.711 1 = G.723 2 = G.729 3 = G.722 Default is 0	Priority of voice encoding method.
84-01-34	---Not Used---		
84-01-35	---Not Used---		

Program/Item No.	Description/ Selection	Assigned Data	Comments
84-01-36	H.323 Trunk Basic Information Setup – The Maximum FAX transmission Rate	0 = V.27ter, 2400bps 1 = V.27ter, 4800bps 2 = V.29, 7200bps 3 = V.29, 9600bps 4 = V.17, 12000bps 5 = V.17, 14400bps Default is 5	
84-01-37	H.323 Trunk Basic Information Setup – FAX FIFO Considering Delay Time	0~600ms Default is 300	
84-01-38	H.323 Trunk Basic Information Setup – Size of FAX Packet	20~48 bytes Default is 20	
84-01-39	H.323 Trunk Basic Information Setup – FAX Modem Transmission Level	0~13 (0dBm ~ -13dBm) Default is 9	
84-01-40	H.323 Trunk Basic Information Setup – FAX Modem Carrier Signal Detection Threshold	0 = -26dBm 1 = -33dBm 2 = -43dBm Default is 1	
84-01-41	H.323 Trunk Basic Information Setup – FAX Communication no Communication Time-Out	10~32000 seconds Default is 30	
84-01-43	H.323 Trunk Basic Information Setup – High-speed Signal Data (fax picture signal) Packet Length	1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms Default is 4	
84-01-44	H.323 Trunk Basic Information Setup – Low-speed Signal Data (FAX Procedure Signal)	0~5 Default is 0	
84-01-45	H.323 Trunk Basic Information Setup – High-speed Signal Data (FAX Procedure Signal)	0~2 Default is 0	

Program/Item No.	Description/ Selection	Assigned Data	Comments
84-01-46	H.323 Trunk Basic Information Setup – TCF Operation Setting	1 = Training signal (TCF) of the fax modem (local). 2 = Training signal (TCF) of the fax modem. Default is 1	
84-01-47	H.323 Trunk Basic Information Setup – The Maximum, Low-speed Signal Data (Size of Packet)	1~65535 bytes Default is 1	
84-01-48	H.323 Trunk Basic Information Setup – Network Transmission Time-out	10~32000 seconds Default is 150	
84-01-49	H.323 Trunk Basic Information Setup – Eflag Beginning Timer	0~65535 Default is 2600	
84-01-50	H.323 Trunk Basic Information Setup – Eflag Stop Timer	Need Information Default is 2300	
84-01-51	H.323 Trunk Basic Information Setup – The Former Line Substitution of Scanning Line	0 = Disable 1 = Enable Default is 1	
84-01-52	H.323 Trunk Basic Information Setup – Eflag Setting at Head DIS	0 = Disable 1 = Enable Default is 1	
84-01-53	H.323 Trunk Basic Information Setup – TFOP Protocol	0 = Disable 1 = Enable Default is 1	
84-01-54	H.323 Trunk Basic Information Setup – NSF Superscription	0 = Disable 1 = Enable Default is 0	
84-01-55	H.323 Trunk Basic Information Setup – ECM (Error Correction Mode)	0 = Disable 1 = Enable Default is 1	

Program/Item No.	Description/ Selection	Assigned Data	Comments
84-01-56	H.323 Trunk Basic Information Setup – Enable Modified Read Code	0 = Disable 1 = Enable Default is 1	
84-01-57	H.323 Trunk Basic Information Setup – NSF Country Code Setting	0~65535 Default is 0	
84-01-58	H.323 Trunk Basic Information Setup – NSF Vendor Code Setting	0~65535 Default is 0	
84-01-59	H.323 Trunk Basic Information Setup – FAX Relay Function	0 = Disable 1 = Enable 2 = Each port mode Default is 0	
84-01-60	H.323 Trunk Basic Information Setup – Echo Canceller Type	0~3 Default is 0	
84-01-61	H.323 Trunk Basic Information Setup – Auto Gain Control	0~5 Default is 0	
84-01-62	H.323 Trunk Basic Information Setup – DTMF Relay Mode	0 = VoIP 1 = RFC2833 2 = H.245 3 = Disable Default is 0	Set up information of VoIP is set by Program 84-06-10.
84-01-63	H.323 Trunk Basic Information Setup –Number of G.722 audio frames	1~4 1 = 10 ms 2 = 20 ms 3 = 30 ms 4 = 40 ms Default is 3	
84-01-64	H.323 Trunk Basic Information Setup – G.722 Voice Activity Detection Mode	0 = Disable 1 = Enable Default is 0	
84-01-65	H.323 Trunk Basic Information Setup – G.722 Jitter Buffer (min)	0~160 Default is 30	

Program/Item No.	Description/ Selection	Assigned Data	Comments
84-01-66	H.323 Trunk Basic Information Setup – G.722 Jitter Buffer (average)	0~160 Default is 60	
84-01-67	H.323 Trunk Basic Information Setup – G.722 Jitter Buffer (max)	0~160 Default is 120	

2.6   H.323 TIE Line Programming Example

Main Site   **Remote Site**

ext 100   *ext 200*

10-12-09 =192.168.1.10
84-26-01 GW1 = 192.168.1.11
84-26-01 GW2 = 192.168.1.12
10-18-01 = 50

10-12-09 =192.168.2.10
84-26-01 GW1 = 192.168.2.11
84-26-01 GW2 = 192.168.2.12
10-18-01 = 51

2.6.1   H.323 TIE Line Programming Example

Program	Main Site (Office A)	Remote Site (Office B)
PRG 10-12-09 *	192.168.1.10	192.168.2.40
PRG 10-12-10 *	255.255.255.0	255.255.255.0
PRG 11-01-01 Digit 1x	3 Digits, Extension	Program 11-01-01 Digit 1x 3 Digits, F-Route
PRG 11-01-01 Digit 2x	3 Digits, F-Route	Program 11-01-01 Digit 2x 3 Digits, Extension
PRG 11-02-01	100, 101, etc.	200, 201, etc.
PRG 14-05-01	Set all IP Trunks to Trunk Group 10	
PRG 22-02-01	Set all IP Trunks to TIE	
PRG 44-02-01 (Table 1)	4	1
PRG 44-02-02 (Table 1)	F-Route	F-Route
PRG 44-02-03 (Table 1)	1	1
PRG 44-05-01 (F-Route 1)	Trunk Group 10	Trunk Group 10
PRG 44-05-09 (Max Digits)	3	3

Program	Main Site (Office A)	Remote Site (Office B)
PRG 84-26-01 (DSP 1)	192.168.1.11	192.168.2.11
PRG 84-26-01 (DSP 2)	192.168.1.12	192.168.2.12
H.323 Settings		
PRG 10-40-01	Enable H.323 Trunk Availability	Enable H.323 Trunk Availability
PRG 10-40-02	Assign the number of Ports (4 Ports)	Assign the number of Ports (4 Ports)
PRG 10-03-02	Set IP Trunks to H.323	Set IP Trunks to H.323
PRG 10-17-01	Manual	Manual
PRG 10-17-02	192.168.1.10	192.168.2.10
PRG 10-18-01	50	51
PRG 10-23-01	Enable System Interconnection (Yes)	
PRG 10-23-02	Assign the IP Address of the Distant System (192.168.2.10)	Assign the IP Address of the Distant System (192.168.1.10)
PRG 10-23-04	Starting digit of the Distant System Extension Number	Starting digit of the Distant System Extension Number

* CD-CP00-US reset is required after making changes to this Programs.

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NEC iSIP

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

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.

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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.

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 .

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.
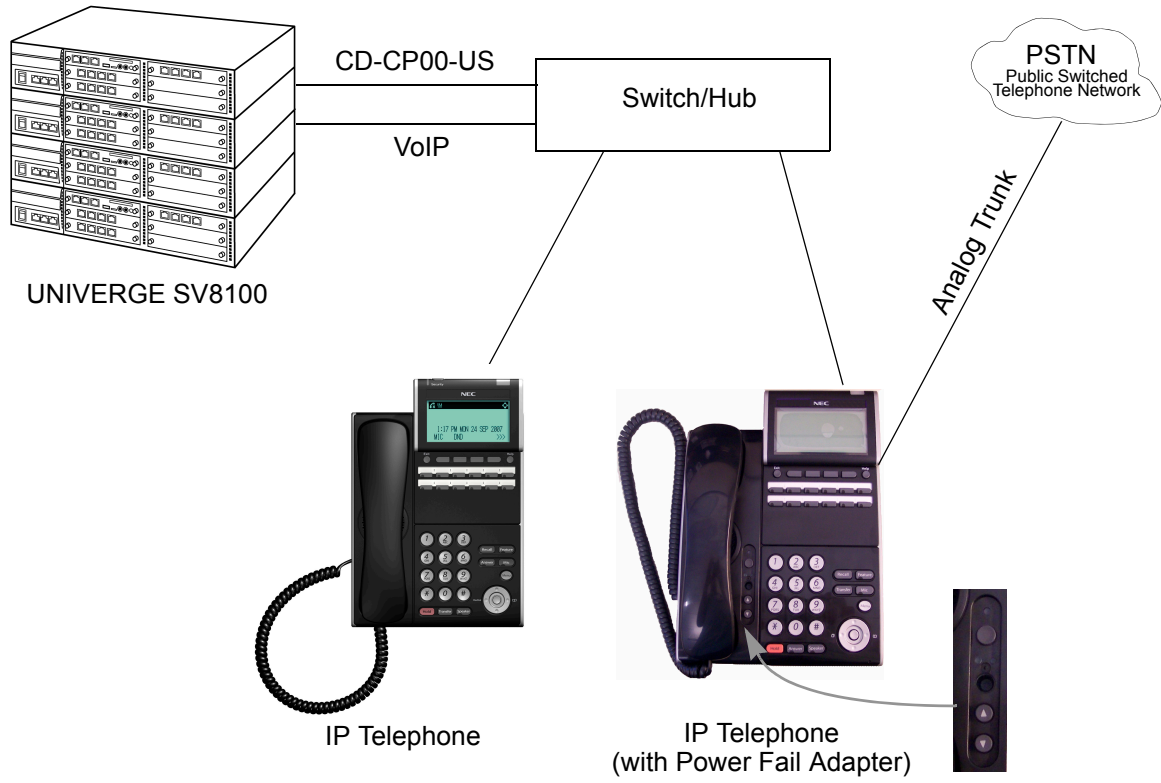
Figure 8-2 Power Fail Adapter Connection
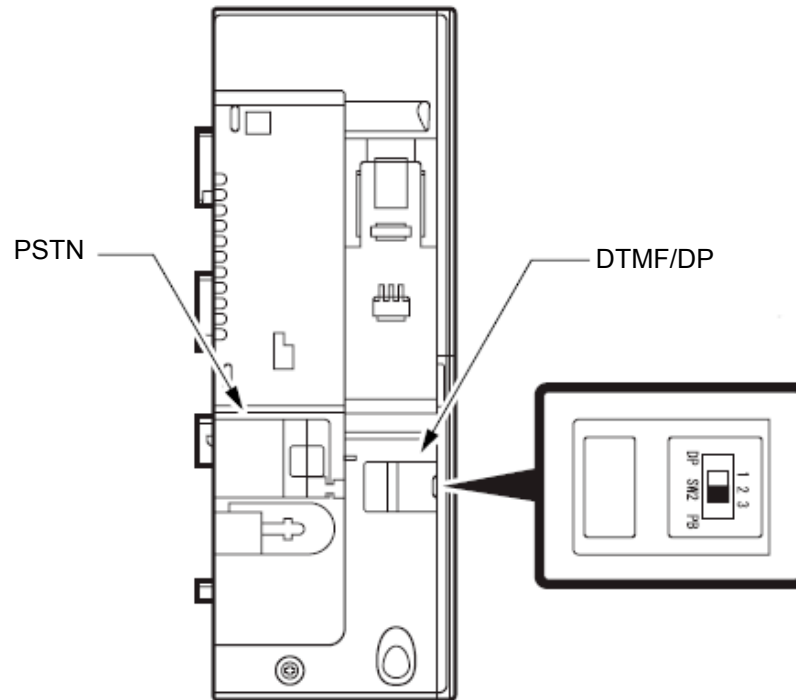
Figure 8-3 IP Telephone Connection

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/hub port.
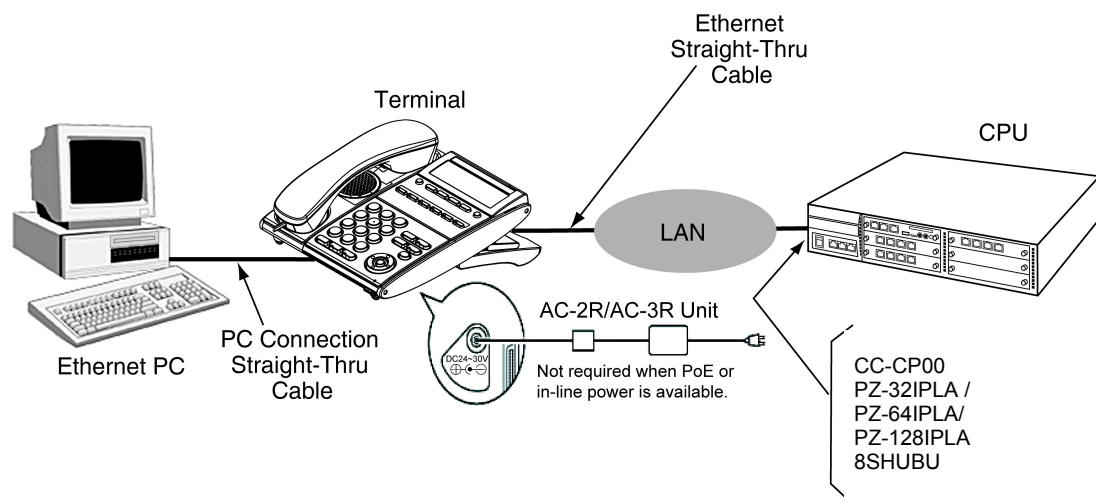
Figure 8-4 Typical Network IP Connection

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/hub 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 a 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   Powered Patch Panel

A powered patch panel has two RJ-45 connectors per IP Phone. One port connects to the switch/hub, and the other port connects to the IP Phone. The patch panel has an integral power supply that adds power to the spare pairs of the RJ-45.

When the IP Phone is connected to the powered patch panel, it automatically receives its power via the spare pairs on the CAT-5 cable. There is no need for a local power adapter.

5.3   Power Over Ethernet (PoE)

A PoE switch is a switched hub 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 10-26-01 – Peer to Peer.

SECTION 7   PROGRAMMING

The first step to connecting IP telephones to the UNIVERGE SV8100 system is to connect the UNIVERGE SV8100 system to the customer data network. Refer to Chapter 3 General IP Configuration. Next, program the PZ-32IPLA/PZ-64IPLA/PZ-128IPLA and associated IP telephone settings. To complete the installation, program the IP telephone.

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

➲   **10-12-02   CD-CP00-US Network Setup - Subnet Mask**
    Select the Subnet Mask to use (default: 255.255.0.0).

➲   **10-12-03   CD-CP00-US Network Setup - Default Gateway**
    If required, select the default gateway IP address to use when using a router (default: 0.0.0.0).

➲   **10-12-09   CD-CP00-US Network Setup - IP Address**
    Select the IP address for the IP connection (default: Slot 1 = 172.16.0.10 or Slot 4 = 172.16.0.13). A static IP address is required by the PZ-32IPLA/PZ-64IPLA/PZ-128IPLA daughter board.

➲ **10-26-01   Peer-to-Peer**

Enable/Disable the Peer-to-Peer feature between IP Stations.

✎   Disabling this feature results in IP Station-to-IP Station calls using DSP Resources.

➲ **15-05-15   IP Telephone Terminal Basic Data Setup - CODEC Type**

For each IP telephone, set which CODEC Type to use.

➲ **84-11-28   D^{term} IP CODEC Information Basic Setup - Audio Capability Priority**

For each Type, set the CODEC to use (default: G.711).

➲ **84-26-01   IPL Basic Setup - IP Address**

For each PZ-32IPLA/PZ-64IPLA/PZ-128IPLA, enter the IP address for the IPLA blade (default: Slot (DSP) 1 = 172.16.0.20, Slot (DSP) 2 = 172.16.0.44).

➲ **84-26-02   IPL Basic Setup - RTP Port Number**

For each PZ-32IPLA/PZ-64IPLA/PZ-128IPLA, enter the RTP port number (default: 10020).

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

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).

8.1 Example Configuration 1 - Static IP Addressing, One LAN

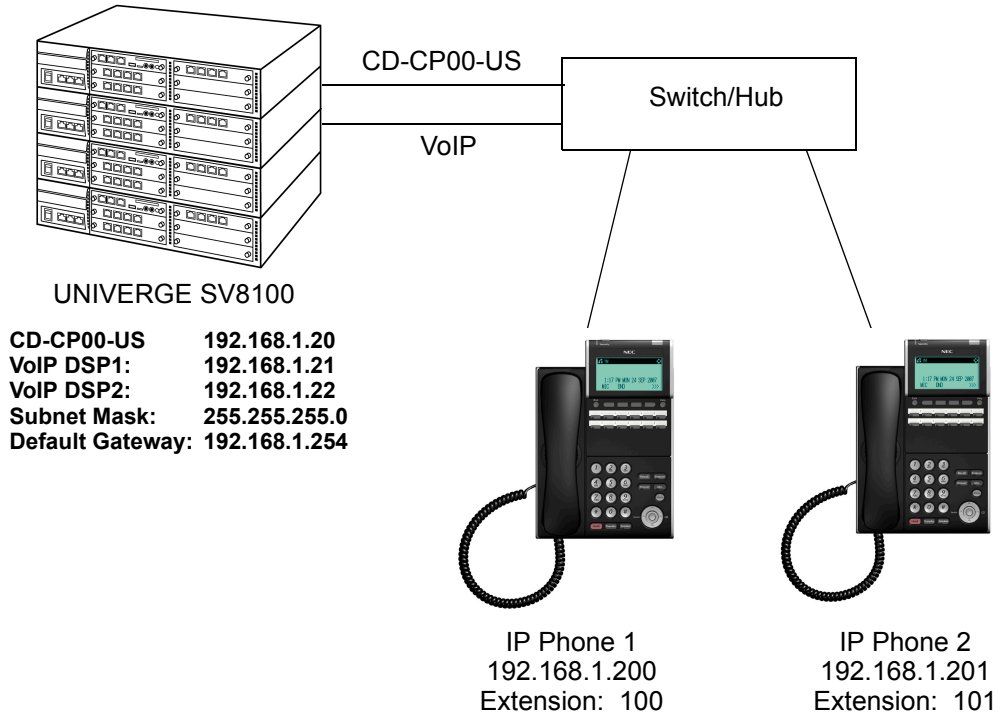This example shows IP Phone connected to a single LAN (no routers), with static IP Addresses.

UNIVERGE SV8100

CD-CP00-US	**192.168.1.20**
VoIP DSP1:	**192.168.1.21**
VoIP DSP2:	**192.168.1.22**
Subnet Mask:	**255.255.255.0**
Default Gateway:	**192.168.1.254**

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)	Leave default
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
84-26-01 : IPL Basic Setup - IP Address	172.16.0.20

8.2   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 2003 server)

❍   inDHCP internal DHCP server

In this case, additional programming would be required. Refer to Chapter 3 General IP Configuration.

UNIVERGE SV8100

CD-CP00-US   **192.168.1.20**
VoIP DSP1:   **192.168.1.21**
VoIP DSP2:   **192.168.1.22**
Subnet Mask: **255.255.255.0**
Default Gateway: 192.168.1.254

IP Phone 1
192.168.1.200
Extension: 200

IP Phone 2
192.168.1.201
Extension: 201
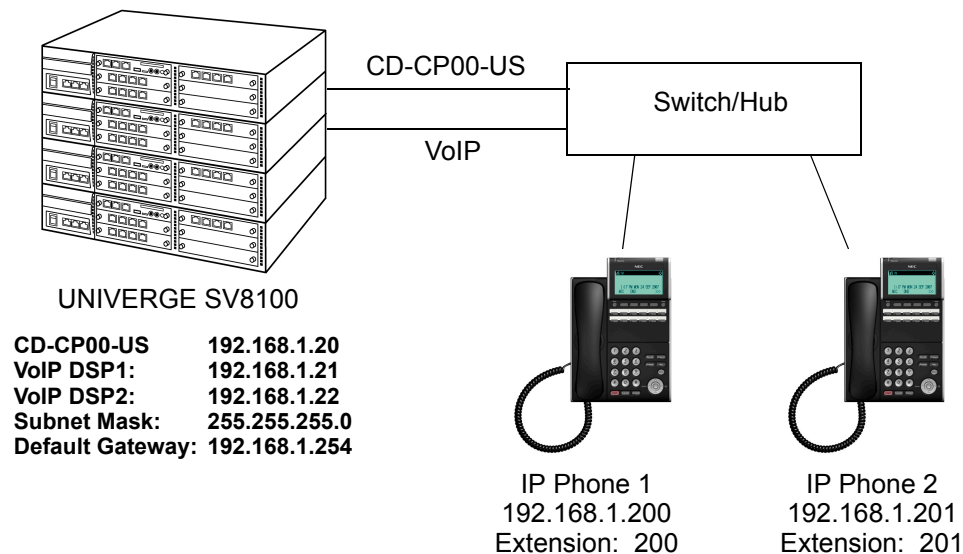
Figure 8-6 Example Configuration 2 - Dynamic IP Addressing, One LAN

Programming - CD-CP00-US:

10-12-01 : CD-CP00-US Network Setup - IP Address (for CD-CP00-US)	leave default
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	192.168.1.200 Min
10-14-01 : Managed Network Setup (Scope 1)	192.168.1.250 Max

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

10-16-35 SIP Server Code	120
10-16-36 SIP Server IP Address	192.168.1.20
84-26-01 : IPL Basic Setup - IP Address	192.168.1.21
	192.168.1.22

Programming - IP Phones:

DHCP Mode (Ext. 200):	Enabled

8.3   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.
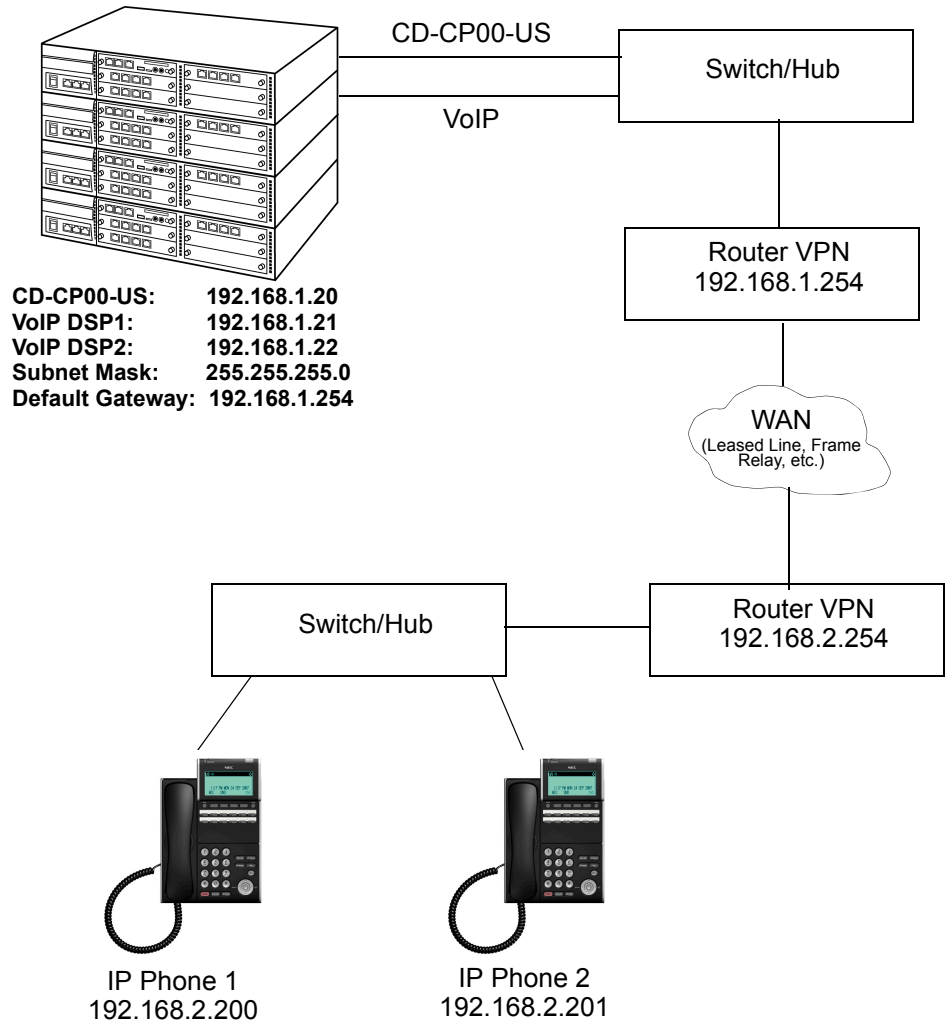
Figure 8-7 Example Configuration 3 - Static IP Addressing, Routed WAN

Programming - CD-CP00-US:

10-12-01 : CD-CP00-US Network Setup - IP Address (for CD-CP00-US)	leave default
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
	198.168.2.254

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 2003 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 8.2 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 address. The method for adding this service varies depending on the DHCP server used.

10.1   DHCP Server Configuration Example

The example below shows the necessary steps to add Option 129 to a Windows 2000 Server.

1.   Enter Computer Management and expand **Service and Applications**.

2.   Right click on **DHCP** and click on **Set Predefined Options.** Refer to Figure 8-8 DHCP Server Configuration on page 8-15.

3.   When the Predefined Options and Values dialogue box opens, click on **Add**.
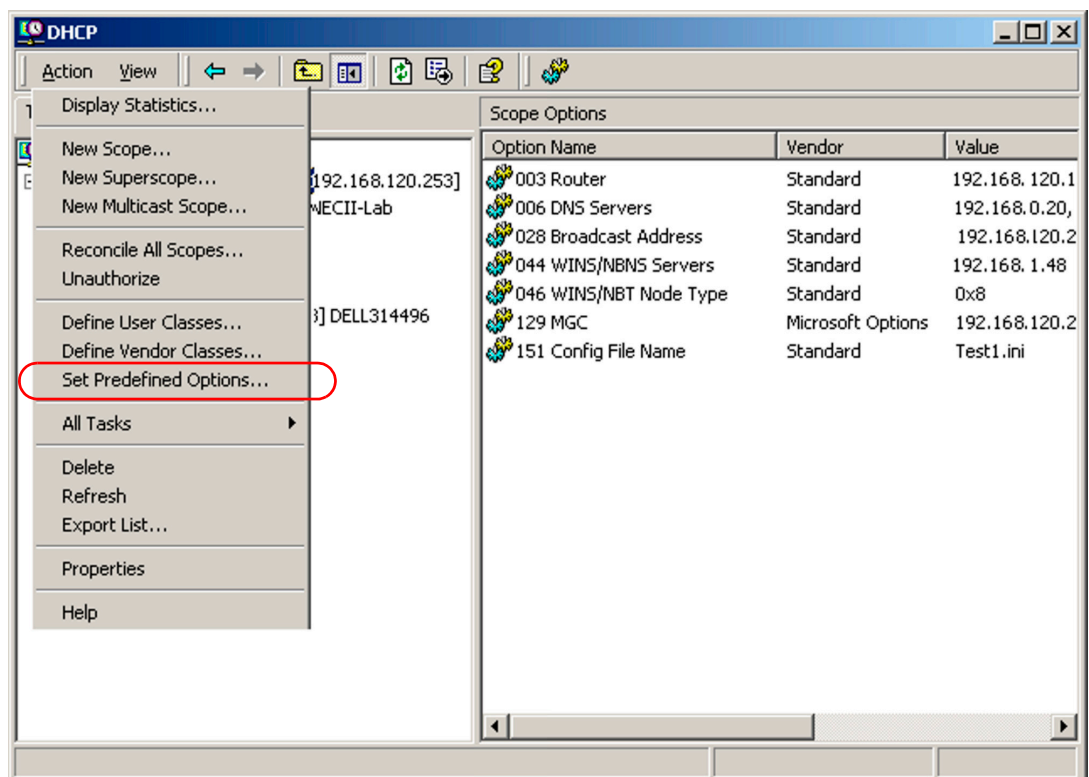
Figure 8-8 DHCP Server Configuration

4.   Enter the information In the Option Type dialog box, and click **OK**.

✎   *A new Option type for the DHCP server is created.*

Figure 8-9 Option Type Dialog Box

5.   To add the Option Type to the DHCP scope and specify the value to use, locate the scope that is currently being used (there may be more than one) in Computer Management.

6.   Right click on **Scope Options,** and click on **Configure Options**.

7.   Scroll down to **Option 129** (created in steps 1 to 5), and check the box.

8.   You are then prompted to enter a value.
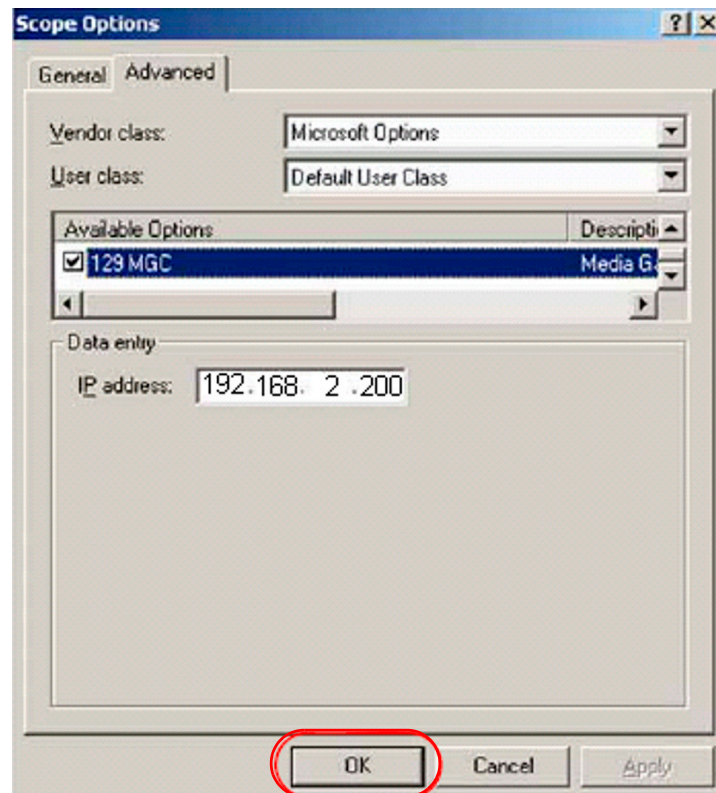
Figure 8-10 Scope Options Dialog Box

9.   Enter the PZ-(X)IPLA address - this is typically the PZ-32IPLA/
        PZ-64IPLA/PZ-128IPLA IP address (Program 84-05-01), and click **OK**.

10.   The DHCP server is now ready to provide IP Phones with the PZ-(X)IPLA
        address.

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 8-11 IP Phone Manager

3.   Click **Terminal**.
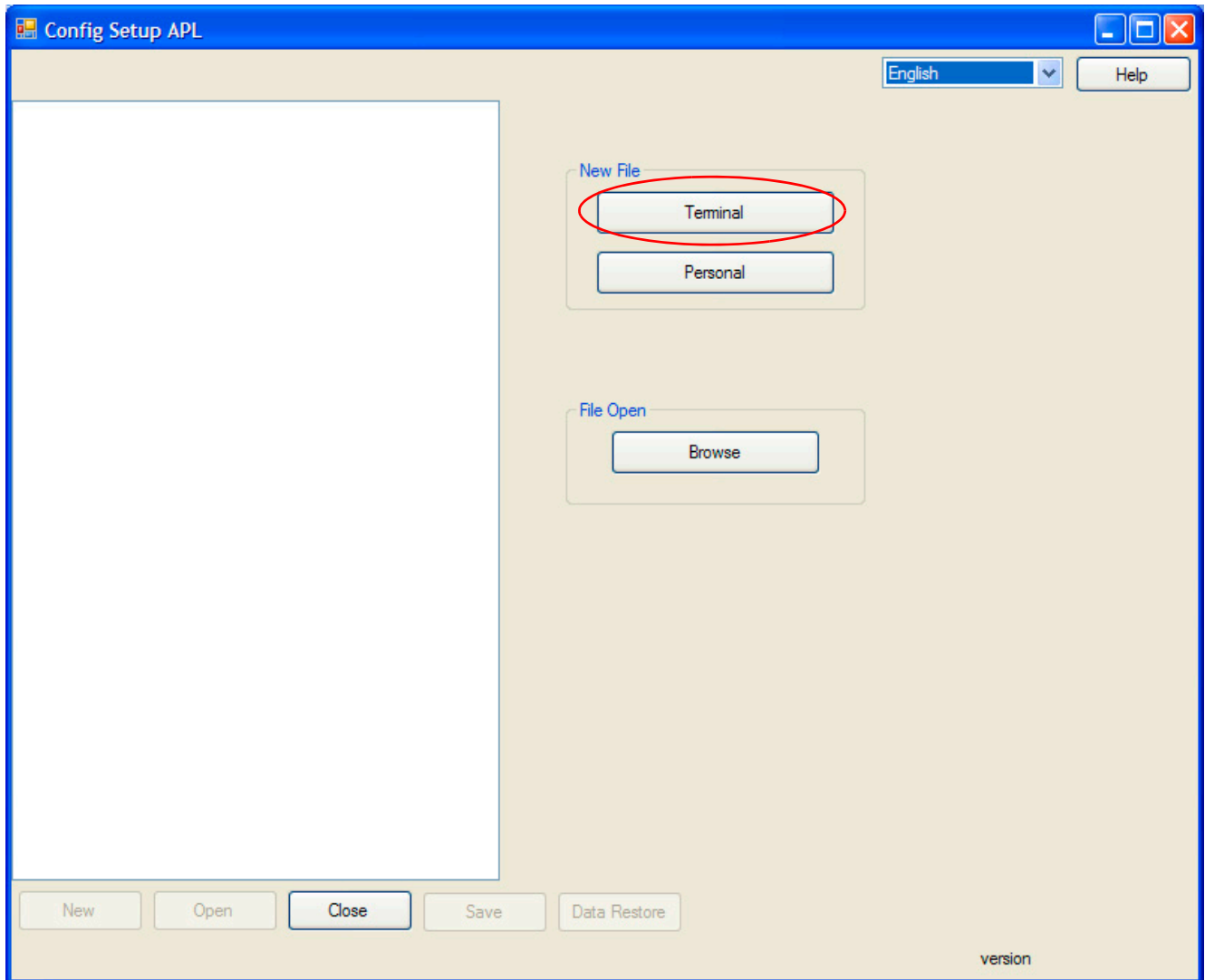
Figure 8-12 Config Setup APL

4.   Click **1st Server Address**.

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

6.   Click **OK**.
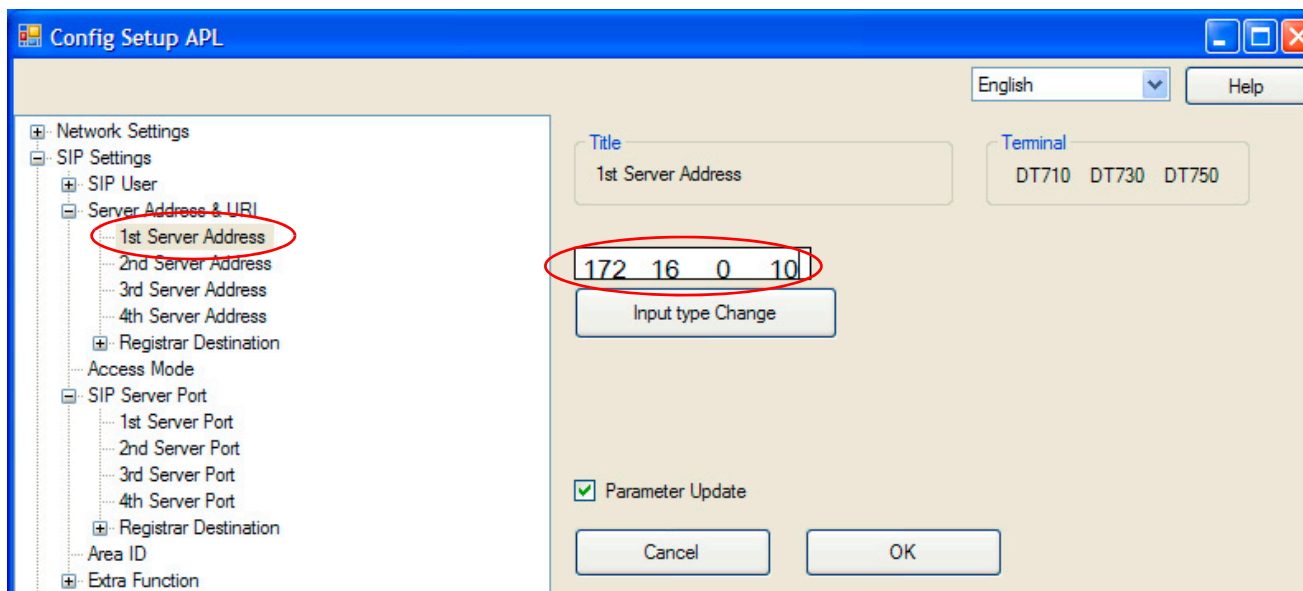
Figure 8-13 Assign IP Address

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

8.   Click **OK**.
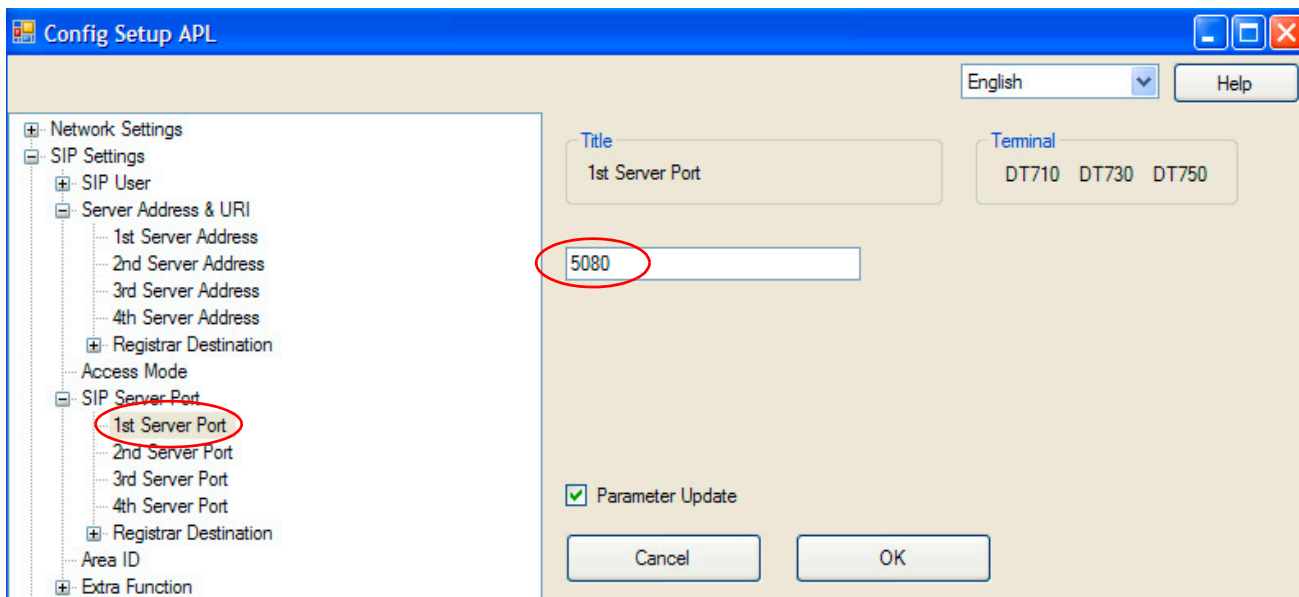
Figure 8-14 Network Settings - 1st Server Port

9.   After the changes are made, click **Save**.

10.   When the Save window opens, click **Save as...**
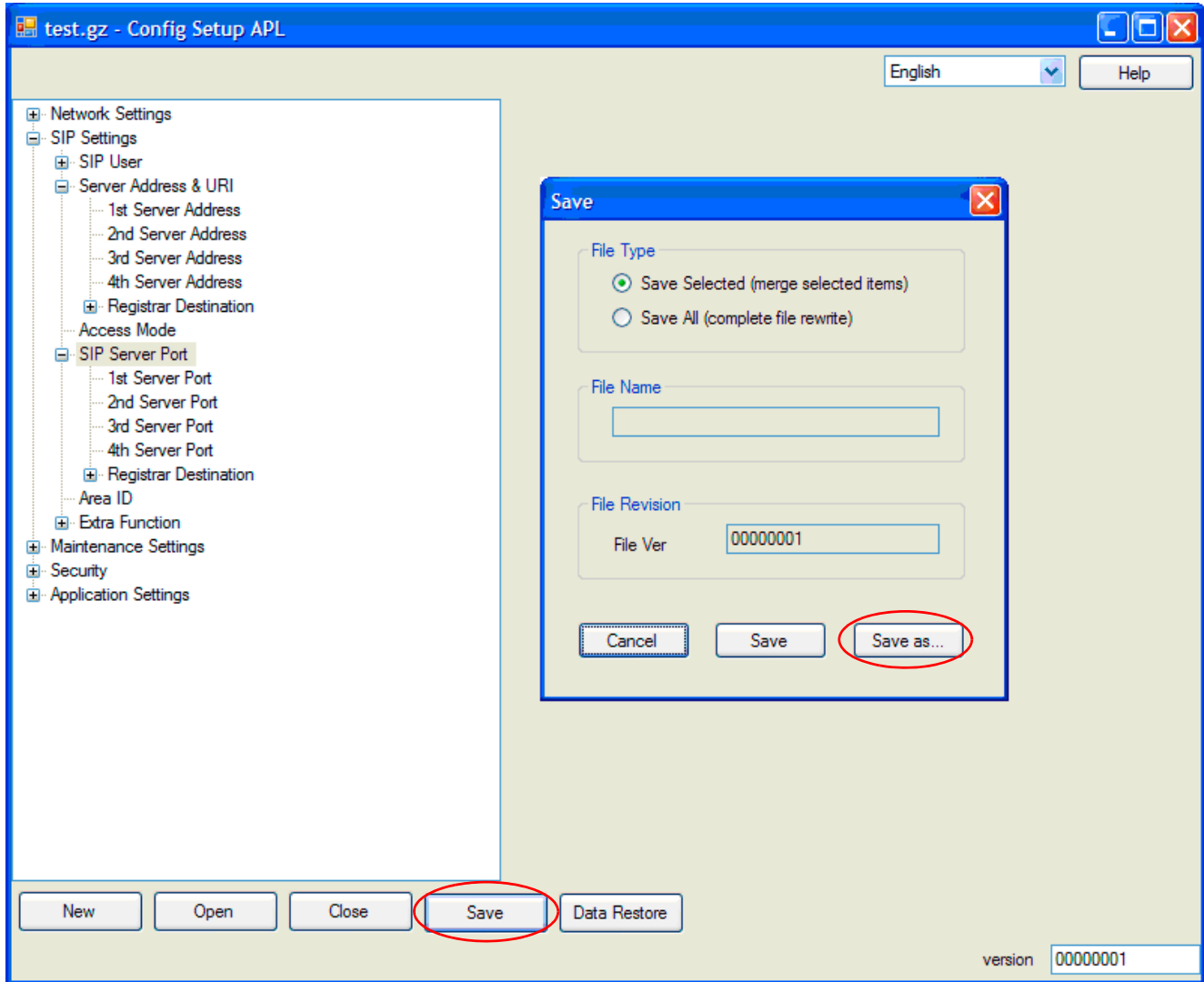
Figure 8-15 Save As/Name File Window

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**.
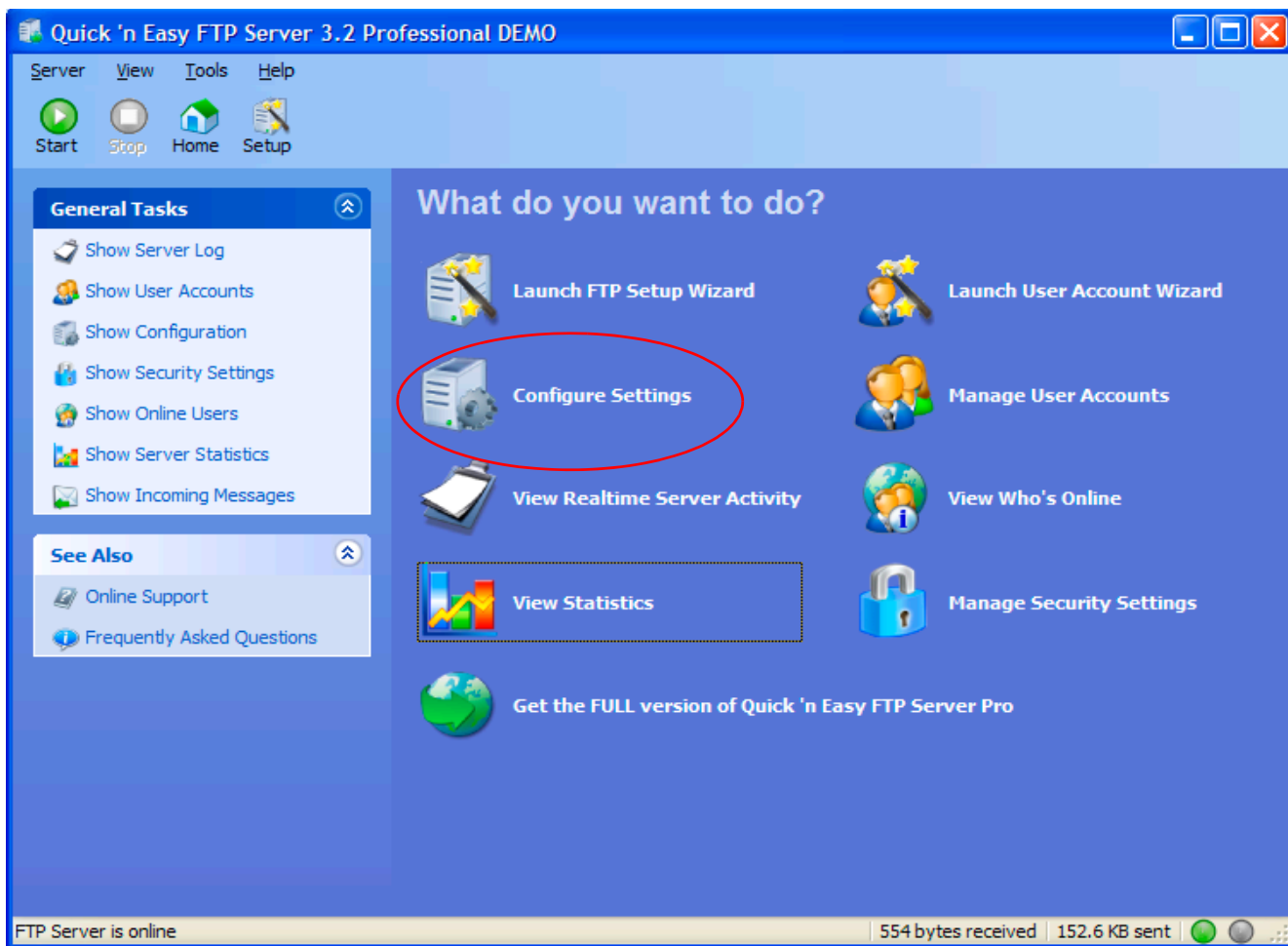
Figure 8-16 Quick 'n Easy FTP Server

2.   Click on the **Advanced** tab to specify the directory to store files.
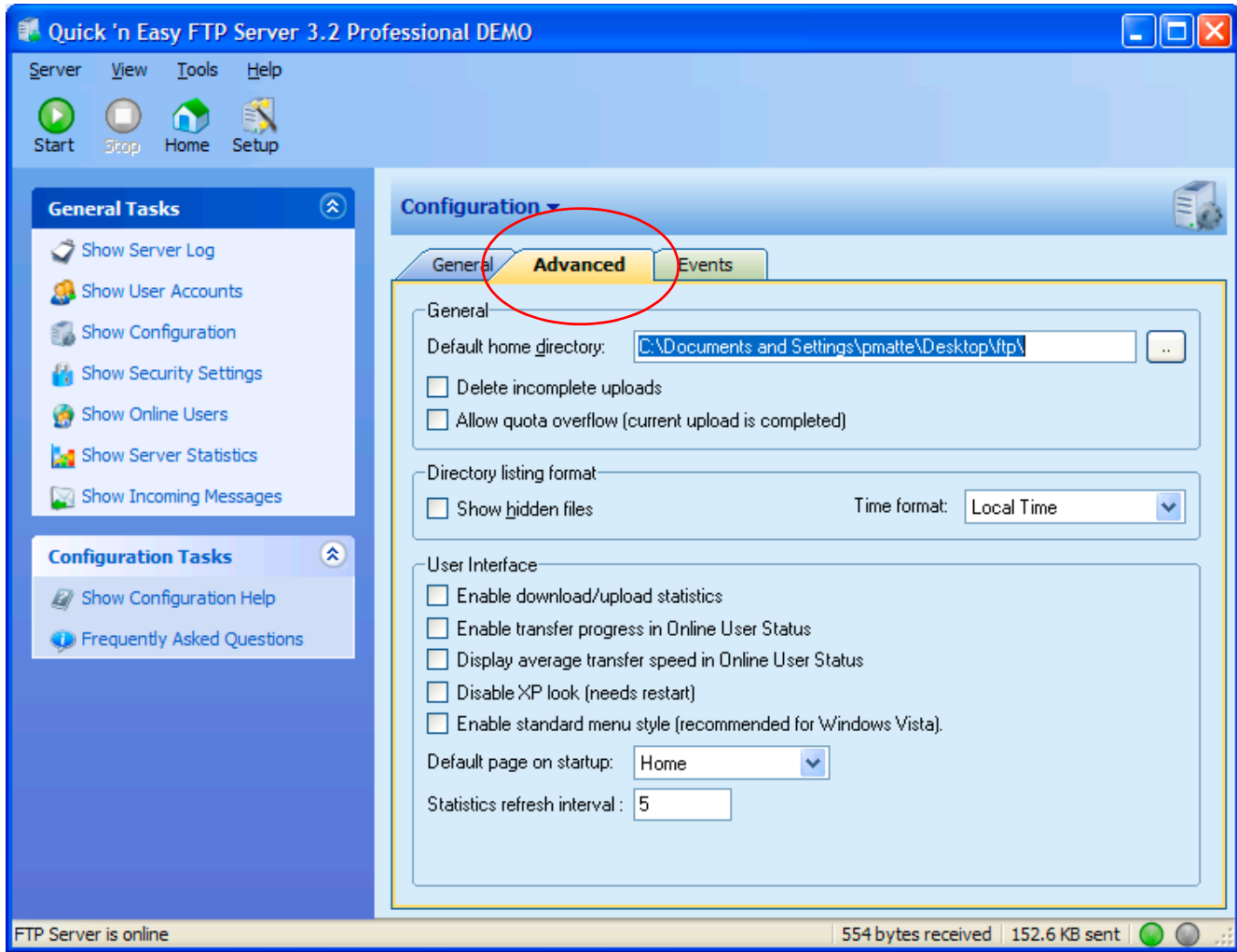
Figure 8-17 Quick 'n Easy FTP Server Configuration

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 2003 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   CONFIGURING QUALITY OF SERVICE

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

The UNIVERGE SV8100 system supports:

❍   IEEE802.1q (VLAN Tagging)

❍   IEEE802.1p (Priority)

With the VLAN tagging mode, the UNIVERGE SV8100 system can handle packets with or without a VLAN tag. If the VLAN ID of a packet is different from the registered one, that packet is dropped.

12.1   Layer 2 Priority Control

Layer 2 priority control can be enabled on an ethernet switch, if it supports VLAN tagging. This allows layer 2 prioritization.
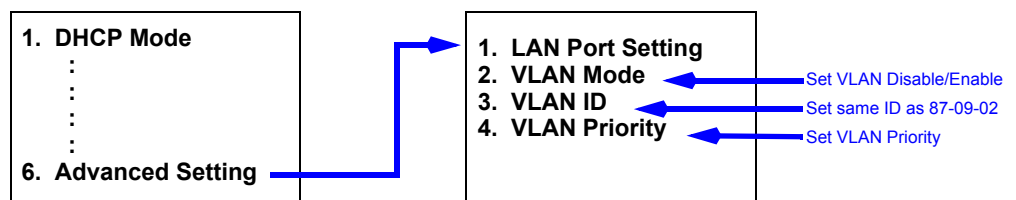
Using a switch that supports 802.1p allows:

❍   priority control

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

12.1.1   Programming Layer 2 Priority Control

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

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

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

```
┌─────────────────────┐          ┌──────────────────────┐
│ 1.  DHCP Mode       │          │ 1.  LAN Port Setting │
│       :             │          │ 2.  VLAN Mode        │ ◄── Set VLAN Disable/Enable
│       :             │          │ 3.  VLAN ID          │ ◄── Set same ID as 87-09-02
│       :             │          │ 4.  VLAN Priority    │ ◄── Set VLAN Priority
│       :             │          │                      │
│ 6.  Advanced Setting│──────►   │                      │
└─────────────────────┘          └──────────────────────┘
```

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

12.2   Layer 3 (ToS) Priority Control

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

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

The UNIVERGE SV8100 system can set the ToS value for each protocol, and Voice Control. This setting allows flexible prioritization.

12.2.1   Programming Layer 3 Priority Control

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

➲   **84-10-01   ToS Setup – ToS Mode**

For the Protocol type (1 = Not Used, 2 = Not Used, 3 = Voice Control, 4 = H323, 5 = RTP/RTCP, 6 = SIP, 7 = CCISoIP, 8 = DT700 MLT, 9 = SIP Trunk, 10 = NetLink), select the ToS mode (Disable, IP Precedence or Diffserv).

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

For the Protocol type (1 = Not Used, 2 = Not Used, 3 = Voice Control, 4 = H323, 5 = RTP/RTCP, 6 = SIP, 7 = CCISoIP, 8 = DT700 MLT, 9 = SIP Trunk, 10 = NetLink), if Program 84-10-01 is set to 2, select the Diffserv priority (0~63).

SECTION 13   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:

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

SECTION 14   SYSTEM **IP** PHONES AND ANALOG TRUNKS

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

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

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

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

SECTION 15   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.)

15.1   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.

15.2   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.

15.3   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:

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.   Set the server mode to FTP/TFTP in Program 84-07-01 and specify the
      IP address of the FTP/TFTP server (this is the PC where the FTP/TFTP
      software is installed) in Program 84-07-02.

SECTION 16   IP STATION (SIP MULTILINE TELEPHONE)

16.1   Description

The UNIVERGE SV8100 system supports IP phones using Session Initiation
Protocol (SIP). These IP telephones have the same look and functionality as
typical multiline telephones, but they are connected to an IPLA card in the
CPU rather than hardwired to a DLCA port.

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

❍   Reduced telephone re-location costs. Unplug an IP telephone at one
      location and plug it into another VoIP ready jack at another location. The
      extension number can stay with the telephone, if programmed to do so.

❍   Multiple users can share the same IP telephone but keep their own
      personal extensions. With an IP phone you have the ability to log the
      phone out, then log back in with another extension number. All of your
      personal settings follow the login ID of your extension.

❍   The cabling infrastructure can be simplified. There is no longer a need
      for separate cabling for the phone system. Built into the IP phones is a 2
      port 10/100 manageable data switch. The data connection for the PC is
      available on the back of the IP phone. This built in data switch also
      supports 802.1Q and 802.1 P VLAN tagging abilities. The data and the
      voice can be tagged separately.

❍   Allows you to connect remote office workers with a telephone that has
      the look and same functionality as the telephones connected at the
      office.

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

The SIP Multiline Telephone interfaces directly to the CCPU that houses the PZ-32IPLA/PZ-64IPLA/PZ-128IPLA daughter board. The IPLA provides the digital signal processors (DSPs) for IP stations and trunks.

❍   PZ-32IPLA provides 32 DSPs

❍   PZ-64IPLA provides 64 DSPs

❍   PZ-128IPLA provides 128 DSPs

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

The following are examples of DSP allocation.

❍   Calling from an IP phone to a TDM phone uses one DSP.

❍   Calling from an IP phone to another IP phone, registered to the same CPU, uses no DSPs.

❍   Calling from a TDM phone to another TDM phone uses no DSPs.

❍   Calling from a TDM phone and out an IP trunk uses one DSP.

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

16.2   IP Addressing

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.

One address is used for the IP phones to register and pass signaling information. The remaining two IP addresses are used for the voice traffic to and from the 32 DSP channels.

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

Card	Minimum IP Addresses	Maximum IP Addresses	Notes
PZ-32IPLA	2	3	One IP address is required for 16 DSP channels. One additional IP Address is required for registration and signaling.
PZ-64IPLA	2	5	
PZ-128IPLA	2	9	

✎   When assigning the IP addresses to the IPLA 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 an IPLA 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 NIC. E.g. PCPro, Web Pro, ACD, etc. all terminate to the IPLA NIC card when installed.

The IPLA 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, IPLA or CPU, to route outside of its own network.

16.3   IP Phone Registration Modes

The SIP MLT supports three different registration modes.

❍   Plug and Play

❍   Automatic Login

❍   Manual Login

Plug and Play Registration

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

Automatic Login

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

Manual Login

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

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

16.4 General IP Configuration

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

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

❍   One way delay must not exceed 100ms.

❍   Round trip delay must not exceed 200ms.

❍   Packet loss must not exceed 1%.

❍   Data switches must be manageable.

❍   Routers must provide QOS.

❍   There must be adequate bandwidth for estimated VoIP traffic. Refer to Section 16.7 Bandwidth on page 8-52.

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.

16.5 VLANs

A VLAN is used to logically break up the network and minimize broadcast domains. Without VLANS, the network must be physically segmented to break up broadcast domains. Each network segment is then connected through a

routing device adding latency and cost. Latency is a delay in the transmission of data and is caused by routing packets from one LAN to another. In a VoIP environment latency must be kept to a minimum.

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

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

Currently the IPLA 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 is plugged into a data switch supporting VLANS, the VLAN tag must be removed before passing the frame onto the CPU/IPLA.

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.

16.5.1   Logging In on the PC

1.   Web browse to the IP address of phone.

2.   To log in, enter default user name: **ADMIN**

3.   Enter default password: **6633222**
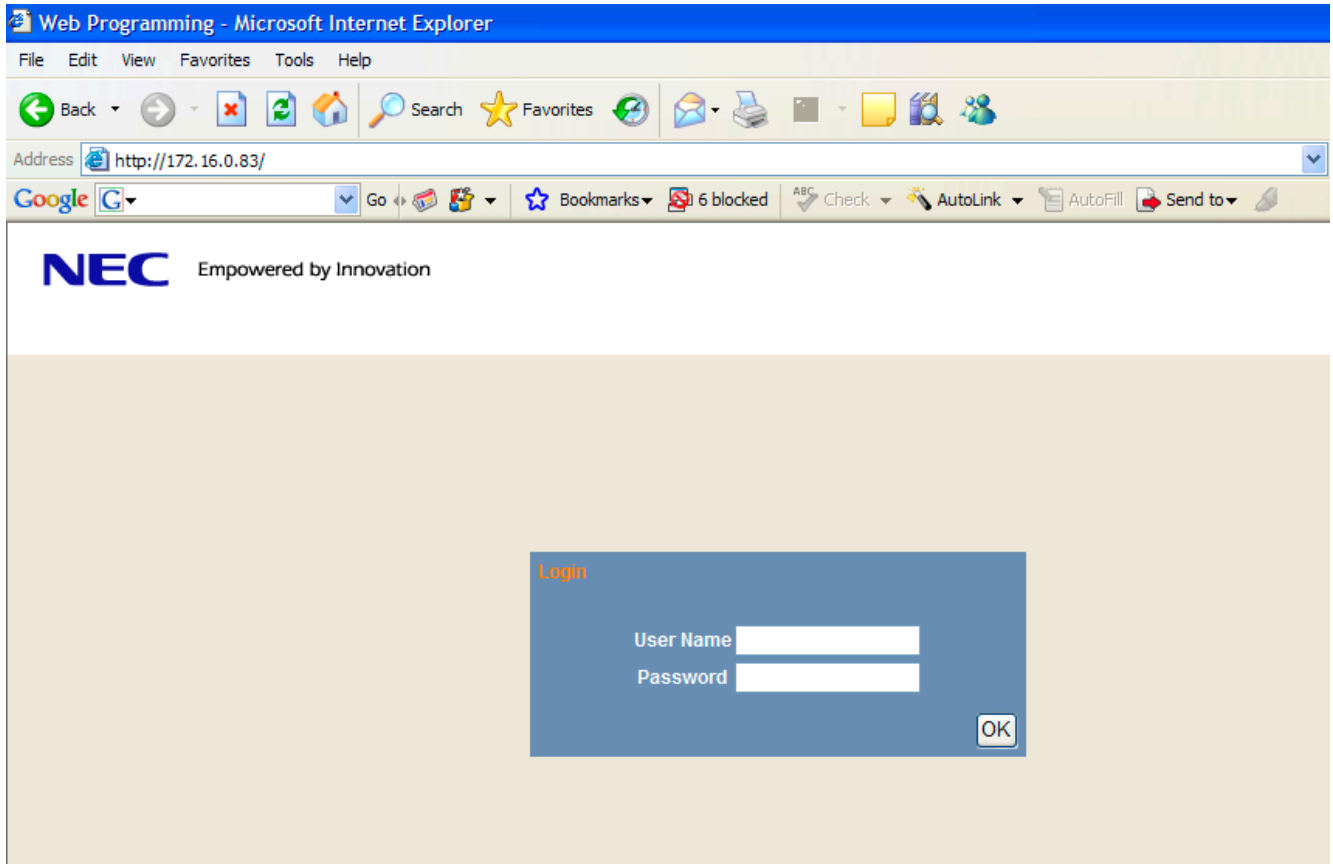
4.   Click **OK**.

Figure 8-18 Log In to IP Phone

16.5.2   Tagging Voice Packets Using IP Phone

1.   Log in. Refer to Section 16.5.1 Logging In on the PC on page 8-36.

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:

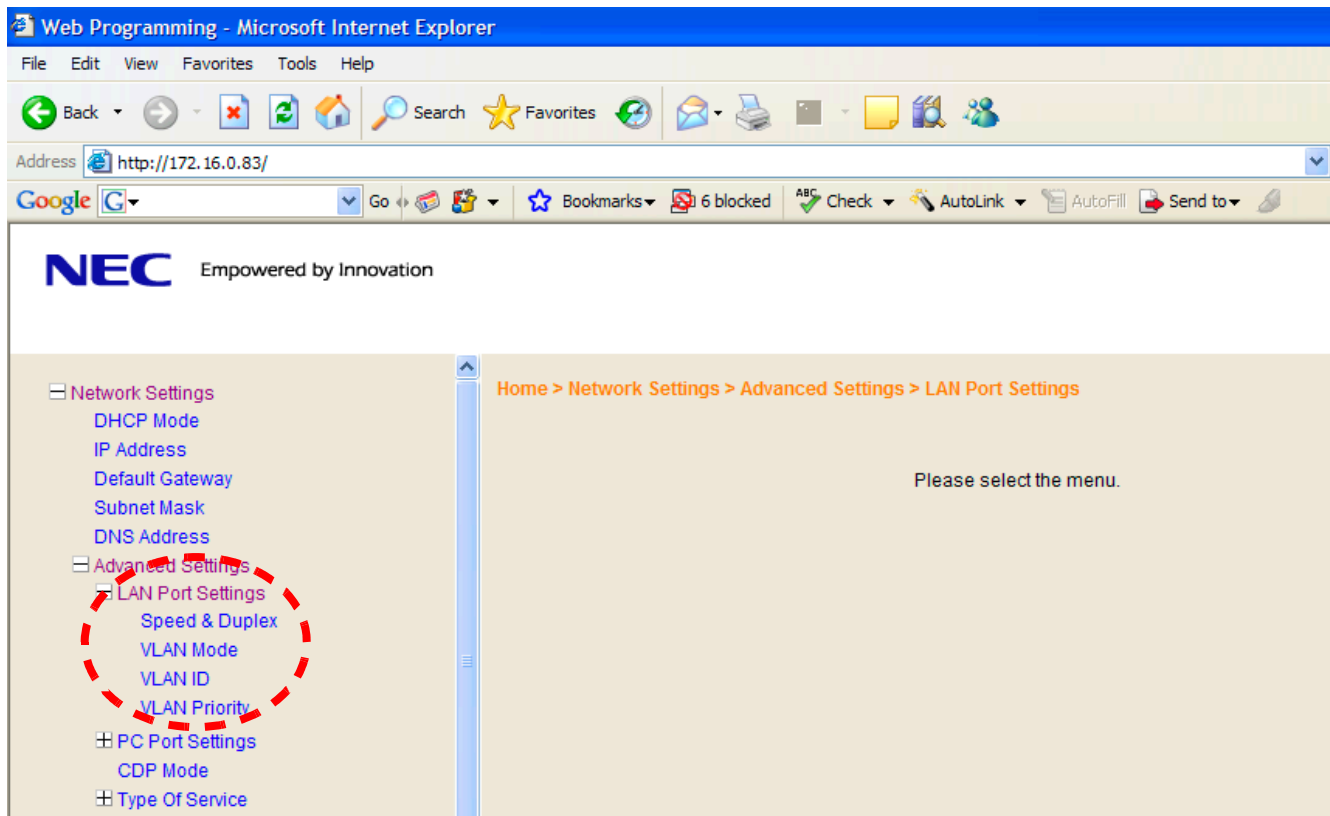   ●   VLAN Mode

   ●   VLAN ID

   ●   VLAN Priority

Figure 8-19 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**.
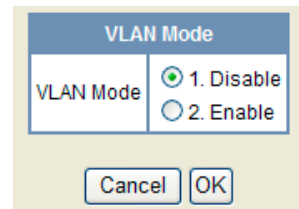
Figure 8-20 VLAN Mode

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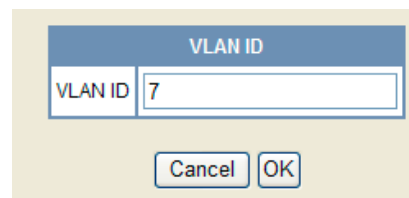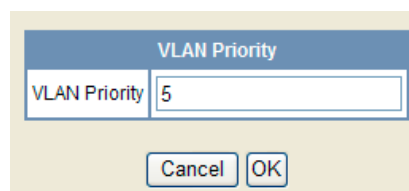
	Enter the VLAN ID and click **OK**.

Figure 8-21 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 8-22 VLAN Priority

16.5.3   Tagging Data Packets Using IP Phone

1.   While logged in to the IP address of the phone on the PC, go to **Network Settings>Advanced Settings>PC Port Settings**. Refer to Section 16.5.1 Logging In on the PC on page 8-36.

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

●   Port VLAN Mode

●   Port VLAN ID
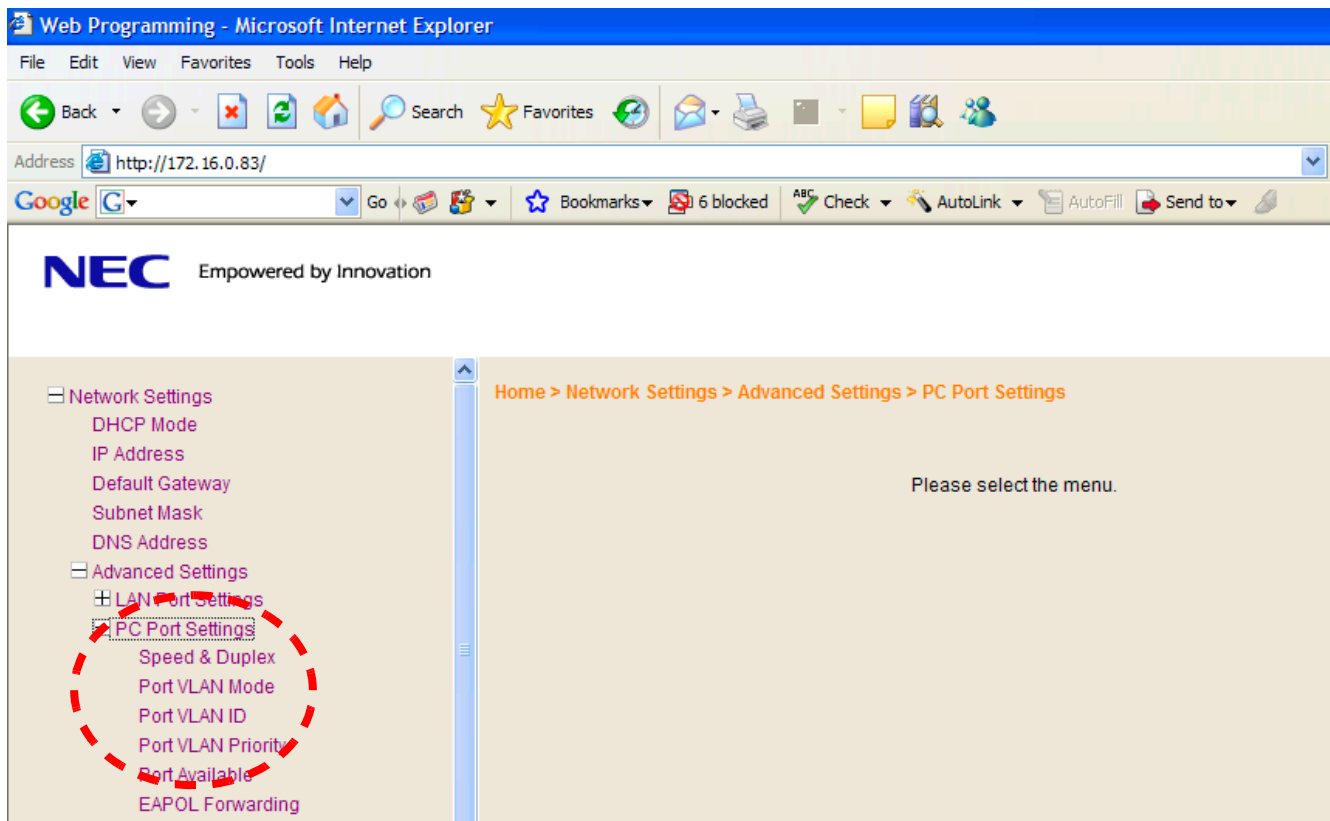
●   Port VLAN Priority.

Figure 8-23 PC Port Settings Window

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

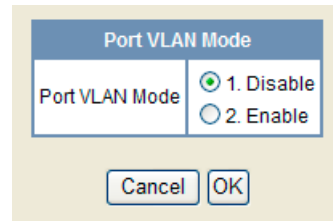4.   Select either Enable or Disable (default) and click **OK.**

Figure 8-24 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.
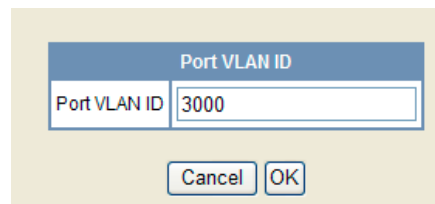
      Enter the VLAN ID, and click **OK**.

Figure 8-25 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**.
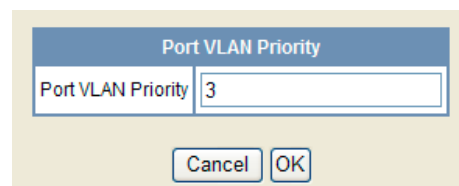

Figure 8-26 Port VLAN Priority

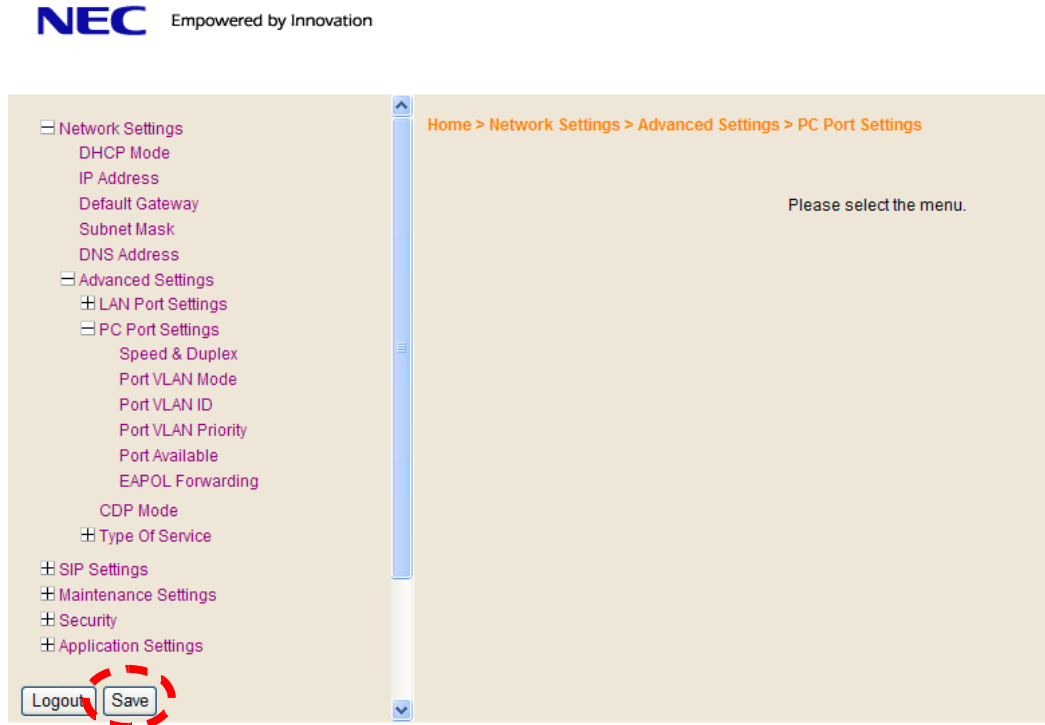7.   Click **Save**.

Figure 8-27 Save Network Settings

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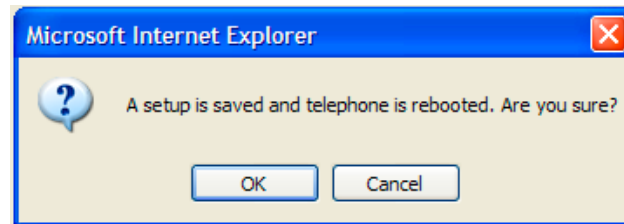   and applies the VLAN settings.

Figure 8-28 Save Confirmation Window

16.5.4   Entering VLAN Settings by Phone (Voice Packets Only)

1.   Log in. Refer to Section .

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.

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

1.   Log in. Refer to Section .

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.

16.6   ToS Settings (Layer 3 QoS)

The marking of packets at layer 3 is done by marking the ToS byte in the IP header of the voice packet. The 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 card. This does not affect the packets sent from the IP terminals.

Figure 8-29 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 8-30 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
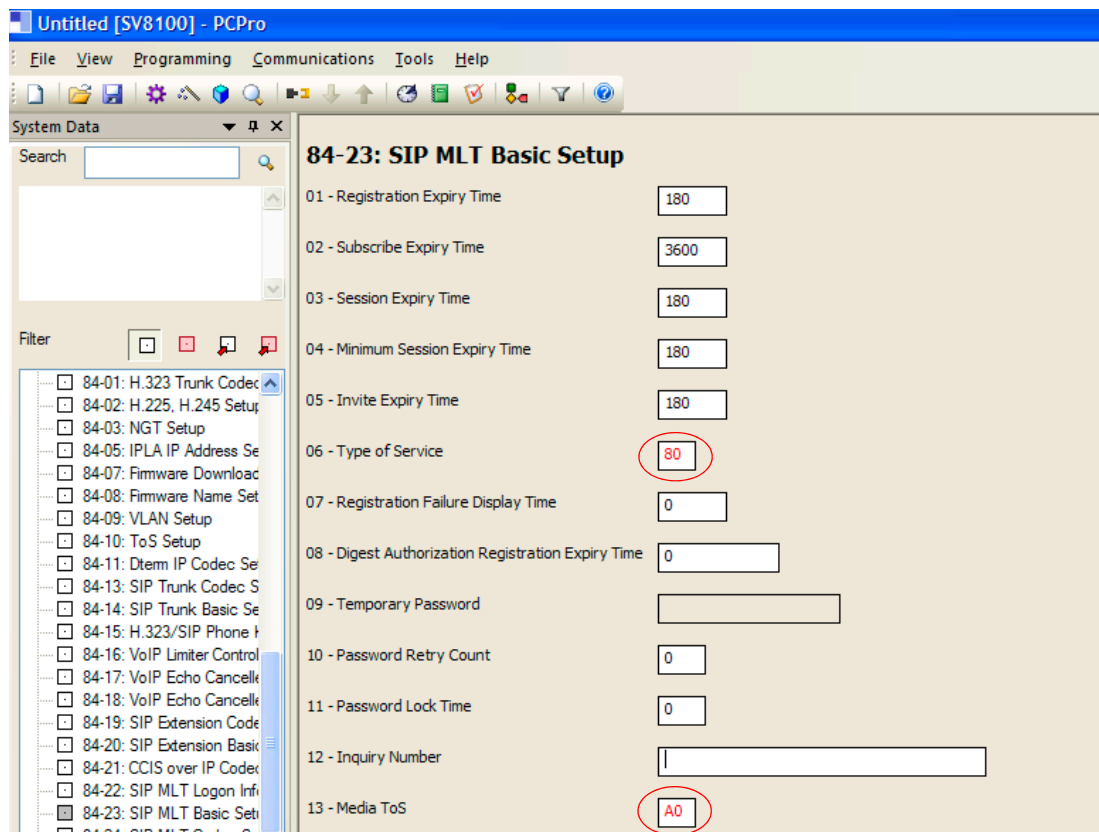
Figure 8-30 SIP MLT Basic Setup

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

Table 8-2 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 16.5.1 Logging In on the PC on page 8-36.

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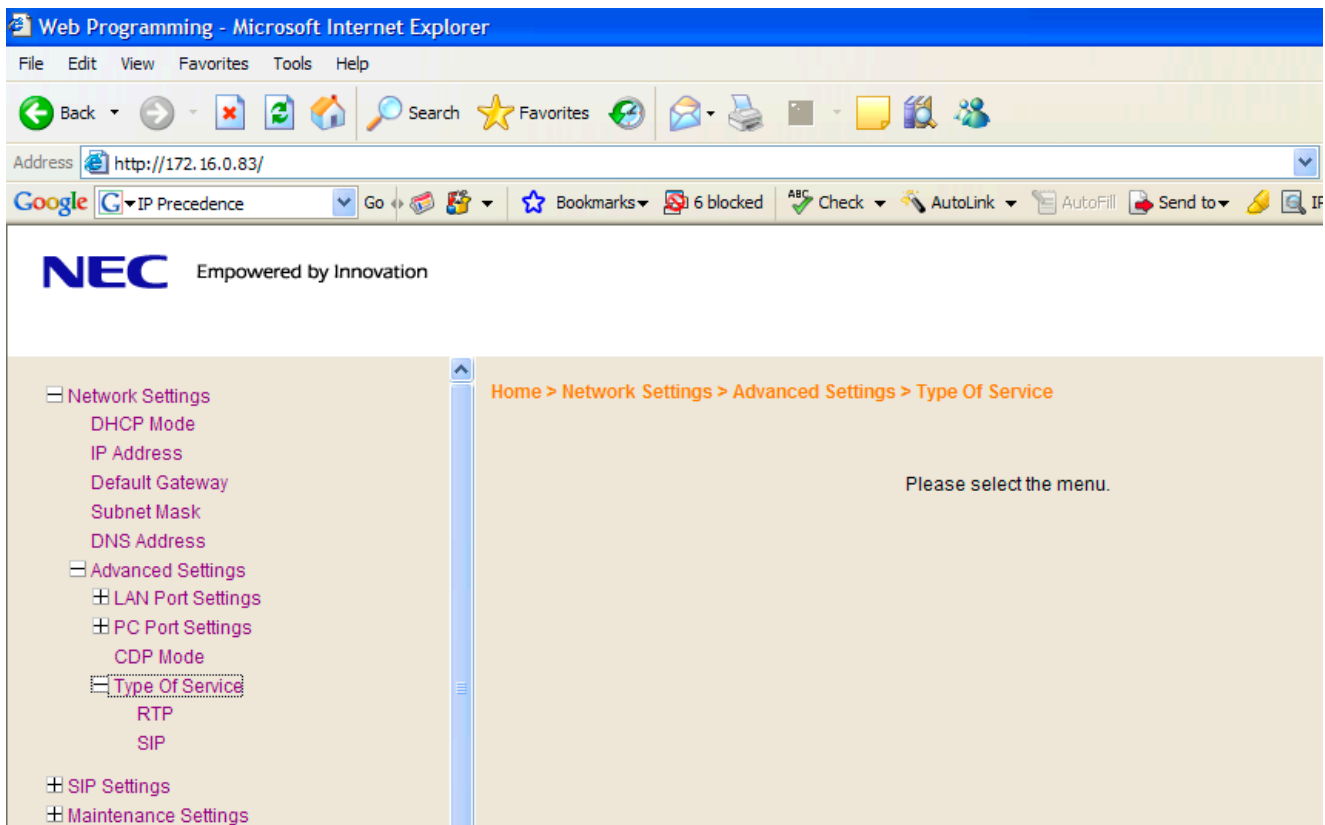

Figure 8-31 Type of Service Window

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 8-2 Common IP Precedence/Diffserv Values and Hexadecimal Equivalent on page 8-49.

Access the following menus to select options:

- RTP - Voice Packets
- SIP - Signalling Packets
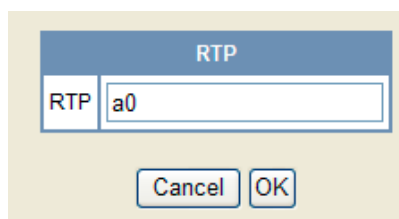
Figure 8-32 RTP - Voice Packets

Figure 8-33 SIP - Signalling Packets
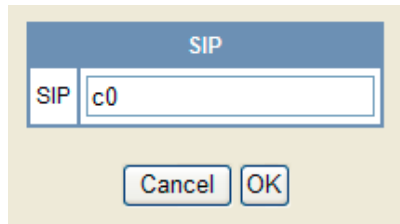
4.   After selecting **Save**, the following message appears.

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.

16.7 Bandwidth

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

❍ Layer 2 media

❍ CODEC

❍ Packet Size

❍ RTP Header Compression

❍ Voice Activity Detection (VAD)

❍ Number of simultaneous calls

❍ 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 compression

Compressed Header 2 ~ 4 bytes	VOICE PAYLOAD

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

Bandwidth Calculations

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

❍   G.711 = 64000bps

❍   G.722 = 64000bps

❍   G.729 = 8000bps

Payload Calculation Voice

❍   (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

16.8   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.

Multiple Gateways

For example, the diagram below shows the SV8100 assigned a default
Gateway of 192.168.1.1/24. The existing network includes two routers in the
same broadcast domain connecting different networks. With regular data
communications, this is not an issue. However the VoIP cannot function
correctly in this type of network environment without specific assignments.
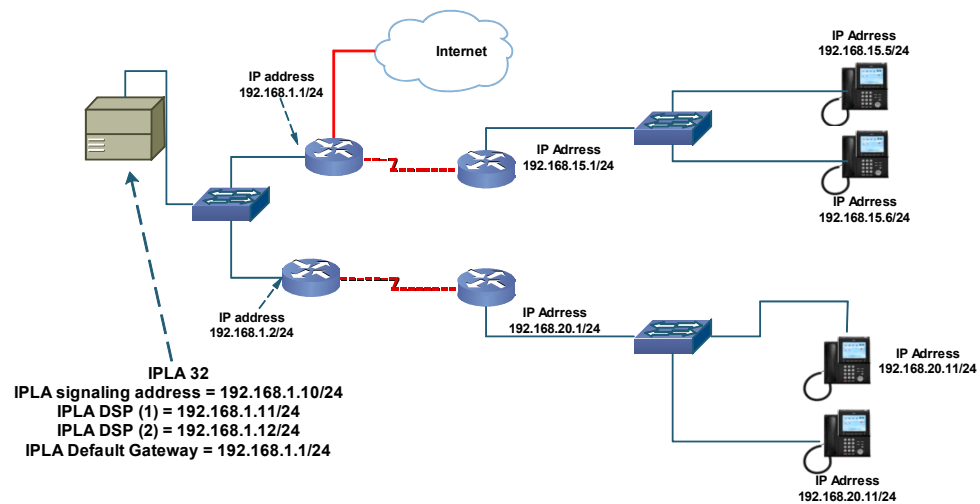
Figure 8-34 SV8100 Network Example No. 1

The above allows for normal operation of IP phones in the 192.168.15.0
network but 192.168.20.0 network IP Phones would have considerable issues.
Symptoms would include one way, or no speech path, cut off, to name a few.
If the default gateway for the SV8100 was changed to 192.168.1.2 then the
reverse would occur with the 192.168.15 network experiencing issues and the
192.168.20.0 network functioning correctly.

The new VoIP routing table feature (available on the SV8100 only) allows for
the assignment of tables to route VoIP packets to different gateway
addresses. In normal operation the SV8100 first checks its VoIP routing table
for a match, if one is found that route is selected. If no match is found, the
default gateway is then used.

The following is an example of a VoIP routing table allowing operation for all the IP phones in

System Data

| | | Grid View | Apply | Cancel | Default | Copy |

10-45: IP Routing Table Setup

Route Table (1~100) 1

Route Table	IP Address	Subnet Mask	Default Gateway
001	192.168.20.0	255.255.255.0	192.168.1.2
002	0.0.0.0	128.0.0.0	0.0.0.0
003	0.0.0.0	128.0.0.0	0.0.0.0
004	0.0.0.0	128.0.0.0	0.0.0.0
005	0.0.0.0	128.0.0.0	0.0.0.0
006	0.0.0.0	128.0.0.0	0.0.0.0
007	0.0.0.0	128.0.0.0	0.0.0.0
008	0.0.0.0	128.0.0.0	0.0.0.0
009	0.0.0.0	128.0.0.0	0.0.0.0

Figure 8-35 VoIP Routing Table Example

Route table 001 has an IP address of 192.168.20.0 and a subnet mask of 255.255.255.0. This states that any IP address that is destined to 192.168.20.1 ~ 192.168.20.255 is routed to the IP address of 192.168.1.2. There is no need to add a statement for the 192.168.15.0 network because if no match is found in this table, the packets destined for a distant network are routed to the default gateway.

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.
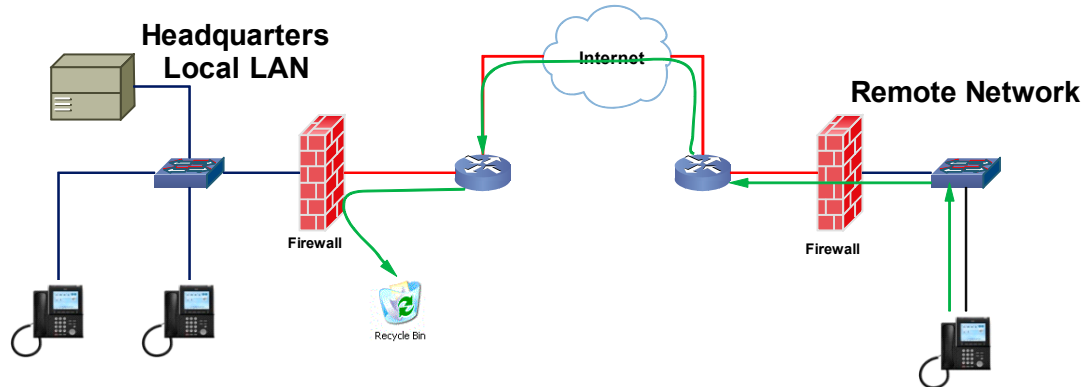
Figure 8-36 SV8100 Network Example No. 1

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 ports are installed.

❍   IPLA 32 open UDP ports **10020 ~ 10083**

❍   IPLA 64 open UDP ports **10020 ~ 10147**

❍   IPLA 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.
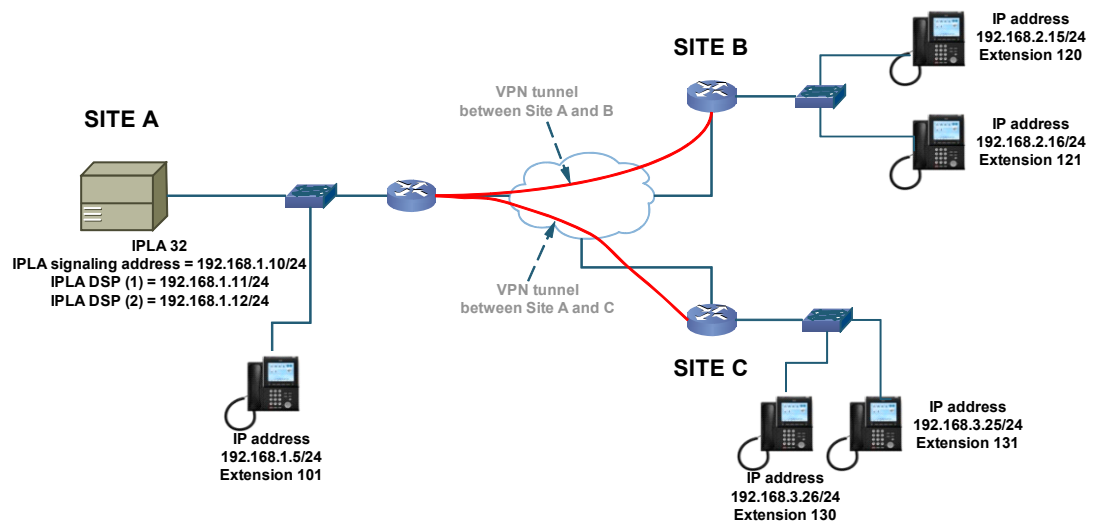
Figure 8-37 SV8100 Network Example No. 3

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).

❍   192.168.1.10 (IPLA) 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). 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) 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).

❍   192.168.1.10 (IPLA) 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 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 8-38 IP System Operation Setup

16.9   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 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 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 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	Assign Layer 3 IP addresses to each of the IPCA Gateways. There are 16 DSP resources to each Gateway. If IPLA 32 is installed first 2 Slots will require assignment. If IPLA64 is installed the first four slots will require assignment. If IPLA 128 is installed all eight Gateway addresses must be assigned.	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	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
15-05-17	IP Telephone Terminal Basic Data Setup – Calling Party Display Info	The part of the Invite message the calling party information is taken from. There are four choices: **Nickname:** Displays the nickname programmed in 15-05-04. **Display Name:** Some SIP phones contain a Field called Display Name. If configured this displays on a call from that station. **User Part**: Some SIP phones contain a field called User Part. If configured this displays on a call from that station. **Extension:** Display shows extension of the SIP phone	0 = Nickname 1 = Display Name 2 = User Part 3 = Extension Default is 0		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
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

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
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
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	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
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
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	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
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	
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	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
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
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	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
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	
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--					

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
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	
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

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
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
84-24-26	DT700 Multiline CODEC Basic Information Setup – TX Gain	Gain setting for IPLA 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 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	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
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
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	

16.10 SIP MLT Quick Startup Guide

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

❍   Plug and Play

❍   Automatic Registration

❍   Manual Registration

16.10.1   Plug and Play

1.   Program 10-12
     Assign the IPLA 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.
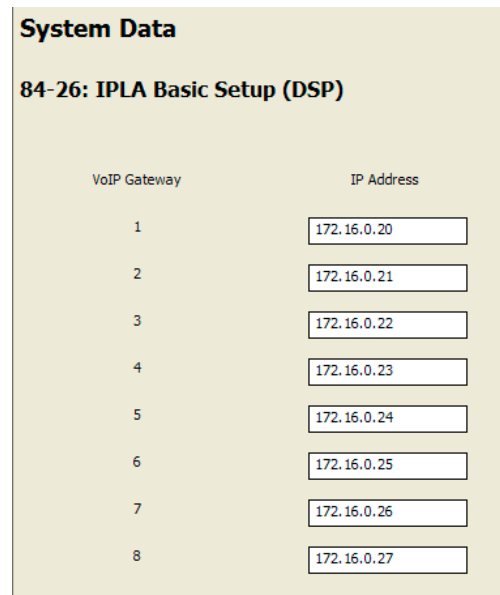

Figure 8-39 System Data 10-12: CD CP00 Network Setup

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. Each DSP provides 16 paths from IP to TDM.

Figure 8-40 System Data 84-26: IPLA 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 8-41 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 PC-Pro, the extensions are available for selection in Program 15 and other station related Programs.

Figure 8-42 PCPro Programming Unregistered Phones

Figure 8-43 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.

16.10.2 Automatic Registration

1.   Steps 1 ~3 are the same as for Plug and Play mode. Step 4 is not optional and MUST be assigned when using Automatic Registration.

2.   Same as Plug and Play mode.

3.   Same as Plug and Play mode.

4.   To enable key data and other station feature programming before IP Phone is brought online, the extensions must be identified as IP Phones. Once checked in the IP Phone List in PC-Pro (see images below), the extensions are available for selection in Program 15 and other station related Programs.
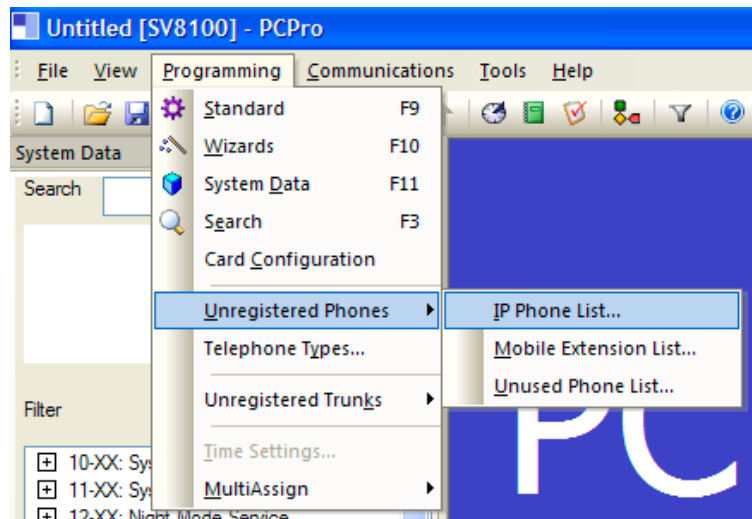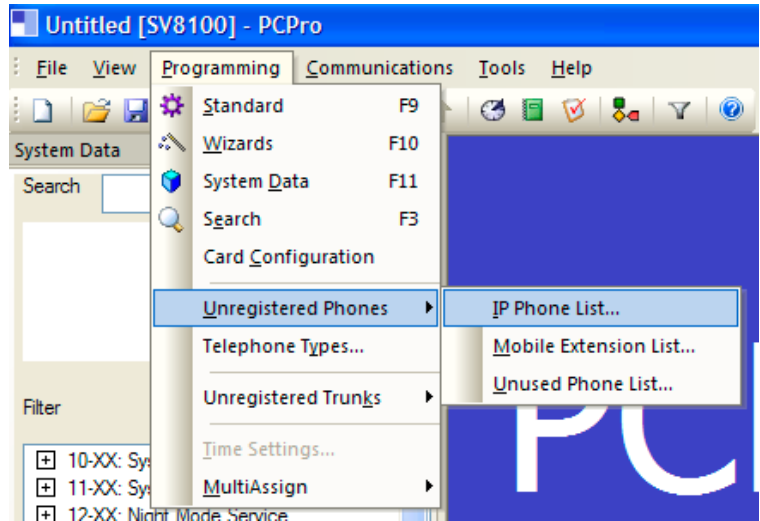

Figure 8-44 PCPro Programming Unregistered Phones
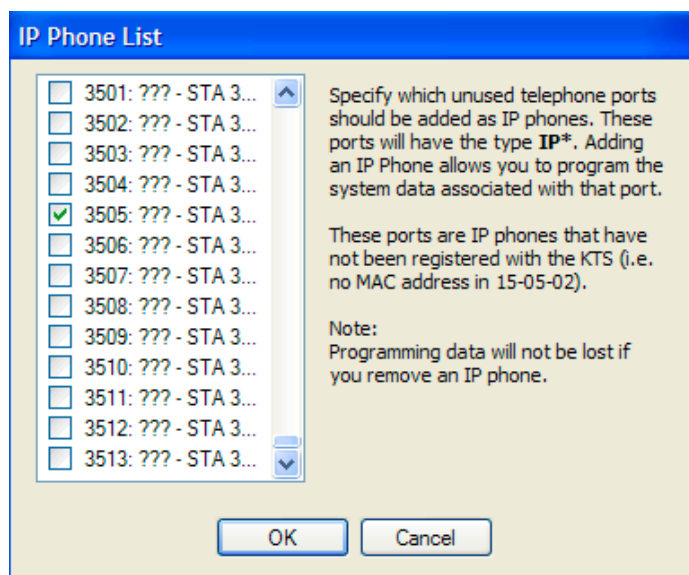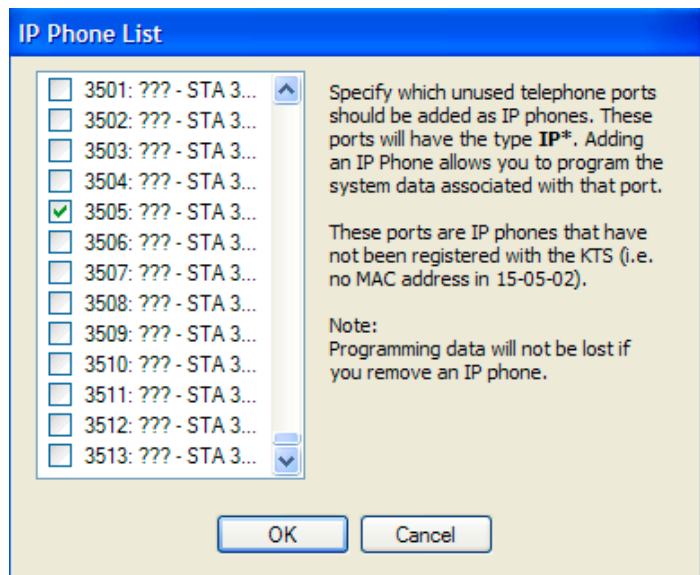

Figure 8-45 IP Phone List

5.   Program 10-46

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

System Data

10-46: SIP Multi-Line Telephone Server Information Setup

01 - Register Mode   Automatic

Figure 8-46 Automatic Registration Setting

6.   Program 15-05-27

Each IP phone requires a unique personal ID index. Valid settings are
1 ~ 512.

System Data   Grid View   Apply   Cancel   Default

15-05: IP Phone Basic Setup

Extension   3505: IP* - STA 3505 - 0.0.0.0 - Port 504

Figure 8-47 Automatic Registration Basic Setup

27 - Personal ID Index   1
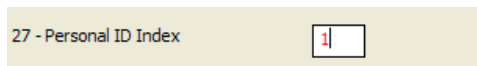
Figure 8-48 Automatic Registration Personal ID Index

7.   Program 84-22-01

Assign the user ID and password to be associated with the Personal ID Index assigned in Step 6.

Figure 8-49 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.

16.10.3 Manual Registration

Steps 1~4 are the same as for Section .

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**.

System Data

10-46: SIP Multi-Line Telephone Server Information Setup

01 - Register Mode   Manual
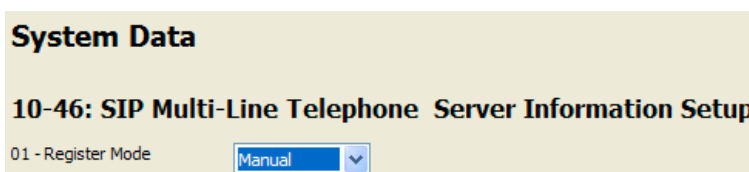
Figure 8-50 Manual Registration

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.

Hold Transfer ✱ #

✎ *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.

16.11 IP Phone Relocation

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

IP Phone Relocation Override

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

Conditions

❍   Multiple IP Phones cannot use the same user ID and the same password at the same time.

❍   When a user is using multiple IP Phones at the same time, the user ID and password must be different for each phone.

❍   When a user is using SoftPhone (CTI mode) and controlling the IP Phone by this SoftPhone, the user ID and password should be different for the SoftPhone and IP Phone.

❍   An IP Phone (IP Phone and Soft phone) with DSS console cannot override another IP Phone.

❍   An IP Phone (IP Phone and Soft phone) with DSS console cannot be overridden from another IP Phone.

❍   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 8-3 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.
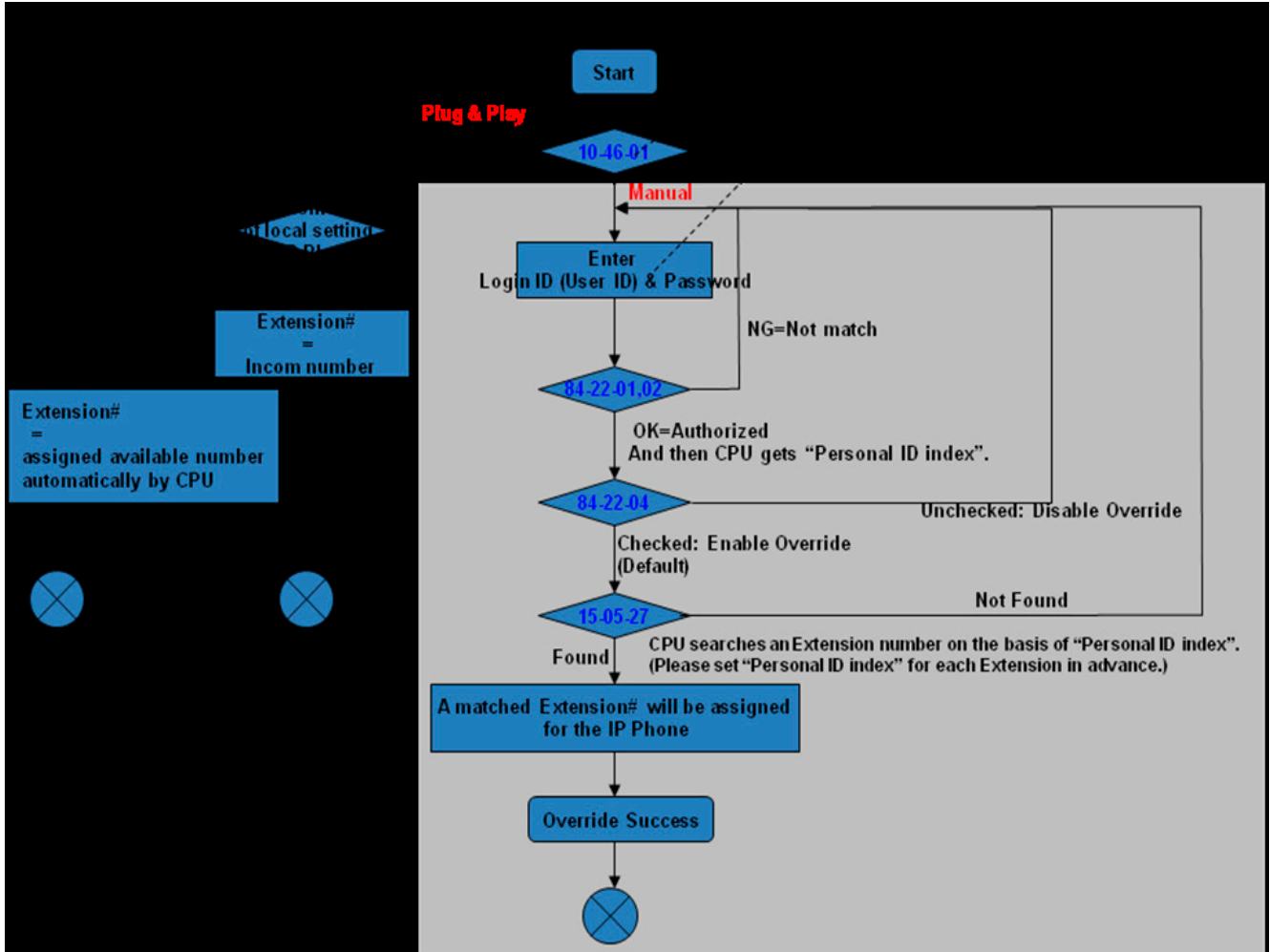
Figure 8-51 IP Phone Relocation Flow Chart

Third-Party SIP

SECTION 1   INTRODUCTION

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

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

The PZ-(X)IPLA is offered in three different types:

❍   PZ-32IPLA – 32 DSP Resources

❍   PZ-64IPLA – 64 DSP Resources

❍   PZ-128IPLA – 128 DSP Resources

With the PZ-128IPLA up to 128 TDM talk paths are supported. This total may be shared among SIP stations or SIP trunks. Registered SIP stations and/or SIP trunks require a one-to-one relation with the PZ-(X)IPLA DSP Resource. This is a required component of SIP implementation in the UNIVERGE SV8100. The UNIVERGE SV8100 PZ-(X)IPLA contains a regular TCP/RTP/IP stack that can handle real-time media and supports industry standard SIP (RFC3261) communication on the WAN side.

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

A maximum of (1) PZ-(X)IPLA can be installed per UNIVERGE SV8100 System, supporting the maximum of 512 IP stations.

The UNIVERGE SV8100 supports the following codecs that are considered to provide toll-quality equivalent speech path.

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

❍   G.729. Low bandwidth requirement used on most Wide Area Network links.

❍   G.711. μ Law – High bandwidth requirement usually used on Local Area Networks.

❍   G.723. Low bandwidth requirement not commonly used.

❍   G.722 This CODEC is useful in fixed network, Voice over IP applications, where the required bandwidth is typically not prohibitive.

❍   G.726 is an ITU-T ADPCM speech coded standard covering the transmission of voice at rates of 16, 24, 32, and 40Kbps.

❍   ILBC (Internet Low Bitrate CODEC) Narrow band speech. ILBC enables graceful speech quality degradation in the case of lost frames, which occurs in connection with lost or delayed IP packetsG.711 uLaw - Highest Bandwidth.

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

❍   Draft-IETF-sipping-service-examples-09.txt.

❍   Draft-IETF-sipping-cc-transfer-05.txt.

❍   IETF RFC is defined as: Internet Engineering Task Force (RFC) Request for Comments.

❍   The SIP Station feature set supports the Message Waiting Indication (MWI) based on RFC3842.

SECTION 2   PROGRAMMING

2.1   Blade Setup

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
10-03-01	ETU Setup – IPLA Pkg DB	Trunk Logical Port Number	0~200 READ ONLY Default is 0	X		
10-03-02	ETU Setup – VoIP DB	Trunk Type	0 = H.323 1 = SIP Default is 1	X		

2.2   Configure VoIPDB Networking Information

The VoIPDB should be connected to the same IP Subnet as the CD-CP00-US. It also requires a valid IP address.

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

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
10-12-03	CD-CP00-US Network Setup – Default Gateway	Assign the IP Address for the Router.	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 the IP Address of the VoIPDB.	0.0.0.0 ~ 126.255.255.254 128.0.0.13~191.255.255.254 192.0.0.1~223.255.255.254 Default is 172.16.0.10	X		
10-12-10	CD-CP00-US Network Setup – Subnet Mask	Assign the Subnet Mask of the VoIPDB.	255.255.0.0 Default is 255.255.0.0	X		
10-40-01	IP Trunk Availability – IP Trunk Availability	Assign the SIP Trunk availability.	0 = Disable 1 = Enable Default is 0	X		
10-40-02	IP Trunk Availability – Number of Ports	Assign the number of SIP trunks.	0~128 Default is 0	X		
10-40-04	IP Trunk Availability – Number of IP CCIS Ports	Assign the number of IP – K-CCIS trunks.	0~128 Default is 0	X		

2.2.1   VoIPDB (DSP) Basic Setup

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
84-26-01	IPL Basic Setup – IP Address	Assign IP Addresses for each DSP resource to be used. IP Addresses must be assign in groups of two and one IP Address must be assigned for every 16 DSP Resources used. Note: The VoIPDB IP Addresses must be statically assigned.	Default: 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	Assign the RTP Port Number.	0~65534 Default for DSP1 is 10020		X	
84-26-03	IPL Basic Setup – RTCP Port Number (RTCP Port Number + 1)	Assign the RTCP Port Number.	0~65534 Default for DSP1 is 10021		X	

2.2.2   VoIP ToS Setup

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

2.2.3   SIP Peer to Peer

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
10-26-01	IP System Operation Setup – Peer to Peer Mode	Disabling 10-26-01 results in SIP MLT Station-to-SIP MLT Station calls using a DSP resource.	0 = Off 1 = On Default is 1		X	
10-26-02	IP System Operation Setup – RTP Forwarding Mode		0 = Disable 1 = Enable Default is 0		X	
10-26-03	IP System Operation Setup – SIP Peer to Peer Mode		0 = Off 1 = On Default is 1		X	

2.2.4   IP Extension Numbering

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
11-01-01	System Numbering	Refer to the SV8100 Programming Manual for all options and default settings.	1 = 3 Digit; Intercom Default is 3 Digit; Intercom	X		
11-02-01	Extension Numbering	Assign up to eight digits for extension numbering	Dial (Up to eight digits) Default Ports 1~99 = 101~199 Ports 100~512 = 3101~3513	X		

2.2.5   SIP Extension CODEC Information

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
84-19-01	SIP Extension CODEC Information Basic Setup – Number of G.711 Audio Frames	Define the G.711 audio Frame Size	1~4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms Default is 2			
84-19-02	SIP Extension CODEC Information Basic Setup – G.711 Voice Activity Detection Mode	Enable/Disable Voice Activity Detection for G.711.	0 = Disable 1 = Enable Default is 0			
84-19-03	SIP Extension CODEC Information Basic Setup – G.711 Type	Define the G.711 Type – μ-law is recommended in USA.	0 = A-law 1 = μ-law Default is 1			
84-19-04	SIP Extension CODEC Information Basic Setup – G.711 Jitter Buffer (min)	Define G.711 Jitter Buffer minimum accepted value.	0~160ms Default is 20			
84-19-05	SIP Extension CODEC Information Basic Setup – G.711 Jitter Buffer (Average)	Define G.711 Jitter Buffer setting.	0~160ms Default is 40			
84-19-06	SIP Extension CODEC Information Basic Setup – G.711 Jitter Buffer (max)	Define G.711 Jitter Buffer maximum accepted value.	0~160ms Default is 80			
84-19-07	SIP Extension CODEC Information Basic Setup – Number of G.729 Audio Frames	Define the G.729 audio Frame Size.	1~6 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms 5 = 50ms 6 = 60ms Default is 2			
84-19-08	SIP Extension CODEC Information Basic Setup – G.729 Voice Activity Detection Mode	Enable/Disable Voice Activity Detection for G.729.	0 = Disable 1 = Enable Default is 0			
84-19-09	SIP Extension CODEC Information Basic Setup – G.729 Jitter Buffer (min)	Define G.711 Jitter Buffer minimum accepted value.	0~270ms Default is 20			

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
84-19-10	SIP Extension CODEC Information Basic Setup – G.729 Jitter Buffer (average)	Define G.729 Jitter Buffer setting.	0~270ms Default is 40			
84-19-11	SIP Extension CODEC Information Basic Setup – G.729 Jitter Buffer (max)	Define G.729 Jitter Buffer maximum accepted value.	0~270ms Default is 80			
84-19-17	SIP Extension CODEC Information Basic Setup – Jitter Buffer Mode	Define the Jitter Buffer mode – supported Static or Immediate.	1 = Static 2 = Adaptive during Silence 3 = Adaptive Immediately Default is 3			
84-19-18	SIP Extension CODEC Information Basic Setup – VAD Threshold	Define the VAD Threshold.	0~30 Default is 20			
84-19-26	SIP Extension CODEC Information Basic Setup – TX Gain	Define TX Gain Values. Consult the SV8100 Programming Manual to set acceptable values.	0~40 (-20dBm ~+20dBm) 0 = -20dBm 1 = -19dBm : 20 = 0dBm : 39 = +19dBm 40 = +20dBm Default is 20			
84-19-27	SIP Extension CODEC Information Basic Setup – RX Gain	Define RX Gain Values – Adjusting this value increases or decreases volume levels for the sending party. Consult the SV8100 Programming Manual for Transmit Gain scale to set acceptable value.	0~40 (-20dBm ~ +20dBm) 0 = -20dBm 1 = -19dBm : 20 = 0dBm : 39 = +19dBm 40 = +20dBm Default is 20			

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
84-19-28	SIP Extension CODEC Information Basic Setup – Audio Capability Priority	Define Audio capability priority.	0 = G.711_PT 1 = G.723_PT 2 = G.729_PT 3 = G.722 4 = G.726 5 = Not Used Default is 0			
84-19-31	SIP Extension CODEC Information Basic Setup – DTMF Payload Number	Define the DTMF Payload Number.	96~127 Default is 96			
84-19-32	SIP Extension CODEC Information Basic Setup – DTMF Relay Mode		0 = Disable 1 = RFC2833 Default is 0			
84-19-33	SIP Extension CODEC Information Basic Setup – Number of G.722 Audio Frames		1~4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms Default is 3			
84-19-34	SIP Extension CODEC Information Basic Setup – G.722 Voice Activity Detection Mode		0 = Disable 1 = Enable Default is 0			
84-19-35	SIP Extension CODEC Information Basic Setup – G.722 Jitter Buffer (min)		0~160ms Default is 30			
84-19-36	SIP Extension CODEC Information Basic Setup – G.722 Jitter Buffer (Average)		0~160ms Default is 60			
84-19-37	SIP Extension CODEC Information Basic Setup – G.722 Jitter Buffer (max)		0~160ms Default is 120			
84-19-38	SIP Extension CODEC Information Basic Setup – Number of G.726 Audio Frames		1~4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms Default is 3			
84-19-39	SIP Extension CODEC Information Basic Setup – G.726 Voice Activity Detection Mode		0 = Disable 1 = Enable Default is 0			

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
84-19-40	SIP Extension CODEC Information Basic Setup – G.726 Jitter Buffer (min)		0~160ms Default is 30			
84-19-41	SIP Extension CODEC Information Basic Setup – G.726 Jitter Buffer (Average)		0~160ms Default is 60			
84-19-42	SIP Extension CODEC Information Basic Setup – G.726 Jitter Buffer (max)		0~160ms Default is 120			
84-19-43	---Not Used---					
84-19-44	---Not Used---					
84-19-45	---Not Used---					
84-19-46	---Not Used---					
84-19-47	---Not Used---					
84-19-48	---Not Used---					

2.2.6   SIP Extension Basic Information Setup

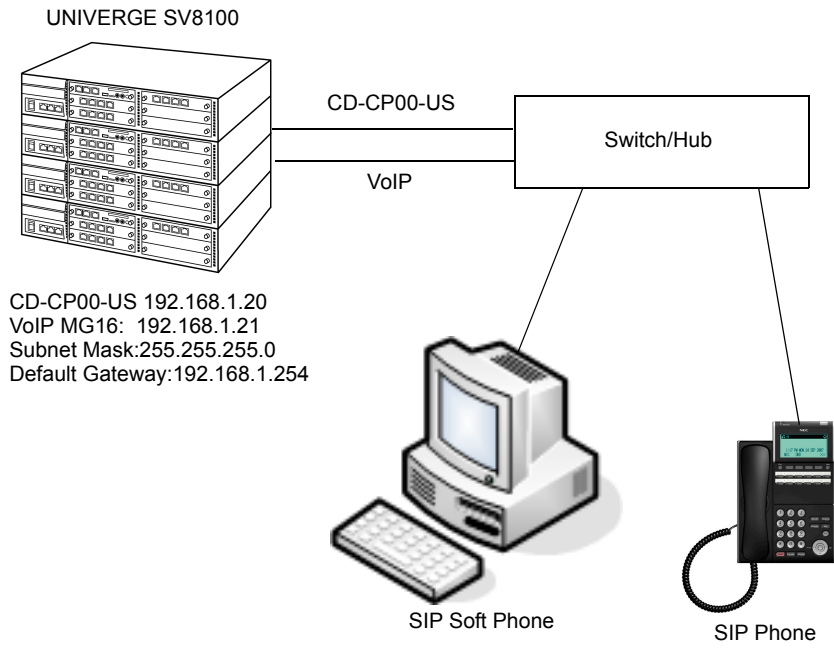| Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
84-20-01	SIP Extension Basic Information Setup – Registrar/Proxy Port	Define SIP station Proxy Port.	1~65535 Default is 5070			
84-20-02	SIP Extension Basic Information Setup – Session Timer Value	Define the periodic refresh time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

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
84-20-05	SIP Extension Basic Information Setup – Expire Value of Invite	Define the time out response value for SIP invite.	0~3600 seconds Default is 180			

2.2.7   IP Phone Configuration

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
15-05-01	IP Telephone Terminal Basic Data Setup – Terminal Type	Review the type protocol support by the IP Phone. Viewing Only – No changes permitted.	0 = NGT 1 = H.323 2 = SIP 3 = MEGACO 4 = SIP-MLT READ ONLY Default = 0			
15-05-02	IP Telephone Terminal Basic Data Setup – IP Phone Fixed Port Assignment	MAC Address of registered MLT SIP phone is stored and/or can input the MAC address of an MLT SIP phone so when it comes online it is provided with the extension in which the MAC address matches.	MAC address 00-00-00-00-00-00 to FF-FF-FF-FF-FF-FF Default is 00-00-00-00-00-00			
15-05-07	IP Telephone Terminal Basic Data Setup – Using IP Address	Informational Only registered IP Phones	0.0.0.0~255.255.255.255 Default is 0.0.0.0			
15-05-15	IP Telephone Terminal Basic Data Setup – CODEC Type	Assign the CODEC Type of the MLT SIP.	1-Type 1 2-Type 2 3-Type 3 4-Type 4 5-Type 5 Default is 1			
15-05-16	IP Telephone Terminal Basic Data Setup – Password	Assign the authentication password for SIP single line telephones.	Up to 24 characters Default not assigned			

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
15-05-17	IP Telephone Terminal Basic Data Setup – Calling Party Display Info	The part of the Invite message the calling party information is taken from. There are four choices: Nickname: Displays the nickname programmed in 15-05-04. Display Name: Some SIP phones have a Field called Display Name. If configured, in the SIP phone, this displays on a call from that station. User Part: Some SIP phones have a field called User Part. If configured in the SIP phone, this displays on a call from that station. Extension: Display shows extension of the SIP phone.	0 = Nickname 1 = Display Name 2 = User Part 3 = Extension Default is 0			
15-05-18	IP Telephone Terminal Basic Data Setup – IP Duplication Allowed Group	If an adapter has one IP address coming into it but multiple extensions off of it, assign all extensions to a group so the CPU knows that the one IP address is assigned to multiple extensions.	0 = Not Used 1 = Group 1 2 = Group 2 3 = Group 3 4 = Group 4 5 = Group 5 6 = Group 6 7 = Group 7 8 = Group 8 9 = Group 9 10 = Group 10 Default is 0			

2.3   SIP Phone Example

UNIVERGE SV8100

CD-CP00-US

VoIP

Switch/Hub

CD-CP00-US 192.168.1.20
VoIP MG16: 192.168.1.21
Subnet Mask:255.255.255.0
Default Gateway:192.168.1.254

SIP Soft Phone

SIP Phone

NetLink

**10**

SECTION 1   INTRODUCTION

NetLink Networking provides a seamless connection, using an IP network, to join multiple SV8100 communication servers into what would appear to be a single communications server. With a unified numbering plan, users can access any extension in the network as if they were in the same location.

With NetLink Networking, multiple locations are linked, allowing only one operator and a shared voice mail in the network. Up to 16 nodes (locations) can be connected as part of the NetLink network.

With NetLink, one communication server has the main database and controls other communication servers. The communication server is called Primary System. All call control is basically done by the Primary System. All slots, trunks and stations belong to the Primary System. No need to consider the numbering plan in the other communication servers. It is easy to consider how to grab a trunk in another communication server. There is no feature limitation when using NetLink. Any terminal in any node operates the same as if they were located in the Primary System.

An added feature of NetLink is the Fail-Over feature. If the Primary System is turned off or disconnected, without the Fail-Over feature, all communication servers would stop working. With Fail-Over, the server locates one of the Secondary Systems as based on the SV8100 programming, and it takes over the Primary System database allowing the other linked nodes to continue functioning.

NetLink

Primary
System

IP Network

Secondary
System 1

Secondary
System 2
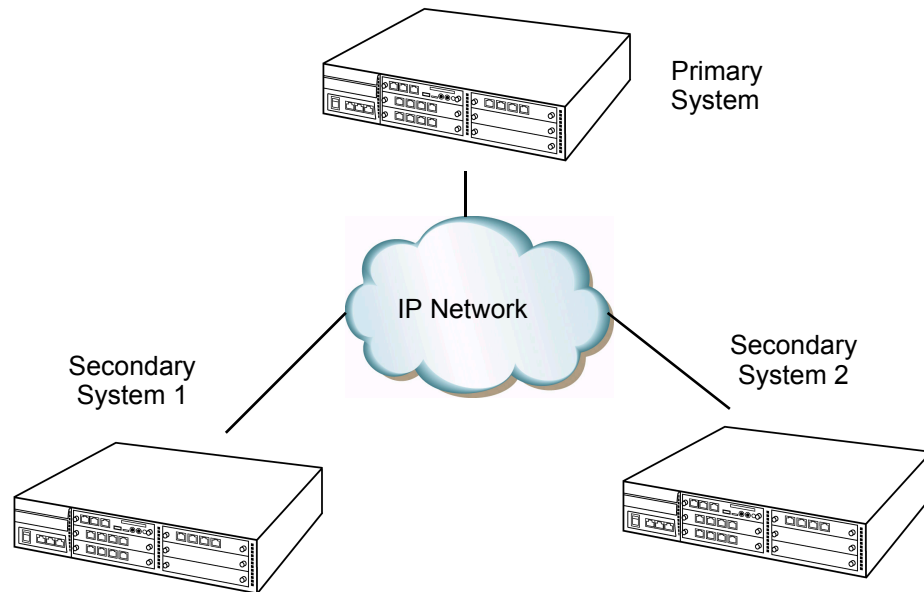
SECTION 2   SV8100 REQUIREMENTS

2.1   Main Software

NetLink is supported with the V1100/V1.12 or higher release of the SV8100.

2.1.1   Hardware

Each SV8100 Communications Server needs the following items:

- ❍   CD-CP00-US (V1100/V1.12 or higher)

- ❍   MEMDB

- ❍   VOIPDB

Install the station/trunk blades as required for your configuration.

Invalid data is displayed in the LCD of the terminal if Program 51-01 is enabled and no MEMDB is installed.

2.1.2   Capacity

- ❍   A maximum of 16 nodes (connected sites).

- ❍   512 Extensions/200 Trunk ports (as allowed by hardware and licensing).

2.1.3   License

  ❍ A NetLink License is required for the Primary System.

  ❍ All licenses must be registered into the Primary System.

  ❍ Refer to Section 10 Licensing Control for more details.

2.1.4   Protocol

TCP/IP IPv4 (IPv6 available in a future release)

2.1.5   Determining System ID

NetLink allows connecting a maximum of 16 communication servers. To identify the communication server, each communication server must have one unique number between 1 and 16.

2.1.6   Determining Primary System

To determine the Primary System which should be controlling the NetLink network, the Primary System is selected based on System ID. The lowest ID number becomes Primary System.

If you add a new communication server to NetLink once the Primary System is already determined and functioning, the new communication server becomes a Secondary System.

SECTION 3 INSTALLATION

3.1 NetLink Connection

One valid IP address must be assigned for every 16 DSPs that are used. In addition to the DSP IP address, another IP address is required for signaling.

For example, to realize the full functionality of the PZ-32IPLA, three valid IP addresses in the same subnet must be assigned. One address is used for signaling information. The remaining two IP addresses are used for the voice traffic to and from the 32 DSP channels.

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

Card	Minimum IP Addresses	Maximum IP Addresses	Notes
32 IPLA	2	3	One IP address is required per 16 DSP channels. One additional IP Address is needed for signaling.
64 IPLA	2	5	
128 IPLA	2	9	

When assigning the IP addresses to the IPLA card, the addresses must be in the same network (subnet). If the CPU is to be connected also to the network, it requires a separate IP address in a different network (subnet).

When you have an IPLA card attached to the CPU, the CPU NIC is no longer required. All connections that previously terminated to the CPU NIC card can now be terminated to the IPLA NIC.

For example, PC Pro, Web Pro, and ACD terminate to the IPLA NIC card, when installed. Both the IPLA and CPU NIC share the same gateway assignment. The default gateway command in Program 10-12-03 is used by both NICs, allowing only one device, IPLA or CPU, to route outside of its own network.

General IP Configuration

The voice quality of VoIP depends on variables such as available bandwidth, network latency, and quality of service initiatives (QoS), 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 users IP based voice solution. Therefore, NEC recommends connecting the VoIP equipment through a local area network using private IP addresses.

For a network to be suitable for VoIP it must pass specific requirements. To make sure that the site meets these requirements, 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

❍   Adequate bandwidth for estimated VoIP traffic (refer to 1.5.3 Bandwidth)

❍   Depending on how QoS policies are built in the network, assignments might be needed in the CPU.

Some Network Considerations

Before adding the UNIVERGE SV8100 to a customer network, a detailed
network diagram of the existing network must be obtained from the customer.
This diagram provides information about any network conditions that can
prevent or hinder the VoIP equipment from functioning correctly.

Multiple Gateways

Figure 10-1 Multiple Gateways shows the UNIVERGE SV8100 assigned a
default Gateway of 192.168.1.1/24. The existing network has two routers in
the same broadcast domain connecting different networks. With regular data
communications this is not an issue. However, the VoIP cannot function
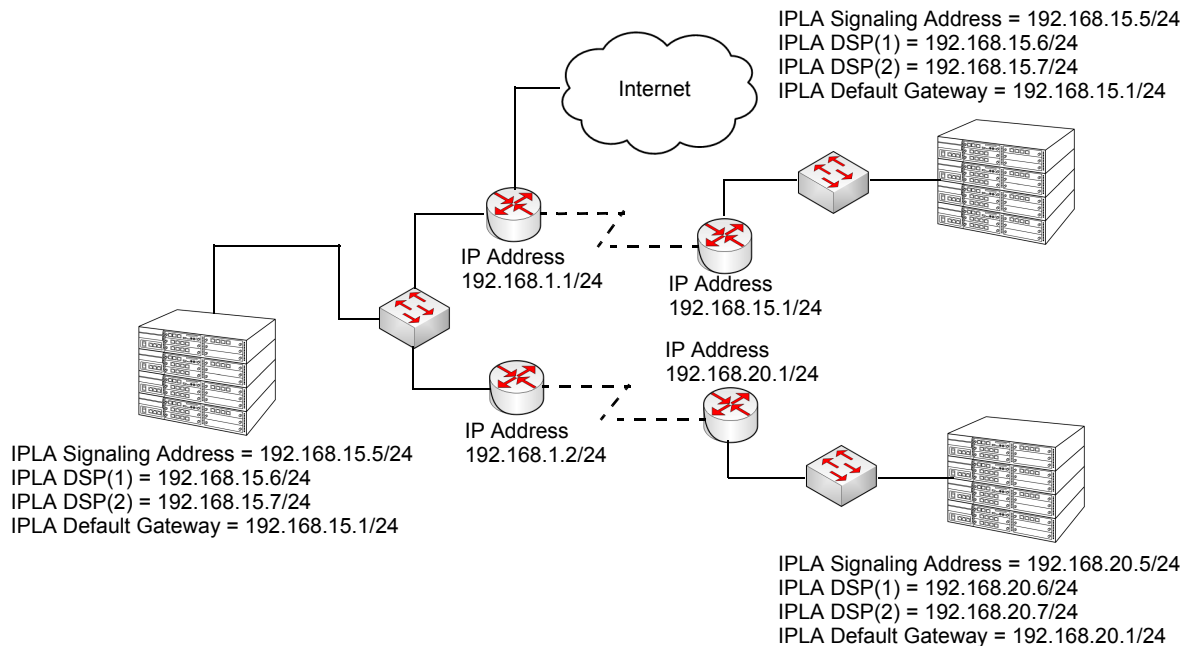correctly in this type of network environment without specific assignments.

IPLA Signaling Address = 192.168.15.5/24
IPLA DSP(1) = 192.168.15.6/24
IPLA DSP(2) = 192.168.15.7/24
IPLA Default Gateway = 192.168.15.1/24

Internet

IP Address
192.168.1.1/24

IP Address
192.168.15.1/24

IP Address
192.168.20.1/24

IP Address
192.168.1.2/24

IPLA Signaling Address = 192.168.15.5/24
IPLA DSP(1) = 192.168.15.6/24
IPLA DSP(2) = 192.168.15.7/24
IPLA Default Gateway = 192.168.15.1/24

IPLA Signaling Address = 192.168.20.5/24
IPLA DSP(1) = 192.168.20.6/24
IPLA DSP(2) = 192.168.20.7/24
IPLA Default Gateway = 192.168.20.1/24

Figure 10-1 Multiple Gateways

Figure 10-1 Multiple Gateways shows a configuration that allows for normal
operation of SV8100 NetLink from the 192.168.1.0 network to the
192.168.15.0 network, but the 192.168.20.0 network can have considerable
issues. Symptoms can include one way, or no speech path, cut-offs, to name
a few. If the default gateway for the SV8100 is changed to 192.168.1.2, then
the reverse occurs with the 192.168.15.0 network experiencing issues, and
the 192.168.20.0 network functioning correctly.

The new VoIP routing table feature (available on the SV8100 only) allows for the assignment of tables to route VoIP packets to different gateway addresses. In normal operation, the SV8100 first checks the VoIP routing table for a match. If one is found, that route is selected. If no match is found, the default gateway is then used.

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, to provide 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.

Figure 10-2 Two SV8100 Systems Connected Via the WAN shows two SV8100 systems. One on the corporate local LAN and one on a Remote network connected via the WAN. The remote site cannot call the MAIN site, therefore, it is not working.
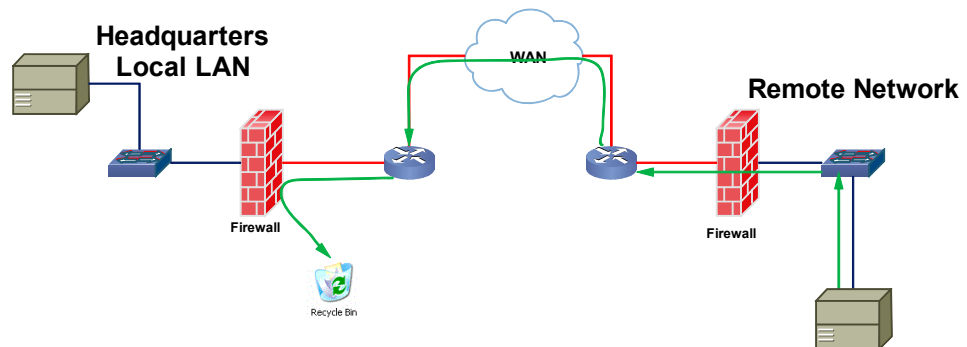

Figure 10-2 Two SV8100 Systems Connected Via the WAN

The green arrow in Figure 10-2 Two SV8100 Systems Connected Via the WAN represents the data packets leaving the REMOTE IPLA card destined for the SV81000 on the Headquarters LAN. The firewall on the Headquarters network is not configured to recognize the TCP/UDP ports utilized by the NEC

equipment, thus blocking them 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, 58000 and 58002 (TCP) for signaling and the voice ports, are required to be open at each location. This depends on how many IPLA ports are installed.

❍   IPLA 32 open UDP ports 10020 ~ 10083

❍   IPLA 64 open UDP ports 10020 ~ 10147

❍   IPLA 128 open UDP ports 10020 ~ 10275

VPN

Another common feature is to use 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 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.

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.
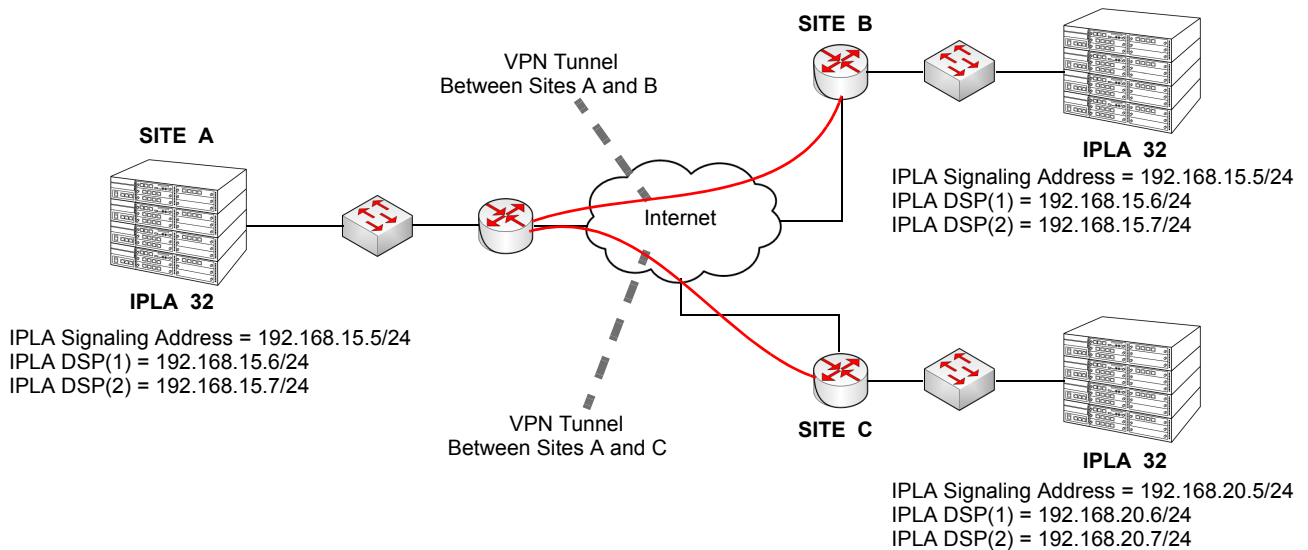

Figure 10-3 VPN Network

Site A can communicate with phones on sites B and C. Phones on sites B and C cannot communicate with each other. To correct this issue, another VPN connection between sites B and C is required.

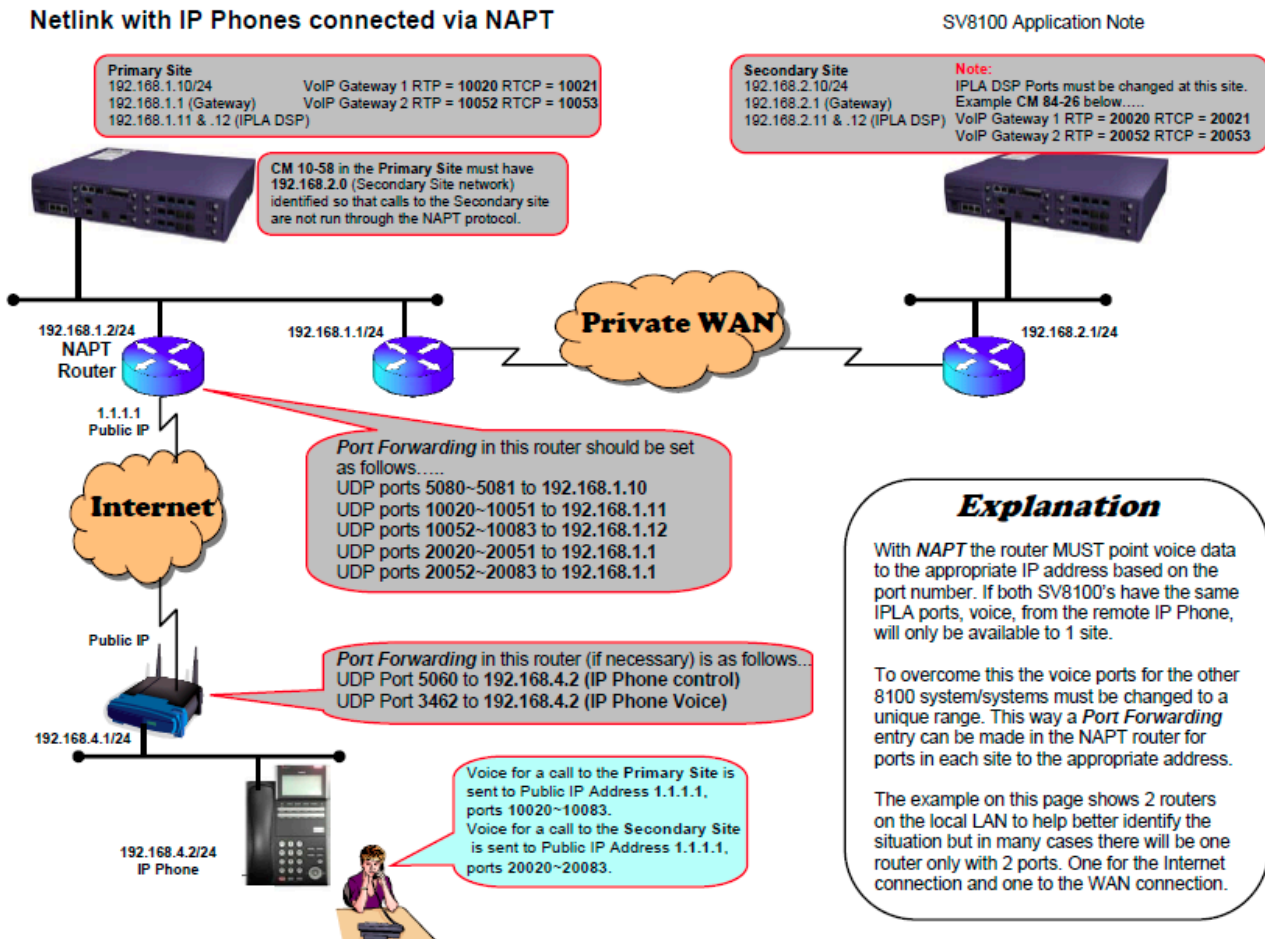**NetLink with IP Phones connected via NAPT**

Figure 10-4 NetLink with IP Phones connected via NAPT

3.2   Capacity

❍   A maximum of 512 Extension ports (as allowed by hardware and licensing)

❍   A maximum of 200 Trunk ports (as allowed by hardware and licensing)

The maximum number of ports, receivers, conference groups, etc., is the same as stand-alone communication server.

3.3   Hardware Information

Each SV8100 Communications Server needs the following items:

❍   CD-CP00-US (V1100/V1.12 or higher)
❍   MEMDB
❍   VOIPDB

Install the station/trunk blades as required for your configuration.

✎   When installing a Secondary System in a NetLink network, and the Secondary System has CD-LCA blades, the CD-LCA blades will come online and get assigned ports **before** any CD-DLCA blades.

The VOIPDB daughter boards in the network do not need to have the same capacity. However, any chassis which may be used as a Primary must have a VOIPDB large enough to handle the required network resources. It is recommended with this feature that the VOIPDB be the same as the largest installed VOIPDB (32, 64, 128 channel) in the network.

3.4   Software Information

All communication servers in the NetLink network should be at the same SV8100 software level.

After the NetLink network is set up, a user can press **Feature** + **5** on their terminal and review the following information:

❍   The Terminal System ID

❍   Primary or Secondary Assignment

❍   IP Address

❍   System ID of Current Primary System

3.5   Limitations

3.5.1   Features

Synchronous Ringing via NetLink is not supported.

3.5.2   IP Terminal

All IP terminals connect to the Primary System.

- o   IP Terminal

- o   Wireless LAN Terminal

- o   Softphone

3.5.3   IP Trunks

All IP trunks belong to the Primary System.

3.5.4   General limitations

- ❐   The Primary System (Main Site) requires the appropriate NetLink licenses dependant upon the number of nodes in the NetLink network.

- ❐   Up to 16 Nodes can be supported in a NetLink network.

- ❐   A maximum of 240 Virtual slots are supported.

- ❐   Port assignment is performed sequentially by the requested order from the Secondary Systems.

- ❐   All nodes in a NetLink network should have the same main CD-CP00-US software.

- ❐   When a Terminal is placed on hold, the Music on Hold comes from the system where the Terminal resides.

- ❐   When a trunk is placed on hold, the Music on Hold comes from the system where the trunk resides.

- ❐   External Paging uses an output on the CD-CP00-US of the Primary System.

- ❐   A PGD(2)-U10 ADP must be used if External Paging is required in the Secondary Systems.

- ❐   License information in the Primary will be copied to the Secondary site when doing database duplication.

- ❐   Duplicate license information in the Secondary System is available for only seven days.

❐ After seven days, the license expires. To renew the license, a connection to the original Primary site must be re-established. (Once the connection to the Primary is recovered, if fail-over occurs again, the license is once again enabled for seven days to the new Primary System.)

❐ If the original Primary site is in the NetLink network as a temporary secondary, the license information is available.

❐ If a user wants to enter another additional feature license, it needs to be entered on the original Primary System.

❐ When fail-over occurs, the Primary System is changed to another communication server. The IP applications do not know the new primary IP Address, so the following features are disabled after fail-over:

- SMDR
- ACD-MIS
- SIP Terminal
- Soft Phone
- IP K-CCIS

❐ The following Programs are not updated by the Primary System during fail-over :

❐ Program 10-01, Program 10-02, Program 10-12, Program 10-13, Program 10-14, Program 10-15, Program 10-16, Program 10-45, Program 51-01, Program 90-01 or Program 90-09.

❐ Data in SRAM area is not transferred to the Secondary Systems during fail-over, therefore when fail-over occurs DND and Caller ID History may be lost.

❐ Mobile Extension is supported only when using the PRI installed in the Primary System.

❐ When a Secondary System with an ETIA or RTB assigned is added to a NetLink network, the Primary Systems database does not replicate the data in Programs 10-55 or 90-61.

❐ When using Netlink and UCB, the trunks and extensions on the remote sites are not supported for UCB (**Version 2.50 or lower**).

❐ When External MOH is assigned, the CN8 or CN9 on the front of the CD-CP00-US can only be utilized at the Primary Site. All Secondary sites must provide the External MOH input via an ACI Input [PGD(2)-U10 ADP]. (**Version 2.51 or lower**).

❒   With **Version 3000 or higher**, both the Primary Site and Secondary Sites can have their own local MOH source connected to the CN8 or CN9 on the front of the CD-CP00-US.

❒   With **Version 3000 or higher**, the T-1 CCTA and PVA-CCIS blade is supported in the primary system and/or secondary systems.

❒   When using Netlink and UCB, the trunks and extensions on the remote sites are supported for UCB (**Version 2.51 or higher**).

❒   When using Mobile Extension in a NetLink Network, the ISDN/PRI must be used in the Primary System.

❒   InMail is not supported in a NetLink network.

❒   When using Set Relocation, and a terminal is re-located from one physical system to another physical system, route programming must be made accordingly for 911 calls.

❒   In a NetLink environment, IPK/IPK II Migration can only be used in the Primary system. Secondary systems do not support IPK/IPK II Migration.

❒   FoIP (Fax over IP) is supported in a NetLink Network.

❒   For a network to be suitable for VoIP it must pass specific requirements. To ensure the site meets these requirements, 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

   - Adequate bandwidth for estimated VoIP traffic

   - Depending on how QoS policies are built in the network, assignments may be needed in the CPU

Table 10-1 VoIP Resource Chart

		Primary System				Secondary 1		Secondary 2	
		TDM Terminal	IP Terminal (DT700)	CO Analog /Digital	IP Trunk	TDM Terminal	CO Analog /Digital	TDM Terminal	CO Analog /Digital
Primary System	TDM Terminal	0	P:1	0	P:1	P:1 S1:1	P:1 S1:1	P:1 S2:1	P:1 S2:1
	IP Terminal (DT700)	P:1	0	P:1	P:2	S1:1	S1:1	S2:1	S2:1
Secondary System 1	TDM Terminal	P:1 S1:1	S1:1 P:2	P:1 S1:1	P:2 S1:1	0	0	P1:1 S2:1	S1:1 S2:1
Secondary System 2	TDM Terminal	P:1 S2:1	S2:1 P:2	P:1 S2:1	P:2 S2:1	S1:1 S2:1	S1:1 S2:1	0	0

P = Primary

S1 = Secondary System #1

S2 = Secondary System #2

❒ The number of conference blocks in a NetLink network is the same as a stand-alone system.

❒ **Invalid data** is displayed in the LCD of the terminal if Program 51-01 is enabled and a PZ-ME50-US is not installed.

Restrictions:

● All IP Trunks and IP Terminals must be connected to the Primary System.

● The number of total ports depends on the Primary System.

● System ID (Program 51-01-02) must be unique for each system in a NetLink network.

● ACD/MIS can connect to the Primary (Main) site only.

● Only one Voice Mail can be installed in a NetLink network.

● In-Mail and VRS use the VMDB of the Primary (Main) site.

● APSU can be installed in the Secondary (Remote) System, however the NetLink time zone follows the Primary (Main) System.

❒ When installing a Secondary System in a Netlink network and the Secondary System has CD-LCA blades installed, the CD-LCA blades will come online and assign ports before any CD-DLCA blades are assigned.

❒ Secondary systems must follow the primary CPU software settings for Mu-law/A-law within the country where the primary system is located. SV8100 NetLink connections are only supported within the same countries/areas. For example, the SV8100 can be connected via NetLink between the US and Canada, however the SV8100 **cannot** be connected via NetLink with sytems in other countries (e.g., Mexico or the U.K.).

❒ Synchronous Ringing via NetLink is not supported.

❒ Always connect to the Primary System when using PCPro.

❒ When using K-CCIS - T1 in a Netlink system, the CCTA blade is only supported in the primary system.

❒ The following programs require a reset after making a change using PCPro, WebPro or Handset programming:

Table 10-2 CPU Reset Programs

Program	When Changed Using
10-12-01	Handset, WebPro, PCPro
10-12-02	Handset, WebPro, PCPro
10-12-09	Handset, WebPro, PCPro
51-01-01	Handset, WebPro, PCPro
51-15	WebPro, Handset Programming
84-03-06	Handset, WebPro, PCPro
90-04	Handset Programming
90-58	Handset Programming
10-46-07	PcPro
84-23	Handset, WebPro, PCPro

3.6   DSP

(PB Receiver, Dial Tone Detector, Caller ID Detector, Caller ID Sender, MF Receiver).

○ A DSP is used on the system CD-CP00-US which has the call. For instance, when a trunk on System ID 2 uses a receiver, a DSP on System ID 2 is used.

○ The Primary System must control DSP resources on the Secondary Systems but it cannot control all of them.

❍ There are five zones to handle DSP resources in NetLink. The zone
number depends on the System ID number.

Zone #	System ID
1	1, 6, 11, 16, 21, 26, 31, 36, 41, 46
2	2, 7, 12, 17, 22, 27, 32, 37, 42, 47
3	3, 8, 13, 18, 23, 28, 33, 38, 43, 48
4	4, 9, 14, 19, 24, 29, 34, 39, 44, 49
5	5, 10, 15, 20, 25, 30, 35, 40, 45, 50

In the same Zone, the communication servers must share the DSP
resource number. For instance, if a user in System ID-1 is using DSP
resource #1, resource #1 becomes busy in Zone #1 (System ID 1, 6, 11,
16, 21, 26, 31, 36, 41, 46).

❍ All communication servers follow the DSP Setting in Program 10-09 in
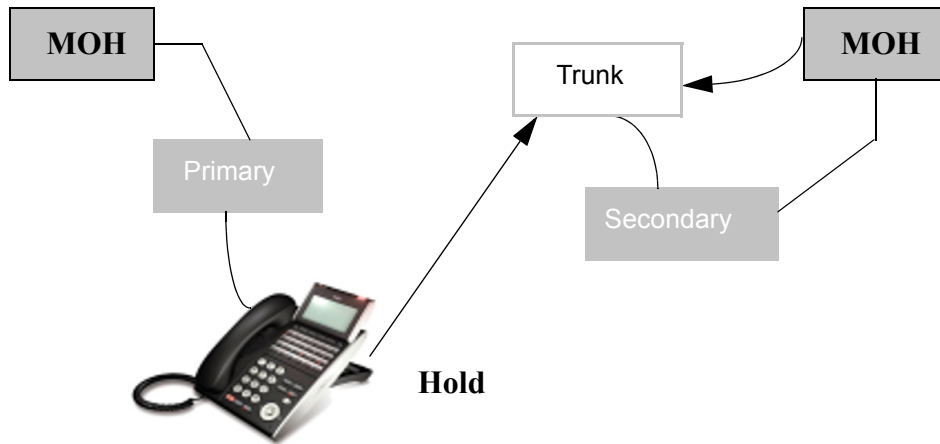the Primary System.

3.7   E911

A trunk is required for each communication server for 911 calls and the ARS
routing table must be set up to seize the proper trunk. If no DSP resource is
available, one of the active DSPs is dropped and the communication server
uses that DSP for the 911 call.

3.8   Music on Hold

❍ When External MOH is assigned, the CN8 or CN9 on the front of the
CD-CP00-US can only be utilized at the Primary Site. All Secondary
sites must provide the External MOH input via an ACI Input [PGD(2)-U10
ADP]. With Version 2.51 or lower.

❍ With Version 3000 or higher, both the Primary Site and Secondary Sites
can have their own MOH source connected to the CN8 or CN9 on the
front of the CD-CP00-US. Same rules below apply.

❍ If the external MOH setting is defined in the Primary System, each
communication server needs to have an external MOH source.
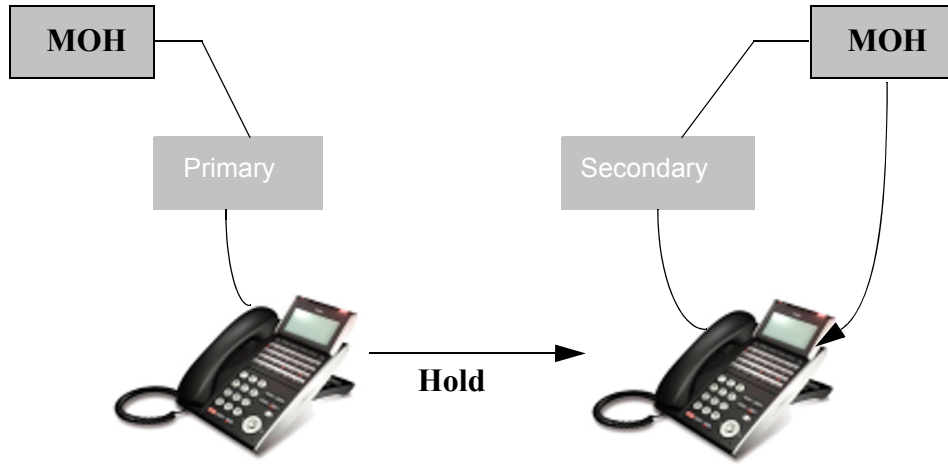
When a terminal is placed on hold, MOH comes from the communications server in which the terminal resides.

When a trunk is placed on hold, MOH comes from the communications server in which the trunk resides.
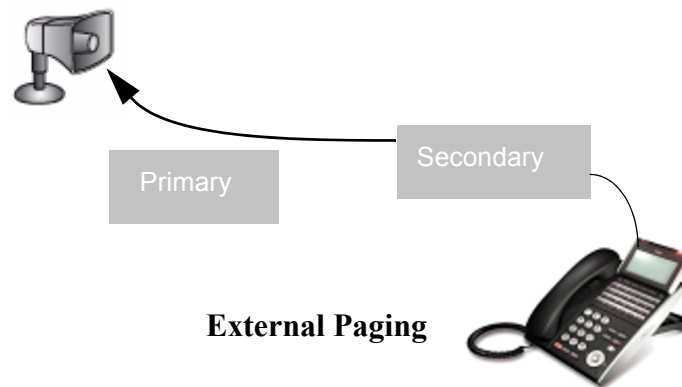
External Paging

✎   External Paging uses an output on the CD-CP00-US of the Primary
     System.

✎   A PGD(2)-U10 ADP must be used if External Paging is required in the
     Secondary Systems.

3.9   Bandwidth

The following example shows the approximate bandwidth used for the NetLink
signaling:

❍   Bandwidth signaling for 1 call = 4kb

❍   Bandwidth of 1 voice call (G.711, VIF=20ms) = 90kbps

If there are 100 calls (average talk time of five minutes) between the linked
communication servers for one hour:

❍   Signaling data total is 4kb x 1000 calls divided by 3600 seconds =
     1.11kbps

❍   Voice data total is 1000 calls divided by 60/5 x 90Kbps = 7.5Mbps

Using the above calculation, the signaling data is smaller than the voice data.
So, in determining the bandwidth needed, using the voice bandwidth number
for both the voice and data allows adequate bandwidth.

3.10   Conference Block

❍   The number of conference blocks in NetLink is the same as with a
     stand-alone communication server.

3.11   Voice Mail

❍   VM8000 InMail is not supported with NetLink.

❍   One UM8000 Voice Mail can be installed in NetLink.

❍   UM8000 Voice Mail can be installed in the Primary or Secondary System.

3.12   FoIP

❍   FoIP (Fax over IP) is supported in a NetLink network.

SECTION 4   LIST OF SUPPORTED FEATURES IN SECONDARY SYSTEM

The following table lists supported features.

Table 10-3 Supported Features in Secondary System

Feature Name	Supported in Secondary System	Comments
Account Code – Forced/Verified/ Unverified	Yes	
Account Code Entry	Yes	
Alarm	Yes	
Alarm Reports	Conditions	Programs 90-10-80 ~ 90-10-95 are not supported.
Alphanumeric Display	Yes	
Analog Communications Interface (ACI)	Yes	
Ancillary Device Connection	Yes	
Answer Hold	Yes	
Answer Key	Yes	
Attendant Call Queuing	Yes	
Automatic Call Distribution (ACD)	Conditions	**Fail-Over:** VRS card must be installed in Primary system. MIS monitor must be connected to the Primary system. If you are logged in and Fail-over occurs the agent will not be logged back in when the Secondary system takes over. The agent will have to manually log back in to continue taking calls.
Automatic Release	Yes	
Automatic Route Selection	Conditions	**Fail-Over:** ARS Class Of Service must be programmed for each System in a NetLink Network to ensure proper call routing during Fail-Over.

Table 10-3 Supported Features in Secondary System

Feature Name	Supported in Secondary System	Comments
Background Music	Conditions	If using Background Music, each site must provide its own music source. Primary and Secondary Systems cannot share a source. If the Primary System is set to Music off of the CD-CP00-US, then all Secondary Systems also must get their music off of the CD-CP00-US.
Barge-In	Yes	
Battery Backup – System Memory	N/A	
Battery Backup – System Power	N/A	
Call Appearance (CAP) Keys	Yes	
Call Arrival (CAR) Keys	Yes	
Call Duration Timer	Yes	
Call Forwarding	Yes	
Call Forwarding – Centrex	Yes	
Call Forwarding – Park and Page	Yes	
Call Forwarding with Follow Me	Yes	
Call Forwarding, Off-Premise	Yes	
Call Forwarding/ Do Not Disturb Override	Yes	
Call Monitoring	Yes	
Call Redirect	Yes	
Call Waiting/ Camp-On	Yes	
Callback	Yes	

Table 10-3 Supported Features in Secondary System

Feature Name	Supported in Secondary System	Comments
Caller ID	Yes	
Caller ID Call Return	Yes	
Central Office Calls, Answering	Yes	
Central Office Calls, Placing	Yes	
Class of Service	Yes	
Clock/Calendar Display	Yes	
CO Message Waiting Indication	Yes	
Code Restriction	Yes	
Code Restriction Override	Yes	
Code Restriction, Dial Block	Yes	
Conference	Yes	
Conference, Voice Call/Privacy Release	Yes	
Continued Dialing	Yes	
Cordless DECT Terminals	Yes	
Cordless Telephone Connection	Yes	
Data Line Security	Yes	
Delayed Ringing	Yes	
Department Calling	Yes	
Department Step Calling	Yes	

Table 10-3 Supported Features in Secondary System

Feature Name	Supported in Secondary System	Comments
Dial Pad Confirmation Tone	Yes	
Dial Tone Detection	Yes	
Dialing Number Preview	Yes	
Digital Trunk Clocking	Yes	
Direct Inward Dialing (DID)	Yes	
Direct Inward Line (DIL)	Yes	
Direct Inward System Access (DISA)	Yes	
Direct Station Selection (DSS) Console	Yes	
Directed Call Pickup	Yes	
Directory Dialing	Yes	
Distinctive Ringing, Tones and Flash Patterns	Yes	
Do Not Disturb	Yes	
Door Box	Yes	
Drop Key	Yes	
D^{term} Cordless II Terminal	Yes	
D^{term} Cordless Lite II Terminal	Yes	
E911 Compatibility	Yes	
Facsimile CO Branch Connection	Yes	

Table 10-3 Supported Features in Secondary System

Feature Name	Supported in Secondary System	Comments
Flash	Yes	
Flexible System Numbering	Yes	
Flexible Timeouts	Yes	
Forced Trunk Disconnect	No	
Group Call Pickup	Yes	
Group Listen	Yes	
Handset Mute	Yes	
Handsfree and Monitor	Yes	
Handsfree Answerback/ Forced Intercom Ringing	Yes	
Headset Operation	Yes	
Hold	Yes	
Hot Key-Pad	Yes	
Hotel/Motel	Yes	
Hotline	Yes	
Howler Tone Service	Yes	
Intercom	Yes	
IPK/IPK II Migration	Conditions	If IPK/IPK II Migration is used in a NetLink environment, it must be the Primary System.
IP Multiline Station (SIP)	Conditions	All IP Terminals connect to the Primary System.
IP Single Line Telephone (SIP)	Conditions	All IP Terminals connect to the Primary System.
IP Trunk - (SIP) Session Initiation Protocol	Conditions	All IP Trunks connect to the Primary System.

Table 10-3 Supported Features in Secondary System

Feature Name	Supported in Secondary System	Comments
IP Trunk - H.323	Conditions	All IP Trunks connect to the Primary System.
ISDN Compatibility	Yes	
K-CCIS - IP	Conditions	All IP Trunks connect to the Primary System.
K-CCIS - IP with PVA	Conditions	With V3000 or higher, the PVA-CCIS blade is supported in the primary system and/or secondary systems.
K-CCIS - T1	Conditions	With V3000 or higher, the CCTA blade is supported in the primary system and/or secondary systems.
Last Number Redial	Yes	
Licensing	Conditions	All licenses must be registered into the Primary System. During Fail-over, the Secondary System will contain all licenses for seven days.
Line Preference	Yes	
Long Conversation Cutoff	Yes	
Maintenance	Conditions	Web Pro and PC Pro can be used to view the individual Card Configuration screens of all Systems in a NetLink Network, however the System Data is for all Systems combined.
Meet Me Conference	Yes	
Meet Me Paging	Yes	
Meet Me Paging Transfer	Yes	
Memo Dial	Yes	
Message Waiting	Yes	
Microphone Cutoff	Yes	
Mobile Extension	Conditions	Mobile Extension is only supported when using the ISDN/PRI installed in the Primary System.

Table 10-3 Supported Features in Secondary System

Feature Name	Supported in Secondary System	Comments
Multiple Trunk Types	Yes	
Music on Hold	Conditions	Refer to Section 3.8, Music on Hold.
Name Storing	Yes	
Night Service	Yes	
Off-Hook Signaling	Yes	
One-Touch Calling	Yes	
Operator	Yes	
(OPX) Off-Premise Extension	Yes	
Paging, External	Conditions	Refer to Section 3.8, Music on Hold.
Paging, Internal	Yes	
Park	Yes	
PBX Compatibility	Yes	
PC Programming	Conditions	WebPro and PCPro can be used to view the individual Card Configuration screens of all Systems in a NetLink Network, however the System Data is for all Systems combined. You must be connected to the Primary System.
Power Failure Transfer	Yes	
Prime Line Selection	Yes	
Private Line	Yes	
Programmable Function Keys	Yes	
Programming from a Multiline Terminal	Yes	
Pulse to Tone Conversion	Yes	
Redial Function	Yes	

Table 10-3 Supported Features in Secondary System

Feature Name	Supported in Secondary System	Comments
Remote (System) Upgrade	Yes	
Repeat Redial	Yes	
Resident System Program	Yes	
Reverse Voice Over	Yes	
Ring Groups	Yes	
Ringdown Extension, Internal/External	Yes	
Room Monitor	Yes	
Save Number Dialed	Yes	
Secondary Incoming Extension	Yes	
Secretary Call (Buzzer)	Yes	
Secretary Call Pickup	Yes	
Selectable Display Messaging	Yes	
Selectable Ring Tones	Yes	
Serial Call	Yes	
Single Line Telephones, Analog 500/2500 Sets	Yes	
SLT Adapter	Yes	
Softkeys	Yes	
Speed Dial – System/Group/ Station	Yes	

Table 10-3 Supported Features in Secondary System

Feature Name	Supported in Secondary System	Comments
Station Hunt	Yes	
Station Message Detail Recording	Yes	
Station Name Assignment – User Programmable	Yes	
Station Relocation	Conditions	When using Set Relocation, and a terminal is relocated from one physical system to another, route programming must be made accordingly for 911 calls.
SV8100 Communications Analyst Enterprise	Conditions	The SV8100 Communications Analyst Server must connect to the Primary System.
SV8100 Desktop Applications	Conditions	The SV8100 Desktop Applications must connect to the Primary System.
SV8100 Internal Router	Yes	
SV8100 PoE Gigabit Switch	Yes	
SV8100/SV8300 Terminals	Yes	
Synchronous Ringing	No	
T1 Trunking (with ANI/DNIS Compatibility)	Yes	
Tandem Ringing	Yes	
Tandem Trunking (Unsupervised Conference)	Yes	
TAPI Compatibility	Yes	
Tone Override	Yes	
Traffic Reports	No	
Transfer	Yes	

Table 10-3 Supported Features in Secondary System

Feature Name	Supported in Secondary System	Comments
Trunk Group Routing	Yes	
Trunk Groups	Yes	
Trunk Queuing/ Camp-On	Yes	
UCB (Unified Communications for Business)	No	UCB is not supported with NetLink.
UM8000 Mail	Yes	
Uniform Call Distribution (UCD)	Yes	
Uniform Numbering Network	Yes	
UNIVERGE Multimedia Conference Bridge	Yes	
Universal Slots	Yes	
User Programming Ability	Yes	
Virtual Extensions	Yes	
VM8000 InMail	No	The VM8000 InMail is not supported with NetLink.
Voice Mail Integration (Analog)		
Voice Mail Message Indication on Line Keys	Yes	
Voice Over	Yes	
Voice Response System (VRS)	Yes	
Volume Controls	Yes	

Table 10-3 Supported Features in Secondary System

Feature Name	Supported in Secondary System	Comments
Warning Tone For Long Conversation	Yes	
Wireless DECT (SIP)	Conditions	All IP Terminals connect to the Primary System.

SECTION 5   NETLINK FUNCTIONS

5.1   Programming

The following programs require a CD-CP00-US reset after making a change via PCPro, Web Pro and handset programming.

Program	When Changed via
10-12-01	Handset, Web Pro, PCPro
10-12-02	Handset, Web Pro, PCPro
10-12-09	Handset, Web Pro, PCPro
10-12-10	Handset, Web Pro, PCPro
51-01-01	Handset, Web Pro, PCPro
51-15	Handset, Web Pro
84-03-06	Handset, Web Pro, PCPro
90-04	Handset Programming
90-58	Handset Programming
10-46-07	PCPro
84-23	Handset, Web Pro, PCPro

✎   *When Program 90-08-01 System Reset is performed from any site in a NetLink Network, the Primary System will be reset.*

Minimum Required Programming

❍   **10-12-09 : CD-CP00-US Network Setup – VoIP Daughter Board IP Address**

Set the LAN IP address for the 32, 64 and 128 channel VoIP daughter boards. The IP number increases by one for each increase in the slot number.

❍   **10-12-10 : CD-CP00-US Network Setup – VoIP Daughter Board Subnet Mask**

Define the subnet mask for the 32, 64 and 128 channel VoIP daughter boards. The setting of Subnet Mask errors when all Host Addresses are 0.

❍   **51-01-01 : NetLink System Settings – NetLink System ID**

This is the ID number (0 = Disabled, 1-50) that identifies each SV8100 in the NetLink network. Each SV8100 must have a unique number in the network.

When this option is set to 0, NetLink is disabled.

✎   *The SV8100 must be reset when changing this option.*

❍ **51-03-01 : NetLink Internet Protocol Address List Setting – Internet Protocol Address List**

The communications server sees the Primary SV8100 from in this list. When no Primary SV8100 is seen or Fail-Over occurs, the Node List is referred to to establish a new link. This setting is necessary when Program 51-01-03 is set to 0 or Program 51-05-02 is other than 0.

Enter IP addresses for any communications server which is included in the network (especially the Primary SV8100). Once the communications server connects to the Primary SV8100, this setting is updated by the Primary SV8100 when Program 51-13-01 is enabled. This allows any new or changed SV8100s to be added automatically.

An IP address cannot be defined more than once.

Required Programming for the Secondary System Setup

❍ **51-04-01 : IP Address for Top Priority Primary System – Internet Protocol Address for Top Priority Primary System**

Enter the IP address of the Primary SV8100. This setting is needed to use the Primary SV8100 Automatic Integration Feature (Program 51-06-01). If the secondary flag is set in Program 51-01-03 secondary SV8100s connect with this IP address.

❍ **51-05-01 : Timer Settings for NetLink – Keep Alive Interval**

Set the interval (1~3600) for the secondary SV8100 to send the Keep Alive packet to the Primary SV8100 to confirm communication.

❍ **51-05-02 : Timer Settings for NetLink – Keep Alive Response Waiting Time**

The secondary SV8100 waits this time (0, 5~10800) for response to the Keep Alive packet to confirm communication with Primary SV8100. If no response is received, Fail-Over occurs. A setting of 0 disables this and the SV8100 waits indefinitely for a response from the Primary SV8100. For Fail-Over to occur, this time must be set to an entry other than 0.

❍ **51-05-03 : Timer Settings for NetLink – Primary Search Packet Sending Interval**

This timer determines the time between packets the SV8100 waits when searching for a new Primary SV8100 with the Fail-Over feature (1~3600).

❍ **51-05-04 : Timer Settings for NetLink – Top Primary Search Time**

Set the time (5~10800) between packet sending when the SV8100 is reviewing priority levels for a new higher-priority Primary SV8100.

❍ **51-05-05 : Timer Settings for NetLink – Top Priority Primary Detection Packet Sending Interval**

When the current Primary SV8100 is not the Top Priority Primary SV8100, the SV8100 sends packets at this interval (1-3600) to check if a higher priority SV8100 exists

❍ **51-05-06 : Timer Settings for NetLink – Primary System Seeking Time with Forced Change Primary**

When the Forced Change Primary command is executed, the SV8100 searches for the new Primary SV8100 for this time (1~10800).

❍ **51-05-07 : Timer Settings for NetLink – Communications Socket Refresh Timer**

If the IP connection becomes unstable and communication is lost (keep-alive function does not work), the SV8100 retries the connection at this time (20~3600).

❍ **51-06-01 : NetLink Primary System Automatic Integration Setting – Primary**
Automatic Integration

When Fail-Over occurs, multiple Primary SV8100s may be established. When the connection is recovered, with this option enabled (1), the NetLink feature is automatically reestablished around the top priority Primary SV8100.

❍ **51-06-02 : NetLink Primary System Automatic Integration Setting – Blade Reset Timing Option**

When the Primary Automatic Integration reestablishes the NetLink network, the blades in the secondary SV8100s are reset. This option determines if the secondary SV8100 blades are reset only when all blades are idle (0) or anytime (1).

Other Feature Programming

Refer to the specific NetLink functions which follow for additional programs.

❍ **10-01-01 ~ 10-01-07 : Time and Date**

Change the SV8100 Time and Date through SV8100 programming. Extension users can also dial Service Code 828 to change the Time if allowed by an extension Class of Service.

❍ **51-01-02 : NetLink System Settings – Primary Candidate Order**

When the Primary SV8100 is turned off or disconnected from the network, this value (1~50) is used to select the new Primary SV8100 with the Fail-Over feature. The smaller the number, the higher the priority for the SV8100. When the value for two SV8100s is the same, lower NetLink System ID (51-01-01) is selected as the Primary SV8100.

❍ **51-01-03 : NetLink System Settings – Secondary System**

The link between SV8100s is established based on this setting (0, 1). When enabled, the SV8100 connects with the top priority Primary SV8100 (address set in Program 51-04-01). If the SV8100 is not found within the defined time (Program 51-05-02), the Fail-Over feature of the SV8100 searches for the Primary SV8100 as if this option were set to 0.

The link between SV8100s is dynamically established based on the node list set in Program 51-03-01. The Primary SV8100 is selected in the order in which the SV8100s wake up.

❍ **51-02-01 : NetLink System Individual Setting – System Name**

Set the desired site name (up to 20 characters) for ease of maintaining information. Once the SV8100 is connected to the Primary SV8100, this setting is updated by the Primary System data.

❍ **51-02-02 : NetLink System Individual Setting – Time Zone – Hour**

Determine the offset hours from the Primary SV8100. This setting affects the time display on display telephones (0~24 = -12 ~ +12 hours).

❍ **51-02-03 : NetLink System Individual Setting – Time Zone – Minute**

Determine the offset minutes from the Primary SV8100. This setting affects the time display on display telephones (0 ~ 120 = -60 ~ +60 minutes).

❍ **51-02-04 : NetLink System Individual Setting – System Authentication MAC Address**

When Program 51-13-03 is enabled, the SV8100 checks this MAC address against the MAC address of the connecting CCPU. If the value is different, the connection is refused.

❍ **51-07-01 : NetLink Forced Change of Primary System Settings – Allow Forced Change of Primary System**

When Fail-Over occurs, you can manually change the Primary SV8100 using Program 51-08, if this option is enabled. Program 51-06-01 must be set to 0.

❍ **51-07-02 : NetLink Forced Change of Primary System Settings – Blade Reset Timing Option**

When the Forced Change of Primary Settings is performed, the blades in the secondary communication servers are reset. This option determines if the secondary SV8100 blades are reset only when all blades are idle (0) or anytime (1).

❍ **51-08-01 : New Primary System Setting – Internet Protocol Address of New Primary**

When forcing the communications server to update to a new Primary SV8100, the communications server using the IP address defined here as the new Primary.

✎   After a Forced Change of Primary SV8100 is done, this entry is erased.

❍ **51-08-02 : New Primary System Setting – System ID of New Primary**

If you already have an IP address registered in Program 51-11-03, you can execute a Forced Change Primary SV8100 by entering the system ID. If this ID is set to 0, the Top Priority SV8100 is selected as the new Primary.

❍ **51-09-01 : NetLink TCP Port Settings – Primary Port**

Define the port the Primary SV8100 uses to communicate with the Secondary SV8100 (0~65535).

❍ **51-09-02 : NetLink TCP Port Settings – Communication Port**

Define the port used to communicate between other networked communications servers (0~65535).

❍ **51-09-03 : NetLink TCP Port Settings – Secondary Port**

This setting defines the port used by the Secondary SV8100 to communicates to the Primary SV8100 (0~65535). If 0 is entered, the port is selected dynamically.

❍ **51-09-04 : NetLink TCP Port Settings –Primary Search Port**

When Fail-Over occurs, each SV8100 communicates with the other communication servers using the port number specified in this entry (0~65535). If 0 is entered, the port is selected dynamically.

If an entry other than 0 is made, up to 50 ports (depending in the number of networked systems) are continuously used from the specified port number. (Ex: If 5000 is entered, 5001~5049 are used.)

❍ **51-09-05 : NetLink TCP Port Settings – Primary Detection Port**

Enter the port number to search for the Top Priority Primary SV8100. If 0 is entered, the port is selected dynamically (0~65535).

❍ **51-09-06 : NetLink TCP Port Settings – Database Replication Secondary System Listening Port**

Define the listening port used so that the Secondary SV8100 can replicate the Primary SV8100 database (0~65535).

❍ **51-09-07 : NetLink TCP Port Settings – Database Replication Primary System Detection Port**

Define the port used for communication so that the Primary SV8100 may synchronize the Secondary SV8100 with the program data (0~65535). If 0 is entered, the port is selected dynamically.

❍ **51-10-01 : Remaining Virtual Slots**

View the remaining number of slots which can be controlled by NetLink. The NetLink feature can control up to 240 virtual slots maximum. (The physical slots in the NetLink network are maintained as virtual slots by the SV8100.) This option is not user-definable and is view-only.

❍ **51-11-01 - 51-11-06 : NetLink System Information**

View the SV8100 information for communication servers connecting with the Primary SV8100. These options are not user-definable and are view-only.

❍ **51-12-01 - 51-12-06 : Primary System Information**

View the information for the Primary SV8100 in a NetLink network. These options are not user-definable and are view-only.

❍ **51-13-01 : NetLink Option Settings – Automatic IP Address List Update**

If this option is enabled (1), the Internet Protocol address list is updated to include the IP address of a Secondary SV8100 upon connection.

❍ **51-13-02 : NetLink Option Settings – Time Zone Enhancing**

When enabled (1), the time zone is applied to the following items: LCD Clock Display, Caller ID History, VRS Time Announce, Time and Date Set by Service Code, Alarm Clock, Hotel Mode Wake-Up Call (time announce included).

When disabled (0), the time zone is only applied to the LCD Clock Display and Caller ID History.

Program 51-02-02 must also be set for this option.

❍ **51-13-03 : NetLink Option Settings – MAC Address Authentication**

When enabled (1), connection authentication of the SV8100 is done with the MAC address set in Program 51-02-04. The system compares its own MAC address and if the address does not match, the Primary System rejects the connection.

❍ **51-14-01 : NetLink System Control – Delete System Information**

Delete the SV8100 and slot information for a particular communications server using the NetLink feature. To use this program, the communications server must not be connected to the network.

❍ **51-15-01 : Easy Set Command**

Automatically set the minimum settings for the NetLink feature. This program sets up to four communication servers and applies the minimum settings required to Programs 10-12-09, 51-01-01, 51-03-01, 51-04-01, and 51-05-02.

❍ **51-16-01 : NetLink System Data Replication Mode Setting – System Data Replication Mode**

Set the replication mode (0 = Disabled, 1 = Setting Time Mode, 2 = Interval Mode). An entry of '1 replicates the data at the time set in 51-16-02. If this option is set to 2, replication occurs at the interval set in 51-16-03.

❍ **51-16-02 : NetLink System Data Replication Mode Setting – System Data Replication Time Setting**

If Program 51-16-01 is set to 1, set the time to replicate the program data (0000~2359).

❍ **51-16-03 : NetLink System Data Replication Mode Setting – System Data Replication Interval Setting**

If Program 51-16-01 is set to 2, set the interval time between replicating the program data (15-1440 minutes).

❍ **51-16-04 : NetLink System Data Replication Mode Setting – Replication Time Stamp**

This program displays the last time the program data was replicated. This is automatically updated when the replication occurs. This option is view-only.

❍ **51-16-05 : NetLink System Data Replication Mode Setting – System Data Replication Wait Time**

When a NetLink network is created, this option determines the time the SV8100 waits until replication is started (1~86400 seconds).

❍ **51-16-06 : NetLink System Data Replication Mode Setting – System Data Replication Interval**

Set the time to start replication to the next node after replication has completed to one node (1~86400 seconds).

SECTION 6   FAIL-OVER

With NetLink, a connection to the Primary System may be lost (disconnected or powered down) from the TCP/IP connection. When this occurs, there could be a loss of control at all Secondary Systems.

The Fail-Over feature works when NetLink has lost connection to the Primary System for longer than the defined keep-alive timer. It finds one of the Secondary Systems to take over the Primary Systems database so that NetLink can still work. The networked communication servers reboot (this takes approximately 30 seconds), and when they come back up are linked to the new Primary System.

The Primary System is selected based on the Primary Candidate Order and System ID. The decision for the new Primary System is based first on the value of the Primary Candidate Order. The lower the number, the higher the priority. If the value is the same number, the System ID is then checked.

If the communication server database is set up as follows, System ID 2 becomes the new Primary System.

System ID Program 51-01-01	Primary Candidate Order Program 51-01-02
1	30
2	30
3	30
4	30

If communication server database is set up as follows, System ID 4 becomes the new Primary System.

System ID Program 51-01-01	Primary Candidate Order Program 51-01-02
1	30
2	30
3	30
4	1

Required Programs:

❍ **51-01-01 : NetLink System Settings - NetLink System ID**

This is the ID number (0 = Disabled, 1-50) that identifies each SV8100 in the NetLink network. Each SV8100 must be a unique number in the network. When this option is set to 0, NetLink is disabled.

✎ *The SV8100 must be reset when changing this option.*

❍ **51-01-02 : NetLink System Settings – Primary Candidate Order**

When the Primary SV8100 is turned off or disconnected from the network, this value (1~50) is used to select the new Primary SV8100 with the Fail-Over feature. The smaller the number, the higher the priority for the SV8100. When the value for two SV8100s is the same, lower NetLink System ID (51-01-01) is selected as the Primary SV8100.

❍ **51-03-01 : NetLink Internet Protocol Address List Setting – Internet Protocol Address List**

The communications server sets the Primary SV8100 using this list. When no Primary SV8100 is seen or Fail-Over occurs, the Node List is used to establish a new link. This setting is necessary when Program 51-01-03 is set to 0 or Program 51-05-02 is other than 0.

Enter IP address for any communications server included in the network (especially the Primary SV8100). After the communications server connects to the Primary SV8100, this setting is updated by the Primary SV8100 when Program 51-13-01 is enabled. This allows any new or changed SV8100s to be added automatically.

An IP address cannot be defined more than once.

- OR -

❍ **51-01-03 : NetLink System Settings – Secondary System**

The link between SV8100s is established based on this setting (0, 1). When enabled, the SV8100 connects with the top priority Primary SV8100 (address set in Program 51-04-01). If the SV8100 is not found during the defined time (Program 51-05-02), the Fail-Over feature of the SV8100 searches for the Primary SV8100 as if this option were set to 0.

The link between SV8100s is dynamically established based on the node list set in Program 51-03-01. The Primary SV8100 is selected in the order in which the SV8100s wake up.

❍ **51-04-01 : IP Address for Top Priority Primary System – Internet Protocol Address for Top Priority Primary System**

Enter the IP address of the Primary SV8100. This setting is needed to use the Primary SV8100 Automatic Integration Feature (Program 51-06-01). If the secondary flag is set in Program 51-01-03 secondary SV8100s connect with this IP address.

❍ **10-12-09 : CD-CP00-US Network Setup – VoIP Daughter Board IP Address**

Set the LAN IP address for the 32, 64 and 128 channel VoIP daughter boards. The IP number increases by one for each increase in the slot number.

Related Programs:

❍ **51-09-01 : NetLink TCP Port Settings – Primary Port**

Define the port the Primary SV8100 uses to communicate with the Secondary SV8100 (0~65535).

❍ **51-09-02 : NetLink TCP Port Settings – Communication Port**

Define the port used to communicate between other networked communications servers (0~65535).

❍ **51-09-03 : NetLink TCP Port Settings – Secondary Port**

This setting defines the port used by the Secondary SV8100 to communicates to the Primary SV8100 (0~65535). If 0 is entered, the port is selected dynamically.

❍ **51-09-04 : NetLink TCP Port Settings – Primary Search Port**

When Fail-Over occurs, each SV8100 communicates with the other communication servers using the port number specified in this entry (0~65535). If 0 is entered, the port is selected dynamically.

If an entry other than 0 is made, up to 50 ports (depending in the number of networked systems) are continuously used from the specified port number. (Ex: If 5000 is entered, 5001~5049 are used.)

The system ID for each communication server must be unique. Otherwise, the connection is refused. The information for the refused connection is included in the communication server alarm report.

If a communication server loses connection and another communication server with the same ID then tries to connect to the NetLink, it is recognized as the first communication server (which lost the connection) and is allowed to connect.

If a Secondary System loses connection, it does not affect the operation of the Primary System.
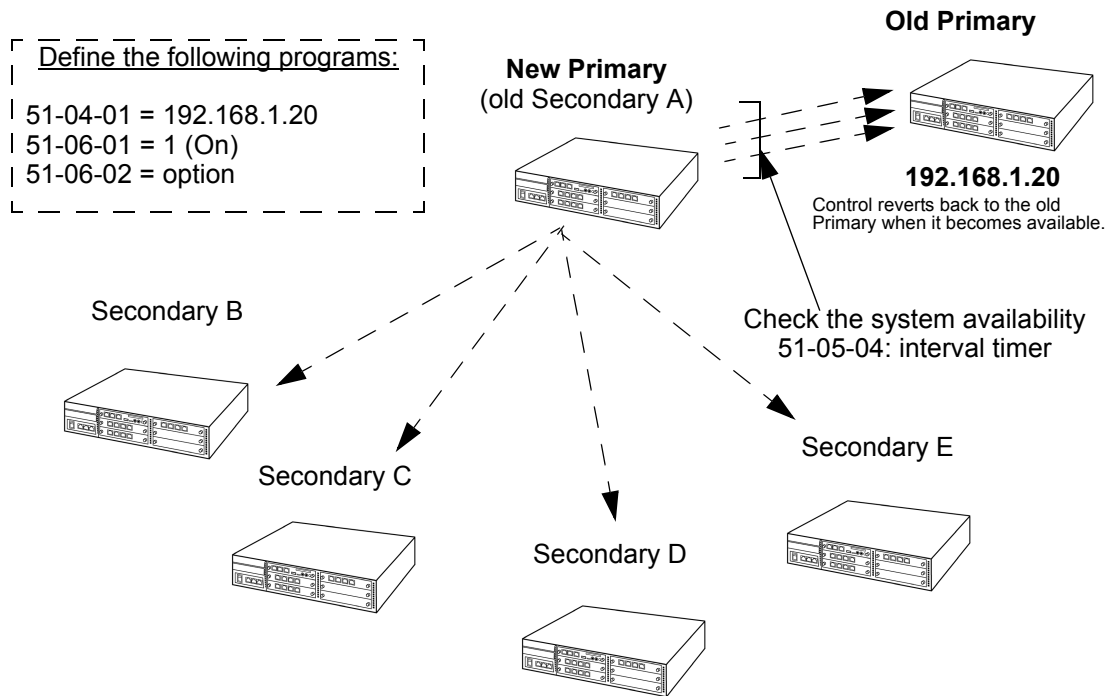
Conditions

❍   Any chassis which may be used as a Primary must have a VOIPDB daughter board large enough to handle the required network resources. It is recommended that the VOIPDB be the same as the largest installed VOIPDB (32, 64, 128 channel) in the network.

❍   When the Primary System is switched, all terminals and blades are reset so the change of the Primary System can be made. This process takes approximately 30 seconds.

❍   Only the voice mail connected with the Primary System can be used. Since the recording data of the voice mail is not synchronized, the content originally recorded by the Primary cannot be succeeded, though the voice mail of the Secondary System can be used after it becomes the Primary after Fail-Over.

❍   Synchronous Ring and NetLink cannot be used together.

❍   The system IP address set can be connected as Primary by turning on Program 51-01-03 instead of Program 51-03-01 and specifying the IP address of the Primary using Program 51-04-01.

❍   During Fail-Over, the IP Multiline terminals can connect to the Secondary System, however when the Primary System comes back online, a 60-second reset of the Secondary System is required. IP Terminal Programming is required.

SECTION 7   TOP PRIORITY PRIMARY SYSTEM (PRIMARY SYSTEM AUTOMATIC INTEGRATION)
When the Primary System is temporarily disconnected/powered down, another communications server can become the new Primary System. To have the original Primary System restored as the Primary when it is reconnected/powered up, enable this function.

❍   The IP address of the Primary communication server must be defined.

❍   By default, this function is **disabled**.

Define the following programs:

51-04-01 = 192.168.1.20
51-06-01 = 1 (On)
51-06-02 = option

New Primary
(old Secondary A)

Old Primary

192.168.1.20
Control reverts back to the old
Primary when it becomes available.

Check the system availability
51-05-04: interval timer

Secondary B

Secondary C

Secondary D

Secondary E

Related Programming:

○ **51-04-01 : IP Address for Top Priority Primary System – Internet Protocol Address for Top Priority Primary System**

Enter the IP address of the Primary SV8100. This setting is needed to use the Primary SV8100 Automatic Integration Feature (Program 51-06-01). If the secondary flag is set in Program 51-01-03 secondary SV8100s connect with this IP address.

○ **51-06-01 : NetLink Primary System Automatic Integration Setting – Primary Automatic Integration**

When Fail-Over occurs, multiple Primary SV8100s may be established. When the connection is recovered, with this option enabled (1), the NetLink feature automatically reestablishes around the top priority Primary SV8100.

○ **51-06-02 : NetLink Primary System Automatic Integration Setting – Blade Reset Timing Option**

When the Primary Automatic Integration reestablishes the NetLink network, the blades in the secondary SV8100s are reset. This option determines if the secondary SV8100 blades are reset only when all blades are idle (0) or anytime (1).

○ **51-09-05 : NetLink TCP Port Settings – Primary Detection Port**

Enter the port number to search for the Top Priority Primary SV8100. If 0 is entered, the port is selected dynamically (0~65535).

7.1 Forced Change Primary System

If you need to change the Primary System, you can specify the communication server manually. While the Top Priority Primary System is running, this feature is not available. To use this feature, define the IP address or System ID of the target communication server. From telephone programming, the user can change the Primary System. When a new Primary System is manually selected, for the IP terminals to reregister to the new Primary, the existing Primary System must be powered down and then powered up.

Related Programming:

❍ **51-05-06 : Timer Settings for NetLink – Primary System Seeking Time with Forced Change Primary**

When the Forced Change Primary command is executed, the SV8100 searches for the new Primary SV8100 for this time (1~10800).

❍ **51-06-02 : NetLink Primary System Automatic Integration Setting – Blade Reset Timing Option**

When the Primary Automatic Integration reestablishes the NetLink network, the blades in the secondary SV8100s are reset. This option determines if the secondary SV8100 blades are reset only when all blades are idle (0) or anytime (1).

❍ **51-07-01 : NetLink Forced Change of Primary System Settings - Allow Forced Change of Primary System**

When Fail-Over occurs, you can manually change the Primary SV8100 using Program 51-08, if this option is enabled. Program 51-06-01 must be set to "0".

❍ **51-07-02 : NetLink Forced Change of Primary System Settings – Blade Reset Timing Option**

When the Forced Change of Primary Settings is performed, the blades in the secondary communication servers are reset. This option determines if the secondary SV8100 blades are reset only when all blades are idle (0) or anytime (1).

❍ **51-08-01 : New Primary System Setting – Internet Protocol Address of New Primary**

When forcing the communications server to update to a new Primary SV8100, the communications server uses the IP address defined here as the new Primary.
✎   *After a Forced Change of Primary SV8100 is done, this entry is erased.*

❍ **51-08-02 : New Primary System Setting - System ID of New Primary**

If you already have an IP address registered in Program 51-11-03, you can execute a Forced Change Primary SV8100 by entering the system ID. If this ID is set to 0, the Top Priority SV8100 is selected as the new Primary.
✎   *Blade reset is required.*
When the Primary System is changed by Automatic Integration or Forced Change Primary System, all blades are reset automatically.

7.2   IP Settings

When Fail-Over occurs, the Primary System is changed to another communication server. The IP applications do not know the new primary IP Address, so the following features are disabled after Fail-Over.

❍   SMDR

❍   ACD-MIS

❍   SIP Terminal

❍   Soft Phone

❍   K-CCIS-IP

SECTION 8   COMMUNICATION SERVER DATABASE SYNCHRONIZATION

When a communication server is connected to the NetLink, the databases of the Secondary Systems are updated by the Primary System. Should the Primary System be taken off line, it is possible for any other communication server to take over as the Primary System position at Fail-Over.

By default, synchronization starts three minutes after the Secondary System is connected to the Primary System. Each Secondary System is updated one by one.

The databases are synchronized based on the schedule defined in the database programming.
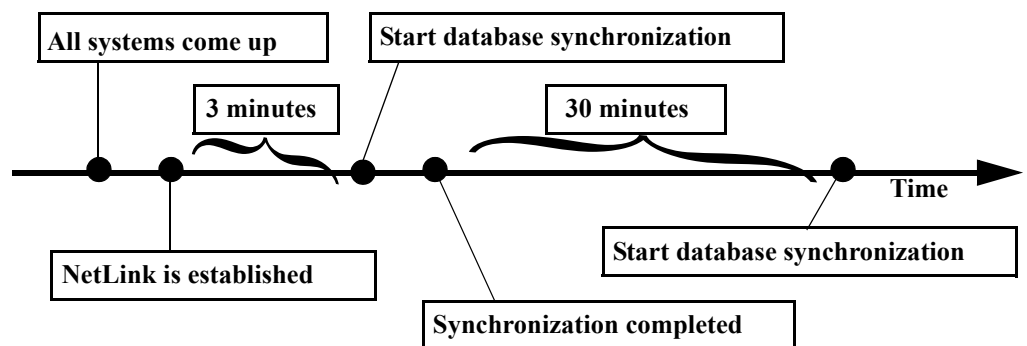
Be careful when changing the communication server database. If the Primary System is changed before synchronization, the edited data is lost.

8.1   Asynchronous Settings

The following program settings are not updated by the Primary System:

10-01, 10-02, 10-12, 10-13, 10-14, 10-15, 10-16, 10-45, 51-01, 90-01, 90-09.

8.2   SRAM Database

Data in SRAM area is not transferred to Secondary Systems. When Fail-Over occurs, a terminal may loose DND, Caller ID History, etc.

Related Programming:

❍   **51-09-06 : NetLink TCP Port Settings – Database Replication Secondary System Listening Port**

Define the listening port used so the Secondary SV8100 can replicate the Primary SV8100 database (0~65535).

❍   **51-09-07 : NetLink TCP Port Settings – Database Replication Primary System Detection Port**

Define the port used for communication so that the Primary SV8100 may synchronize the Secondary SV8100 with the program data (0~65535). If 0 is entered, the port is selected dynamically.

❍   **51-16-01 : NetLink System Data Replication Mode Setting – System Data Replication Mode**

Set the replication mode (0 = Disabled, 1 = Setting Time Mode, 2 = Interval Mode). An entry of 1 replicates the data at the time set in Program 51-16-02. If this option is set to 2, replication occurs at the time set in Program 51-16-03.

❍   **51-16-02 : NetLink System Data Replication Mode Setting – System Data Replication Time Setting**

If Program 51-16-01 is set to 1, set the time to replicate the program data (0000~2359).

❍   **51-16-03 : NetLink System Data Replication Mode Setting – System Data Replication Interval Setting**

If Program 51-16-01 is set to 2, set the time between replicating the program data (15-1440 minutes).

❍   **51-16-04 : NetLink System Data Replication Mode Setting – Replication Time Stamp**

This program displays the last time the program data was replicated. This is automatically updated when the replication occurs. This option is view-only.

❍   **51-16-05 : NetLink System Data Replication Mode Setting – System Data Replication Wait Time**

When a NetLink network is created, this option determines the time the SV8100 waits before replication is started (1~86400 seconds).

❍ **51-16-06 : NetLink System Data Replication Mode Setting – System Data Replication Interval**

Set the time to start replication to the next node after replication has completed to one node (1~86400 seconds).

SECTION 9   CREATE RECOVERY DATABASE

Before assigning a System ID, you should create a Recovery Database. The Recovery Database can be used when you want to disconnect the communication server from the NetLink.

✎   *Communication server reset is required.*

Related Programming:

❍ **90-57-01 : Backup Recovery Data**

Back up the SV8100 data file preserved in the flash memory on the CCPU. This data is used for recovering data if required. Up to five recovery files can be preserved in the flash memory on the CCPU.

❍ **90-58-01 : Restore Recovery Data**

If required, use this option to restore one Backup Recovery Data file (files 1-5, saved in Program 90-57-01) to restore the previous SV8100 data. After executing this option, the SV8100 restarts automatically.

❍ **90-59-01 : Delete Recovery Data**

Delete the Backup Recovery Data files (files 1~5, saved in Program 90-57-01), if needed.

SECTION 10   LICENSING CONTROL

For the NetLink feature, a license with an activation code is required. This license determines how many communication servers can be connected.

All licenses for all options (applications, adapters, ports, etc.) must be registered in the Primary System.

When Fail-Over occurs:

The Primary System gives a copy of the activation codes to the Secondary Systems. This allows the features to continue even if the Primary System fails and a Secondary System becomes the new Primary System.

However, the license of Secondary System is temporary – lasting seven days. After seven days, the communication server is reset and the licenses are removed.

If the link is recovered and original Primary System is back in service, the licenses are returned to use without any expiration time.

❍ License information in the Primary is copied to the Secondary site when doing database duplication.

❍ Duplicate license information in the Secondary System is available for only seven days.

❍ After seven days, the license(s) expire. To renew the license(s), a connection to the original Primary site must be reestablished. (After the connection to the Primary is recovered, if Fail-Over occurs again, the license(s) are once again enabled for seven days to the new Primary System.)

❍ When the Forced Change of the Primary System (Program 50-07-01) is used to manually force the original Primary System to become one of the Secondary Systems in a Netlink Network, the license information will still be available.

❍ If a user wants to enter another additional feature license, it must be entered on the original Primary System.

SECTION 11   MAC ADDRESS AUTHORIZATION

NetLink can reject the connection from unauthenticated communication servers.

The Primary System finds the MAC address of the communication server which is trying to connect, and if the Primary System does not know the MAC address, the connection is refused.

Related Programming:

❍ **51-02-04 : NetLink System Individual Setting – System Authentication MAC Address**

When Program 51-13-03 is enabled, the SV8100 checks this MAC address against the MAC address of the connecting CCPU. If the value is different, the connection is refused.

❍ **51-13-03 : NetLink Option Settings – MAC Address Authentication**

When enabled (1), connection authentication of the SV8100 is done with the MAC address set in Program 51-02-04. The system compares its own MAC address and if the address does not match, the Primary System rejects the connection.

SECTION 12   MAINTENANCE FEATURES

Confirming NetLink Basic Information

From a multiline terminal, press **Feature + 5**. The LCD display shows the System ID, Primary/Secondary, IP Address of the VOIPDB (not CD-CP00-US), and the Primary System ID.

Confirming Communication Server Status

If an installer needs to know the status of each communication server, on the Primary System, access Program 51-11 from telephone programming. The following is displayed:

❍   IP Address

❍   MAC Address (CD-CP00-US)

❍   Communication Server Name

❍   Software Version

❍   Primary Priority

❍   Connect Status
   – OFF: The communication server is disconnected.
   – ON: The communication server is connected as secondary.
   – Primary: The communication server is connected as primary.

SECTION 13   PORT ASSIGNMENT

Port assignment starts from the Primary System lowest slot number. The port numbers for the Secondary Systems are assigned in the order in which the communication servers are connected. The next port number is assigned to the first Secondary System which connects to the NetLink SV8100.

(Example)

Primary System

Slot	CD-CP00-US	
Slot	16ESIU	Ext port: 1 16
Slot	4COIU	Trunk port: 1 4
Slot	4BRIU	Trunk port: 5 12

Secondary System 1 Connected First

Slot	CD-CP00-US	
Slot	16ESIU	Ext port: 17 32
Slot	8SLIU	Ext port: 33 40
Slot	4BRIU	Trunk port: 13 20
Slot 5		

Secondary System 2 Connected Second

Slot	CD-CP00-US	
Slot	16ESIU	Ext port: 41 56
Slot	2BRIU	Trunk port: 21 24

As described above, port assignment is performed in the order which the communication server connects. With this condition, if a 4COIU is installed to the Communication Server 1 – slot 5, the trunk ports are from 25 to 28. To predefine the slots (and not have them automatically assigned by the communication server), use PCPro.

SECTION 14   SLOT CONTROL

When performing a Remote Upgrade, you must connect to each individual system in a NetLink Network. A Remote Upgrade to the Primary System does not upgrade the Secondary Systems.

In the NetLink, the Primary System needs to control all the slots of the Secondary Systems. For the Primary System to handle more slots, there is a Virtual Slot Function in NetLink.

This function allows the Primary System to control 240 slots and, users can check what kind of blade is installed in each communication server and how many blades are available in the link.

Virtual Slot Image
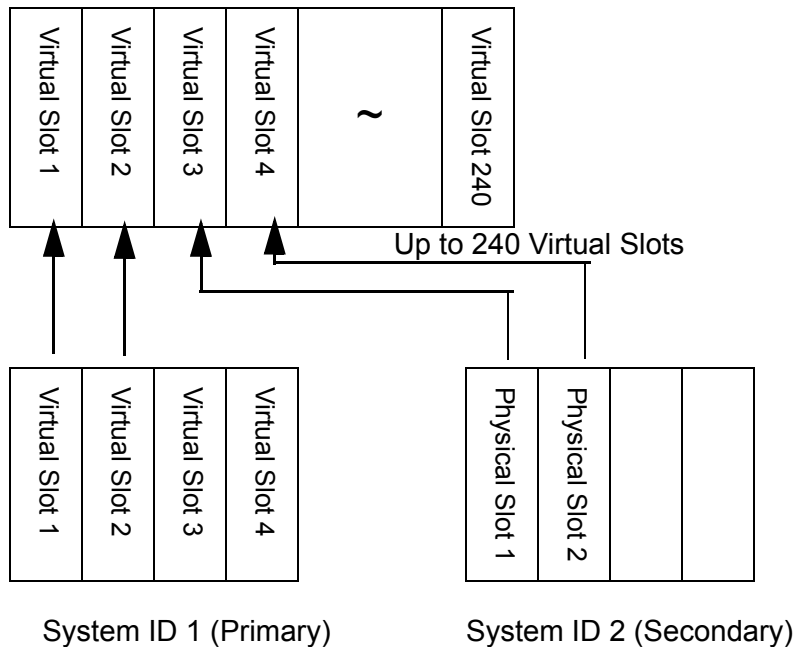
Related Programming:

❍ **51-10-01 : Remaining Virtual Slots**

View the remaining number of slots which can be controlled by NetLink. The NetLink feature can control up to 240 virtual slots maximum. (The physical slots in the NetLink network are maintained as virtual slots by the SV8100.) *This option is not user-definable.*

Slot Deletion

1. Unplug the blade from the communications server chassis.

2. Use **Program 90-05 : Slot Control** and select menu #1 (delete command).

3. Specify the System ID and Slot Number.

✎ *The information in the virtual slot is deleted as well.*

Communication Server Information Deletion

❍ **Using Program 51-14-01 : NetLink System Control – Delete System Information, Specify the System ID.**

To use this program, the communications server must not be connected to the network.

All virtual slot information of the communication server is deleted. The communication server no longer appears in Program 51-11.

SECTION 15   TIME ZONE SETUP

If a communication server in the network is in a different location, the time zone may be different. To adjust the clock display on multiline terminals, the user can define the offset from the Primary System.

The following features are adjusted when the offset is enabled in Program 51-13-02:

❍ Terminal Time Display

❍ Incoming History/Redial History

❍ VRS Recording Announce

❍ Date and Time Setting by Service Code

❍ Calendar Setting by Service Code

❍ Alarm Clock Setting

❍ Hotel/Motel Wake-Up Call

Related Programming:

❍   **10-01-01 - 10-01-07 : Time and Date**

Change the SV8100 Time and Date through SV8100 programming. Extension users can also dial Service Code 728 to change the Time, if allowed by an extension Class of Service.

❍   **51-02-02 : NetLink System Individual Setting – Time Zone – Hour**

Determine the offset hours from the Primary SV8100. This setting affects the time display on display telephones (0~24 = -12 ~ +12 hours).

❍   **51-02-03 : NetLink System Individual Setting – Time Zone – Minute**

Determine the offset minutes from the Primary SV8100. This setting affects the time display on display telephones (0 ~ 120 = -60 ~ +60 minutes).

❍   **51-13-02 : NetLink Option Settings – Time Zone Enhancing**

When enabled (1), the time zone is applied to the following items:
LCD Clock Display, Caller ID History, VRS Time Announce, Time and Date Set by Service Code, Alarm Clock, Hotel Mode Wake-Up Call (time announce included).

When disabled (0), the time zone is applied only to the LCD Clock Display and Caller ID History.

Program 51-02-02 must also be set for this option.

SECTION 16   VOIP HANDLING

Using VoIP equipment is permitted with NetLink. A maximum of VoIP 128 channels can be controlled.

When an analog/digital terminal talks via NetLink, a VoIP resource in the communications server to which the terminal is connecting is used.

When an IP terminal talks to an analog/digital terminal via NetLink:

❍   For an IP terminal, a VoIP resource is not required.

❍   For an analog/digital terminal, one VoIP resource is required.

When IP terminals talk to an IP terminal, they follow Peer-to-Peer setting:

❍   Peer to Peer: VoIP Resource is not required.

❍   Not Peer to Peer: Two VoIP resources are required.

When an analog/digital terminal uses a CO line via NetLink:

❍   The chassis which has the analog/digital terminal needs one VoIP resource, and the chassis which has the CO line needs one VoIP resource.

When an IP terminal uses a CO line via NetLink:

❍   The chassis which has the CO line needs one VoIP resource.

When an IP terminal talks to an IP trunk:

❍   The Primary System requires two VoIP resources.

When an analog/digital terminal talks to an IP trunk via NetLink:

❍   The chassis which has the analog/digital terminal needs one VoIP resource.

❍   The Primary System requires two VoIP resources.

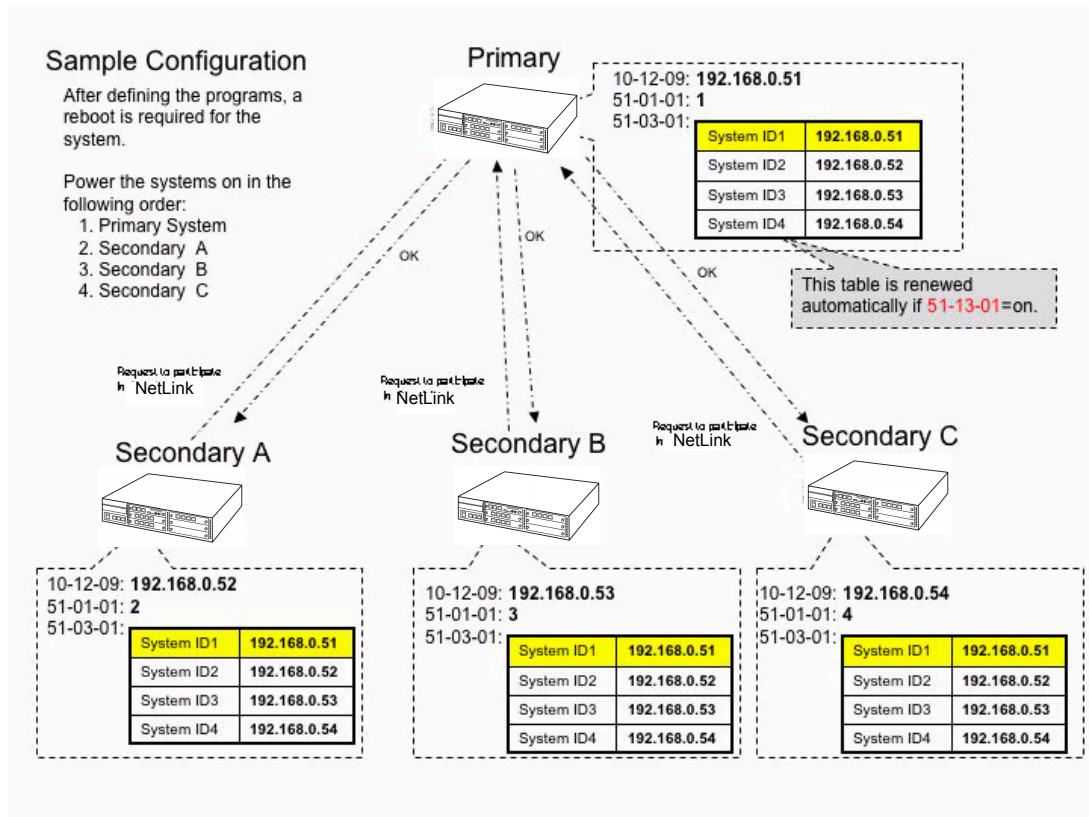The number in each box indicates how many VoIP resources are required to talk.

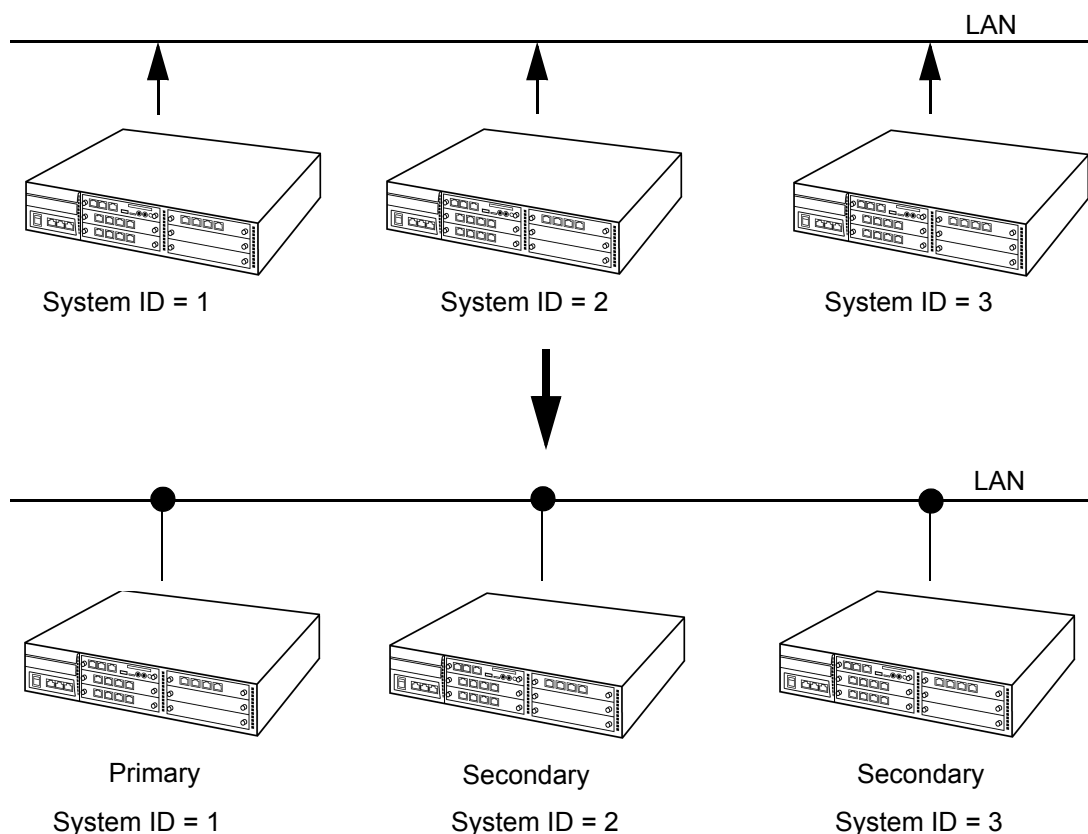|		Primary System				Secondary 1		Secondary 2	
		TDM Terminal	IP Terminal DT700	CO Analog/ Digital	IP Trunk	TDM Terminal	CO Analog/ Digital	TDM Terminal	CO Analog/ Digital
Primary System	TDM Terminal	0	P:1	0	P:1	P :1 S1 :1	P :1 S1 :1	P :1 S2 :1	P :1 S2 :1
	IP terminal DT700)	P:1	0	P:1	P:2	S1 :1	S1 :1	S2 :1	S2 :1
Secondary System 1	TDM Terminal	P:1 S1:1	S1:1 P:2	P:1 S1:1	P :2 S1 :1	0	0	P1 :1 S2 :1	S1 :1 S2 :1
Secondary System 2	TDM Terminal	P:1 S2:1	S2:1 P:2	P:1 S2:1	P :2 S2 :1	S1 :1 S2 :1	S1 :1 S2 :1	0	0

✎   *P: Primary System*
      S1: Secondary System #1
      S2: Secondary System #2

About VOIP DSP Settings

Program 84-26 determines the VoIPDB IP address and port number. The settings are programmed by the Primary System.

If you assign a VoIPDB IP address in the Secondary System before connecting the Primary System, the Primary System reflects the data for the VoIPDB settings.

Sample Configuration

After defining the programs, a reboot is required for the system.

Power the systems on in the following order:
1. Primary System
2. Secondary A
3. Secondary B
4. Secondary C

System ID = 1   System ID = 2   System ID = 3

Primary   Secondary   Secondary

System ID = 1   System ID = 2   System ID = 3

Configuration Flow (Case 1) – Primary Based on System ID

1. Create a Recovery Database **– _Recommended_**.
   Because each communication server database and communication server con-figuration are updated by the Primary System, creating a recovery database is recommended. This may be needed if the communication server goes back to a stand-alone server.
   Refer to the Section 8 Communication Server Database Synchronization on page 10-41 and Section 9 Create Recovery Database on page 10-43 sections.

   1. Before programming the NetLink feature, general IP configuration must be performed.

      ◆ Program 10-12-09 : VoIPDB IP Address
      ◆ Program 10-12-10 : Subnet Mask
      ◆ Program 84-26-01 : VoIPDB DSP IP Address
      ◆ Program 10-12-03 : Default Gateway

2.   For NetLink, the minimum programs must be set:

  ◆   System IP Address List (Program 51-03-01)
  ◆   System ID (Program 51-01-01)

3.   After entering the System ID, the communication server asks to reboot the communication server. Reboot.

4.   When the communication server comes up, the Primary System and Secondary Systems are determined automatically.

      If you need to change the Primary System to another server, use 7.1 Forced Change Primary System on page 10-40.

      When connection cannot be made to the original Primary System, and a new Primary is selected, it can automatically return the original server as the Primary System after connection/power is restored. Refer to the Section 7 Top Priority Primary System (Primary System Automatic Integration) on page 10-38 function.

      ✎   *If the VM8000 voice mail is installed, it must be in the Primary System (Site A).*
            Assigning Top Priority Primary System and Primary System Automatic Integration are recommended.

Specify the communication server in Site-A for the Top Priority Primary System
Site-A becomes the Primary System. Without this function, either Site-A or Site-B can become the Primary System.

Enable Primary System Automatic Integration
When Site-A is disconnected from the LAN, Fail-Over works and Site-B becomes the Primary.

If you would rather stop the Fail-Over function, you can disable it by setting Program 51-05-02 to 0. With it disabled, Site-B is locked while the Primary System is down.

If Site-A connects to the LAN again, the Automatic Integration function reestablishes the NetLink and Site-A becomes the Primary. Without this function, Site-B remains the Primary.

Configuration Flow (Case 2) – Primary Defined Based on Programming

After you decide which communication server should become the Primary:

1.   Create a Recovery Database **– *Recommended***.
      Because each communication server database and communication server configuration is updated by the Primary System, creating a recovery database is recommended. This may be needed if the communication server goes back to a stand-alone server.
      Refer to the Section 8 Communication Server Database Synchronization on page 10-41 and Section 9 Create Recovery Database on page 10-43 sections.

5.   Before programming the NetLink feature, general IP configuration must be performed.

  ◆   Program 10-12-09 : VOIPDB IP Address
  ◆   Program 10-12-10 : Subnet Mask
  ◆   Program 84-26-01 : VOIPDB DSP IP Address
  ◆   Program 10-12-03 : Default Gateway, etc.

6.   Program the following NetLink settings:

Primary - Site A	Secondary - Site B
51-01-01 System ID	51-01-01 System ID
51-04-01 Top Priority System	51-04-01 Top Priority System
Enter own IP address	Enter site A (Primary) IP Address
51-06-01 Enable Automatic Integration	51-01-03 Set Secondary Flag

7.   After entering the System ID, the communication server asks to reboot the communication server. Reboot.

8.   When the communication servers come up, the Primary System and Secondary Systems are connected and start working.

9.   For database synchronization, wait three minutes (or more).

Programming

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

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

The reverse type (white on black) just beneath the Description heading is the program access level. You can use the program only if your access level meets or exceeds the level the program requires. Refer to Section 17 How to Enter Programming Mode on page 10-54 for a list of the system access levels and passwords.

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

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

1. Enter the programming mode.

2. 15-07-01

```
15-07-01    TEL
KY01 = *01
  ←                    →
```

tells you to enter the programming mode, dial 150701 from the telephone dial pad. After you do, you see the message 15-07-01 TEL on the first line of the telephone display. This indicates the program number (15-07), item number (01), and that the options are being set for the extension. The second row of the display KY01 = *01 indicates that Key 01 is being programmed with the entry of *01. The third row allows you to move the cursor to the left or right, depending on which arrow is pressed. To learn how to enter the programming mode, refer to Section 17 How to Enter Programming Mode below.

SECTION 17   HOW TO ENTER PROGRAMMING MODE

To enter programming mode:

1. Go to any working display telephone.

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

2. *Do not* lift the handset.

3. Press **Speaker**.

4. **# * # *** .

    ```
    Password
    ```

5. Dial the system password + **Transfer**.
    Refer to the following table for the default system passwords. To change the passwords, use Program 90-02: Programming Password Setup.

Password	User Name	Level	Programs at this Level
47544	**necii**	1 (MF)	Manufacturer (MF): All programs
12345678	**tech**	2 (IN)	Installation (IN): All programs in this section not listed below for SA and SB

Password	User Name	Level	Programs at this Level
0000	**ADMIN1**	3 (SA)	System Administrator – Level 1 (SA): 10-01, 10-02, 10-12, 10-13, 10-14, 10-15, 10-16, 10-17, 10-18, 10-22, 12-02, 12-03, 12-04, 15-01, 15-07, 15-09, 15-10, 15-11, 20-16, 21-07, 21-14, 22-04, 22-11, 25-08, 30-03, 32-02, 40-02, 41-02, 41-03, 41-04, 41-05, 41-06, 41-07, 41-08, 41-09, 41-10, 41-11, 41-12, 41-13, 41-14, 41-15, 41-16, 41-17, 41-18, 90-03, 90-04, 90-06, 90-07, 90-18, 90-19
9999	**ADMIN2**	4 (SB)	System Administrator – Level 2 (SB): 13-04, 13-05, 13-06

SECTION 18   HOW TO EXIT PROGRAMMING MODE

To exit the programming mode:

When you are done programming, you must be out of a program option to exit (press **Answer** to exit the program option).

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

```
                    Program Mode
        Base  Service  OP1   OP2
```

2.   Press **Speaker**. If changes were made to system programming, Saving System Data is displayed.

3.   The display shows Complete Data Save when completed and exits the telephone to idle.

   ✎   *To save a customer database, a blank USB Drive is required. Insert the USB Drive into the CD-CP00-US and, use Program 90-03, to save the software to the USB Drive. (Use Program 90-04 to reload the customer data if necessary). A USB Drive can hold only one customer database. Each database to be saved requires a separate drive.*

SECTION 19   USING KEYS TO MOVE AROUND IN THE PROGRAMS

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

Table 10-4 Keys for Entering Data

Keys for Entering Data	
Use this key...	**When you want to...**
0~9 and ∗	Enter data into a program.
Transfer	Complete the programming step you just made (e.g., pressing **Enter** on a PC keyboard). When a program entry displays, press **Transfer** to bypass the entry without changing it.
Recall	Delete the entry to the left (e.g., pressing **Backspace** on a PC keyboard).
Hold	Delete or clear all characters to the right of the cursor.
Answer	Exit one step at a time from the program window currently being viewed. For example, if programming item 5 in 15-03, press **Answer** to enter a new option in program 15-03. Press **Answer** again to select a new program in the 15-XX series. Press **Answer** a third time to enter a new program beginning with **1**. Press **Answer** one last time to bring you to the beginning program display to allow you to enter any program number.
MIC	Press **MIC** to switch between the different input data fields. The cursor moves up to the top row of the display. Press **MIC** again to move the cursor back to the middle row.
LINE KEYS	Use programmed settings to help with the program entry. These settings vary between programs from LINE 1 = 0 (off) and LINE 2 = 1 (on) to preset values for timers where LINE 1 = 5, LINE 2 = 10, LINE 3 = 15, etc. For programs with this option, the line key, which currently matches the programmed setting, lights steady. The display can also indicate Softkey, which will allow you to select the values as well (-1 and +1 will step through these pre-programmed settings.)
LINE KEY 1	Program a pause into a Speed Dialing bin.
LINE KEY 2	Program a recall/flash into a Speed Dialing bin.
LINE KEY 3	Program an @ into a Speed Dialing bin.

Table 10-4 Keys for Entering Data (Continued)

Keys for Entering Data	
Use this key...	**When you want to...**
VOL ▲	Scroll backward through a list of entry numbers (e.g., from extension etc.) or through entries in a table (e.g., Common Permit Table). If you enter data and then press this key, the system accepts the data before scrolling forward.
VOL ▼	Scroll forward through a list of entry numbers (e.g., from extension etc.) or through entries in a table (e.g., Common Permit Table). If you enter data and then press this key, the system accepts the data before scrolling backward.

SECTION 20   PROGRAMMING NAMES AND TEXT MESSAGES

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

Table 10-5 Keys for Entering Names

Use this keypad digit . . .	When you want to. . .
1	Enter characters: **1 @ [¥] ^ _ ` { | } Æ ¨ Á À Â Ã Ç É Ê ì ó**
2	Enter characters: **A-C, a-c, 2**.
3	Enter characters: **D-F, d-f, 3**.
4	Enter characters: **G-I, g-i, 4**.
5	Enter characters: **J-L, j-l, 5**.
6	Enter characters: **M-O, m-o, 6**.
7	Enter characters: **P-S, p-s, 7**.
8	Enter characters: **T-V, t-v, 8**.
9	Enter characters: **W-Z, w-z, 9**.
0	Enter characters: 0 ! " **#** $ % & ' ( ) ô Õ ú ä ö ü α ε θ
*	Enter characters: ∗ + , - . / : ; < = > ? π Σ σ Ω □ ¢ £

Table 10-5 Keys for Entering Names

Use this keypad digit . . .	When you want to. . .
#	**#** = Accepts an entry (only required if two letters on the same key are needed – ex: TOM). Pressing **#** again = Space. (In system programming mode, use the right arrow Softkey instead to accept and/or add a space.)
Conf	Clear the character entry one character at a time.
Hold	Clear all the entries from the point of the flashing cursor to the right.

SECTION 21   USING SOFTKEYS FOR PROGRAMMING

Each UNIVERGE SV8100 display telephone provides interactive Softkeys for intuitive feature access. The options for these keys automatically change depending on where you are in the system programming. Press the Softkey located below the option you want, and the display changes accordingly.

```
–
          Program Mode
Base      Service   OP1     OP2
   ◯        ◯      ◯       ◯
```

Press the VOLUME <Webdings>5 or VOLUME <Webdings>6 to scroll between the menus.

```
–
          Program Mode
CCIS        Hard   Mtnance
   ◯        ◯      ◯       ◯
```

SECTION 22   WHAT THE SOFTKEY DISPLAY PROMPTS MEAN

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

Table 10-6 Softkey Display Prompts

<table>
<tr><th colspan="2">Softkey Display Prompts</th></tr>
<tr><th>If you press this Softkey . . .</th><th>The system will. . .</th></tr>
<tr><td>back</td><td>Go back one step in the program display.
You can press VOLUME ▲ or VOLUME ▼ to scroll forward or backward through a list of programs.</td></tr>
<tr><td>↑</td><td>Scroll down through the available programs.</td></tr>
<tr><td>↓</td><td>Scroll up through the available programs.</td></tr>
<tr><td>select</td><td>Select the currently displayed program.</td></tr>
<tr><td>←</td><td>Move the cursor to the left.</td></tr>
<tr><td>→</td><td>Move the cursor to the right.</td></tr>
<tr><td>-1</td><td>Move back through the available program options.</td></tr>
<tr><td>+1</td><td>Move forward through the available program options.</td></tr>
</table>

Program 10 : System Configuration Setup
10-12 : CD-CP00-US Network Setup

Level:
SA

Description

Use **Program 10-12 :** CD-CP00-US **Network Setup** to setup the IP Address, Subnet-Mask, and Default Gateway addresses.

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

Input Data

Item No.	Item	Input Data			Default	Description
01	IP Address	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.254.255.254 192.0.0.1 ~ 223.255.255.254			192.168.0.10	Set for CD-CP00-US.
02	Subnet Mask	128.0.0.0 240.0.0.0 254.0.0.0 255.192.0.0 255.248.0.0 255.255.0.0 255.255.224.0 255.255.252.0 255.255.255.128 255.255.255.240 255.255.255.254	192.0.0.0 248.0.0.0 255.0.0.0 255.224.0.0 255.252.0.0 255.255.128.0 255.255.240.0 255.255.254.0 255.255.255.192 255.255.255.248 255.255.255.255	224.0.0.0 252.0.0.0 255.128.0.0 255.240.0.0 255.254.0.0 255.255.192.0 255.255.248.0 255.255.255.0 255.255.255.224 255.255.255.252	255.255.255.0	The setting of Subnet Mask is invalid when all Host Addresses are 0. If the network section is: 0, 127, 128.0, 191.255, 192.0.0, 223.255.255 The setting of Subnet Mask is invalid.
03	Default Gateway	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.254.255.254 192.0.0.1 ~ 223.255.255.254			0.0.0.0	IP Address for Router.

Input Data (Continued)

Item No.	Item	Input Data	Default	Description
04	Time Zone	0~24 (0 = -12 Hours and 24 = +12 Hours)	+7 (-5 hours)	Determine the offset from Greenwich Mean Time (GMT) time. Then enter its respective value. For example, Eastern Time (US and Canada) has a GMT offset of -5. The program data would then be 7 (0= -12, 1= -11, 2= -10, 3= -9, 4= -8, 5= -7, 6= -6, 7= -5, ……24= +12)
05	**NIC Interface**	0 = Auto Detect 1 = 100Mbps, Full Duplex 2 = 100Mbps, Half Duplex 3 = 10Mbps, Full Duplex 4 = 10Mbps, Half Duplex	0	NIC Auto Negotiate (CD-CP00-US)
06	**Network Address Port Translation (NAPT) Router Setup**	0 = No (Disable) 1 = Yes (Enable)	0	If using an external NAPT Router or not.
07	**NAPT Router IP Address (Default Gateway [WAN])**	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0	Set the IP address on the WAN side of router.
08	**ICMP Redirect**	0= (Enable) 1= (Disable)	0	When receiving ICMP redirect message, determine if the IP Routing Table updates automatically or not.
09	**IP Address**	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	172.16.0.10	Set for IPLA.

Input Data (Continued)

Item No.	Item	Input Data			Default	Description
10	**Subnet Mask**	128.0.0.0	192.0.0.0	224.0.0.0	255.255.0.0	Set for IPLA.
		240.0.0.0	248.0.0.0	252.0.0.0		
		254.0.0.0	255.0.0.0	255.128.0.0		
		255.192.0.0	255.224.0.0	255.240.0.0		
		255.248.0.0	255.252.0.0	255.254.0.0		
		255.255.0.0	255.255.128.0	255.255.192.0		
		255.255.224.0	255.255.240.0	255.255.248.0		
		255.255.252.0	255.255.254.0	255.255.255.0		
		255.255.255.128	255.255.255.192	255.255.255.224		
		255.255.255.240	255.255.255.248	255.255.255.252		
		255.255.255.254	255.255.255.255			
11	**NIC Setup**	0 = Auto Detect			0	Set for IPLA.
		1 = 100Mbps, Full Duplex				
		2 = 100Mbps, Half Duplex				
		3 = 10Mbps, Full Duplex				
		4 = 10Mbps, Half Duplex				
		5 = 1 Gbps, Full Duplex				
		6 = 1 Gbps, Half Duplex				

Conditions

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

Feature Cross Reference

❑ Voice Over Internet Protocol (VoIP)

Program 51 : NetLink Service
51-01 : NetLink System Property Setting

Level:
IN

Description

Use **Program 51-01 : NetLink System Property Setting** to define the parameters of the NetLink feature.

✎   *Each system must be set with its own information.*

✎   *When the NetLink System ID is changed (Item 01), the system must be reset.*

Input Data

Item No.	Item	Input Data	Default
01	**NetLink System ID** This is the ID of each NetLink system. Setting should insure that no overlap occurs between nodes.	0~50 (0 = No operation)	0
02	**Primary Candidate Order** When the Primary system is turned off or disconnect from network, this value is used to select a new Primary system. Smaller number is higher priority. If this value is the same number, the System ID (Program 51-01-01) is referred, and the system which has the smaller number is selected as Primary system.	1~50	30
03	**Secondary System Flag** 0: NetLink is dynamically established based on Node List in Program 51-03-01. Primary System is selected in the order which the system wakes up. 1: The system connects with Top Priority Primary System. If Top Priority Primary System is not found, the system searches Primary System as if this setting is 0.	0 = Disable 1 = Enable	0

Conditions
None

Feature Cross Reference

None

Program 51 : NetLink Service
51-02 : NetLink System Individual Setting

Level:
IN

Description

Use **Program 51-02 : NetLink System Individual Setting** to set system data for each NetLink system.

✎ *Program 51-02-03 is not used in US, but is used in other countries.*

Input Data

System ID	1~50

Item No.	Item	Input Data	Default
01	**System Name** This name is given to each system.	Up to 20 characters.	blank
02	**Time Zone (Hour)** Determine the time offset from the Primary system. (0 = -12, 1 = -11, 2 = -10.... 12 = 0 13 = +1, 14 = +2, 24 = +12) This setting affects Time Display on MLT (see Program 51-13-02).	0~24	12
03	**Time Zone (Minute)** Determine the time offset from the Primary system. (0 = -60, 1 = -59, 2 = -58.... 120 = +60) This setting affects Time Display on MLT (see Program 51-13-02). This program is not used in the US, but is used in other countries.	0~120	60
04	**Authenticate System MAC Address** To use this function, set Program 51-13-03 to 1 (enable), NetLink systems reject the connection from unauthenticated system access.	00-00-00-00-00-00~ FF-FF-FF-FF-FF-FF	00-00-00-00-00-00

Conditions
None

Feature Cross Reference

None

Program 51 : NetLink Service
51-03 : NetLink Internet Protocol Address List Setting

Level:
IN

Description

Use **Program 51-03 : NetLink Internet Protocol Address List Setting** to set the IP address of the NetLink system.

Input Data

List ID	1~50

Item No.	Item	Input Data	Default
01	**Internet Protocol Address List** The system seeks the Primary system based on this list. When there is no Primary system yet, or Fail-Over occurs, Node List is referred to establish new link. This setting is necessary when Program 51-01-03 is 0, or Program 51-05-02 is other than 0. Once the system connects to the Primary System, this setting is updated by the Primary system when Program 51-13-01 is On. So, enter IP address of the systems which may become Primary at least.	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0

Conditions

❍   The system seeks Primary system based on this list.

❍   When there is no Primary System yet, or Fail-Over occurs, Node List is referred to establish new link.

❍   This setting is necessary when Program 51-01-03 is 0, or Program 51-05-02 is other than 0. Once the system connects to the Primary System, this setting is updated by the Primary system when Program 51-13-01 is on. So, enter IP address of the systems which may become Primary.

Feature Cross Reference

None

Program 51 : NetLink Service
51-04 : IP Address Setting of Top Priority Primary System of NetLink

Level:
IN

Description

Use **Program 51-04 : IP Address Setting of Top Priority Primary System of NetLink** to set the IP address of the new Primary System.

Input Data

List ID	1~50

Item No.	Item	Input Data	Default
01	**Internet Protocol Address of Top Priority Primary** Enter the IP address of the Top Priority Primary System. To use this feature, set Program 51-06-01 to 1(On).	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0

Conditions
None

Feature Cross Reference

None

Program 51 : NetLink Service
51-05 : NetLink Timer Settings

Level:
IN

Description

Use **Program 51-05 : NetLink Timer Settings** to set the various timers within the NetLink system.

Item No.	Item	Input Data	Default
01	**Keep Alive Sending Interval** This is the Keep Alive timer sending time from the Secondary system to confirm communication with the Primary system.	1~3600	5
02	**Keep Alive Response Waiting Time** This is the time the Secondary system waits for a response from the Primary system before cutting off communication.	0, 5~10800 (0 = infinity)	20
03	**Primary Search Packet Sending Interval** While searching the Primary system, the system sends a packet at this interval.	1~3600	5
04	**Primary Search Time Maximum Value** Total time of Primary system seek time.	5~10800	20
05	**Top Priority Primary Detection Packet Sending Interval** When current Primary system is not Top Priority Primary System, the system sends packet to check if Top Priority System exists.	1~3600	10
06	**Primary Compulsion Specification Trial Maximum Time** When the forced change Primary command is executed, the system searches the new Primary system for this time.	1~10800	30
07	**Socket Refresh Time** For some reason, the IP connection may become unstable. Then keep-alive function does not work. To avoid this, if there is no data traffic for this time, the socket is refreshed.	20~3600	40

Conditions
None

Feature Cross Reference

None

Program 51 : NetLink Service
51-06 : NetLink Primary Automatic Integration Setting

Level:
IN

Description

Use **Program 51-06 : NetLink Primary Automatic Integration Settings** to set the automatic integration of the Primary system.

Input Data

Item No.	Item	Input Data	Default
01	**Primary Integration Right or Wrong** When LAN cable was divided, multiple Primary systems may appear. If the LAN connection is recovered, multiple Net-Links exist in the network. When this option is enabling, NetLink is composed around Top priority Primary System.	0 = Off 1 = On	0
02	**Package Reset Timing Option** When Primary System Automatic Integration is done, all packages of secondary systems reset. Select the timing of package reset.	0 = Reset when all packages are idle. 1 = Anytime	0

Conditions
None

Feature Cross Reference

None

Program 51 : NetLink Service
51-07 : NetLink Primary Compulsion Specification Setting

Level:
IN

Description

Use **Program 51-07 : NetLink Primary Compulsion Specification Setting** to set compulsion specification of the Primary system.

Input Data

Item No.	Item	Input Data	Default
01	**Forced Change Primary System Enabling** Set this item whether the Forced Change Primary is available or not.	0 = Disable 1 = Enable	1
02	**Package Reset Timing Option** When Forced Change Primary System is done, all packages reset. Select the timing of package reset. 0 = Reset when all packages are idle, otherwise reject Primary System Integration. 1 = Anytime	0 = On 1 = Off	0

Conditions
None

Feature Cross Reference

None

Program 51 : NetLink Service
51-08 : Primary NetLink Setting

Level:
IN

Description

Use **Program 51-08 : Primary NetLink Setting** to set the IP address and system ID of the compulsory specification of the Primary system.

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

Input Data

Item No.	Item	Input Data	Default
01	**IP Address of New Primary System** Enter target IP address for New Primary system. When the Forced Change Primary system is done, this setting is erased.	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0
02	**System ID of New Primary System** When set to 0, top priority Primary system is assumed to be the new Primary system.	0~50	No setting

Conditions
None

Feature Cross Reference

None

<div style="text-align: right">

Program 51 : NetLink Service
51-09 : NetLink Communication Port Settings

</div>

Level:
IN

Description

Use **Program 51-09 : NetLink Communication Port Settings** to set the various communication ports used on the system.

Input Data

Item No.	Item	Input Data	Default
01	**Primary Waiting Port** Set the communication port that the Primary system uses to communicate with the Secondary system.	0~65535	58000
02	**Communication Waiting Port** Port used to communicate between nodes. It is always opened by all nodes.	0~65535	58001
03	**Secondary Communication Port** Secondary system communicates with Primary system at this port number. If 0 is specified, temporary port is selected dynamically.	0~65535	0
04	**Primary Search Port** When Fail-Over occurred, each system communicates with other system at this port number. If 0 is set, temporary port is selected dynamically. If 0 is not specified, the number and continuous maximum 50 number is used. (Ex. 5000 is specified 5001, 5002…5049 are used).	0~65535	0
05	**Primary Detection Port** Enter port number to seek the Top Priority Primary system. If 0 is specified, temporary port is selected dynamically.	0~65535	0
06	**Database Replication Communication Listening Port** Replicate the database.	0~65535	58002

Input Data

Item No.	Item	Input Data	Default
07	**Database Replication Primary Detection Port** Replicate database. If 0 is specified, temporary port is selected dynamically.	0~65535	0

Conditions
None

Feature Cross Reference

None

Program 51 : NetLink Service
51-10 : Virtual Slot Setting

Level:
IN

Description

Use **Program 51-10: Virtual Slot Setting** to view the number of Virtual slots that are remaining in a NetLink network. There can be up to 240 virtual slots available in NetLink.

Input Data

Item No.	Item	Input Data	Default
01	**Number of Available Virtual Slots** 240 slots can be controlled in NetLink. This command can check how many slots are available.		

Conditions

❍ This Program is Read Only.

Feature Cross Reference

None

Program 51 : NetLink Service
51-11 : NetLink System Information

Description

Use **Program 51-11: NetLink System Information** to reference information about other systems in the NetLink network.

Input Data

System ID	1~50

Item No.	Item	Input Data	Default
01	**System Name**	For reference only.	blank
02	**Connected State**	For reference only.	0
03	**IP Address**	For reference only.	000.000.000.000
04	**MAC Address**	For reference only.	00:00:00:00:00:00
05	**Primary Priority Level**	For reference only.	0
06	**Main Software Version**	For reference only.	XX.XX

Conditions

❍ This Program is Read Only.

Feature Cross Reference

None

Program 51 : NetLink Service
51-12 : Primary System Information

Level:
IN

Description

Use **Program 51-12: Primary System Information** to reference information about the Primary System in the NetLink network.

Input Data

Item No.	Item	Input Data	Default
01	**System ID**	For reference only.	0
02	**System Name**	For reference only.	blank
03	**IP Address**	For reference only.	000.000.000.000
04	**MAC Address**	For reference only.	00:00:00:00:00:00
05	**Primary Priority Level**	For reference only.	0
06	**Main Software Version**	For reference only.	XX.XX

Conditions

❍ This Program is Read Only.

Feature Cross Reference

None

<div align="right">

Program 51 : NetLink Service
51-13 : NetLink Options

</div>

Level:
IN

Description

Use **Program 51-13: NetLink Options** to enable automatic IP address List Operation updates, time zone information, and MAC address authorization.

Input Data

Item No.	Item	Input Data	Default
01	**Automatic IP Address List Operation Update** When set to 1, the list in Program 51-03-01 is automatically updated.	0 = Disable (Off) 1 = Enable (On)	1
02	**Time Zone Option** When set to 0, the following features are affected: Clock Display, Incoming/Outgoing History List. When set to 1, the following features are affected: VRS Time Announce, Date and Time Setting Service Code, Alarm Clock setting, and Hotel mode wake-up call.	0 = Disable (Off) 1 = Enable (On)	0
03	**MAC Address Authorization Enable** Refer to Program 51-02-04 to set MAC address.	0 = Disable (Off) 1 = Enable (On)	0

Conditions
None

Feature Cross Reference

None

Program 51 : NetLink Service
51-14 : NetLink System Control

Level:
IN

Description

Use **Program 51-14: NetLink System Control** to delete system and slot information.

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

Input Data

System ID	1~50

Menu Number	1 = System information deletion

Item No.	Item	Input Data	Default
01	**Delete System Information** Delete system information and the slot information. The system must be disconnected.	1~50	1

Conditions
None

Feature Cross Reference

None

<div align="right">

Program 51 : NetLink Service
51-15 : Demonstration Setting

</div>

Level:
IN

Description

Use **Program 51-15: Demonstration Setting** to automatically set the minimum setting values in NetLink. A system reset occurs after this command is executed.

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

Input Data

Menu Number	1 = Primary automatic setting 2 = Secondary 1 - automatic operation setting 3 = Secondary 2 - automatic operation setting 4 = Secondary 3 - automatic operation setting

Conditions
None

Feature Cross Reference

None

Program 51 : NetLink Service
51-16 : NetLink System Data Replication Mode Setting

Description

Use **Program 51-16: NetLink System Data Replication Mode Setting** to set the system data replication between the Primary and Secondary systems.

Input Data

Item No.	Item	Input Data	Default
01	**System Data Replication Mode** Set the synchronous mode of the system data. When set to 1, the systems are synchronized at the time set in Item 02 below. When set to 2, the systems are synchronized at regular time intervals set in Item 03 below.	0 = Disable 1 = Setting Time Mode 2 = Interval Mode	1
02	**System Data Replication Time Setting** Set the time of day that both systems synchronize database (when Item 01 is set to 1.)	0000~2359	0200
03	**System Data Replication Interval Setting** Set the time interval that both systems synchronize database (when Item 01 is set to 2).	15~1440 (minutes)	30 (min)
04	**Replication Time Stamp** Show next replication time. (Read-Only)	Month: 0~12	–
		Day: 0~31	–
		Hour: 00~23	–
		Minute: 00~59	–
05	**System Data Replication Wait Time** Set the wait time until replication starts when NetLink is created.	1~86400 (seconds)	180 sec
06	**System Data Replication Interval** Set time to start replication to the next node after replication to one node is completed.	0~86400 (seconds)	1 sec

Conditions
None

Feature Cross Reference

None

Program 90 : Maintenance Program
90-57 : Backup Recovery Data

Level:
IN

Description

Use **Program 90-57 : Backup Recovery Data** to backup the system data in the flash memory on the CD-CP00-US and to make the recovery data.

Input Data

Data ID	1~5

Item No.	Item	Input Data
01	**Backup Recovery Data**	[Backup?] : Dial **1**+ press **Transfer** (Press **Transfer** to cancel.)

Conditions
None

Feature Cross Reference

None

Program 90 : Maintenance Program
90-58 : Restore Recovery Data

Level:
IN

Description

Use **Program 90-58 : Restore Recovery Data** to select the recovery data stored in the flash memory of the CD-CP00-US. After this command is executed, the system restarts automatically.

Input Data

Data ID	1~5

Item No.	Item	Input Data
01	**Restore Recovery Data**	[Restore & Reset?] : Dial **1**+ press **Transfer** (Press **Transfer** to cancel.)

Conditions
None

Feature Cross Reference

None

Program 90 : Maintenance Program

90-59 : Delete Recovery Data

Level:
SA

Description

Use **Program 90-59 : Delete Recovery Data** to select and delete the recovery data stored in the flash memory of the CD-CP00-US.

Input Data

Data ID	1~5

Item No.	Item	Input Data
01	**Delete Recovery Data**	[Delete?] : Dial **1**+ press **Transfer** (Press **Transfer** to cancel.)

Conditions
None

Feature Cross Reference

None

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WL1700-MH240 Deployment

SECTION 1   OVERVIEW

This chapter provides information for setting up a UNIVERGE WL1700-MH240 Mobility System for the small and middle business (SMB) market, and helps you build your Mobility System with less effort. However, the Mobility System is normally part of an IP network. The success of the installation depends on the structure and components in the IP network. Make sure that you have sufficient knowledge of the customer IP network.

The Mobility System is also a wireless voice and data communication system. This requires knowledge of radio signal propagation. The success of the installation also depends on the radio signal propagation. Make sure that you have sufficient knowledge about this subject as well.

SECTION 2   INTRODUCTION

Wi-Fi telephony enables the convergence of wireless voice and data applications using a common wireless local area network (WLAN). Wi-Fi telephony bridges traditional telecommunications, data communications and mobile technologies, resulting in ease-of-use and cost savings. A Wi-Fi enabled telephone is a WLAN client device, using the same network technology as wireless laptops and PDAs. A Wi-Fi enabled telephone can be functionally equivalent to a wired telephone, and allowing for configuration and management from the local enterprise telephone system. These benefits can result in substantial cost savings over other similar wireless technologies by leveraging the Wi-Fi infrastructure and by eliminating recurring charges or redundant costs.

However, a Wi-Fi telephone does impose different requirements on the network that result in deployment considerations that differ from networks which are optimized for wireless data. Voice and data applications have different attributes and therefore, different network requirements. Because a Wi-Fi telephone is a mobile communication device, it requires special considerations for continuous high-quality connections as a user moves throughout the coverage area. Another significant difference is the tolerance for network errors and delays.

WL1700-MH240 Deployment

While data applications are designed to accept frequent packet delays and retransmissions, voice quality suffers with more than 150 milliseconds of delay or a very small percentage of lost packets. In addition, data applications are typically bursty in terms of bandwidth use, while a telephone conversation uses a consistent and relatively small Network bandwidth.

Using a WLAN for voice is not complex, but there are some aspects that must be considered, particularly for enterprise applications. A critical objective in deploying enterprise Wi-Fi telephony is to maintain similar voice quality, reliability and functionality as is expected in a wired telephone environment.

This chapter gives you the information to set up a Wi-Fi network where the key issues such as coverage, capacity, quality of service (QOS), and wireless security are all cleared.

SECTION 3   WL1700-MH240 SYSTEM GENERAL DESCRIPTION

Refer to Figure 11-1 WL1700-MH240 System. WL1700 Controller is equipment that controls WL1500 Access Points (APs), its own internal AP, and creates an IEEE802.11-standard wireless network. MH240 is a wireless phone that complies with IEEE 802.11b/g specifications and uses a Wi-Fi network consisting of a WL1700 Controller and WL1500 APs distributed throughout an enterprise environment.
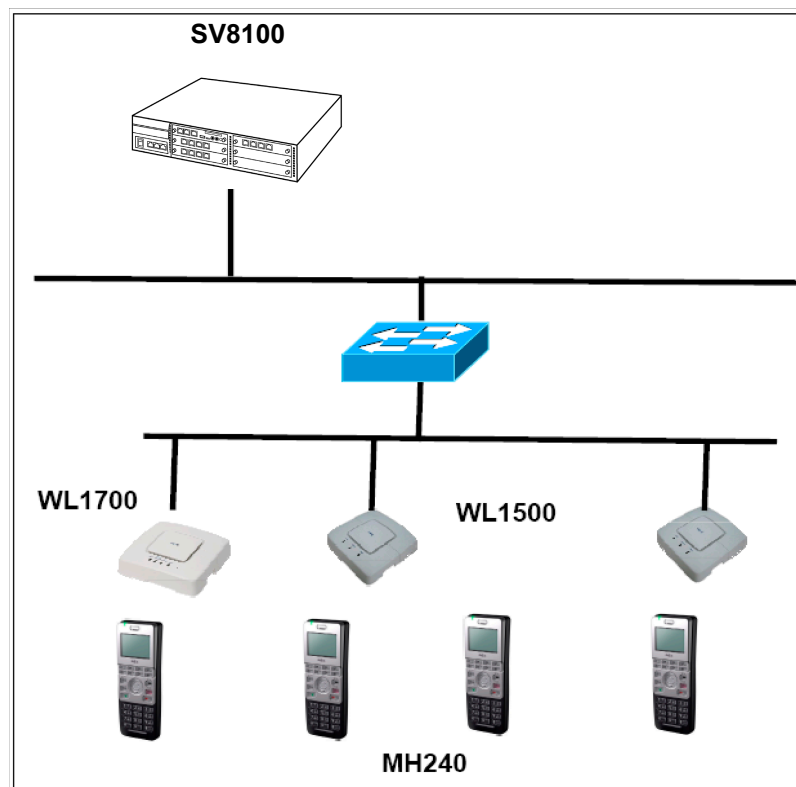
Figure 11-1 WL1700-MH240 System

3.1   Physical Connections

WL1700-MH240 infrastructure components must connect to a facility LAN using WL1700-MS. WL1700-MS must be provided power from Power over Ethernet (PoE) Feeding Adaptor or PoE Hub/Switch. Use straight cables to connect to the Ether connector.

Network wiring is an important component of any Ethernet-based system, and is subject to local and state building code specifications. CAT 5 or better, 4-pair, 10 Base-T/100 Base-TX Ethernet cabling should be used for WL1700-MH240 infrastructure equipment.

3.2   Assigning IP Addresses

An MH240 operates as a LAN client device and, therefore, requires an IP address to operate in the network. An IP address can be assigned dynamically through the SV8100 using standard DHCP protocol.

3.3   Software Configuration/Update Using TFTP

WL1700-MH240 is field-upgradable in terms of new software features or abilities and bug fixes. WL1700-MH240 uses a TFTP to automatically download new code when available.

3.4   Call Admission Control (CAC)

The combination of WL1700-MH240 provides Call Admission Control, which regulates addition of new VoIP sessions on AP radios. You can use CAC to ensure VoIP session by limiting the number of active voice sessions a radio can have. AP does not permit to create a new voice session before ensuring the sufficient bandwidth. This function is enabled by Voice-Extension setting.

3.5   Security

Proper security provisions are critical for any enterprise Wi-Fi network. Wireless technology does not provide any physical barrier from malicious attackers, since radio waves penetrate walls and can be monitored and accessed from outside a facility. The extent of security measures used is typically proportional to the value of the information accessible on the network. The security risk for Wi-Fi telephony is not limited to the typical wired telephony concerns of eavesdropping on telephone calls or making unauthorized toll calls, but is equivalent to the security risk of the data network that connects to the APs.

Several different security solutions can be implemented with MH240 handset. Determining the proper level of security should be based on identified risks, corporate policy and an understanding of the pros and cons of the available security methods.

3.5.1   Wired Equivalent Privacy (WEP)

MH240 Handset supports Wired Equivalent Privacy (WEP) encryption as defined by the 802.11 standard. The handset can use either 40-bit, or 128-bit key lengths. WEP is provided for backward compatibility of laptops, but is not recommend as a security measure for MH240.

3.5.2   Wi-Fi Protected Access (WPA/WPA2)

Recognizing the need for stronger security standards beyond WEP, the IEEE developed and ratified the 802.11i standard, which includes stronger encryption, key management, and authentication mechanisms. Wi-Fi protected Access 2 (WPA2) is the Wi-Fi Alliance's specification and certification program based on the 802.11i standard. WPA2 includes the Advanced Encryption Standard (AES), which is widely accepted as one of the most powerful forms of encryption available. WPA2 has two different authentication modes: **Enterprise Mode** uses 802.1x EAP-based authentication and **Personal Mode** uses a pre-shared key (PSK). 802.1x authentications employ a RADIUS authentication server and an EAP-based key exchange sequence. In the WL1700-MH240 network, it is recommended to use WPA2/PSK.

SECTION 4   AP DEPLOYMENT DESIGN

MH240 uses a Wi-Fi network consisting of WLAN access points (APs) distributed throughout an enterprise environment. The required number and placement of APs in a given environment is driven by several factors, including intended coverage area, system capacity, power output, physical environment, and radio mode.

4.1   Coverage and Positioning

One of the most critical considerations in deployment of WL1700-MH240 is to ensure sufficient wireless coverage. Enterprise Wi-Fi networks are often initially laid out for data applications and may not provide adequate coverage for Wi-Fi telephony. Such networks may be designed to only cover areas where data terminals are used and may not include coverage in other areas such as stairwells, break rooms or building entrances – all places where telephone conversations are likely to occur.

The overall quality of coverage is more important with telephony applications. Coverage that may be suitable for data applications may not be seamless enough to support the requirements of Wi-Fi telephony. Most data communication protocols provide a mechanism for retransmission of lost or corrupted packets. Delays caused by retransmissions are not harmful or even discernible, for most data applications. However, the real-time nature of a full-duplex telephone conversation requires that voice packets be received correctly within tens of milliseconds of their transmission. There is little time for retransmission, and lost or corrupted packets must be discarded. In areas of poor wireless coverage, the performance of data applications may be acceptable due to retransmission of data packets, but for real-time voice, audio quality will likely suffer.

Another factor to consider when determining the coverage area is the device usage. Wireless telephone devices are used differently than wireless data terminals. Wireless telephone users tend to walk as they talk, while data users are usually stationary or periodically nomadic. Wireless voice requires full mobility while data generally requires simple portability. MH240 is typically held close to the user's body, thus introducing additional radio signal attenuation. Data terminals are usually set on a surface or held away from the body. This factor may result in reduced range for a wireless telephone as compared with a data device, therefore the WLAN layout should account for some reduction of radio signal propagation. We recommend deploying an AP in a 15-meter square.
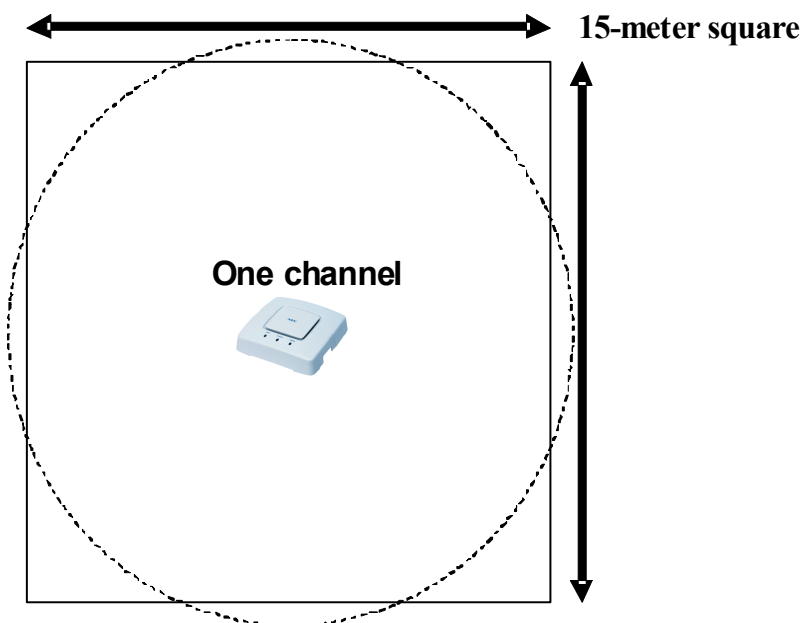
Figure 11-2 Coverage and Positioning

In terms of positioning, it is also one of the critical factors when you place APs. There are basically two kinds of positions. One is the **horizontal** positioning, and the positional relationship with pillar is very important. If you deploy an AP in the center of four pillars, two or more pillars overlap diagonally. It causes more coverage holes. Thus, you should deploy the AP displacing a little from the center of the four pillars.

Another factor is the **vertical** positioning. The more AP is deployed on a higher place, the less the coverage hole occurs. So we can see a lot of cases that APs are deployed on the ceiling. Thus, you must pick up the final deployment position concerning; which area you must assure the communication, where the RF obstacles are, which area you must get the coverage hole away, where you can get the maximum data rate. If possible, removing RF obstacles gives you an optimized radio area. The following figures show how you should implement AP and what the obstacles can be.
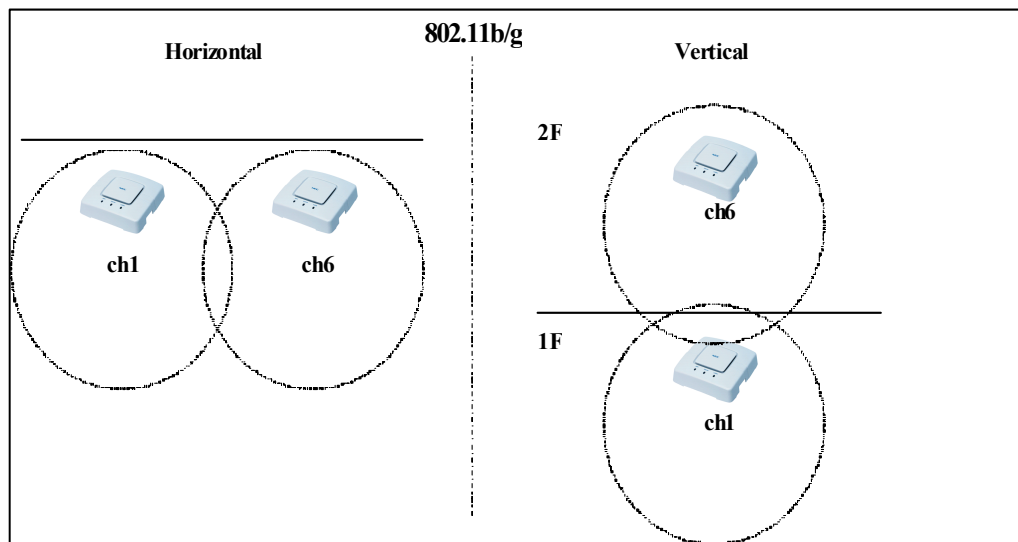
Figure 11-3 Horizontal and Vertical Positioning Examples
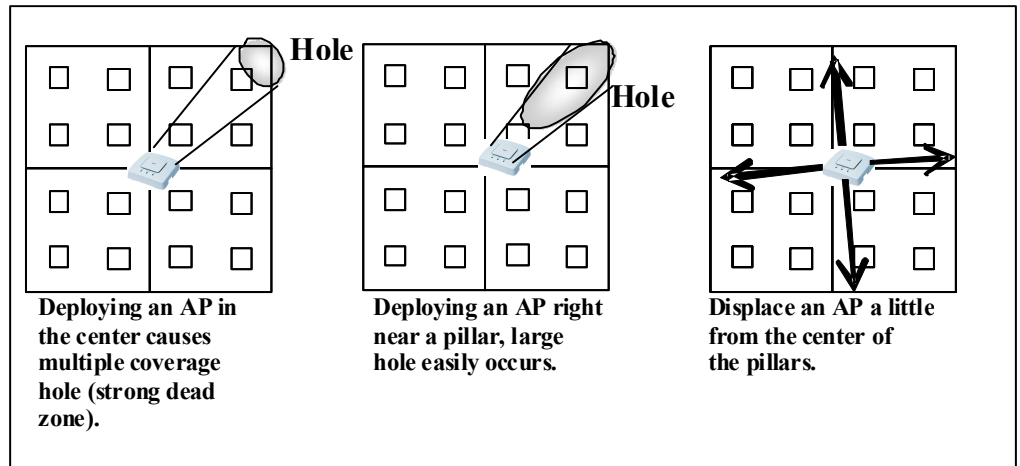
Figure 11-4 Horizontal Positioning
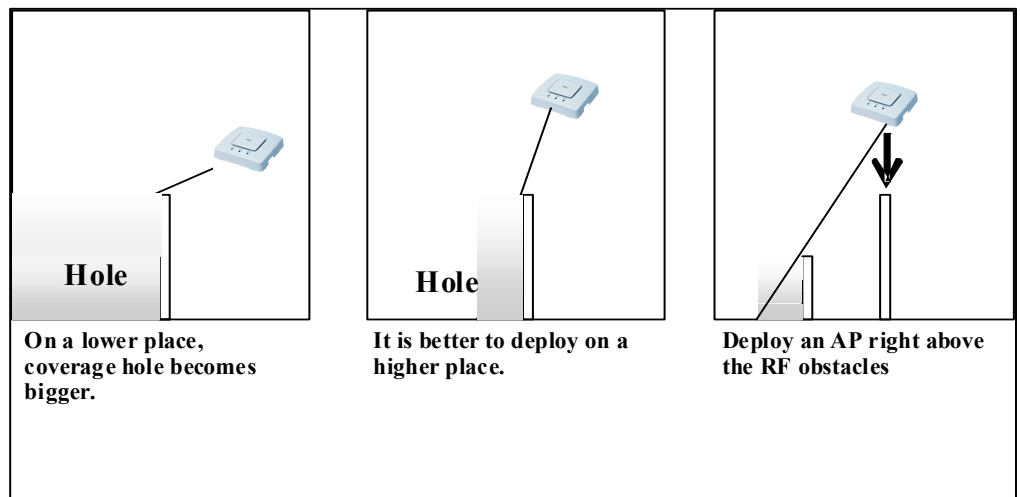
Figure 11-5 Vertical Positioning

Table 11-1 Signal Obstacle Table

Material	Obstacle Level	Example
Metal	Very high	Steel desk, metal partition, steel cabinet, metal door, wire netting
Concrete	High	Floor, outdoor wall
Water	Fair	Tank, Cistern
Human body	Fair	Seat, pedestrian
Block	Fair	Indoor wall
Paper	Fair	File, bookshelf
Wood	Low	Partition
Plaster board	Low	Indoor wall
Chemical material	Low	Partition
Glass	Low	Window

4.2   Overlapping Coverage and Channel Selection

To provide comprehensive coverage for Wi-Fi telephony applications, APs must be positioned with sufficient overlapping coverage to ensure there are no coverage gaps, or dead spots, between them. As MH240 handsets move about the workplace, they seek out other APs to handoff to and re-associate with to maintain the most reliable network connection. A properly designed Wi-Fi network provides a seamless handoff between APs, ensuring excellent voice quality throughout the facility.

The WLAN layout must factor in the transmission settings that are configured in the APs. The transmission of voice requires relatively low data rates and a small bandwidth compared to other applications. The IEEE 802.11 standard includes automatic rate switching abilities so that as a user moves away from the AP, the radio adapts and uses a less complex and slower transmission scheme to send the data. The result is increased range (coverage) when operating at reduced transmission rates. When voice is an application on the WLAN, APs should be configured to allow lower transmission rates to maximize coverage area. If a site requires configuring the APs to only negotiate at the higher rates, the layout of the WLAN must account for the reduced coverage and additional APs that are required to ensure seamless overlapping coverage.

For channel selection, the 802.11b/g standard provides for three noninterfering channels - channels one, six and 11 for North America. Access points within range of each other should always be set to non-interfering channels to maximize the capacity and performance of the wireless infrastructure. The diagram provided below illustrates this deployment methodology.

If adjacent access points are set to the same channel, or use channels with overlapping frequency bands, the resulting interference causes a significant reduction in the network performance and throughput, and degrades overall voice quality.
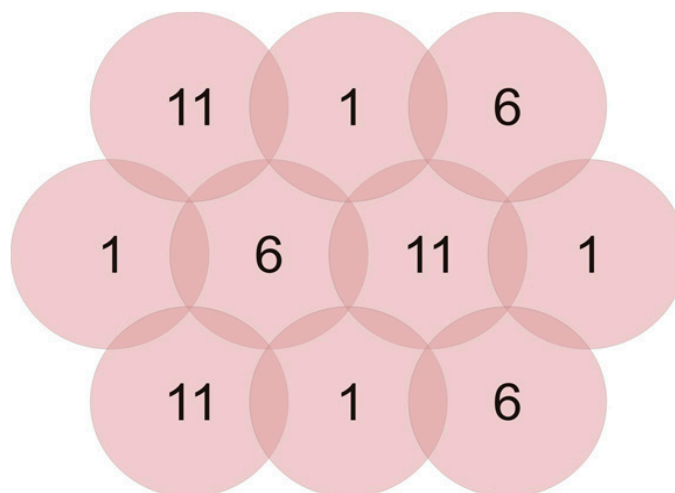
Figure 11-6 802.11b/g Overlapping Cell Coverage Illustration

4.3   Signal Strength, Radio Mode, and Data Rates

To provide reliable service, wireless networks should be engineered to deliver adequate signal strength in all areas where MH240 handsets are used. The required minimum signal strength for MH240 depends on the 802.11 frequency band and modulation used, and the data rates enabled on the AP.

Also, MH240 provides support for 802.11b, 802.11g radio modes. The selection of radio type has significant impact on the overall configuration and layout of the WLAN infrastructure. This fundamental selection determines most other configuration considerations. In the WL1700-MH240 system, NEC provides the recommended signal strength & data rate summarized in the following table to ensure the voice quality according to its development guide.

Table 11-2 Signal Strength and Data Rate Guide

802.11 Radio Mode	Recommended Enabled Data Rates (Mbps)	Required Minimum Signal Strength (RSSI)
802.11g	24, 18	-65dBm

4.4   Transmission Power

The transmission power of APs can be increased or decreased, to provide more or less AP radio coverage. For MH240 handsets, the transmission power setting must be the same for all APs in a facility. This minimizes the opportunity for higher-power devices to interfere with nearby lower-power devices, while providing consistent coverage.

The AP transmit power (TX) should be set so that MH240 receives the required minimum signal strength, as defined in Section 4.3 Signal Strength, Radio Mode, and Data Rates. For deployments with higher AP density, lower TX power settings are typically required to prevent channel interference. Maximum AP power settings vary by band and by channel, and can vary between countries. Local regulations should always be checked for regulatory compliance considerations. In addition, maximum power output levels may vary by AP manufacturer. Keeping the transmit power and of the AP and MH240 at the same level to avoid one-way call occurrence, which is a result of a mismatch between the reach of the signal. There is a case that only raising the transmission power of AP to improve voice quality does not solve the problem. Adversely, if received signal level of handset to AP is too high, it causes bad roaming from an AP to another which leads to the decrease of voice quality. As the result, the following phenomenon may occur: a handset can get packet from an AP but the AP cannot get packet from the handset and tries to resend the packet to the handset. In our WL1700-MH240 system, we set the recommended AP deployment guideline in Section 4 AP Deployment Design, and TX power for WL1700 internal AP in the WL sample configuration to avoid such a mismatch.

4.5   Interference, Multipath, and Signal Distortion

Interference on a wireless network may originate from many sources. Microwave ovens, Bluetooth devices, cordless phones and rogue APs are among the many potential sources. In general, devices that employ or emit radio frequency signals in a given radio coverage area have the potential to cause unwanted interference.

Radio frequency spectrum analyzers can be used to help identify the sources of such interference. Once identified, interference is best mitigated by removing the interfering device(s) from the network area. Otherwise, it may be possible to change the channel setting of the interfering device to avoid conflict with the surrounding APs. If this is also not possible, then the channel of the surrounding APs may be changed to avoid as much frequency overlap with the interfering device.

Multipath distortion is a form of RF interference that occurs when a radio signal has more than one path between the transmitter and the receiver causing multiple signal wave fronts to be heard at the receiver. This is typically caused by the radio signal reflecting off physical barriers such as metal walls, ceilings and other structures and is a very common problem in factories and storage environments. Multiple converging wave fronts may be received as either an attenuated or amplified signal by the receiver. In some instances, although very rare, if the signals arrive exactly out of phase, the result is a complete cancellation of any signal.

4.6   802.11b/g Mixed Channel

Even though 11b and 11g radio uses the same frequency (channel), each modulation method is not the same (11b: DSSS/11g: OFDM). Therefore, 11b radio frame identifies OFDM modulation as interference wave.

To avoid this, 11g compliant device sends CTS (Clear To Send) frame to itself before transmitting OFDM modulated frame to control 11b communication. As those devices do not need to send CTS frames in the wireless network where only 802.11g compliant devices are used, 11g throughput decreases as long as CTS frame is used.

Moreover, as 802.11g device cannot communicate while 802.11b device is communicating with lower data rate as 1/2/5.5/11Mbps, throughput decreases further.

Thus, pay attention when using 11b-only device and 11g device in the same channel that 11g throughput is not enough as its logical throughput.

On the other hand, if only 11g compliant laptops or handheld devices and no 11b devices are also used in the same channel, they can fully use 11g without decreasing throughput.

4.7   Site Surveys

A wireless site survey is highly recommended for any wireless network deployment. However it is especially critical when voice is an application on the network and is essential for large or complex facilities. A site survey can ensure that the wireless network is optimally designed and configured to support voice by confirming RF coverage, cell overlap, channel allocation/ reuse, packet transmission quality and other deployment considerations. While many tools exist that allows customers to perform their own assessment.

To verify coverage of an installed Wi-Fi network, MH240 offers a site-survey mode that can be used to ensure the AP locations and configurations are both correct and adequate. This mode detects the AP signals and displays the signal strength along with the AP channel assignments. The site survey mode may be used to detect areas with poor coverage or interfering channels, check for rogue APs, confirm the Extended Service Set Identification (ESSID) and data rates of each AP and detect AP configuration problems. With MH240, the entire coverage area must be checked to ensure that at least one access point output meets the signal strength requirements.

After a site survey is complete, coverage issues can be resolved by adding and/or relocating APs. Overlap issues may be resolved by reassigning channels or by relocating some access points. When adjustments are made to the WLAN configuration, an additional site survey should be performed to ensure that the changes are satisfactory and have not had adverse impact in other areas of coverage.

4.8   Capacity

The maximum number of simultaneous telephone calls an AP can support is determined by dividing the maximum recommended bandwidth usage by the percentage of bandwidth used for each individual call. Also, network capacity requirements factor into the number of APs required, although in most cases the coverage area is the primary factor. Data traffic is often very bursty and sporadic. This is typically acceptable because data applications can tolerate network congestion with reduced throughput and slower response times. Voice traffic cannot tolerate unpredictable delays, where the bandwidth requirements are much more constant and consistent. Voice traffic can also be predicted using probabilistic usage models, allowing a network to be designed with high confidence in meeting anticipated voice capacity requirements.

For MH240 deployments, the maximum number of active calls must be defined individually for 802.11g. Although 802.11g theoretically provides increased available bandwidth for support of additional simultaneous call volume, the practical call volume limitations depend on many factors such as data rates used, other network traffic and network performance. Also, there are several factors which determine the AP bandwidth utilization during a telephone call. The first is the VoIP protocol used and its characteristics. The type of CODEC used combined with the packet rate determines the size of the voice packets along with any additional overhead information required for the protocol. Payload data generally accounts for 30-50% of a typical voice packet, with 802.11 and IP protocol overhead filling the rest. The 802.11 protocols include timing gaps for collision avoidance, which means bandwidth use is more accurately quantified as a percentage of available throughput rather than actual data throughput. WL1700-MH240 default bandwidth setting is 1229kbps, which is recommended and enables 12 simultaneous calls per AP.

4.9   AP Deployment Steps

1.   Deploy AP on a higher place to reduce the dead zone.

2.   Avoid deploying AP right near pillars.

3.   Avoid deploying AP at the corner on the ceiling.

4.   Avoid deploying AP in the center of four pillars.

5.   Channel interference is most important key point.

6.   Terminal traffic investigation made number of AP and power of AP.

7.   Channel assignment is one of the most important matters from site survey.

8.   Material of office composition relate reachable radio frequency.

9.   WLAN Terminals have interoperability.

10.   Site survey must be performed.

11.   Most WLAN troubles are related to total running environment.

4.10   Site Survey Sequence

```
                    ┌─────────────────────────┐
                    │   1. Clarify customer's  │
                    │        requirement        │
                    └─────────────────────────┘
                                 │
                                 ▼
                    ┌─────────────────────────┐
                    │   2. Check actual status  │
                    └─────────────────────────┘
                                 │
                                 ▼
                    ┌─────────────────────────┐
                    │ 3. Desk design of AP position │
                    └─────────────────────────┘
                                 │
                                 ▼
                    ┌─────────────────────────┐
                    │     4. Site survey        │
                    └─────────────────────────┘
                                 │
                                 ▼
                    ┌─────────────────────────┐
                    │     5. Installation       │
                    └─────────────────────────┘
                                 │
                                 ▼
                    ┌─────────────────────────┐
                    │    6. Voice call test     │
                    └─────────────────────────┘
```
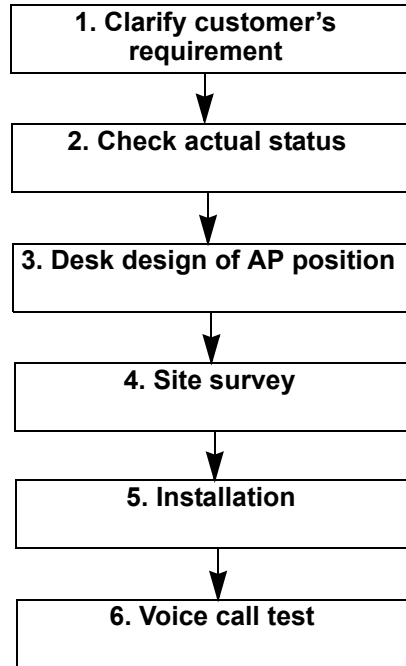
Figure 11-7 Site Survey Sequence

4.11 Interview Sheet

Use the following customer interview sheet to verify requirements.

Table 11-3 Customer Interview

Interview Item	Answer	
Get the floor layout from customer. (Check the office area, location of columns, location of any partition, location of conference room, and special construction such as blow-by, full-height glazing. Check for differences between drawing and present status.	Yes	No
Get the current voice network configuration diagram (logical configuration, physical configuration).	Yes	No
Construction method for floor (reinforced concrete, iron frame) ❍ With regards to the construction method using reinforced concrete, the RF obstacle level is higher than normal concrete. ❍ For iron frame construction method, the radio wave is blocked.	Yes	No
For tenant building:	Yes	No
❍ The other tenants use WLAN?	Yes	No
❍ When WLAN is used by the other tenants.	Yes	No
❍ How did you confirm that existing WLAN is present?		
❍ Is there a microwave oven in use nearby?	Yes	No
❍ Is there any RFID in use?	Yes	No
❍ Is a ham radio in use nearby?	Yes	No
❍ Is a Bluetooth in use nearby?	Yes	No
❍ Is a medical instrument used nearby?	Yes	No
❍ Is industrial RF equipment used?	Yes	No
Verify the following items if the WLAN is already installed: ❍ Obtain a current drawing of the WLAN	Yes	No

Table 11-3 Customer Interview

Interview Item	Answer
If a floor layout is available, answer the following items:	
❍ Where is the existing AP installed?	Yes No
❍ What standard is used in the existing WLAN AP? (802.11a, 802.11b, 802.11g, etc.)	Yes No
❍ Which channel is used in existing AP?	Yes No
❍ Transmission power of existing AP?	Yes No
❍ Vendor/Model	Yes No
❍ What type is the client in existing WLAN?	Yes No
❍ Total number of WLAN terminals in existing WLAN?	Yes No
❍ What type of certification is used in existing LAN? (IEEE802.1X (MS-PEAP, EAP-TLS)	Yes No Yes No
❍ What type of encryption is used in existing LAN? (Static WEP, Dynamic WEP, TKIP)	Yes No
Installed System:	
❍ Service area of VoWLAN?	Yes No
	Does the customer want to cover a special area such as stairway, bathroom, cafeteria, president room, executive room, smoking room, etc.
❍ WLAN devices?	Voice client will be used or not? Yes No Co-operative use between Wi-Fi handset and Softphone? Yes No
❍ Model of WLAN terminal?	MH240 Other ( )
❍ WLAN radio?	Voice: 802.11b Data: 802.11a
❍ How many MH240 sets will be installed?	Sets
❍ Is a DHCP service installed?	Yes No
❍ Is a Radius service installed?	Yes No
❍ (Reference) Network topology for WLAN	Is it the same segment as fixed wired line or WLAN dedicated segment? VLAN, Number of subnets, Number of SSIDs Confirm.

Table 11-3 Customer Interview

Interview Item	Answer
❍ (Reference) Authentication/Encryption method	802.1x WPA WPA2 Other
❍ (Reference) Server	SV8100 CA RADIUS DHCP
❍ (Reference) Active Directory Domain	Fixed LAN and WLAN will be used? Yes No Supplicant? Yes No
❍ Restriction for installation of AP	How far from PoE switch to AP? AP will be installed on ceiling, wall or column?
❍ Redundant Configuration for WL Controller	Yes No

4.12   Deployment Guide per Network Scale

Basically, determine the access point position after designing the floor layout on the desk, and actually measure signal strength using the Site Survey Mode on MH240.

However, what you have to use for survey may vary according to customer network scale. Although this document is focused on small scale (e.g. 1~4 access points in the network) a site survey utility like Ekahau Site Survey is needed to check how strong the radio signal is in the entire wireless network, if the network is bigger than four access point scale.

4.13   Measuring the Actual RF Signal
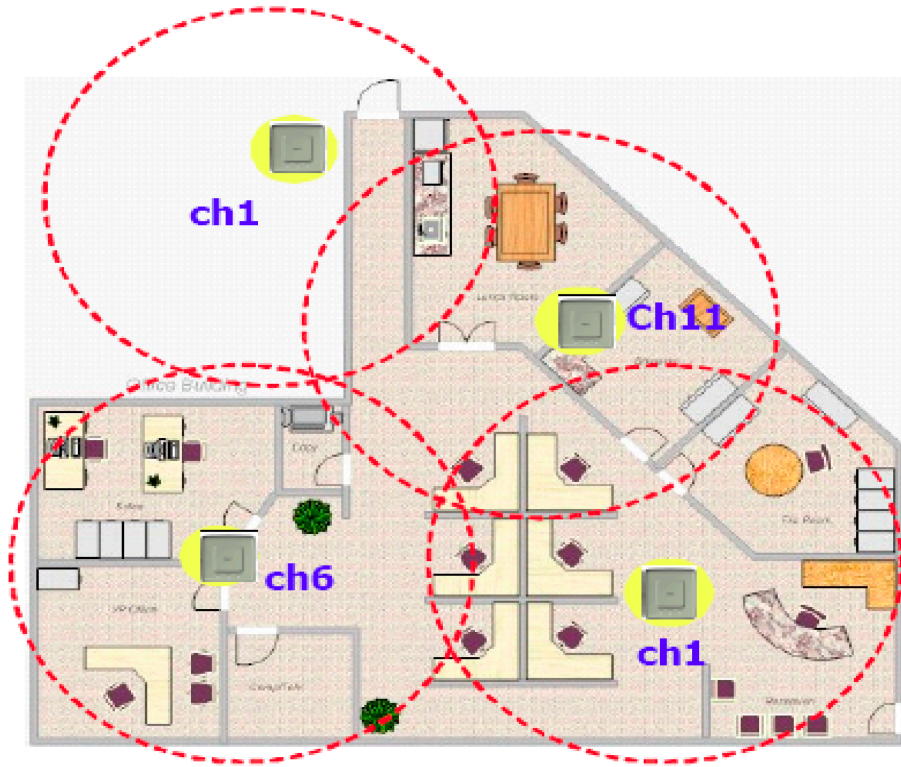
- WL1700 with an internal AP
- Supports 4 AP's with WL1700
- Simple deployment

Logical Diagram
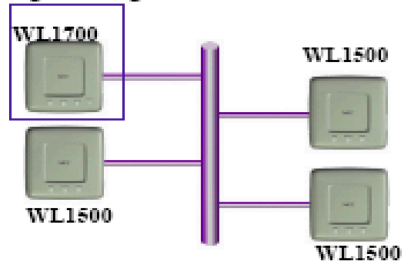
Figure 11-8 SMB Deployment Sample

4.13.1   Minimum Items Needed for RF Measurement

- ❒   WL1700

- ❒   PoE switch (providing PoE via 802.3AF)

- ❒   Tripod (to tentatively fix WL1700 on the ceiling)

- ❒   MH240 (two handsets are better, one handset minimum)

- ❒   LAN cable (to connect WL1700 and PoE switch, long enough to extend to every corner of the coverage area)

4.13.2   How to Measure

1.   Place the tripod at determined estimated deployment position.

2.   Fix WL1700 on the tip of the tripod.

3.   Connect a LAN cable to WL1700 and POE switch.

4.   Turn on the POE switch.

5.   Configure WL1700 based on Easy Setup.
   ✎   Make sure that Cont status of WL1700 LEDs is green, indicating the normal operation.

6.   Run Site Survey Mode on two MH240 handsets and make sure they have associated to service SSID.

7.   Compare both RSSI results and take 'weaker' one. When measuring, stand with your back to WL1700 and sit with your back to WL1700 in a corner of a room, if any.

8.   Continue measuring until the measurement result gets no less than -65dBm. When the result reaches no less than -65dBm, this point is the handoff border between access points.

9.   Place the tripod with WL1700 to next estimated position.

10.   Repeat the procedure 7 to 9.

4.13.3   MH240 Site Survey Utility

MH240 has built-in Site Survey utilities. These tools are helpful for designing Access Point location. The following list shows how to place the MH240 in AP scan mode and explains each item.

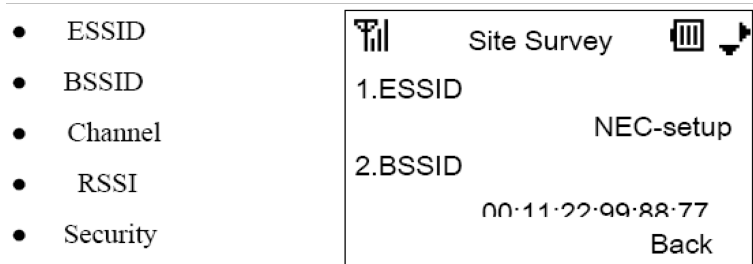1.   Press [MENU] [DEF 3] [1] [1] to enter AP scan mode.
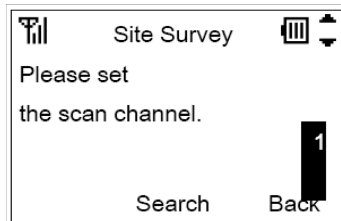
2.   Press ↕ key to select the scan channel, then press search soft key.

3.   Scanning begins.
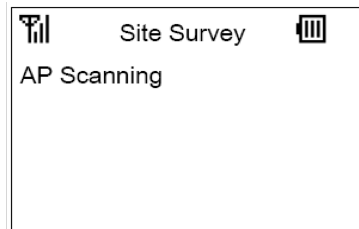
4.   When scan is completed, the result is displayed on an AP basis.
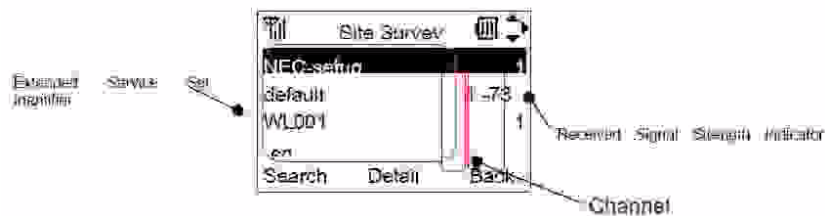
5.   Press ⬍ key to select an AP, then press the Detail soft key to show more detail.

- ESSID
- BSSID
- Channel
- RSSI
- Security

✎   This tool might be used in a small-scale environment where three or four APs are deployed. If you design larger installation, it is recommended to use other tools such as Ekahau. Refer to 4.13.5 Wireless Installation Utility.

4.13.4   Check Voice Quality by Call Test

Basically, you hold a handset with your hand which is farther from the AP. One person makes a call to another. Move around the service area and keep talking until you find out no voice chopping or delay, and see whether the call quality is fine. Note that the voice quality depends on how you hold MH240 and turn your back to the AP that MH240 associates with.

✎   Make sure you set the SIP server for the call test.

4.13.5   Wireless Installation Utility

For larger wireless network installations, it is industry recommended to do a pre- and post-site survey of the Wi-Fi coverage area. During the pre-installation site survey, it helps identify the best AP and antenna placement, channel assignment, and power level assignment for the APs in the coverage area. It also identifies other sources of 802.11 wireless interference in the area of desired coverage. With this knowledge, the network designer can better design the Wi-Fi network to minimize any existing Wi-Fi interference effect on the new wireless installation. The tool recommended for pre-install site survey would either be Ekahau Site Survey tool or AirMagnet Site Survey tool.

Table 11-4 Site Survey Tools

Tool	Product Information
1. Ekahau Site Survey Tool (**preferred**)	http://www.ekahau.com/
2. Air Magnet Site Survey Pro and Planner Tool	http://airmagnet.com/products/survey/

After the pre-install site survey is done, the installer begins to implement the network design created from the pre-install survey. During every installation, there is a possibility that troubles may occur. These troubles can range from manufacturing defects to RF interference not identified during the pre-installation site survey. Troubleshooting tools are always a good device to have to give the installer a chance to identify the source of failure so that installation contingencies can be made. This portion of the troubleshooting process can be handled by using various troubleshooting tools. Such tools include a wireless packet analyzer like Wildpackets OmniPeek WLAN Packet Analyzer and the Cognio Spectrum Analyzer. The Packet Analyzer allows you to analyze the 802.11 Wi-Fi packets to troubleshoot the Wi-Fi process used in 802.11. The Cognio Spectrum Analyzer allows you to look for sources of interference in the same RF spectrum as the installation that may not be an 802.11 device, thus not recognizable by the Packet Analyzer.

In addition to the Packet Analyzer, if using a single listening device or laptop, you can sample only one frequency (or channel) at a time. A tool developed by Cacetech Technologies allows a single listening device to hear multiple channels at the same time and presents the traces into the same trace file for evaluation (and thus the same timed/logical order in which the failure occurred). Without this tool, you can miss packets transmitted on other frequencies or channels than the channel you are listening to.

Table 11-5 Packet Analyzer Tools

Tool	Product Information
1. Wildpackets OmniPeek WLAN Packet Analyzer (**preferred**)	http://www.wildpackets.com/products/omnipeek/overview
2. Air Magnet WLAN Packet Analyzer	http://airmagnet.com/products/laptop_analyzer/

Table 11-6 Multi-channel Listening Device

Tool	Product Information
1. CaceTech's AirPCap Tool	http://www.cacetech.com/products/airpcap-ex.htm

Table 11-7 Spectrum Analyzer Tools

Tool	Product Information
1. AirMagnet Surveyer (powered by Cognio spectrum analyzer) (**preferred**)	http://airmagnet.com/products/spectrum_analyzer/
2. Cogio Spectrum Analyzer	http://metrix.net/cognio-spectrum-expert-for-wifi-20-spectrum-analyzer-p-61.html

SECTION 5   WL1700-M240 EASY SETUP

The Easy Setup guides you how you configure a Wi-Fi network of WL1700-MH240. Here is the summary of the installation.

5.1   Summary of Installation

❍   Use MH240 initial configuration files – The utility that allows you to quickly run WL1700-MH240 system with minimum operation on a computer.

❍   The computer should have TFTP/FTP server program installed.

❍   Access WL1700 with the factory-default IP address, then change the IP address to the new one necessary for the setup.

❍   Input the configuration from WebView.

❍   Manually edit serial ID of APs to configuration template inside WL1700. (It is the only one that you have to edit by yourself).

❍   MH240 must be in range of Setup BSS.

❍   Turn MH240 on and then the files for MH240 setup are automatically downloaded.

❍   Finally service is started.

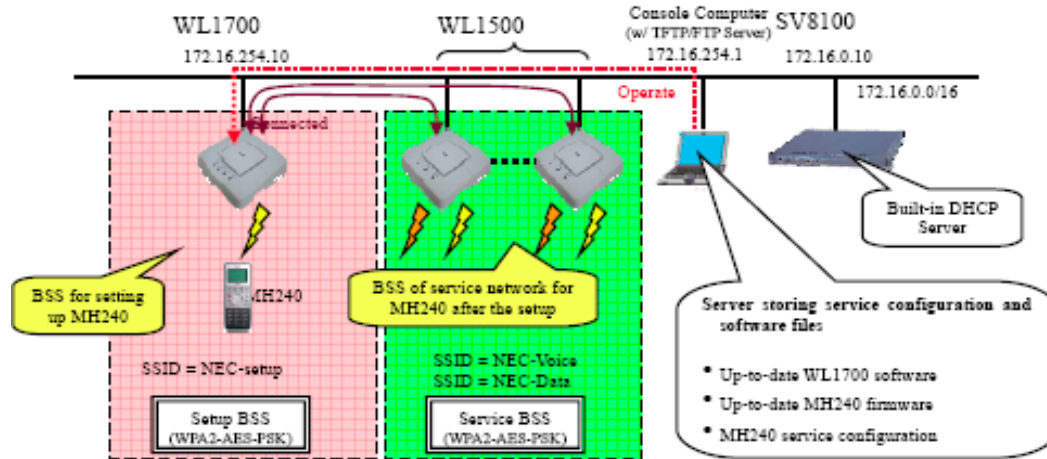### 5.2   WL-1700-MH240 Setup Overview

Figure 11-9 WL1700-MH240 Setup Overview Diagram

✎   You may leave or delete Installation BSS after the setup. It depends on the customer maintenance policy.

✎   SV8100 internal DHCP server shall be used for IP address assignment for MH240 handsets and APs.

1.   Change IP address to 192.168.100.100/16 on the console computer.

2.   Log in to Web console called WebView of WL1700.

3.   From console computer, open the browser and enter https:// 192.168.100.10 in the address bar, and press **Enter**.

4.   Enter **nec** for both Username and password, and click **OK**.

5.   Enter no password for Input Enable Password, and click **OK**. Top page of WebView is displayed.

6.   Input service configuration to WL1700.

7.   Click Commands in the left menu.

8.   Click **Config Management**.

9. Click **Edit** on current configuration file.

10. Current configuration can be seen in Managing Configuration Files. Highlight it all and delete.

11. Copy the following command lines which are used as the Service Configuration.

> **set system ip-address <u>172.16.254.10</u>**
>
> **set system countrycode US**
>
> **set service-profile setup-sp ssid-name NEC-setup**
>
> **set service-profile setup-sp short-retry-count 3**
>
> **set service-profile setup-sp long-retry-count 3**
>
> **set service-profile setup-sp auth-fallthru last-resort**
>
> **set service-profile setup-sp user-idle-timeout 3600**
>
> **set service-profile setup-sp keep-initial-vlan enable**
>
> **set service-profile setup-sp cipher-tkip disable**
>
> **set service-profile setup-sp cipher-ccmp enable**
>
> **set service-profile setup-sp rsn-ie enable**
>
> **set service-profile setup-sp psk-phrase SUMIDA786432**
>
> **set service-profile setup-sp auth-psk enable**
>
> **set service-profile setup-sp auth-dot1x disable**
>
> **set service-profile setup-sp load-balancing-exempt enable**
>
> **set service-profile setup-sp attr vlan-name default**
>
> **set service-profile sp1 ssid-name NEC-Voice**
>
> **set service-profile sp1 no-broadcast enable**
>
> **set service-profile sp1 short-retry-count 3**
>
> **set service-profile sp1 long-retry-count 3**
>
> **set service-profile sp1 auth-fallthru last-resort**
>
> **set service-profile sp1 static-cos enable**
>
> **set service-profile sp1 cos 7**
>
> **set service-profile sp1 cac-mode voice-multiline**
>
> **set service-profile sp1 user-idle-timeout 3600**
>
> **set service-profile sp1 active-call-idle-timeout 300**
>
> **set service-profile sp1 keep-initial-vlan enable**
>
> **set service-profile sp1 cipher-tkip disable**
>
> **set service-profile sp1 cipher-ccmp enable**
>
> **set service-profile sp1 rsn-ie enable**
>
> **set service-profile sp1 psk-phrase 0246802197531**

set service-profile sp1 auth-psk enable

set service-profile sp1 auth-dot1x disable

set service-profile sp1 load-balancing-exempt enable

set service-profile sp1 cac-bw-call 1229

set service-profile sp1 transmit-rate 11g mandatory 18.0,24.0 disabled 1.0,2.0,5.5,6.0,9.0,11.0,12.0,36.0,48.0,54.0

set service-profile sp1 attr vlan-name default

set service-profile sp2 ssid-name NEC-Data

set service-profile sp2 short-retry-count 3

set service-profile sp2 long-retry-count 3

set service-profile sp2 auth-fallthru last-resort

set service-profile sp2 user-idle-timeout 3600

set service-profile sp2 keep-initial-vlan enable

set service-profile sp2 cipher-tkip disable

set service-profile sp2 cipher-ccmp enable

set service-profile sp2 rsn-ie enable

set service-profile sp2 psk-phrase 0246802197531

set service-profile sp2 auth-psk enable

set service-profile sp2 auth-dot1x disable

set service-profile sp2 load-balancing-exempt enable

set service-profile sp2 attr vlan-name default

set radio-profile rp1 dtim-interval 5

set radio-profile rp1 auto-tune channel-config disable

set radio-profile rp1 active-scan disable

set radio-profile rp1 qos-mode voice-extension

set radio-profile rp1 max-voip-bw 1229

set radio-profile rp1 service-profile sp1

set radio-profile rp2 auto-tune channel-config disable

set radio-profile rp2 active-scan disable

set radio-profile rp2 service-profile sp2

set radio-profile setup-rp dtim-interval 5

set radio-profile setup-rp auto-tune channel-config disable

set radio-profile setup-rp active-scan disable

set radio-profile setup-rp service-profile setup-sp

set ap auto mode disable

set ap 1 serial-id h3tzf0017 model WL1700-MS(AP)

set ap 1 radio 1 channel 1 tx-power 13 radio-profile setup-rp mode enable

set ap 2 serial-id h2tzub0301 model WL1500-AP

set ap 2 radio 1 channel 6 tx-power 10 radio-profile rp1 mode enable

set ap 2 radio 2 channel 36 tx-power 10 radio-profile rp2 mode enable

set ap 3 serial-id h3tzub1036 model WL1500-AP

set ap 3 radio 1 channel 11 tx-power 10 radio-profile rp1 mode enable

set ap 3 radio 2 channel 40 tx-power 10 radio-profile rp2 mode enable

set ap 4 serial-id h2tzub1527 model WL1500-AP

set ap 4 radio 1 channel 1 tx-power 10 radio-profile rp1 mode enable

set ap 4 radio 2 channel 44 tx-power 10 radio-profile rp2 mode enable

set ip telnet server enable

set load-balancing mode disable

set vlan 1 port 1

set interface 1 ip <u>172.16.254.10.255.255.0.0</u>

✎   Change the IP addresses underlined on the above command lines according to customer's network environment.

12.   Paste it into Managing Configuration Files.

13.   In the set ap <AP Number> commands, edit the parameter of Serial ID of each AP.
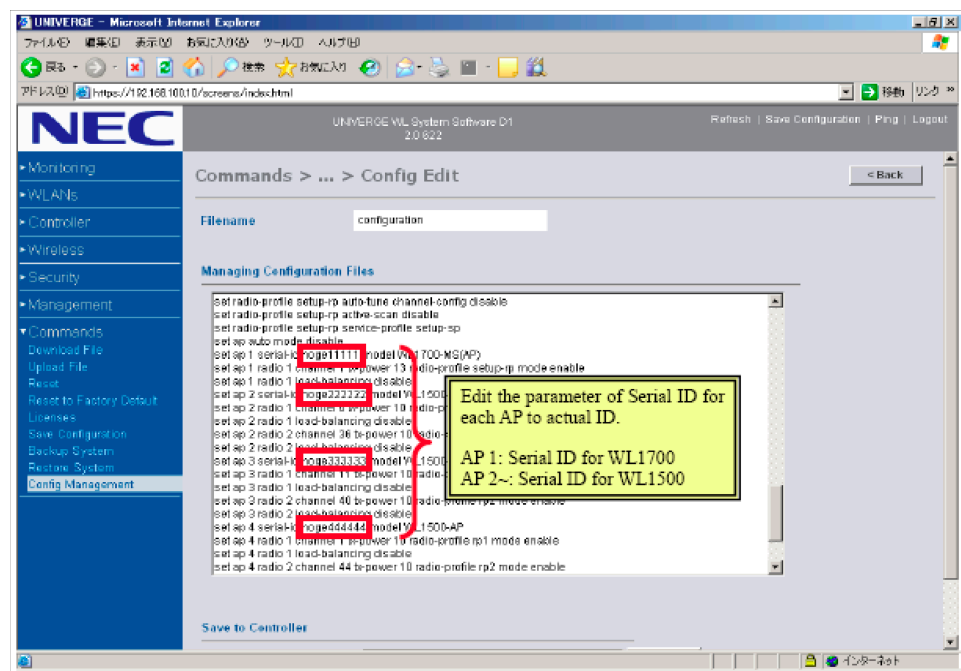
Figure 11-10 Configuration Edit Screen

14.   Enter **Initial.conf** for the Destination URL.

15.   Press **Enter**.

16.   Click **Save**.

17.   Click **OK**.

18.   Select **Initial.conf** in Configuration Files.

19.   Click **Set Next Boot**.
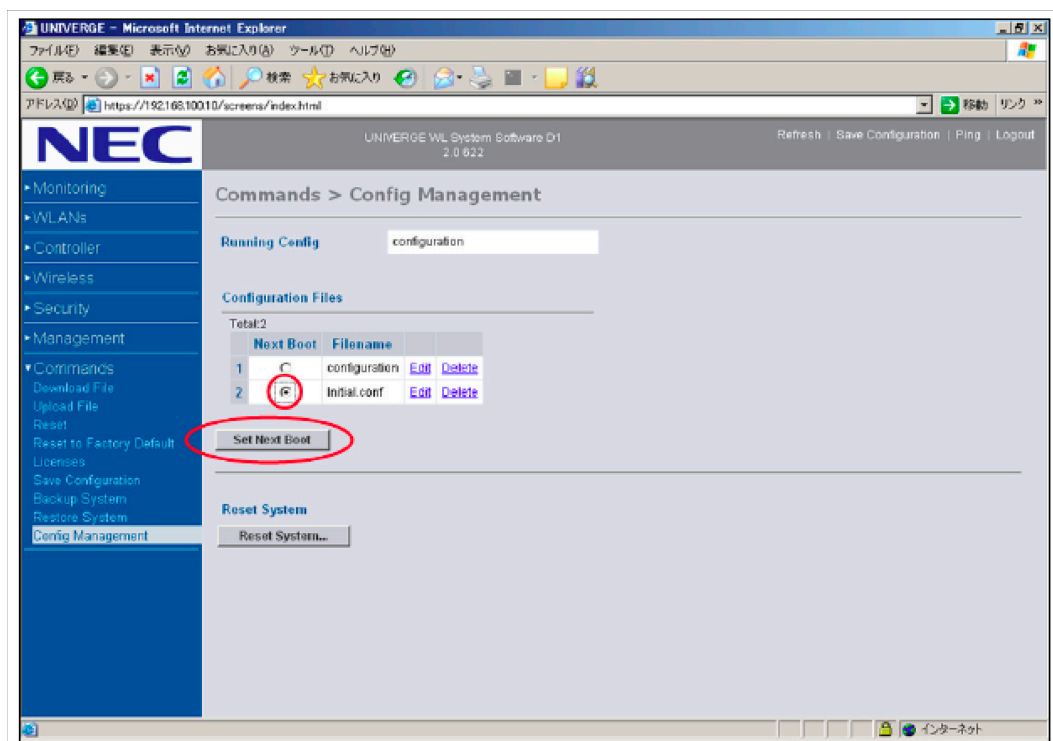
Figure 11-11 Config Management Screen

20.   Move WL1700 and console computer to service network.

21.   Move WL1700 and console computer from the network 192.168.100.0/
      24 to service network 172.16.0.0/16. At the time, WL1700 can be turned
      off.

22.   Change IP address to 172.16.254.1/16 on the console computer.

5.2.1   Setting Up the MH240 Handset

1.   Start up MH240 in the factory-default state, and turn on in Easy Startup mode.

```
┌─────────────────────────────────────────┐
│            Easy Startup                  │
│                                          │
│     ───►  1. FW Upgrade                  │
│           2. Config DL                   │
│           3. All                         │
│                                          │
└─────────────────────────────────────────┘
```
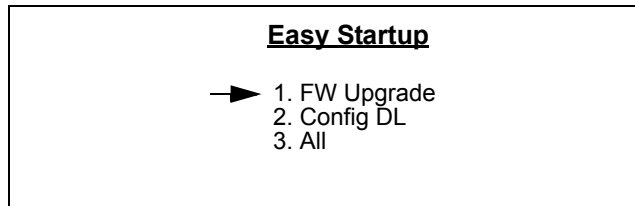
Figure 11-12 Easy Startup Menu

2.   Select **2. Config DL**. Config DL mode is being initialized.

3.   Wait while the handset is downloading Config file. (Config downloading is displayed.)

4.   Wait while the handset is downloading service configuration file. (Config downloading is displayed.)

5.   All eight LEDs blink green twice to inform you that the download is done.

6.   Select **2.Reboot** to take changes effect.

☞   Make sure that TFTP server is up when you download Config file.

☞   MH240 handset automatically accesses TFTP server with IP address: 172.16.254.1 by default to download necessary file. If you want to change the IP address according to customer network environment, the following procedure changes the TFTP IP address set on the MH240.

■   Press MH240 Menu button.

■   Select **5.Setting**.

■   Select **2.Download**.

■   Select **2.Download Address**.

■   Input IP address on TFTP server.

5.2.2   Make sure the Handset is Working
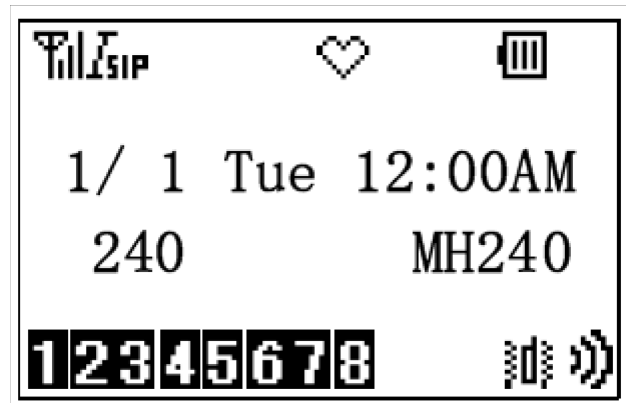
When the ▊▊ and ▊▊ icons can be seen, the handset is ready to send and receive calls.

SECTION 6   TROUBLESHOOTING

6.1   Trouble Summary

Occasionally, you may run into transmission problems due to any number of factors originating from the wireless LAN. NEC MH240 Mobile Handsets can exhibit transmission problems in several ways. They can cease functioning properly, display error messages, or display incorrect data. When using and troubleshooting handsets, consider the following problem sources to determine the best method of approaching any specific situation.

6.1.1   Access Point Problems

Most, but not all, handset audio problems have to do with AP range, positioning, and capacity. Performing a site survey as described in this document can isolate the AP causing these types of problems. If the handset itself is suspected, conduct a parallel site survey with a handset that is known to be properly functioning.

6.1.2   In Range/Out-of-Range

Service is disrupted if a user moves outside the area covered by the wireless LAN APs. Service is restored if the user moves back within range. If a call drops because a user moves out-of-range, the handset recovers the call if the user moves back into range within a few seconds.

6.1.3   Capacity

In areas of heavy use, the call capacity of a particular AP may be filled. If this happens, the user hears busy tone from the handset. The user can wait until another user terminates a call or move within range of another AP and try the call again.

6.1.4   Transmission obstructions

Prior to system installation, the best location for APs for optimum transmission coverage should have been determined. However, small pockets of obstruction may still be present, or obstructions may be introduced into the facility after system installation. This loss of service can be restored by moving out of the obstructed area or by adding/rearranging APs.

6.1.5   WL Controller AP/MH240 Configuration Problems

Certain problems are associated with improper configuration of either the WL system or the handset. Configuration problems are generally corrected by changing the configuration on the WL system or on the handset.

6.2   MH240 Message Definitions and Suggestions

The following table shows the definitions and suggestions of messages that may be displayed on your MH240.

Table 11-8 MH240 Handset Display Messages

Message	Definition and Suggestion
[Password] Unauthorized!	The password entered is incorrect. Enter the correct password.
[Low Battery] Please charge it	Battery charge level is low. Charge the battery.
[Directory] No Input Name	Directory name is not input. Edit the name.
[Handy Key Lock] Handy Key Lock is enabled	Displayed when your phone is locked.
[Handy Key Lock] Handy Key Lock is disabled	Displayed when your phone is unlocked.
[Password Lock] Password Lock enabled	Displayed when your phone is locked with a password.
[Setting] X to Y is input	Entered value is out of range. Enter the value of X and Y.

6.3   MH240 Tips and Suggestions

The following table lists suggestions on how to resolve MH240 operation problems. Confirm the following before making a call:

❍   Is the power on?

❍   Is the battery charged?

❍   Is the Antenna icon 📶 displayed?

Table 11-9 MH240 Operation Problems and Suggestions

Message	Definition and Suggestion
Cannot power on the phone.	– Press the 🔴 key and hold more than two seconds. – Insert the battery correctly into the telephone. – Charge the battery.
Cannot operate keys on the telephone.	– Unlock your phone if the lock icon 🔒 is displayed. – Unlock you phone if "Handy Key Lock" is displayed.

Table 11-9 MH240 Operation Problems and Suggestions

Message	Definition and Suggestion
Communication is lost, audio breakup occurs.	– Check the receiving signal strength. – Check the battery when ▯ is indicated. – Place your telephone more than 50cm from the Access Point.
Cannot see Antenna icon. 📶	– When you are in an area where sufficient signal cannot be obtained, move to other area. – SIP server registration is incomplete. If this occurs, please contact the network administrator.
Cannot sustain battery power.	– Insert the battery correctly into the telephone. – Properly place the battery in the cradle. – Replace the battery.
Cannot make or receive a call.	– Confirm the telephone number of your phone. To change the phone number, please contact your network administrator.
The telephone does not ring.	– When your telephone is placed in courtesy mode, cancel it. The Courtesy icon ♡ appears when your telephone is in this mode. – Check ring volume and adjust (do not turn to "Off").

6.4   WL Troubleshooting

If the WL1700 Controller is suspected, system logs provide a history of WL System events. Traces display real-time messages from the WL System. The following information should be collected to find the cause of the problem that you face.

Priority: High

❍   Field Trouble Report Ticket

❍   Network Diagram

   It should have IP (MAC) address.

❍   Floor Layout

   This shows where WL controller, AP and MH240 are located.

❍   WL Controller Information (only CLI)

   You can get the CLI command, show tech support.

❍   Core File (only CLI)

This file is necessary if WL Controller is rebooted.

Priority: Medium

❍   WL Controller Configuration (CLI/WEB)

❍   Log Buffer (only CLI)

❍   FDB (WEB/CLI)

❍   ARP (only CLI)

❍   Sessions Network (only CLI)

❍   VoIP Summary (only CLI)

Priority: Low

❍   Log Trace (Debug Log)

6.5   WL Controller information (only CLI)

❍   You can get the information by using PC/Terminal software to log in WL
controller.

❍   This information includes WL controller basic configuration and software
version, several logs, etc.

❍   Command → show tech-support.

6.6   Core File (only CLI)

❍   You can get the core file by using PO/Terminal software to log in WL
Controller.

❍   This information is useful for finding the cause of reboot.

❍   Command → dir.

❍   Core File name core:xxxxx.core (xxxxx is discretion).

6.6.1   How to get Core File

❐   PC/TFTP Server (You can have a PC including TFTP server)

❐   Command → copy core file name tftp://x.x.x.x/[sub directory]/
[filename] (**x.x.x.x** = IP address of TFTP Server).

❒   As the character, : is not available as [filename], make sure not
     to put core: as the file name.

6.7   WL Controller configuration

WebView

❍   PC with TFTP Server installed

❍   Commands → Upload File → File:Configuration.

❍   TFTP Server IP address/File Path/Filename.

CLI

❍   You can get the configuration file by using PC/Terminal software to log in
     WL Controller.

❍   Displays all information of WL controller configuration.

❍   Command → show configuration all.

6.8   Log buffer (only CLI)

❍   You can get the log buffer by using PC/Terminal software to log in WL
     Controller.

❍   Command → show log buffer.

6.9   FDB information (only CLI)

❍   You can get the information by using PC/Terminal software to log in WL
     Controller.

❍   Command → show fdb.

6.10   ARP information

❍   You can get the information by using PC/Terminal software to log in WL
     Controller.

❍   Command → show arp.

6.11   Client session information

❍   You can get the information by using PC/Terminal software to log in WL Controller.

❍   Command → show sessions network.

6.12   VoIP information

❍   You can get the information by using PC/Terminal software to log in WL Controller.

❍   Command → show voip summary.

6.13   Log trace

❍   You can get the log by using PO/Terminal software to log in WL Controller.

❍   The trace shows event information regarding processes happening in real time.

❍   This is set in advance, and it should be turned off after you got the data because you can have adverse effects on system performance.

❍   Check the trace setting: show trace.

❍   Display the log trace: show log trace.

❍   Delete log trace: clear log trace.

❍   Delete running trace commands: clear trace [trace-area].

SECTION 7   LEDS OF WL1700

The following table will also assist in your troubleshooting efforts.

Table 11-10 LEDs of WL1700

LED	Indication	Status
Cont Status Lamp (indicates status of the controller portion).	Lights green	Running normally.
	Lights red	– Hardware is faulty (system cannot start up). – System detected an error by the diagnosis function while starting up.
	Off	System is starting up.
AP Status Lamp (indicates status of the access point portion).	Lights green	Running normally when meeting the following conditions: – Access point portion has started up. – Access point is connected to WL controller normally. – Any of the radios are enabled or active scanning is enabled while any radio is disabled.
	Lights orange	Access point is not connected to WL controller.
	Flashes green and orange alternately	Radio is disabled while the access point is connected to the WL controller or access point is starting up.
5GHz Lamp 2.4GHz Lamp	Lights green	Any client is associating to radio.
	Flashes green	Any associated client is communicating.
	Lights orange	Both radio and active scanning are disabled.
	Flashes orange	Any unassociated client is communicating.
	Flashes green and orange alternately	Radio cannot transmit. – Bad radio environment. – Radio is defective.
	Off	When meeting these conditions: – Active scanning is enabled while radio is disabled. – No client is associated while radio is enabled.

SECTION 8   PEGCOUNT

Pegcount provides traffic measurement and statistics of lost-call rate for MH240 accommodated on SIP server. In the WL1700 Controller System, traffic data for the MH240 is collected on AP and SSID basis. WL1700 Controller System stores the setup/end and succeeded/failed result of MH240 calls as Pegcount on the internal memory. Accessing the internal memory of WL1700 Controller, CLI reads and clears Pegcount. The function can be useful when you call but hear the busy tone. In this situation, the reservation bandwidth assigned for an AP might be exceeded.

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NAPT

12

SECTION 1   **NAPT**

1.1   Introduction

The NAPT (**N**etwork **A**ddress **P**ort **T**ranslation) feature gives the
SV8100 the ability to "traverse" its own subnet. With NAPT used in
the SV8100, the network administrator can place the CD-CP00-US
and the IPLA (VoIPDB) in the customers LAN while still making it
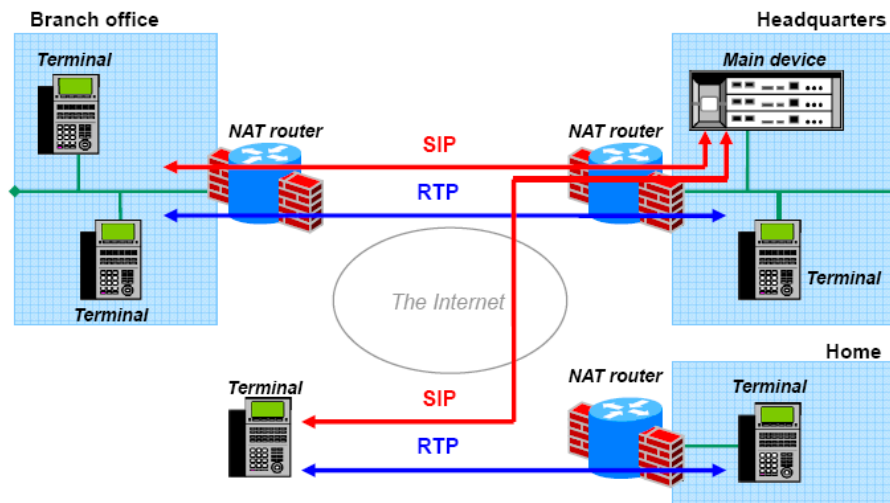accessible to users outside the local LAN.

NAPT

Figure 12-1 NAPT Configuration Example

1.2   SV8100 Requirements

The following information provides requirements for NAPT.

1.2.1   Main Software

NAPT is supported with the V3000/V3.00 or higher release of the SV8100.

1.2.2   Hardware

The SV8100 requires the following hardware:

☐   CD-CP00-US (V3000/V3.00 or higher)

☐   PZ-32IPLA, PZ-64IPLA, PZ-128IPLA

1.2.3   Capacity

512 Extensions (as allowed by hardware and licensing).

1.2.4   License

☐   The system must be licensed for this feature (License 0031).

☐   The system must be licensed for DT700 ITL Terminals.

1.3   Installation

The following settings have been added for NAT traversal.

1.3.1   Settings for terminals

Setting location: 0. Config/ 2. SIP Settings/ 8. NAT Traversal [Eco, Value, Sophisticated]

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

Number and Name of Setting	Setting Value	Default Value	Factory Value	Auto Config	Description
STUN Settings					Currently not used.

Setting location: 0. Config/ 2. SIP Settings/ 8. NAT Traversal/ 3. WAN Settings [Eco, Value, Sophisticated]

Number and Name of Setting	Setting Value	Default Value	Factory Value	Auto Config	Description
1. WAN Mate IP Address	IP Address	0.0.0.0	Available	Available	WAN Address of the router that the SV8100 resides behind. This setting must match what is programmed in PRG 10-12-07.
2. WAN SIP Mate Port	1024~65535	5060	Available	Available	Port number the SV8100 uses for SIP registration in PRG 10-46-06.
WAN Self IP Address	IP Address	0.0.0.0	Available	Available	Only used when Static NAT is enabled. This setting is the WAN address of the router that the NAPT Terminal resides behind.

Setting location: 0. Config/ 1. Network Settings/ 6. Advanced Settings/ 5. Self Port Settings [Eco, Value, Sophisticated]

Number and Name of Setting	Setting Value	Default Value	Factory Value	Auto Config	Description
1. RTP Self Port	1024~65528 (Even numbers only)	3462	Available	Not available	The number of the port receiving RTP data. Note: This setting applies to the ports (sending/receiving) used by an IP Terminal.
2. SIP Self Port	1024~65535	5060	Available	Not available	The number of the port receiving SIP data. Note: This setting applies to the ports (sending/receiving) used by an IP Terminal.

1.4   Network Configuration

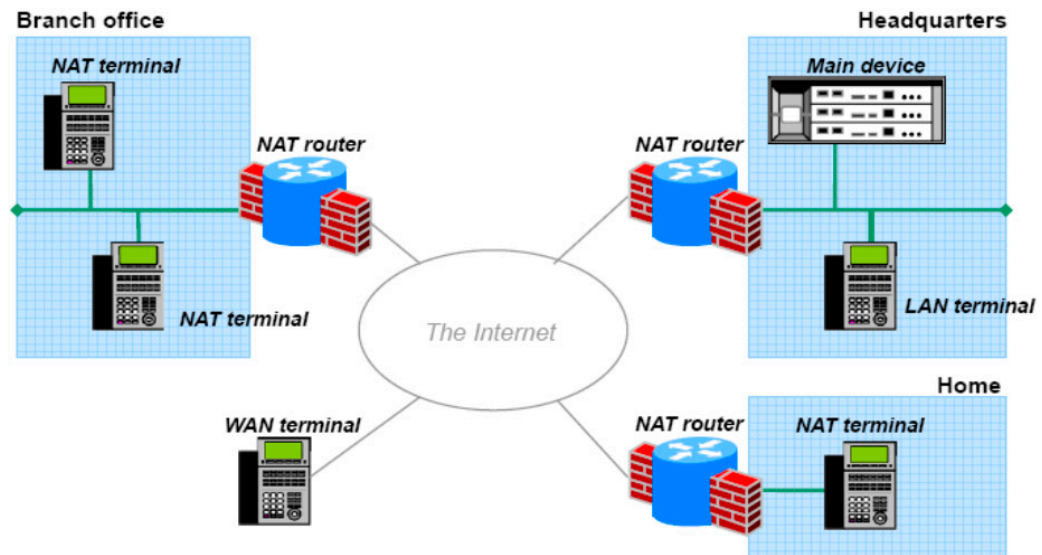A network configuration diagram is shown below.

Figure 12-2 Network Configuration Diagram

Depending on their locations, terminals are classified into the following three categories:

❍   LAN terminals

Terminals installed on the LAN where the main device is installed.

❍   WAN terminals

Terminals directly installed in the global address space.

❍   NAT terminals

Terminals connected to the main device through a NAT router.

1.5   Static NAT

This method is used to convert an address (a WAN-side address for the NAT router) previously specified in the terminal configuration. For this method, a conversion table is statically set up for the NAT router, and then a packet for a specific port is transferred to a specific terminal by using the port forwarding function of the NAT router.

To traverse over NAT by using static NAT, the following settings are required.

❍   Avoiding port overlap

When installing multiple terminals in a NAT domain, allocate the numbers of the SIP communication port and RTP communication port for each terminal so as to avoid overlapping the port numbers.

❍   Configuration settings for terminals

Specify an allocated port number and WAN-side global address in the configuration of each terminal. Terminals act as though there is a terminal on the WAN side by replacing the local IP address of the SIP message to a global address.

❍   Setting up a conversion table for a router

Statically register a conversion table for the terminal IP address in the NAT router domain and the port on the WAN side.
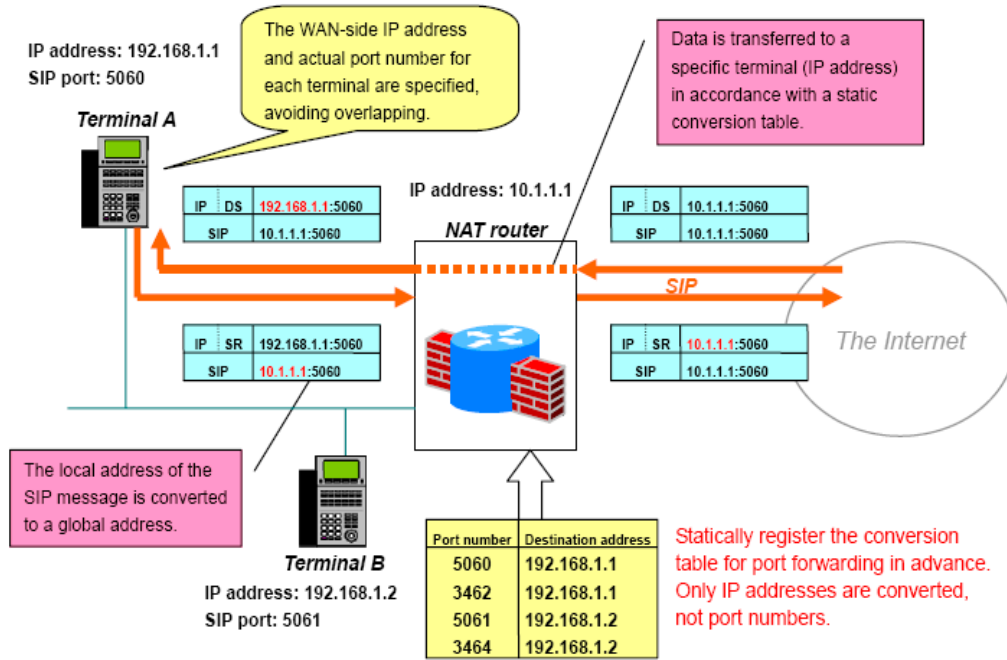

Figure 12-3 Static NAT Example

1.6   Dynamic NAT

This method is used to automatically acquire the WAN-side global address for the NAT router and convert it.
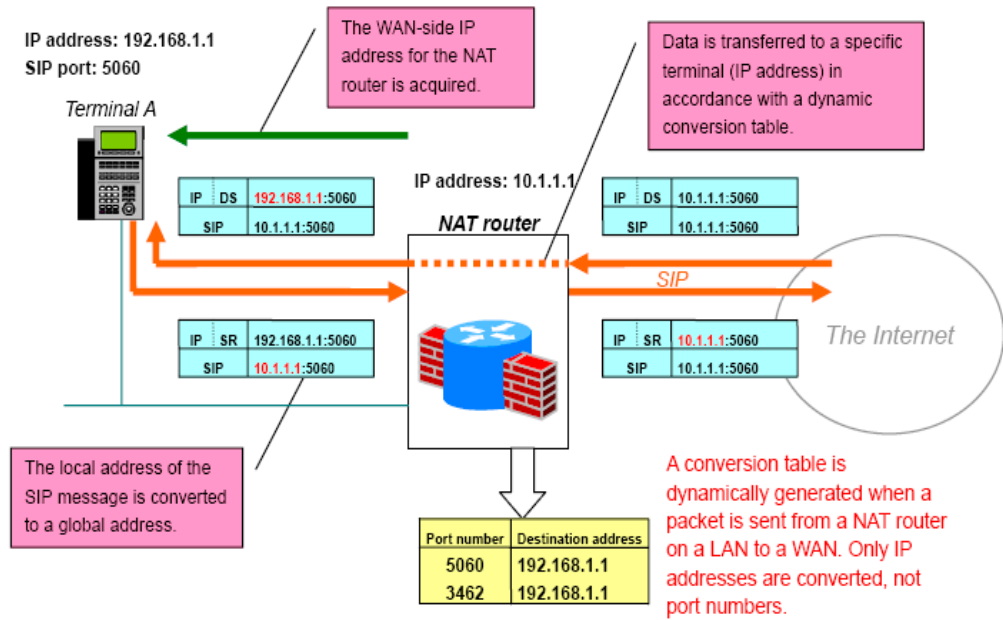

Figure 12-4 Dynamic NAT Example

1.6.1   Acquiring a Global Address from the Main Device

A terminal sends a dummy registration request to the main device and acquires a global address from the response from the main device. More specifically, the terminal acquires the global IP address from the "received" parameter in the Via header of the response to the REGISTER request.
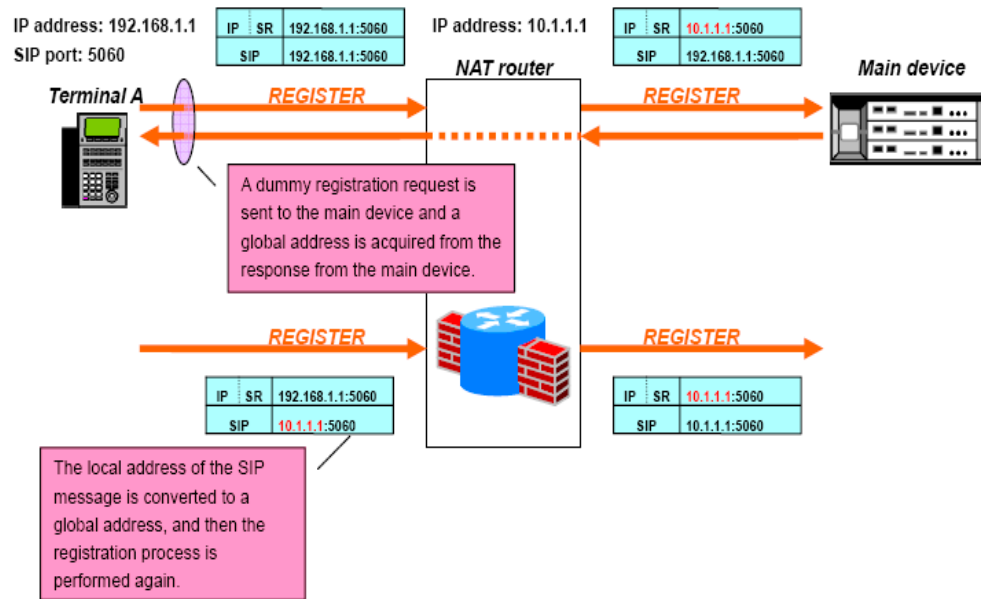

Figure 12-5 Acquiring a Global Address

1.7   Terminal Operations and Settings

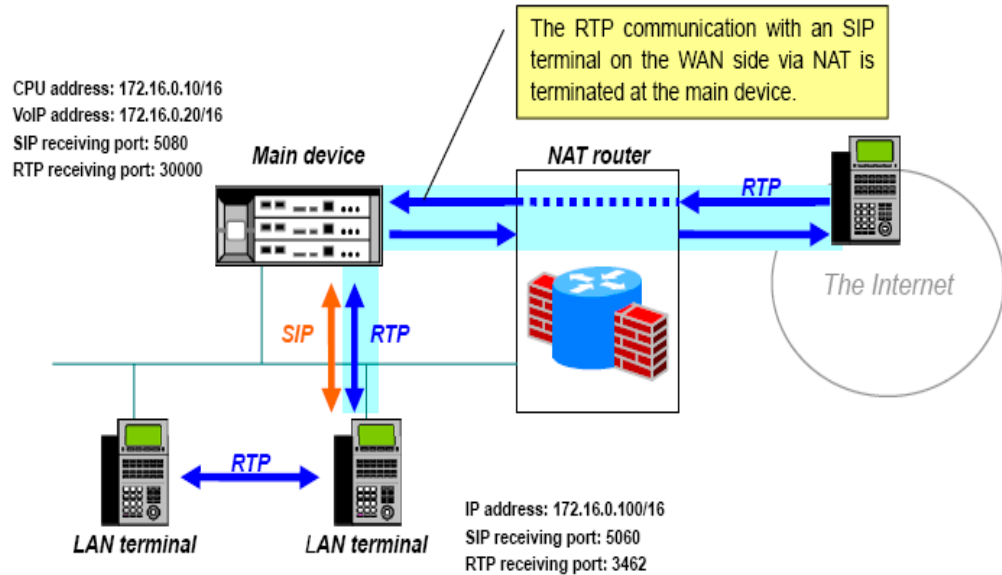### 1.7.1   LAN Terminal Operation

Figure 12-6 LAN Terminal Operation

1.7.2   SIP Communication

Because the main device is installed on the same network (the internal network of the NAT router), communication with devices outside the NAT router is not performed.

1.7.3   RTP Communication

RTP communication between LAN terminals and SIP terminals installed at each location is established as shown below under the control of the main device. Because communication with SIP terminals on the WAN side is established via VoIP of the main device, communication with devices outside the NAT router is not performed.

Remote Terminal/ Local Terminal	LAN Terminal	WAN Terminal	NAT Terminal
LAN Terminal	Peer to Peer	VoIP termination of the main device	VoIP termination of the main device

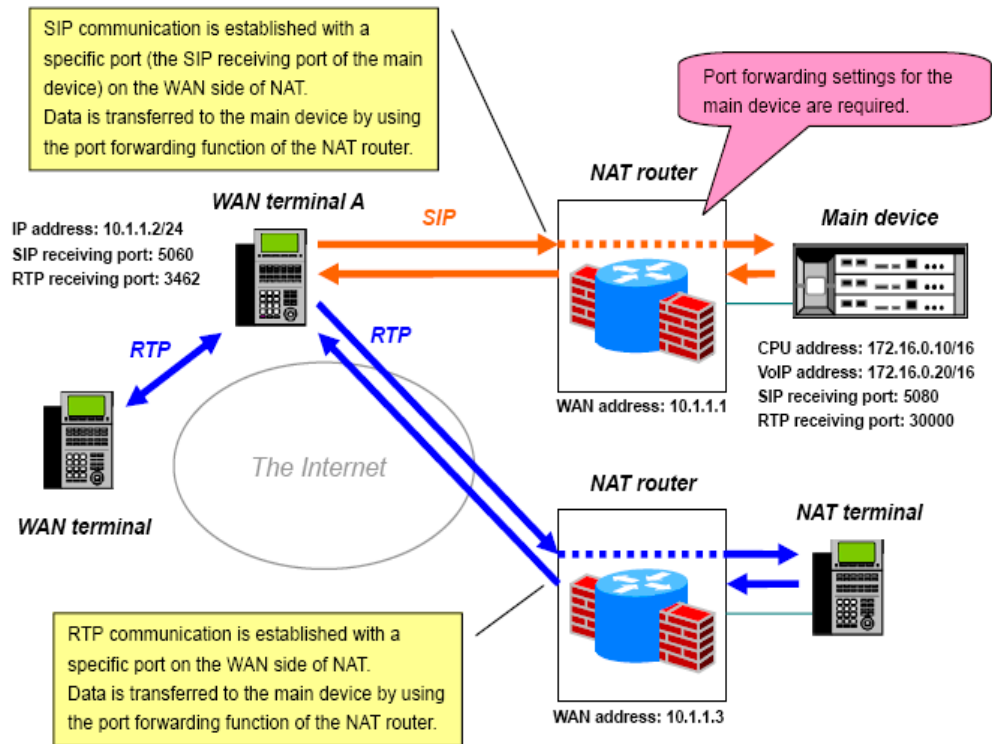### 1.7.4   WAN Terminal Operation

Figure 12-7 WAN Terminal Operation

1.7.5   SIP Communication

Because communication using WAN terminals is established between global address spaces, communication from within the NAT router network to external devices is not performed. If the main device is in a NAT router domain, SIP communication is established with the NAT router, and data is transferred to the main device by using the port forwarding function of the NAT router.

1.7.6   RTP Communication

The communication between WAN terminals and SIP terminals installed at each location is established as follows under the control of the main device.

Remote Terminal/ Local Terminal	LAN Terminal	WAN Terminal	NAT Terminal
WAN Terminal	VoIP termination of the main device	Peer to Peer	Peer to Peer

1.7.7   NAT Terminal Operation (Static NAT)

If you can change the NAT router settings, such as in branch office, you can traverse over NAT by using static NAT.
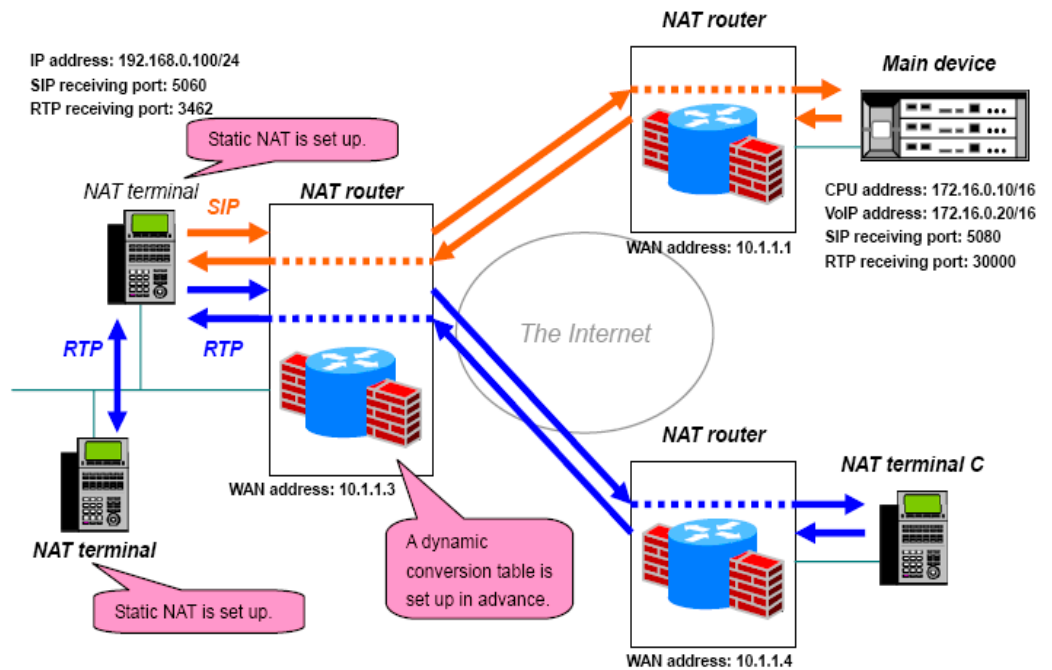

Figure 12-8 NAT Terminal Operation (Static NAT)

1.7.8   SIP Communication

Because this communication is established from within the NAT router to outside, NAT terminals convert addresses of SIP messages to act as if they are installed in the global address space, and then these terminals communicate with the main device.

1.7.9   RTP Communication

The RTP communication between NAT terminals and SIP terminals installed at each location is established as follows under the control of the main device.

Remote Terminal/ Local Terminal	LAN Terminal	WAN Terminal	NAT Terminal
NAT Terminal	VoIP termination of the main device	Peer to Peer	Peer to Peer

1.7.10   NAT Terminal Operation (Dynamic NAT)

If you cannot change the settings for the NAT router, such as in a hotel, you can traverse over NAT by using dynamic NAT.
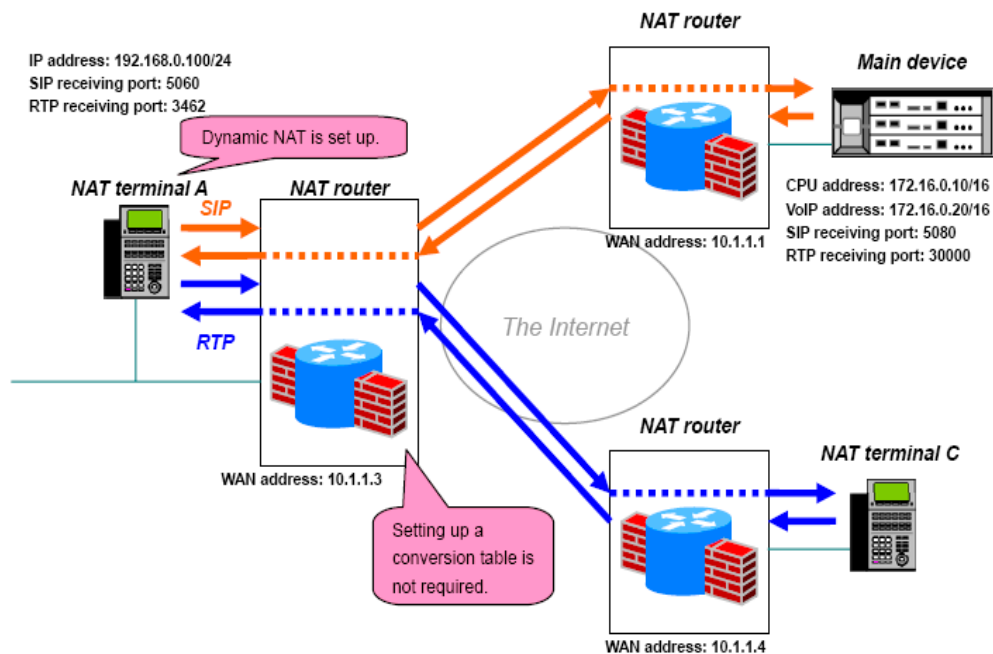

Figure 12-9 NAT Terminal Operation (Dynamic NAT)

1.7.11   SIP Communication

Because this communication is established from within the NAT router to outside, NAT terminals convert addresses of SIP messages to act as if they are installed in the global address space, and then these terminals communicate with the main device.

1.7.12   RTP Communication

The RTP communication between WAN terminals and SIP terminals installed at each location is established as follows under the control of the main device.

Remote Terminal/ Local Terminal	LAN Terminal	WAN Terminal	NAT Terminal
NAT Terminal	VoIP termination of the main device	Peer to Peer	Peer to Peer

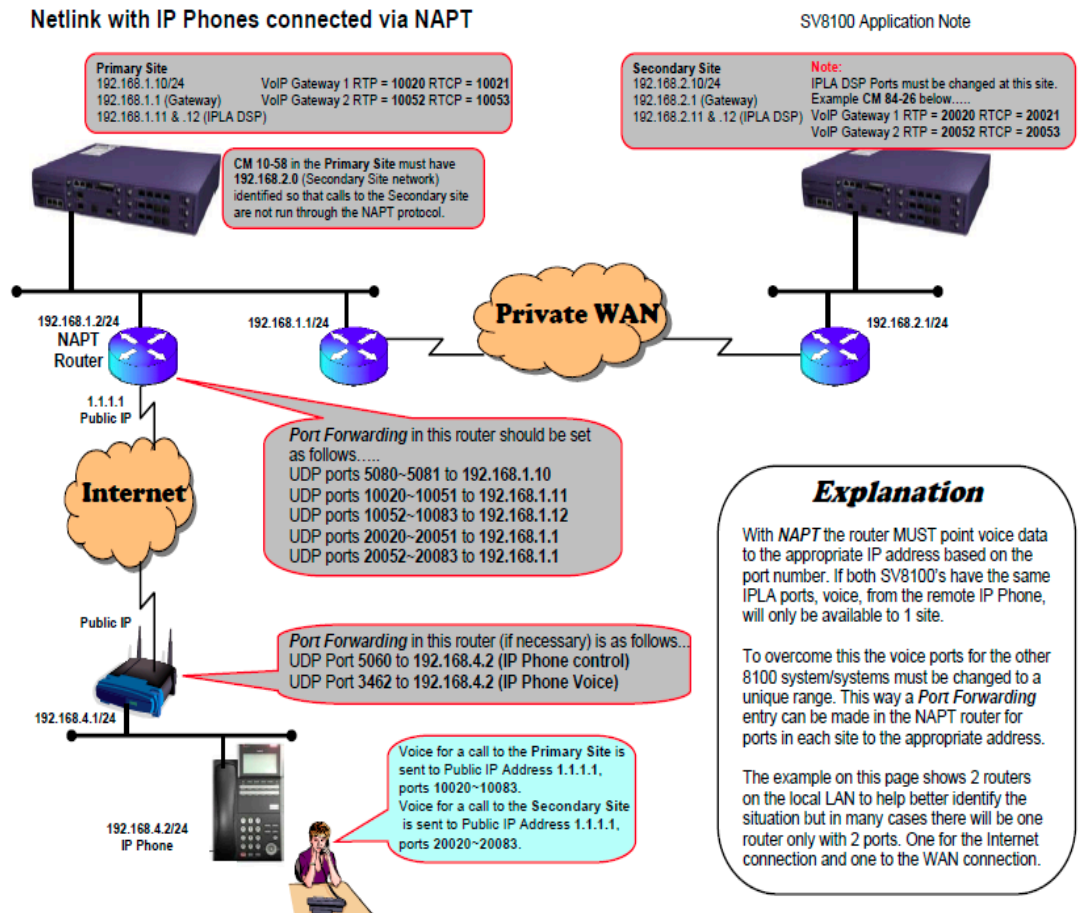### 1.7.13   NetLink with IP Phones connected via NAPT

Figure 12-10 NetLink with IP Phones connected via NAPT

SECTION 2   CONDITIONS

❑   The NAPT feature requires a license on the CD-CP00-US (license ID 0031).

❑   "License Exceeded" will display on the DT700 terminal when trying to register a NAT phone, if the feature is not licensed.

❑   The NAPT feature requires CPU software V3.00 or higher.

❑   Terminals using NAPT must be at firmware V3.0.0.0 or higher.

❑   IP terminals can be connected via NAT router or WAN (direct connection).

❑   The NAT router on the SV8100 side must have a static WAN IP address, UPNP is not supported.

❑   K-CCISoIP does not use NAPT translations.

❑   Softphone (SP310) is not supported.

❑   Wireless phone (MH240) is not supported.

❑   If NAT enabled phones become unresponsive after being idle, the timer in Programs 84-23-01 and 84-23-02 may need to be changed to a shorter interval.

❑   It is necessary to set Program 10-46-14 to **OFF** when the IPLA is assigned a global (public) IP address.

❑   When Program 10-46-14 is set to **ON**, it references programs 10-58-01 and 10-58-02. These programs are used to define any destination networks that do not get sent through the NAPT translations.

❑   UDP ports in the remote routers may be required to be forwarded to the IP Terminals.

❑   NAPT can be used for SIP trunks and terminals on the same system.

SECTION 3   RESTRICTIONS – STATIC NAT

❑   With static NAT, the terminal needs a static IP Address assigned to it, or entries in the DHCP must be made to provide the same IP Address to the terminal.

❑   The NAT router on the terminal side must have the function for setting up static NAT.

❑   A conversion table must be manually set up for the NAT router on the terminal side.

❑   If installing multiple terminals in the domain of the NAT router on the terminal side, the SIP port number and RTP/RTCP port number for each terminal must be specified so as to avoid overlapping.

❑   The SIP server cannot be switched. (Only one address can be registered as the SIP server.)

❑   NetLink Failover is not supported.

Dynamic NAT

❑   The NAT router on the terminal side must have the function for setting up dynamic NAT.

❑   It is assumed that port numbers are not changed by the NAT router on the terminal side. If a port number is changed by NAT router, NEC does not guarantee proper operation.

❑   If installing multiple terminals in the domain of the NAT router on the terminal side, the SIP port number and RTP/RTCP port number for each terminal must be specified so as to avoid overlapping.

❑   The SIP server cannot be switched. (Only one address can be registered as the SIP server.).

❑   NetLink Failover is not supported.

SECTION 4 MINIMUM REQUIRED PROGRAMMING

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

❏ Level 1 – these are the most commonly assigned programs for this feature.

❏ Level 2 – these are the next most commonly assigned programs for this feature.

❏ Level 3 – these programs are not often assigned and require an expert level working knowledge of the system to be properly assigned.

Program Number	Program Name	Description/Comments	Assigned Data	Level 1	Level 2	Level 3
10-12-03	CD-CP00-US Network Setup – Default Gateway	Assign the default gateway IP address for the CD-CP00-US.	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 = 0.0.0.0)	✓		
10-12-07	CD-CP00-US Network Setup – NAPT Router IP Address (Default Gateway [WAN])	Define the IP Address of the WAN side of the router.	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 (default = 0.0.0.0)	✓		
10-12-09	CD-CP00-US Network Setup – IP Address	Set for IPLA.	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 (default = 172.16.0.10)	✓		

Program Number	Program Name	Description/Comments	Assigned Data	Level		
				1	2	3
10-12-10	CD-CP00-US Network Setup – Subnet Mask	Define the Media Gateway Subnet Mask Address.	128.0.0.0 192.0.0.0 224.0.0.0 240.0.0.0 248.0.0.0 252.0.0.0 254.0.0.0 255.0.0.0 255.128.0.0 255.192.0.0 255.224.0.0 255.240.0.0 255.248.0.0 255.252.0.0 255.254.0.0 255.255.0.0 255.255.128.0 255.255.192.0 255.255.224.0 255.255.240.0 255.255.248.0 255.255.252.0 255.255.254.0 255.255.255.0 255.255.255.128 255.255.255.192 255.255.255.224 255.255.255.240 255.255.255.248 255.255.255.252 255.255.255.254 255.255.255.255 (default = 255.255.0.0)	✓		

Program Number	Program Name	Description/Comments	Assigned Data	Level		
				1	2	3
10-20-01	LAN Setup for External Equipment – TCP Port	Define the TCP port (0~65535) when communicating to the SMDR (type 5).	0~65535 default: External Device 1 (CTI Server) = 0 External Device 2 (ACD MIS) = 4000 External Device 5 (SMDR Output) = 0 External Device 6 (DIM Output) = 0 External Device 11 (O&M Server) = 8010 External Device 12 (Traffic Report Output) = 0 External Device 13 (Room Data Output for Hotel Service) = 0 External Device 14 (IP-DECT Directory Access) = 0	✓		
10-46-06	Register Port Assign the port number to which the SIP messages are sent to on the IPLA. This same port number must be assigned in the SIP Multiline terminals. If this command is changed, it requires a CPU reset.	0~65535	5080		✓	
10-46-13	Subscribe Session Port	0~65535	5081		✓	
10-46-14	NAT Mode	Turns On/Off the NAT mode of the system.	0 = Off 1 = On Default = Off	✓		
10-58-01	DT700 Intranet Local Area Network Setup Network Address	This program sets the local area network of the DT700 terminal when the system has a NAT router and a local router.	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254		✓	

Program Number	Program Name	Description/Comments	Assigned Data	Level		
				1	2	3
10-58-02			248.0.0.0 / 252.0.0.0 / 254.0.0.0 / 255.0.0.0 255.128.0.0 / 255.192.0.0 / 255.224.0.0 255.240.0.0 / 255.248.0.0 / 255.252.0.0 255.254.0.0 / 255.255.0.0 / 255.255.128.0 255.255.192.0 / 255.255.224.0 255.255.240.0 / 255.255.248.0 255.255.252.0 / 255.255.254.0 255.255.255.0 / 255.255.255.128 255.255.255.192 / 255.255.255.224 255.255.255.240 / 255.255.255.248 255.255.255.252 / 255.255.255.254		✓	
84-26-01	IPL Basic Setup – IP Address	Assign the IP address for each DSP on the IPLA.	Slot 1 = 172.16.0.20 : Slot 4 = 172.16.0.44 VoIP GW 1~8: 172.16.0.20~ 172.16.0.27	✓		
84-26-02	IPL Basic Setup – RTP Port Number	Assign the RTP port number to be used for each DSP on the IPLA. ✎ *Only even numbered ports are supported.*	VoIP GW1 = 10020 VoIP GW2 = 10052 VoIP GW3 = 10084 VoIP GW4 = 10116 VoIP GW5 = 10148 VoIP GW6 = 10180 VoIP GW7 = 10212 VoIP GW8 = 10244	✓		
84-26-03	IPL Basic Setup – RTCP Port Number (RTP Port Number + 1)	Define the TCP port number for RTCP to use for each DSP.	VoIP GW1 = 10021 VoIP GW2 = 10053 VoIP GW3 = 10085 VoIP GW4 = 10117 VoIP GW5 = 10149 VoIP GW6 = 10181 VoIP GW7 = 10213 VoIP GW8 = 10245		✓	

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All DSP Busy Indication

SECTION 1   **INTRODUCTION**

The All DSP Busy feature is used to alert users via telephone displays and/or Alarm reports when all DSP resources in the system are being used. This can be used to trouble shoot issues or to alert when the current hardware might need to be upgraded to a higher capacity.

Alarm Message Format:

The Alarm message for will vary depending on what type of resource is unavailable and whether the system is stand alone, Netlink or CCISoIP. This will be displayed on telephones and included in printed or emailed Alarm Reports.

Table 13-1 Alarm Types

Parameters	Description
STA	DSP for IP Station Calls Were All Busy.
TRK	DSP for Trunk Calls Were All Busy, includes SIP and CCISoIP Trunks.
NET	DSP for CCISoIP Networking Calls Were All Busy.
LNK	SP for NetLink Calls Were All Busy.

Alarm Report Example:

The report example below shows an alarm for all busy Station and Trunk DSPs.

```
<< ALARM REPORT >>                              01/22/2009  23:37
 LVL  NO  STAT   DATE     TIME      ITEM          UNIT      SLT PRT PARAMETER
----------------------------------------------------------------------------
 MAJ 0068 ERR  01/22/09 09:30 VoIP All DSP Busy   VoIPDB    01  00  STA
 MAJ 0068 ERR  01/22/09 09:31 VoIP All DSP Busy   VoIPDB    01  00  TRK
 MIN 0002 REC  01/22/09 09:32 PKG Installation    PRIU      02  00
```

LCD Display

LCD Display Indication	Note
Clock/Calendar	Non-NetLink System
VoIP DSP All Busy XX	
Soft Key	XX = Slot number of CCPU with IPLA blade.
Clock/Calendar	NetLink System
VoIP DSP All Busy YY-XX	YY = System ID
Soft Key	XX = Slot number of CCPU with IPLA blade.

SECTION 2   SERVICE CONDITIONS

❍   When using IP Phones, the alarm is shown on both terminals involved in that call if they are both on the same system, this includes NetLink systems.

❍   The alarm cannot be displayed on Standard SIP Phones, Soft Phones or Single Line phones.

❍   If a call from a Standard SIP telephone to a Multiline telephone cannot be established due to an All DSP Busy condition, the Multiline telephone will not display the "All DSP" busy message.

❍   If a SIP trunk call is sent to the SV8100 when all DSP resources are busy, the call is rejected but the alarm is not displayed on any system telephone.

❍   The default alarm setting is Minor.

❍   If an all DSP condition is encountered when making a call across a CCISoIP Network, only the calling stations receives the alarm indication.

SECTION 3   RELATED FEATURES

❍   IP Multiline Station (SIP)

❍   IP Trunk – (SIP) Session Initiation Protocol

❍   K-CCIS – IP

❍   SV8100 NetLink

SECTION 4   GUIDE TO FEATURE PROGRAMMING

Program Number	Program Name	Description/Comments	Assigned Data	Level 1	2	3
20-13-52	VoIP All DSP Busy Display	Set on a per station class of service basis, whether the "All DSP Busy" alarm displays on the LCD when the caller makes an IP call and there is no VoIP DSP resource.	0 = Off 1 = On (default = 1)	✓		
90-10-01	System Alarm Setup	Alarm Number 68 is used for All DSP Busy.	0 = Not set 1 = Major 2 = Minor (default = 2)		✓	

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