NEC

UNIVERGE® SV9100

Networking Manual

A50-035910-004 AU
Issue 2.0

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Introduction

IMPORTANT

When the GPZ-IPLE daughter board is installed, half-duplex connections are not supported. For troubleshooting purposes, a managed switch capable of port mirroring is required to capture packet data from the SV9100 IPLE Ethernet port.

SECTION 1 GENERAL OVERVIEW

This manuals provides information for networking for the UNIVERGE SV9100 systems. Networking can be accomplished using one of the following methods:

- SV9100 K-CCIS (US Only)
- SV9100 IP K-CCIS
- SV9100 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 I-1 Common Terms and Associated Abbreviations

Term	Abbreviation	Description
GCD-CP10	GCD-CP10 SV9100 CPU CPU	SV9100 CPU
GPZ-IPLE	VOIPDB VOIP Daughter Board	VoIP Daughter Board, mounted on GCD-CP10

Table I-1 Common Terms and Associated Abbreviations (Continued)

Term	Abbreviation	Description
GCD-2BRIA	BRIU 2BRIU	2-port Basic Rate Interface blade
GPZ-2BRIA	2BRIUDB BRIUDB	2-port basic rate daughter board, mounted on the GCD-2BRIA
GCD-CCTA	CCHU CCIS CCHU	CCIS Trunk Interface blade
GCD-4COTC	4COIU	4-port loop/ground start trunk blade
GPZ-4COTG	4COIUDB 4COIU Daughter Board	4-port loop/ground start daughter board, mount on GCD-4COTC blade
GCD-8DLCA GCD-16DLCA	8ESIU 16ESIU	8 digital station interface blade 16 digital station interface blade
GPZ-8DLCB	8ESIUDB 8ESIU Daughter Board	8 digital station interface daughter board, mounted on the GCD-8DLCA
CD-PVAA	PVAA	Conference Bridge blade
GCD-RGA	RGA	Application Gateway
DT400/DT300 Series	DT400/DT300 Digital Multiline Terminal DTZ/DTL Telephones Digital Multiline Telephones	SV9100/SV9300 Digital Multiline Terminals. Telephone equipment names begin with DTL/DTZ. For example: DTL-12D-1 TEL, DTL-24D-1 TEL, DTL-32D-1 TEL, DTZ-12D-3 TEL, DTZ-24D-3 TEL, etc.
DT800/DT700 Series	DT800/DT700 IP Digital Multiline Terminal ITZ/ITL Telephones IP Digital Multiline Telephones	SV9100/SV9300 IP Digital Multiline Terminals. Telephone equipment names begin with ITL/ITZ. For example: ITZ-12D-3 TEL, ITZ-24D-3 TEL, ITL-32D-1 TEL, etc.
GCD-4DIOPB	DIOPU 4DOIPU	4-port DID/OPX blade
GCD-4ODTB	TLIU 4TLIU	4-port E&M tie line blade

Book 1 – SV9100 K-CCIS

General Information <US Only>

Chapter 1

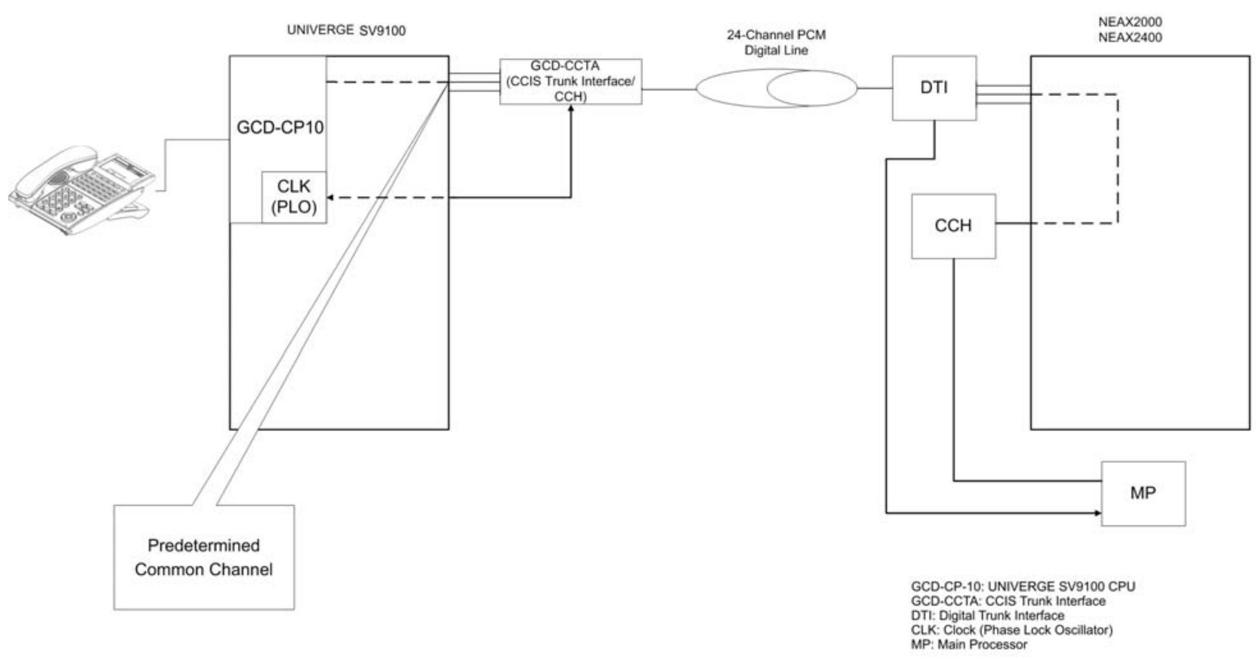
This chapter provides a system outline, the name and functions of the circuit cards required, system capacity, time slot assignments, system specifications and network structure considerations for UNIVERGE SV9100 K-CCIS.

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 (GCD-CCTA) for CCIS Trunk Interface/CCH. The GCD-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 GCD-CP10.

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 . For a diagram of the system outline, refer to Figure 1-1 K-CCIS System Outline.

Figure 1-1 K-CCIS System Outline



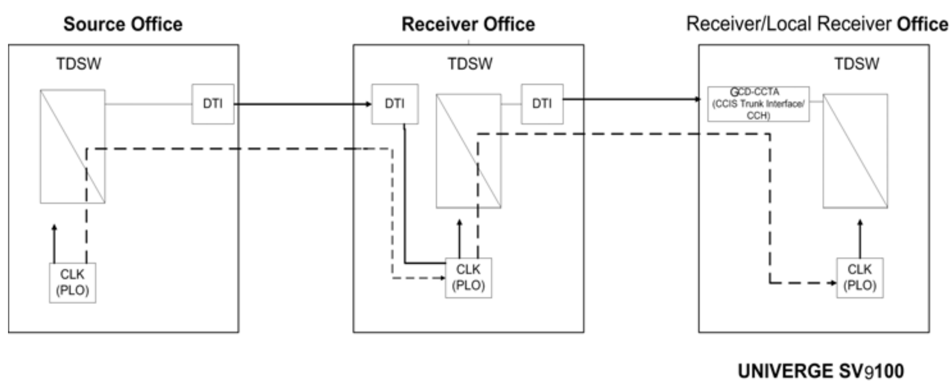
1.1 Common Channel Handler (GCD-CCTA)

The Common Channel Handler (GCD-CCTA) blade provides a common channel signal to a UNIVERGE SV9100 network. It is responsible for signaling between the Key Telephone System (KTS) and the UNIVERGE SV9100 network under control of the GCD-CP10.

1.2 Phase Locked Oscillator (PLO)

The Phase Locked Oscillator (PLO) provides synchronization between the TDSW and other offices and is built into the GCD-CP10. 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 GCD-CCTA.

Figure 1-2 Clock Supply Route



1.3 Common Channel Interoffice Signaling (K-CCIS)

The PBX can provide connection to the UNIVERGE SV9100 using a digital network. The network requires the Common Channel Handler (GCD-CCTA) to control the common signaling between offices.

Digital Network

When UNIVERGE SV9100 is provided through a digital network, the CCH and T-1 functionality is combined on the GCD-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.

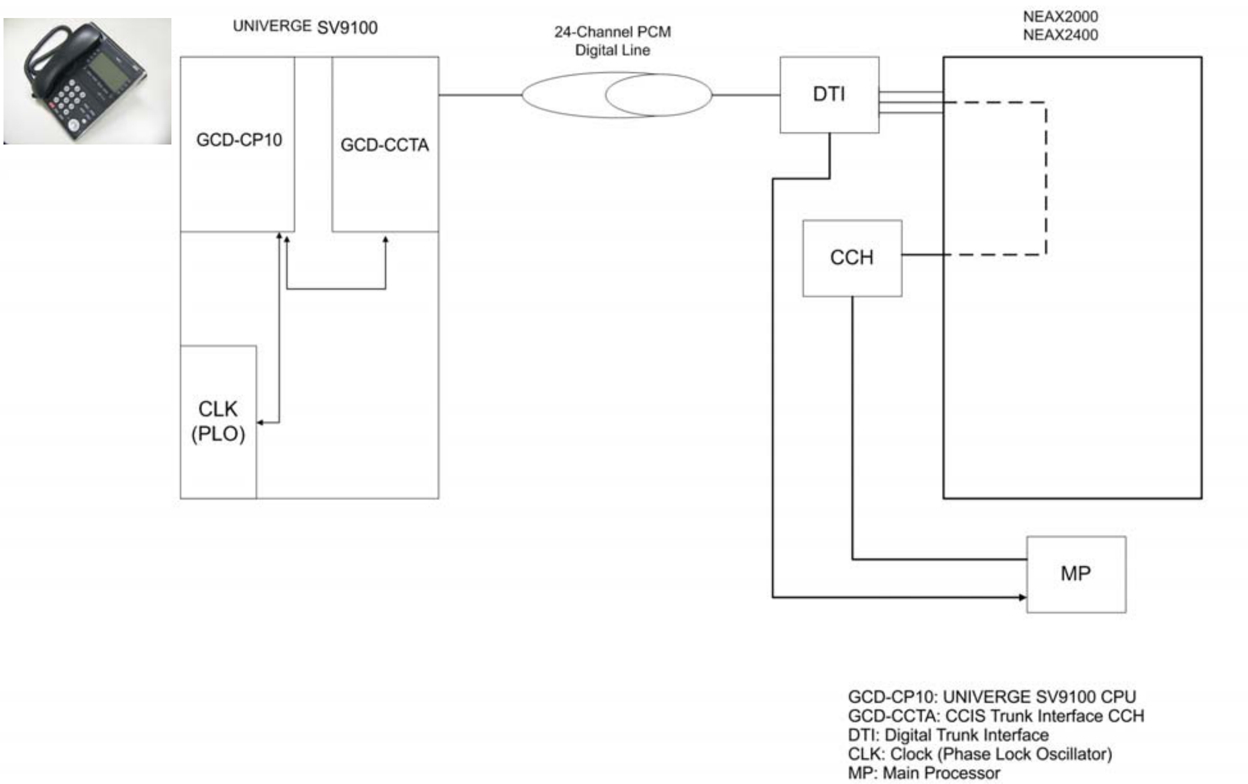


Figure 1-3 K-CCIS System Configuration

Section 2 DTI Specifications

The following specifications apply to the GCD-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 SV9100 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.

Figure 1-4 Frame Configuration of 24-DTI (12 Multiframe)

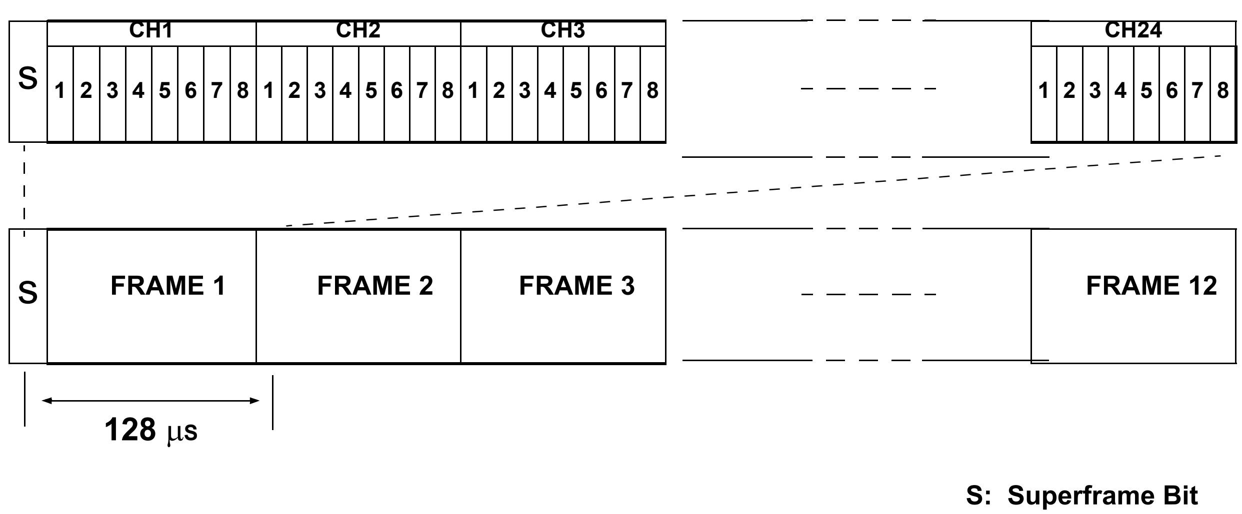


Table 1-1 12-Multiframe Bit Assignment

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)

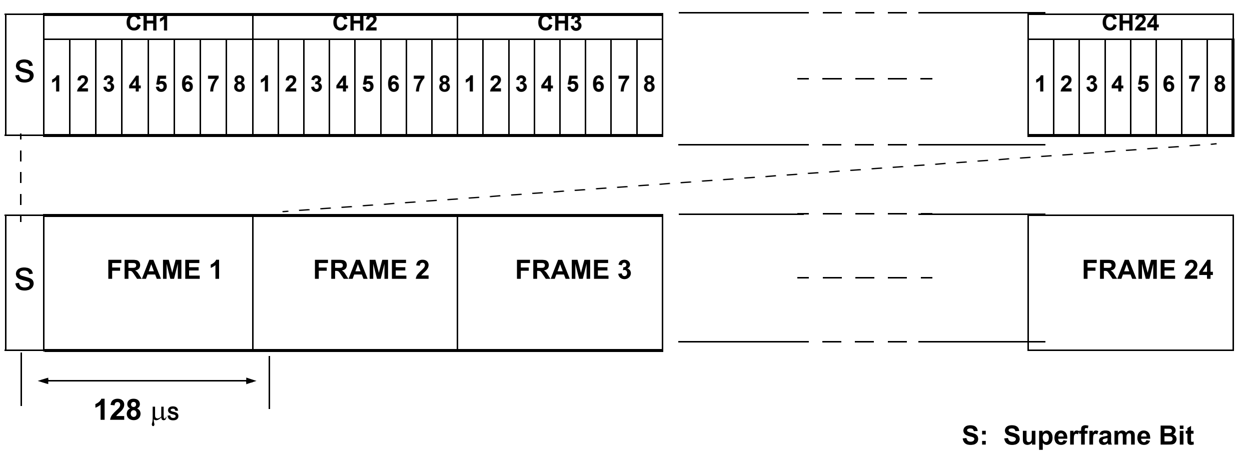


Table 1-2 24-Multiframe Bit Assignment

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
11		m	
12	1		

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

Frame Number	S-Bit		
	Frame Synchronization	4 kbps Data Link	CRC
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 SV9100 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).

Figure 1-6 Star Topology (KTS-to-KTS or PBX-to-KTS)

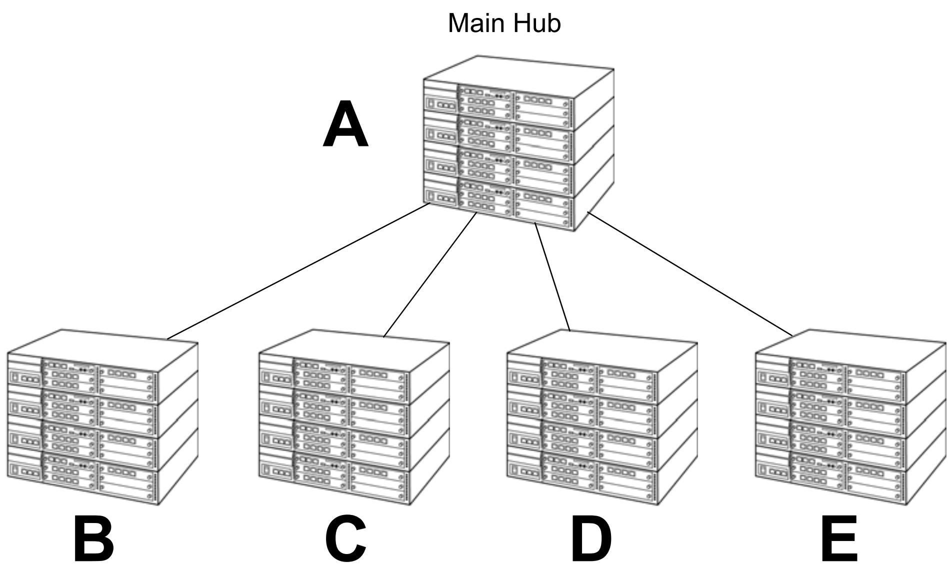
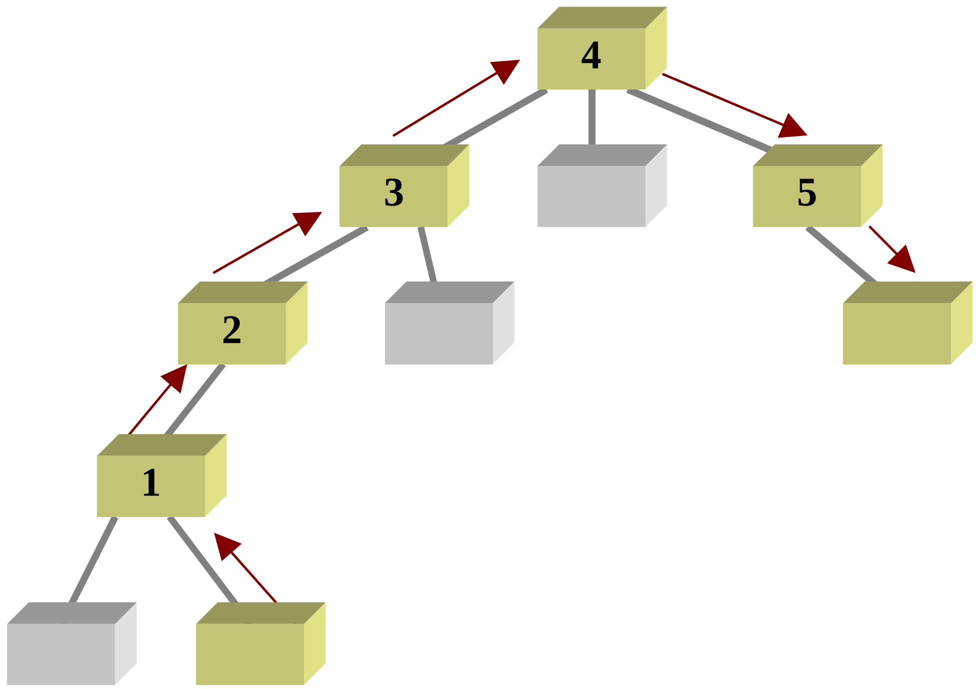


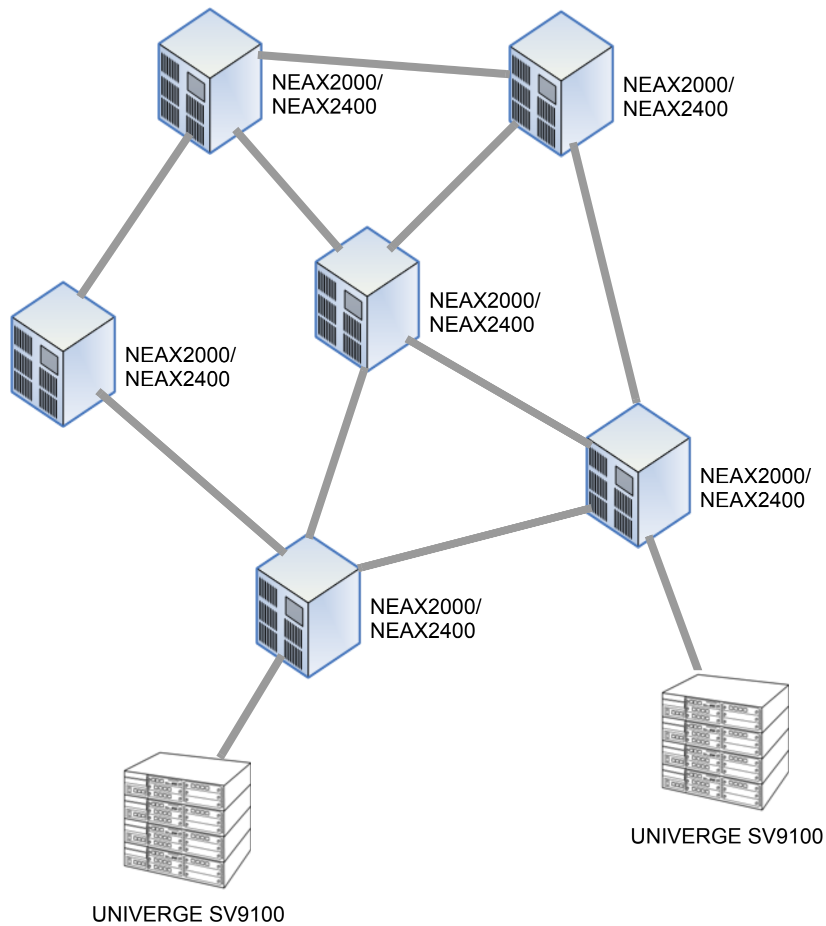
Figure 1-7 Tree Topology (KTS-to-KTS or PBX-to-KTS)



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



A UNIVERGE SV9100 can be connected to a PBX only as a remote office. The UNIVERGE SV9100 software can register a maximum of 255 point codes in the UNIVERGE SV9100 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 SV9100 network.

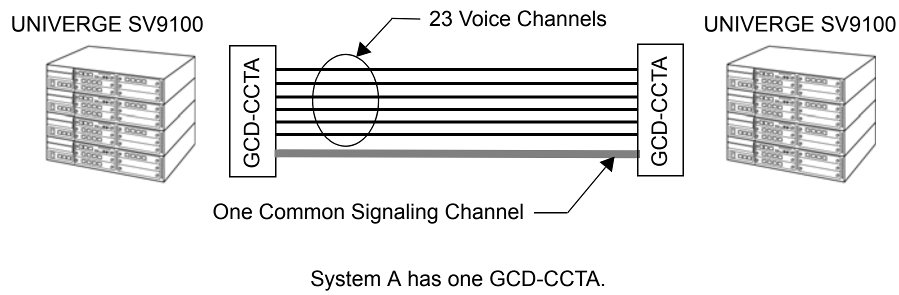
3.2 Determining Number of K-CCIS Routes

When the system is a Central Office or Tandem Office, two or more routes to other offices are required. Each GCD-CCTA can support one K-CCIS link. Up to eight GCD-CCTA blades can be installed in a UNIVERGE SV9100 system. The KTS requires the GCD-CCTA to support a K-CCIS interface.

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

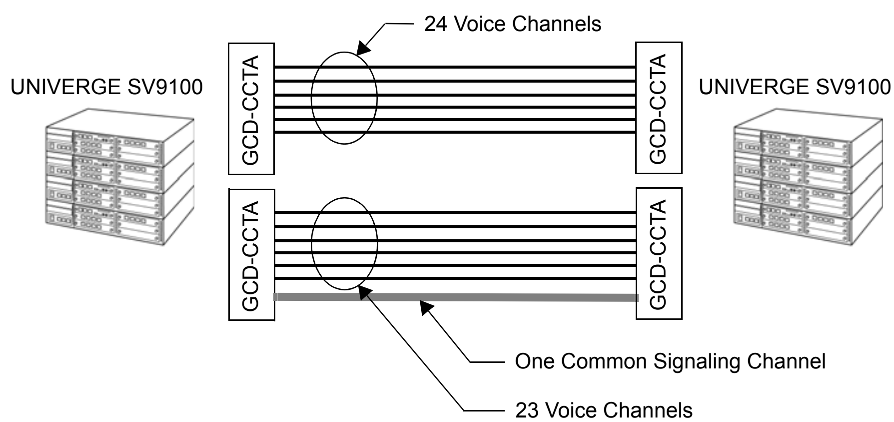
One GCD-CCTA provides 24 channels. One Common Signaling Channel port must be assigned on the Digital Trunk Interface in case two systems are connected by one Digital Link.

Figure 1-9 K-CCIS Routes



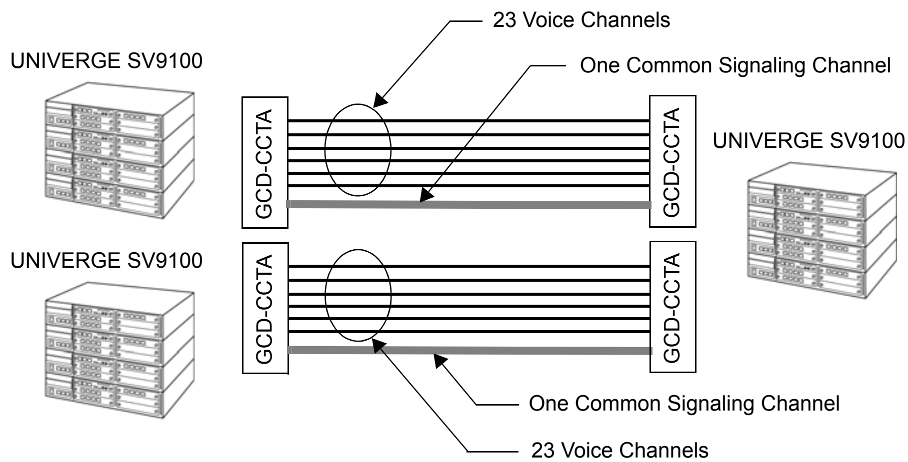
One Common Signaling Channel can be assigned even if two Digital Links are connected between two systems. Only one Common Signaling Channel can be assigned per GCD-CCTA in each system.

Figure 1-10 Assigning One Common Signaling Channel Between Two Systems



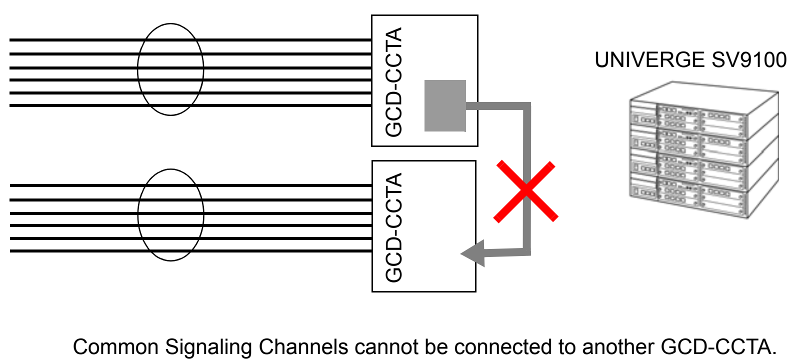
One GCD-CCTA can be assigned to one Common Signaling Channel. Tandem K-CCIS connections require one GCD-CCTA for every connection as shown in Figure 1-11 One GCD-CCTA Assigned per Common Signaling Channel.

Figure 1-11 One GCD-CCTA Assigned per Common Signaling Channel



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

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 SV9100 system, are digital only (GCD-CCTA).

3.4 Determining which Systems Should be the Central Office

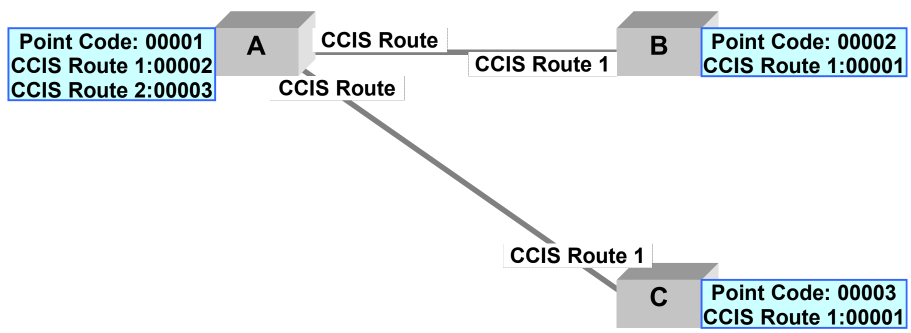
If using a KTS-to-KTS only network and features such as Voice Mail Integration – K-CCIS are used, the key system that has the voice mail system installed, must be programmed as the Central/Originating Office. All other key systems must be programmed as Remote/Destination offices.

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

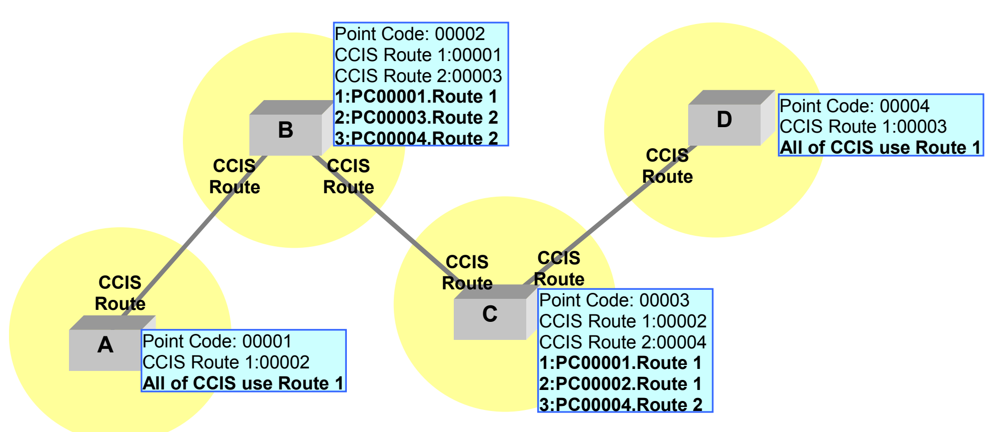
❍ **Data Assignment for System C**

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	

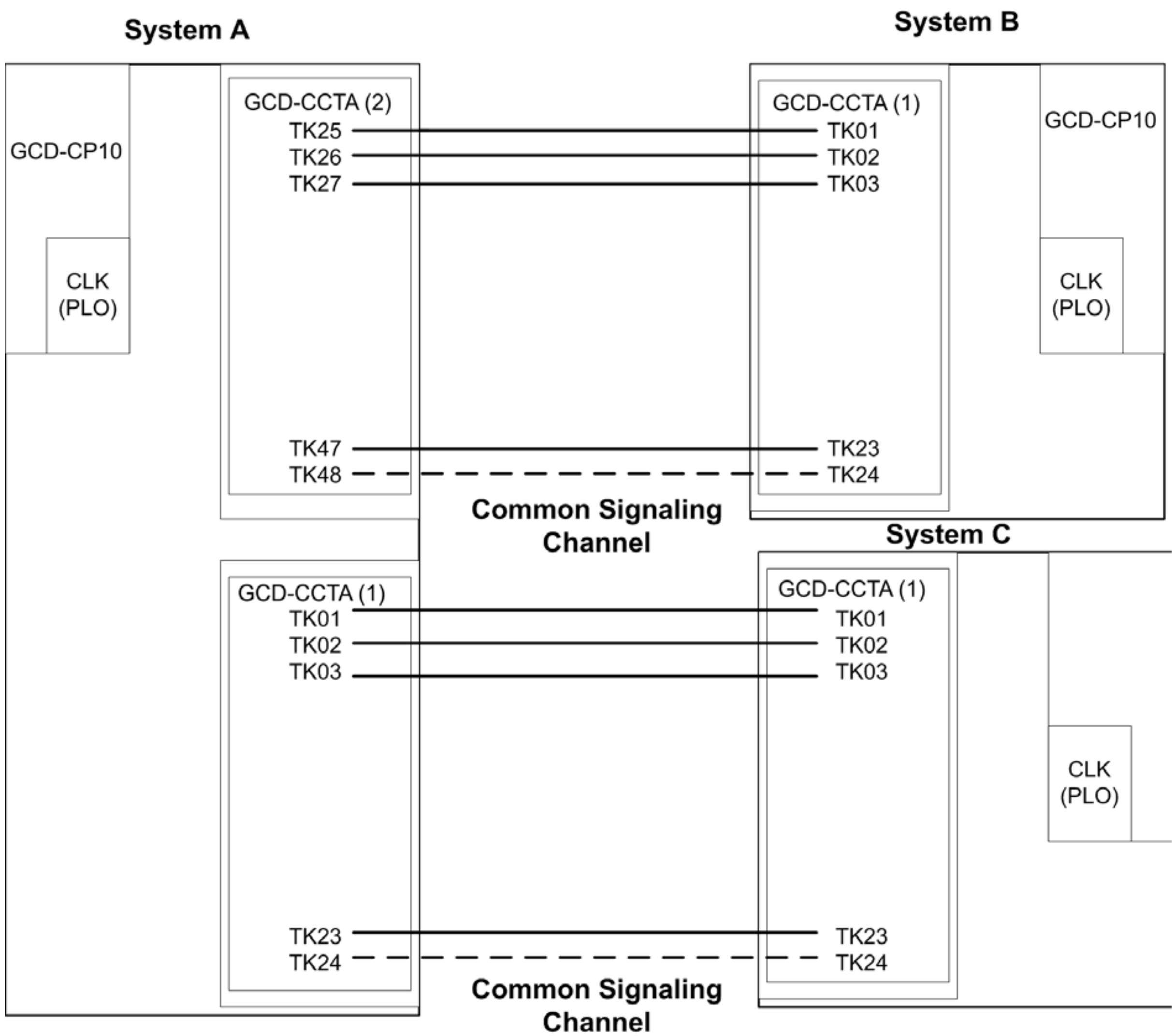
❍ **Data Assignment for System D**

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 GCD-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.

Figure 1-15 Circuit Identification Codes (CIC)



System A:
CIC Numbers for GCD-CCTA (1)TK 01~23 = 001~023
CIC Numbers for GCD-CCTA (2)TK 25~47 = 001~023

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

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

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

3.8 Determining Numbering Plan

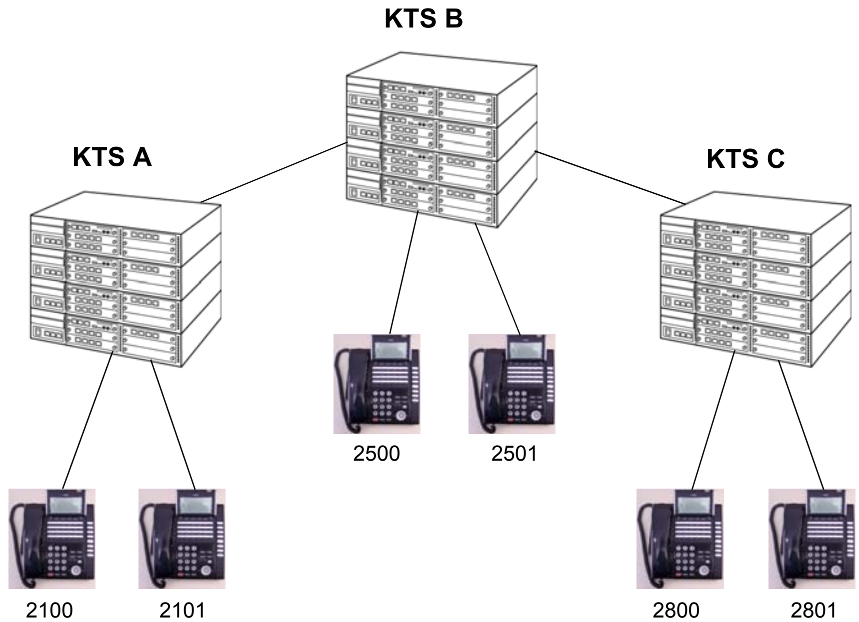
The Uniform Numbering Plan is the numbering plan in the K-CCIS network. The F-Route (Flexible Route Selection) and the Automatic Route Selection (ARS) feature provide the Open Numbering Plan.

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).

Figure 1-16 Closed Numbering Plan Example



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

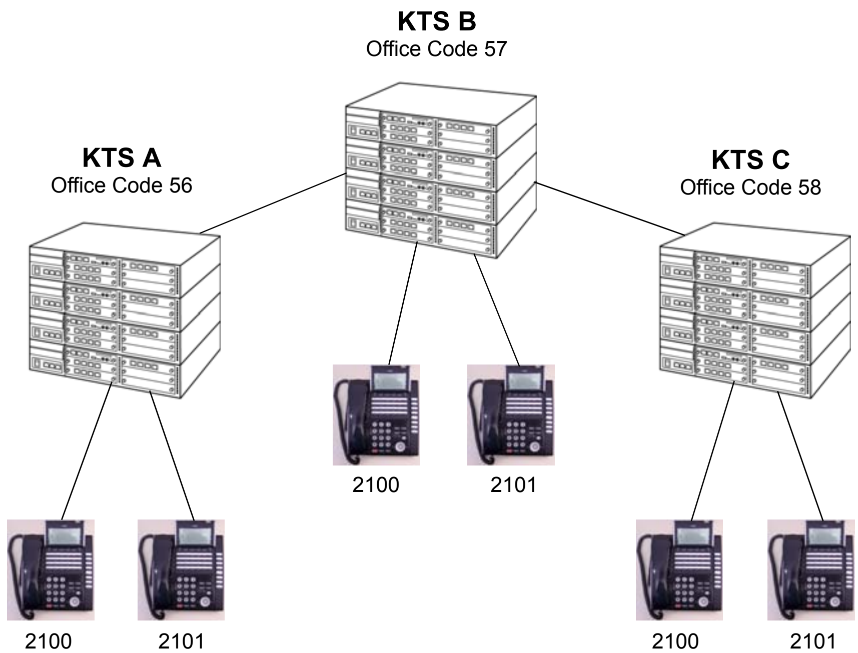
28 00

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 1-17 Open 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

Hardware Installation <US Only>

Chapter 2

SECTION 1 INSTALLATION PRECAUTIONS

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

WARNING

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:

  - GCD-CP10
  - GPZ-BS10 and GPZ-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)

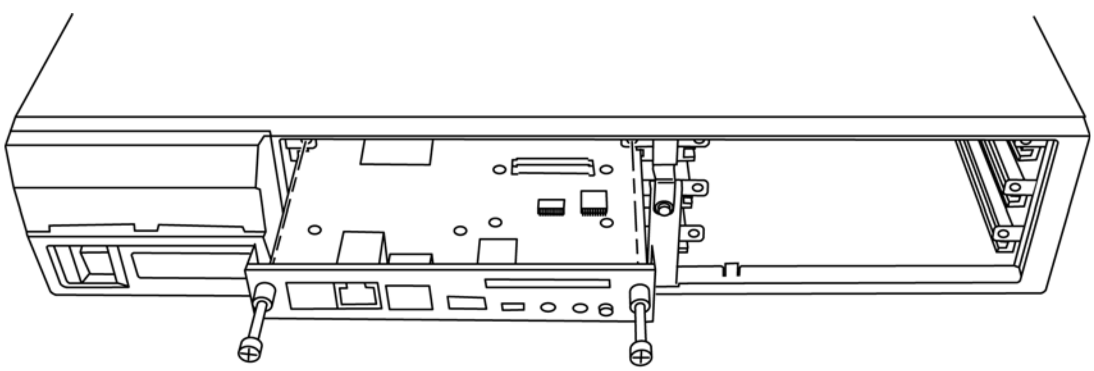
SECTION 2 INSTALLING AN EXTENSION OR TRUNK BLADE

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.

CAUTION

To avoid unexpected extension/trunk numbering if the VoIP or Voice Mail Daughter Board register with the system first, install the GCD-CP10 blade after the other types of extension and trunk blades are installed.

For example, when a digital station blade (GCD-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	GCD-16DLCA	101~116
2	2	GCD-16DLCA	117~132
3	4	GCD-8DLCA GPZ-8LCE	133~148
4	3	GCD-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., GCD-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	GCD-16DLCA	101~116	1	GCD-16DLCA	101~116
2	GCD-8DLCA (no daughter board)	117~124	2	—	—

Table 2-2 Adding Daughter Board to Chassis Example

3	GCD-16DLCA	125~140	3	GCD-16DLCA	125~140
			4	GCD-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, GCD-4ODTA, GCD-4DIOPA, or GCD-2BRIA Blades

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

CAUTION

To avoid unexpected extension/trunk numbering if the VoIP or Voice Mail blades register with the system first, install GCD-CP10 blade after the other types of extension and trunk blades are installed.

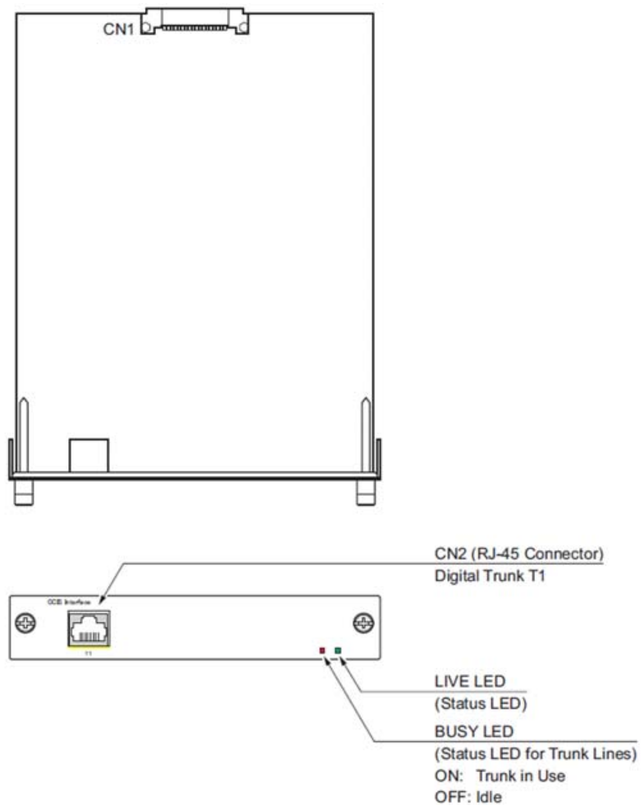
For example, if four blades are installed *in the following order,* the numbering plan below would apply.

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 GCD-CCTA (CCIS Trunk Interface)

Figure 2-2 GCD-CCTA Blade



3.1 Description

The GCD-CCTA blade is common to both UNIVERGE SV9100 and SV9300 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 GCD-CCTA (Common Channel Handler) is an optional blade that provides a common channel signal through the GCD-CCTA to a K-CCIS network and controls the signaling between the KTS and the CP10. Each GCD-CCTA blade supports one K-CCIS link. Eight GCD-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 GCD-CCTA in any universal slot.

3.3 LED Indications

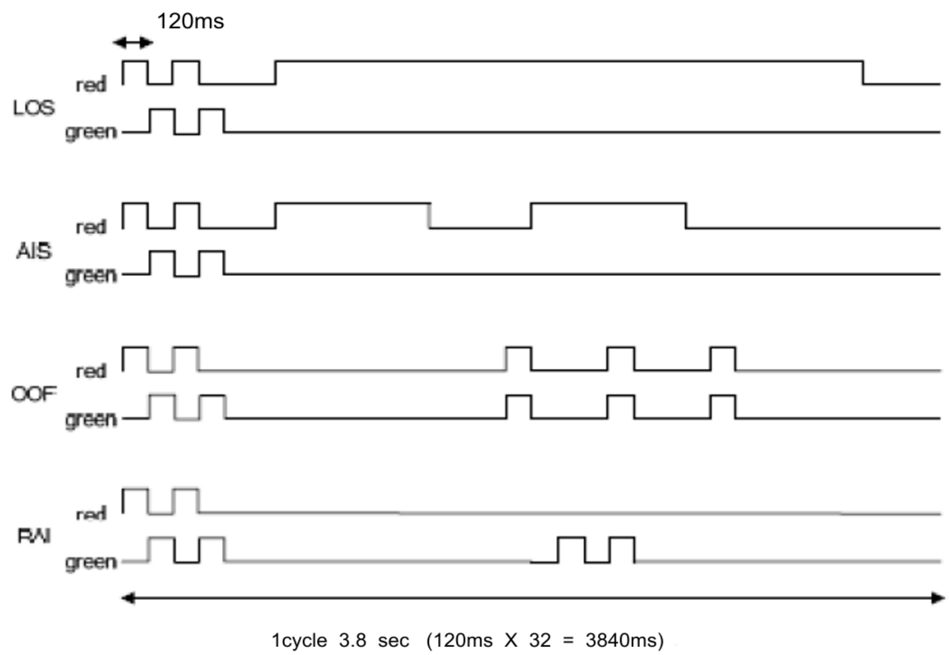
LED indications for the GCD-CCTA are listed in Table 2-4 GCD-CCTA LED Indications. Each LED is listed with its associated function and LED and Operational status. Refer to Figure 2-3 GCD-CCTA LED Indication Pattern of Layer 1 on T1 Unit on page 2-7 for LED pattern information.

Table 2-4 GCD-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.

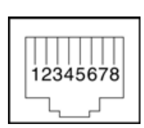
Figure 2-3 GCD-CCTA LED Indication Pattern of Layer 1 on T1 Unit



3.4 Connectors

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

Table 2-5 GCD-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 <US Only>

Chapter 3

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 GCD-CCTA blade is located, the signaling format the GCD-CCTA blade uses, and to assign other information relating to the trunks.

Table 3-1 Digital Trunk Data Programming Assignments

Program/ Item No.	Description/Selection	Assigned Data	Comments
10-03-01	Blade Setup – Logical Port Number	0 ~ 400	The GCD-CP10 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	
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 (default) 1 = External	Define the Master (Internal) or Slave (External) clock source.

Table 3-1 Digital Trunk Data Programming Assignments (Continued)

Program/ Item No.	Description/Selection	Assigned Data	Comments
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 ~ 400	Default priorities for trunks 1 ~ 400 is 1 ~ 400.
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.

Table 3-2 CCIS Programming Assignments

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)	0~400 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 programming 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.

Table 3-3 Numbering Plan Assignments

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 ~ 960: 101 ~ 199 3101 ~ 3961
11-04-01	Virtual Extension Numbering	Assign Station Numbers to Port Number	Defaults for Ports 1 ~ 99 = 201~ 299 100 ~ 512 = Not Assigned

1.4 Programming for Closed Numbering Plan

Use these programs 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.

Table 3-4 Closed Numbering Plan 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 – 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 l 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.

Table 3-4 Closed Numbering Plan Assignments (Continued)

Program/ Item No.	Description/ Selection	Assigned Data	Comments
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	0-100,101-150,255 0 = Not Set 1~100 = Trunk Group from 14-05 101~150 = Networking 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.

Table 3-5 Open Numbering Plan 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 l 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	0-100,101-150,255 0 = Not Set 1~100 = Trunk Group from 14-05 101~150 = Networking 255 = Extension Call Default is 0	Select the trunk group to be used for the outgoing K-CCIS call.

Table 3-5 Open Numbering Plan Assignments (Continued)

Program/ Item No.	Description/ Selection	Assigned Data	Comments
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.
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

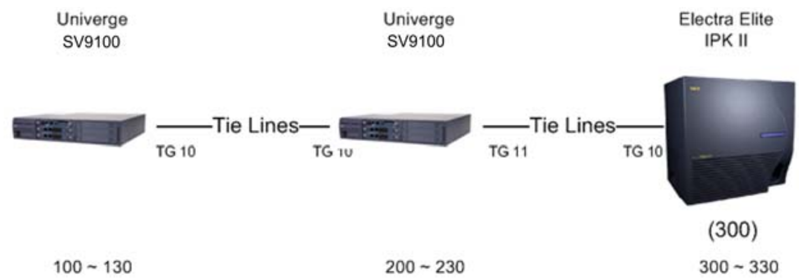
This section provides the steps needed to program a closed numbering plan.

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 SV9100 system is defaulted with the GCD-CCTA card installed.

Abbreviations used in the diagram:

MB = Memory Block	TRK = Trunk	TG = Trunk Group
CC = Clear Channel	TT = Trunk Type	

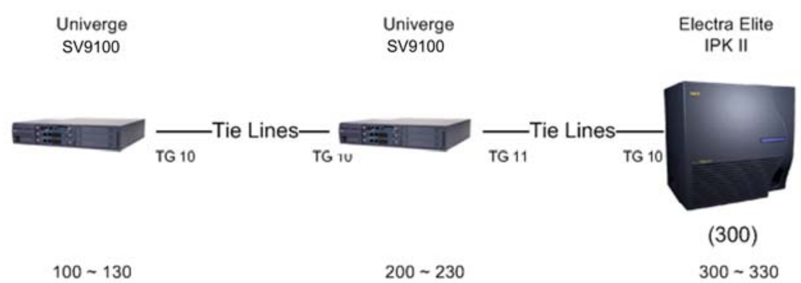


PRG 10-03-03 CC = B8ZS	**PRG 10-03-03** CC = B8ZS	**PRG 10-03-03** CC = B8ZS
PRG 10-03-05 Internal	**PRG 10-03-05** External Internal	**PRG 10-03-05** External
PRG 22-02-01 TT = E&M	**PRG 22-02-01** TT = E&M	**PRG 22-02-01** TT = E&M
PRG 14-05-01 TRK 001~024 = TG 10	**PRG 14-05-01** TRK 001~024 = TG 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	



Calling
200~230
300~330

PRG 11-01-01
Dial 1X = 3 digit; Intercom
Dial 2X = 3 digit; F-Route
Dial 3X = 3 digit; F-Route

PRG 44-02-01
TBL 1 = 2
TBL 2 = 3

PRG 44-02-02
TBL 1 = F-Route
TBL 2 = F-Route

PRG 44-02-03
TBL 1 = F-Route 1
TBL 2 = F-Route 1

PRG 44-05-01
F-Route TBL 1 = TG 10

PRG 44-05-02
F-Route TBL 1 Delete Digit = 0

PRG 44-05-03
F-Route TBL 1 Add Dial = 0

PRG 44-05-09
F-Route TBL 1 Max. Digit = 3

Calling
100~130
300~330

PRG 11-01-01
Dial 1X = 3 digit; F-Route
Dial 2X = 3 digit; Intercom
Dial 3X = 3 digit; F-Route

PRG 44-02-01
TBL 1 = 1
TBL 2 = 3

PRG 44-02-02
TBL 1 = F-Route
TBL 2 = F-Route

PRG 44-02-03
TBL 1 = F-Route 1
TBL 2 = F-Route 2

PRG 44-05-01
F-Route TBL 1 = TG 10
F-Route TBL 2 = TG 11

PRG 44-05-02
F-Route TBL 1 Delete Digit = 0
F-Route TBL 2 Delete Digit = 0

PRG 44-05-03
F-Route TBL 1 Add Dial = 0
F-Route TBL 2 Add Dial = 0

PRG 44-05-09
F-Route TBL 1 Max. Digit = 3
F-Route TBL 2 Max. Digit = 3

Calling
100~130
200~230

PRG 11-01-01
Dial 1X = 3 digit; F-Route
Dial 2X = 3 digit; F-Route
Dial 3X = 3 digit; Intercom

PRG 44-02-01
TBL 1 = 1
TBL 2 = 2

PRG 44-02-02
TBL 1 = F-Route
TBL 2 = F-Route

PRG 44-02-03
TBL 1 = F-Route 1
TBL 2 = F-Route 1

PRG 44-05-01
F-Route TBL 1 = TG 10

PRG 44-05-02
F-Route TBL 1 Delete Digit = 0

PRG 44-05-03
F-Route TBL 1 Add Dial = 0

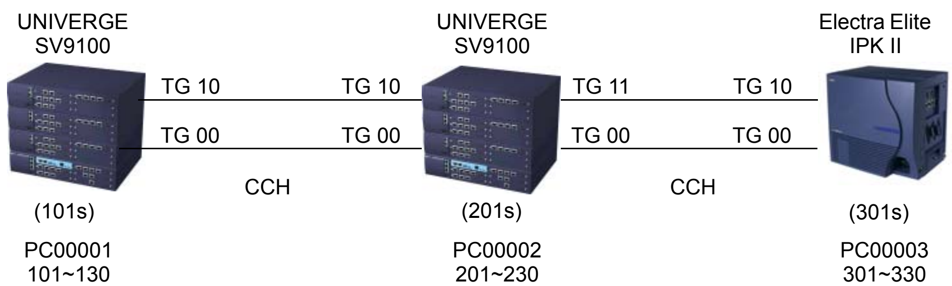
PRG 44-05-09
F-Route TBL 1 Max. Digit = 3

✎ ***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	



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
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 001~024 = 10
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 2
CCIS Route ID2 = Org. Point Code 2

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
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

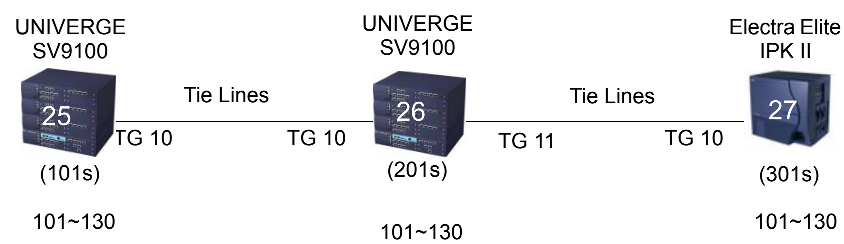
This sections provides the steps needed to program an open numbering plan.

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 SV9100 and Electra Elite IPK II system is defaulted with the GCD-CCTA blades installed.

Abbreviations used in the diagram:

MB = Memory Block	TRK = Trunk	TG = Trunk Group
CC = Clear Channel	TT = Trunk Type	



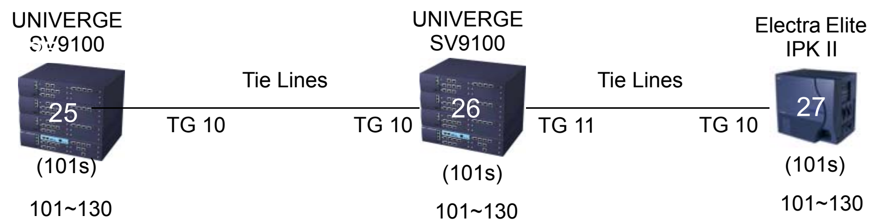
PRG 10-03-05 Internal	**PRG 10-03-05** External Internal	**PRG 10-03-05** External
PRG 10-03-03 CC = B8ZS	**PRG 10-03-03** CC = B8ZS	**PRG 10-03-03** CC = B8ZS
PRG 22-02-01 TRK 001~024 = Tie	**PRG 22-02-01** TRK 001~024 = Tie TRK 025~048 = Tie	**PRG 22-02-01** TRK 001~024 = Tie
PRG 14-05-01 TRK 001~024 = 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
101 ~ 130

PRG 11-01-01
Dial 1X = 3 Digits; Intercom
Dial 8X = 1 Digit, F-Route Access

PRG 11-02-01
Port 1 = 101

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

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

PRG 44-02-03
TBL1 = F-Route TBL1
TBL2 = F-Route TBL2
TBL3 = F-Route TBL2

PRG 44-05-01
TBL1 = TG 255
TBL2 = TG 10

Calling
101 ~ 130

PRG 11-01-01
Dial 1X = 3 Digits; Intercom
Dial 8X = 1 Digit, F-Route Access

PRG 11-02-01
Port 1 = 101

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

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

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

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

Calling
101 ~ 130

PRG 11-01-01
Dial 1X = 3 Digits; Intercom
Dial 8X = 1 Digit, F-Route

PRG 11-02-01
Port 1 = 101

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

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

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

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

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

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

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

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

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

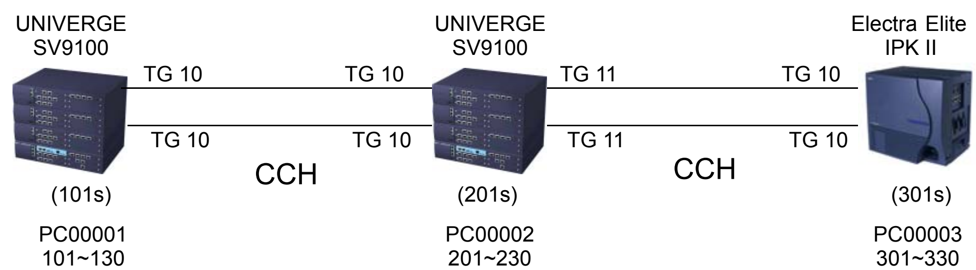
PRG 44-05-09
F-Route TBL2; Max Digit = 6

✎ *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 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 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 1

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

PRG 50-03-01
N/A

PRG 50-04-01
825

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

PRG 50-02-03
CCIS Route ID1 = Org. Point Code 2
CCIS Route ID2 = Org. Point Code 2

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

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

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

PRG 50-04-01
826

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 50-03-01
N/A

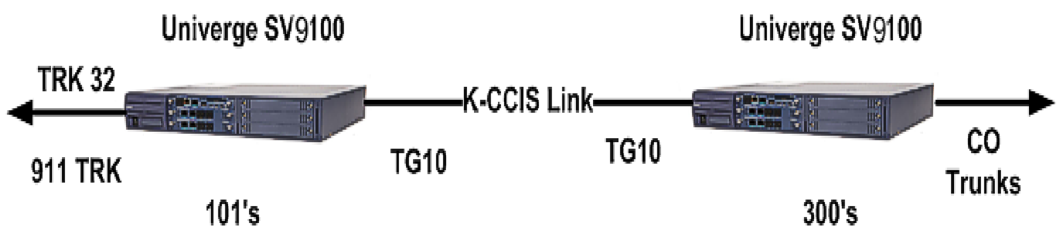
PRG 50-04-01
827

1.8 Dedicated Tandem CO Trunk Calls

The following diagram provides an example of SV9100 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 SV9100 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 11-09-01
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 11-01-01
Dial 9 = 1 Digit; Trunk Access
Dial 3 = 3 Digit; Intercom
Dial 0 = 1 Digit; Operator Access

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

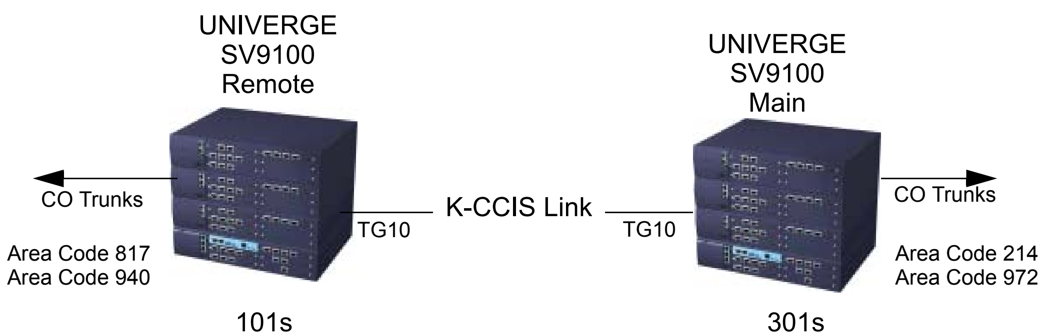
PRG 26-03-01
ARS Treatment TBL1 = D019RE

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 SV9100 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



PRG 11-01-01
Dial 9 = 1 digit: Trunk Access

PRG 11-09-01
Trunk Access Code = 9

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

PRG 26-01-01
ARS Service = On

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

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

PRG 11-01-01
Dial 9 = 1 digit: Trunk Access

PRG 11-09-01
Trunk Access Code = 9

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

PRG 26-01-01
ARS Service = On

PRG 26-02-01
ARS Analysis TBL1 = 1817
ARS Analysis TBL2 = 1940
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

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

Chapter 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 GCD-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:

- ➥ Hot Line – K-CCIS on page 4-76
- ➥ Multiple Call Forwarding – All Calls – K-CCIS on page 4-82
- ➥ Multiple Call Forwarding – Busy/No Answer - K-CCIS on page 4-87
- ➥ Paging Access – K-CCIS on page 4-92
- ➥ Quick Transfer to Voice Mail – K-CCIS on page 4-100
- ➥ Station-to-Station Calling – K-CCIS on page 4-103
- ➥ Uniform Numbering Plan – K-CCIS on page 4-105
- ➥ Voice Call – K -CCIS on page 4-107
- ➥ Voice Mail Integration – K-CCIS on page 4-109

SECTION 2 SYSTEM AVAILABILITY

2.1 Terminal Type

All stations

2.2 Required Components

GCD-CCTA

- OR -

GPZ-IPLE

2.3 Operating Procedures

Normal call handling procedures apply.

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

Chassis Software	GCD-CCTA
SV9100 V1.00	X

X = Compatible
– = Not compatible

Section 3 Service Conditions

General:

- Each UNIVERGE SV9100 system can have up to eight K-CCIS routes.
- One GCD-CCTA is required to support each K-CCIS link. A maximum of eight K-CCIS links are supported.
- 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 SV9100 uses the ARS/F-Route Tables.
- The UNIVERGE SV9100 uses the F-Route Tables to assign an Open Numbering Plan.
- When all K-CCIS voice channels are busy, the UNIVERGE SV9100 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.

Restrictions:

- The UNIVERGE SV9100 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 SV9100 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 SV9100 tandem system

are passed through and supported.

- ❑ An UNIVERGE SV9100 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 SV9100 can have a maximum of 255 codes assigned to distant systems.
- ❑ Using an UNIVERGE SV9100-to-UNIVERGE SV9100 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 SV9100 system in a K-CCIS network must have at least one trunk for Emergency 911 calls.
- ❑ Using a NEAX-to-UNIVERGE SV9100 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 GCD-4ODTA Analog Line interface.
- ❑ Eight GCD-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:

GCD-CCTA

- OR -

GPZ-IPLE

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 SV9100 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.

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:

GCD-CCTA

- OR -

GPZ-IPLE

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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All multiline terminals

Required Components

GCD-CCTA

- OR -

GPZ-IPLE

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 9).
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 0).
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 DT800/DT700 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.(default)
20-09-07	Class of Service Options (Incoming Call Service) – Call Queuing	0 = Off 1 = On Default is 0	Must be Off for Call Forward – Busy to operate.

Program/ Item No.	Description/ Selection	Assigned Data	Comments
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 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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All multiline terminals

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

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 (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.(default)
20-09-07	Class of Service Options (Incoming Call Service) – Call Queuing	0 = Off 1 = On Default is 0	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.

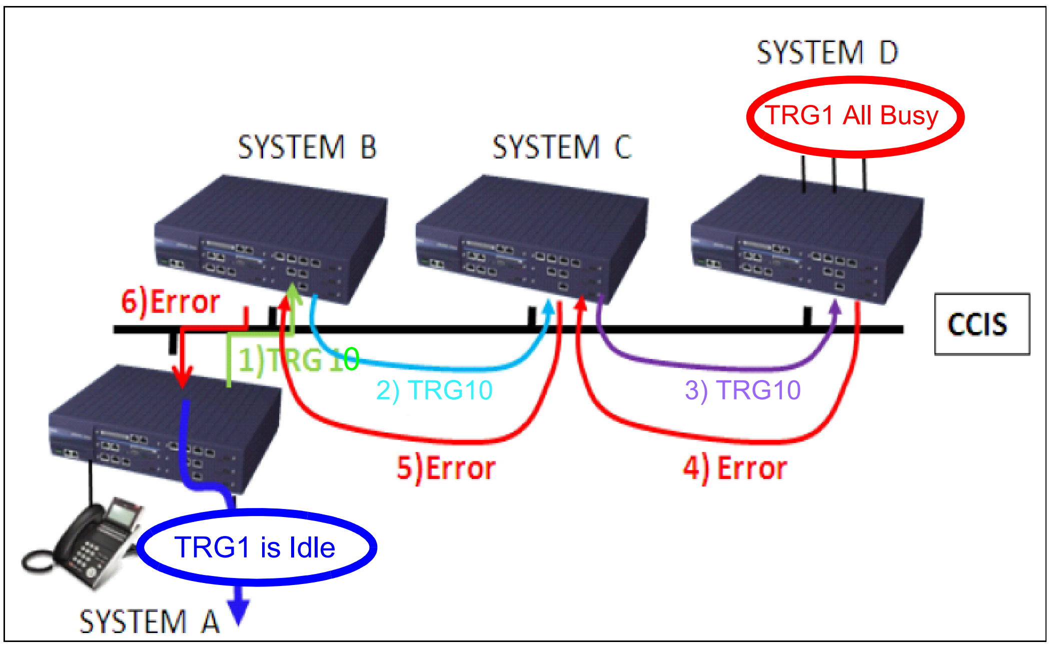
Program/ Item No.	Description/ Selection	Assigned Data	Comments
15-07-01	Programmable Function Keys	10 = Call Forward – Immediate 11 = Call Forward – Busy 12 = Call Forward – No Answer 13 = Call Forward – Busy No Answer	Service Codes: 741 742 743 744
14-01-13	Basic Trunk Data Setup – Trunk-to-Trunk Transfer 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	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.

K-CCIS Call Rerouting

Feature Description

The CCIS Call Rerouting feature allows a system to use multiple call routing priorities when remote system trunks are all busy. The four priorities can be local or remote trunks. For example using ARS and F-Route table priorities the system can try up to four remote systems or up to three remote systems and a local trunk to route the outbound call to its destination. If an outbound route is unavailable for any reason the call will fall through to the next priority.

Figure 4-1 K-CCIS Call Rerouting



System Availability

Terminal Type:

All Terminals

Required Components:

GCD-CP10

GPZ-IPLE

K-CCIS License (0016)

K-CCISoIP License (5012)

SERVICE CONDITIONS

General:

- ❑ The originating system must have a dial treatment of D019RE where 9 is the ARS trunk access code in the destination system for this feature to work.
- ❑ ARS must be enabled in all systems for this feature to work.
- ❑ The CCIS Call Rerouting feature is not supported on stations using 3rd party CTI.
- ❑ The CCIS Call Rerouting feature is not supported on a station which is controlled by 1st party CTI.
- ❑ The CCIS Call Rerouting feature will not work if Line Load Control has been triggered on the destination system.
- ❑ The CCIS Call Rerouting feature is only supported on CCISoIP with SV9100 systems.
- ❑ The CCIS Call Rerouting feature can only use the four routing options contained in one F-Route table.
- ❑ The CCIS Call Rerouting feature cannot fall through from one F-Route table to a second F-Route table.
- ❑ If none of the four F-Route table route priorities are available the outbound call will fail.

Programming Example

The following example will use the first two priorities of System A to route 10 digit local calls out trunk group one of System B and if that fails the call is routed out trunk group one of System A.

This example assumes the following:

- System A has local trunks 1-8 in trunk group 1.
- Both systems are already configured for the LAN on which they reside and have a network route to each other.
- Both systems are configured for K-CCISoIP.
- System A is assigned point code 1 and 1XX extensions.
- System B is assigned point code 2 and 2XX extensions.
- While dial analysis can be configured in all systems, for this example only System A has ARS dial analysis configured.

Table 4-1 Programming Example

Program	Description	System A Setting	System B Setting
10-68-01	IP Trunk Mode CCIS	3	3
10-68-02	Starting CCIS Trunk Number	9	9
10-68-03	Number of CCIS Trunks to Create. This will create CCISoIP trunks 9-16.	8	8
14-05-01	First setting data. Put CCISoIP trunks in trunk group 10.	10	10
14-05-01	Second setting data. Set priorities 1-8 for trunks 9-16.	1~8	1~8
26-01-01	Enable ARS Services	1	1
26-02-01	For table entry 1 enter 5 wildcard characters.	@@@@@	N/A
26-02-02	For table entry 1 set service type to F-Route (2)	2	N/A
26-02-03	For table entry 1 set additional data to F-Route Table 2.	2	N/A
26-03-01	Set ARS Treatment 1 to sent the number including the trunk access code of 9 to System B.	D019RE	N/A
44-02-01	Closed numbering setup - Set to first digit of remote system extension number	2	1

Table 4-1 Programming Example (Continued)

Program	Description	System A Setting	System B Setting
44-02-02	Closed numbering setup – Set to F-Route table (2).	2	2
44-02-03	Closed numbering setup – Set to use F-Route table 1 for intercom calls.	1	1
44-05-01	Closed numbering setup – Set table 1 first priority to use CCIS trunk group 10.	10	10
44-05-09	Closed numbering setup – Set table 1 first priority maximum digit length to 3.	3	3
44-05-10	Closed numbering setup – Set table 1 first priority to remote system point code.	2	1
44-05-01	ARS Routing – Set table 2 first priority to CCIS trunk group 10.	10	N/A
44-05-08	ARS Routing – Set table 2 first priority to dial treatment 1	1	N/A
44-05-09	ARS Routing – Set table 2 first priority maximum dialing digit 11.	11	N/A
44-05-10	ARS Routing – Set table 2 first priority to System B point code (2).	2	N/A
44-05-01	ARS Routing – Set table 2 second priority to local trunk group 1.	1	N/A
44-05-09	ARS Routing – Set table 2 second priority maximum dialing digit 10.	10	N/A

Guide to Feature Programming

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

Program Number	Program Name/Description	Input Data	Default
10-12-01	**GCD-CP10 Network Setup - IP Address** Should be set to 0.0.0.0 when using PRG 10-12-09.	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 = 192.168.0.10
10-12-03	**GCD-CP10 Network Setup – Default Gateway** Assign the default gateway IP address for the GCD-CP10.	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
10-12-09	**GCD-CP10 Network Setup – IP Address** Set IP Address for GPZ-IPLE.	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 = 172.16.0.10
10-19-01	**VoIP DSP Resource Selection** Select type of GPZ-IPLE DSP Resource. This program setting has no affect on the terminal/ trunk port assignment or usage.	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 ~ 256 = 0
10-50-01	**License Information – License Name** Read Only program used to confirm license information that is stored in a system.	N/A	
10-54-01	**License Configuration for Each Package – License Code** Assign VoIP resource licenses (5103) to the GCD-CP10 slot (1).	1 ~ 255 resource licenses	No Setting
10-68-01	**IP Trunk Availability – Trunk Type** Set trunk type to CCISoIP for interconnecting trunks.	0 = None 1 = SIP 2 = H.323 3 = CCIS	Default = 0
10-68-02	**IP Trunk Availability – Start Port** Set the trunk port number to start the assignment from.	0 ~ 400	Default = 0
10-68-03	**IP Trunk Availability – Number of Ports** Set the number of ports to assign from the starting point set in 10-68-02.	0 ~ 400	Default = 0
11-01-01	**System Numbering – Service Code** The Call reroute feature only supports closed numbering. This program is used to customize the system internal (Intercom) numbering plan.	Refer to the Programming Manual for default values.	

Program Number	Program Name/Description	Input Data	Default
11-02-01	**Extension Numbering** Assign extension numbers to extension ports. The telephone programming identity follows the port number – not the extension number.	Maximum of eight digits.	1 101 2 102 ~ ~ 99 199 100 3101 ~ ~ 199 3200 200 3201 ~ ~ 960 3513
14-01-38	**Trunk Basic Setup – Outgoing CLI Selection** Read Only program used to confirm the trunk type.	0: None 1: SIP 2: H.323 3: CCIS	
14-05-01	**Trunk Group – Trunk Group Number** Assign CCISoIP trunks to trunk groups (1~100).	Trunk Group Number: 0~100 Priority Number: 1~400	(default = Trunk Group 1, with priority in ascending order.)
20-09-08	**Class of Service Options (Incoming Call Service) – Calling Party Information** Turn Off or On an extension user ability to have calls queued if a call rings the extension when it is busy.	0 = Off 1 = On	Default COS 1 ~ 15 = 0
20-14-01	**Class of Service Options for DISA/E&M – First Digit Absorption (Delete First Digit Dialed)** This feature must be enabled in the system where the outgoing trunks reside for the Class of Service (COS) sent be the originating system, at default this is COS 1. Turn Off or On a DISA or tie trunk caller ability to dial 9 for Trunk Group Routing or Automatic Route Selection.	0 = Off 1 = On	Default COS 1 ~ 15 = 0
22-02-01	**Incoming Call Trunk Setup** Set the CCISoIP trunks to 5 (Tie Line).	Trunks 1 ~ 400 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

Program Number	Program Name/Description	Input Data	Default
26-01-01	**Automatic Route Selection Service – ARS Service** ARS must be enabled in all system for this feature to work.	0 = Disabled (ARS service is Off) 1 = Enabled (ARS service is On)	Default = 0
26-01-06	**Automatic Route Selection Service – Class of Service Match Access** With the ARS Class of Service Match Access feature, you can determine whether or not the system should allow a call based on the COS assigned to the Dial Analysis Table (Program 26-02). This change can be used to create a tenant-like application. It then uses the trunk group set in the Additional Entry in Program 26-02-03 to place the out-going call. When this feature is enabled, the calls are routed in sequential order, and forward – provided the Class of Service for the trunk groups match.	0 = Disable (Off) 1 = Enable (On)	Default = 0
26-01-07	**Automatic Route Selection Service – F-Route Access COS Reference** Define where the system looks for the Automatic Route Selection (ARS) class of service.	0 = F-Route 1 = ARS	Default = 0
26-02-01	**Dial Analysis Table for ARS/LCR – Dial** Enter the digits (16 digits maximum: 1~9, 0, #, @; 800 separate entries) for the Dial Analysis Table which is analyzed by ARS/LCR. This table is checked after any programmed F-Route operations have completed. The system then refers to Program 26-02-02 and Program 26-02-03 to determine the routing for the call. To enter a wild card/don't care digit, press Line Key 1 to enter an @ symbol. It is important to remember that the system checks the table numbers in numerical order. This means that entries for specific numbers should be entered first (such as your local area codes), then enter the items containing wild card digits. If the system sees an entry of 2@@, any table entries which follow are ignored. For example, if 268, 269, and 270 are local exchanges, these would be the first three table entries which route according to the settings made in Program 26-02-02 and Program 26-02-03 for each of the table entries. If the next entry is 2@@, the system checks no further in this program and routes all other 2xx numbers according to the entries made in Program 26-02-02 and Program 26-02-03 for this table entry.	Dial a maximum of 16 digits (0 ~ 9, #, @)	Default = No Setting

Program Number	Program Name/Description	Input Data	Default
26-02-02	**Dial Analysis Table for ARS – ARS Service Type** For each Dial Analysis Table used (1~2000), select Service Type 2 – F-Route Selected to have the dialed number controlled by the F-Route table. If Service Type 2 is selected and F-Route operation is on, the F-Route table used is determined by Program 44-04.	0 = No Service (Call Restricted) 1 = Route to Trunk Group 2 = Select F-Route Access	Default = 0
26-02-03	**Dial Analysis Table for ARS – Additional Data** For each Dial Analysis Table (1~2000), enter the F-Route table to use (1~500)	F-Route Table 0~500	Default = 0
26-02-04	**Dial Analysis Table for ARS – ARS Class of Service** For each Dial Analysis Table (1 ~ 2000), set the Automatic Route Selection (ARS) Class of Service (0 ~ 50).	Class = 0 ~ 50	Default = 0
26-03-01	**ARS Dial Treatments – Treatment Code** For the originating system a treatment code must be used for any route that will use outbound trunks in a remote system. The recommended treatment code is D019RE where 9 is the trunk access code in the destination system.	Maximum of 24 characters	Default = Blank
26-04-01	**ARS Class of Service** Set class of service for ARS. Used when 26-01-07 is set to ARS.	Class = 0 ~ 50	Default = 0
34-02-01	**E&M Tie Line Class of Service** This program can be used to assign the Tie Line Class of Service (1 ~ 15) for inbound trunks. Use Program 20-14-01 to set the Tie Line Class of Service options. You cannot use Program 20-06 to assign Class of Service to Tie Lines.	Day/Night Mode 1 ~ 8 Class: 1 ~ 15	Default = 1
44-02-01	**Dial Analysis Table for ARS/F-Route Access – Dial** Set the Dial digits for the Pre-Transaction Table for selecting ARS/F-Route (eight digits maximum: 1 ~ 9, 0, #, @). To enter a wild card/don't care digit, press Line Key 1 to enter an @ symbol.	Up to eight digits max.	Default = No Setting
44-02-02	**Dial Analysis Table for ARS/F-Route Access – Service Type** Set the Service Type (0 ~ 3) for the Pre-Transaction Table for selecting ARS/F-Route.	0 = No Setting (None) 1 = Extension Call (Own) 2 = ARS/F-Route Table (F-Route) 3 = Dial Extension Analyze Table (Option)	Default = 0

Program Number	Program Name/Description	Input Data	Default
44-02-03	**Dial Analysis Table for ARS/F-Route Access – Additional Data** If a Service Type is set to F-Route in Program 44-02-02, set which F-Route table to use.	1 = Delete Digit = 0 ~ 255 (255: Delete All Digits) 2 = 0 ~ 500 (0 = No Setting) 3 = Dial Extension Analyze Table Number = 0 ~ 4 (0 = No Setting)	Default = 0
44-05-01	**ARS/F-Route Table – Trunk Group Number** Select the trunk group number to be used for the outgoing ARS call. For this feature you would use trunk groups for the CCIS connections.	0-100,101-150,255 0 = Not Set 1~100 = Trunk Group from 14-05 101~150 = Networking 255 = Extension Call	Default = 0
44-05-04	**ARS/F-Route Table – Beep Tone** Enable this option if the customer wants to hear an audible beep every time a F-Route priority is used.	0 = Off (No Beep) 1 = On (Beep)	Default = 0
44-05-07	**ARS/F-Route Table – ARS Class of Service** For each ARS/F-Route table (1 ~ 500) and each priority number (1 ~ 4) you can assign a Class of Service to be used for ARS calls. This COS is sent to the destination system. Extension ARS COS is determined in Program 26-04-01.	0 ~ 50	Default = 0
44-05-08	**ARS/F-Route Table – Dial Treatment** For each ARS/F-Route table (1 ~ 500) and priority number (1 ~ 4) you must assign a dial treatment to use. The recommended treatment code is D019RE where 9 is the trunk access code in the destination system. The Dial Treatments are defined in Program 26-03-01.	0 ~ 15	Default = 0
44-05-09	**ARS/F-Route Table – Maximum Digit** Set the maximum number of digits to send when using the F-Route.	0 ~ 24	Default = 0
44-05-10	**ARS/F-Route – CCIS over IP Destination Point Code** For each ARS/F-Route table (1 ~ 500). Set the CCIS over IP Destination Point Code (0 ~ 16367).	0 ~ 16367	Default = 0
50-01-01	**CCIS System Setting – CCIS Availability** Any CCIS settings lose functionality if this setting is set to 0.	0 = Disable 1 = Enable	Default = 0
50-02-03	**Connecting System Settings – Originating Point Code** For Route ID 9 set the Origination point code.	0 ~ 16367	Default = 0

Program Number	Program Name/Description	Input Data	Default
50-03-01	**CCIS Destination System Settings – Destination Point Code** Assign the destination transfer point code for Tandem KTS.	0 ~ 16367	Default = 0
50-03-03	**CCIS Destination System Settings – IP Address (IP only)** Assign remote system IP network information	xxx.xxx.xxx.xxx (xxx = 0 ~ 255) Default is 0.0.0.0	Default = 0.0.0.0
84-26-01	**IPLE Basic Setup – IP Address** Assign the IPLE IP Address.	XXX.XXX.XXX.XXX	Default: Slot 1 = 172.16.0.20
84-26-02	**IPLE Basic Setup – RTP Port Number** If needed the RTP port number can be set here.	0 ~ 65534	Default = 10020
84-26-03	**IPLE Basic Setup – RTCP Port Number** If needed the RTCP port number can be set here. (RTP Port Number + 1)	0 ~ 65534	Default = 10021

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:

GCD-CCTA

- OR -

GPZ-IPLE

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 SV9100, but the station connected to the UNIVERGE SV9100 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 SV9100 is connected to the Electra Elite IPK II, the maximum digits assignment in the UNIVERGE SV9100 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.

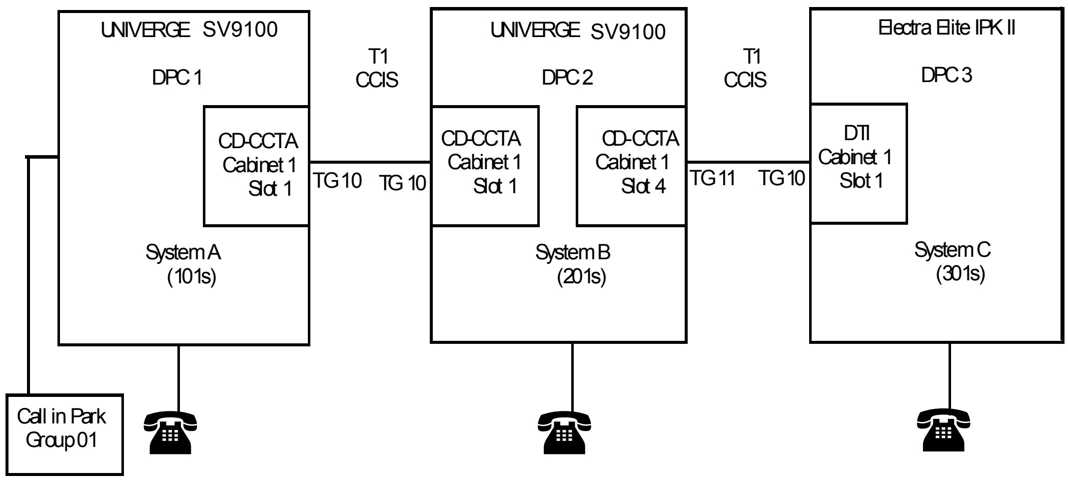
Remote System (Call Park Retrieve)

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.
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.

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 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	0 = Not Set 1 ~ 100 = Trunk Group from 14-05 101 ~ 150 Networking 255 = Extension Call Default is 0	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	Notes
System A	501+05	501 = System A Call Park Retrieve Access Code 05 = Park Orbit Number
System B	502+01+05	502 = System B Call Park Retrieve Access Code 01 = Park Hold Group Number 05 = Park Orbit Number
System C	503+01+05	503 = System C Call Park Retrieve Access Code 01 = Park Hold Group Number 05 = Park Orbit Number

System A (101s)	System B (201s)	System C (301s)
PRG 11-01-01 50 = 2 digits; F-Route	**PRG 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)

System A (101s)	System B (201s)	System C (301s)
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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All multiline terminals

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

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 SV9100 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 0	To enable receiving Call Queuing.
20-06-01	Class of Service for Extensions	0~ 15	Ext.101 is in Class 15. All others are in Class 1.(default)
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.

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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All multiline terminals

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

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 SV9100 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 SV9100 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 1	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 1	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	Ext.101 is in Class 15. All others are in Class 1.(default)
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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All multiline terminals

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

Operating Procedures

Normal call handling procedures apply.

Service Conditions

General:

- ❑ Both the caller/calling station number and name can be displayed on an UNIVERGE SV9100 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 SV9100 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 1	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	Ext.101 is in Class 15. All others are in Class 1.(default)
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:

GCD-CCTA

- OR -

GPZ-IPLE

OPERATING PROCEDURES

Placing a call with CPN:

1. Lift the handset or press Speaker.
2. Dial 0 + 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 ***0*** 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 1	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.

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 SV9100 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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All multiline terminals

Required Components

GCD-CCTA

- OR -

GPZ-IPLE

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 SV9100.
- The information storage capacity of the local UNIVERGE SV9100 office is approximately 300 calls. If the K-CCIS link is down due to network trouble, the billing information is stored by the GCD-CP10. When the maximum calls exceed this amount, the oldest call information is overwritten by the latest (newest) call.
- With the UNIVERGE SV9100, 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 SV9100 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 1	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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All multiline terminals

Attendant Add-On Console

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

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 SV9100 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 SV9100 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.	Identifies 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.
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.

Program/ Item No.	Description/Selection	Assigned Data	Comments
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	Ext.101 is in Class 15. All others are in Class 1.(default)
30-03-01	DSS Console Key Assignment	1~114	0~99 (General Functional) ✱00~✱99 (Appearance Functional Level) 95 = Page Switching

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 SV9100 system is connected to another UNIVERGE SV9100 system, the main office *can* control remote offices.

When connected to a NEAX2400, the UNIVERGE SV9100 can be used only as a remote office.

For more details, refer to the UNIVERGE SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All multiline terminals

Required Components

GCD-CCTA

- OR -

GPZ-IPLE

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	5	Day 2 mode
2	Night 1 mode	6	Night 2 mode
3	Midnight 1 mode	7	Midnight 2 mode
4	Rest 1 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 SV9100 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 SV9100 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.

Program/ Item No.	Description/Selection	Assigned Data	Comments
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.

Program/ Item No.	Description/Selection	Assigned Data	Comments
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.
20-06-01	Class of Service for Extensions	0~15	Ext.101 is in Class 15. All others are in Class 1.(default)

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

GCD-CCTA

- OR -

GPZ-IPLE

Operating Procedures

To use this feature at any terminal:

1. Lift the handset, and wait for internal dial tone.
   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 4-41.

❑ 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.

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 SV9100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

OPERATING PROCEDURES

To call an Attendant Position:

1. Lift the handset or press Speaker.
2. Dial 9 or the operator call code if it is different from 9.

SERVICE CONDITIONS

General:

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

❑ When calling to a UNIVERGE SV9100 Attendant Position, the SV9100 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 SV9100 Attendant Position, the operator office code is included with the name.

❑ When making a call from a UNIVERGE SV9100 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 SV9100 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 SV9100 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 SV9100 station, the UNIVERGE SV9100 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 SV9100 Programming Manual for all options and default settings.

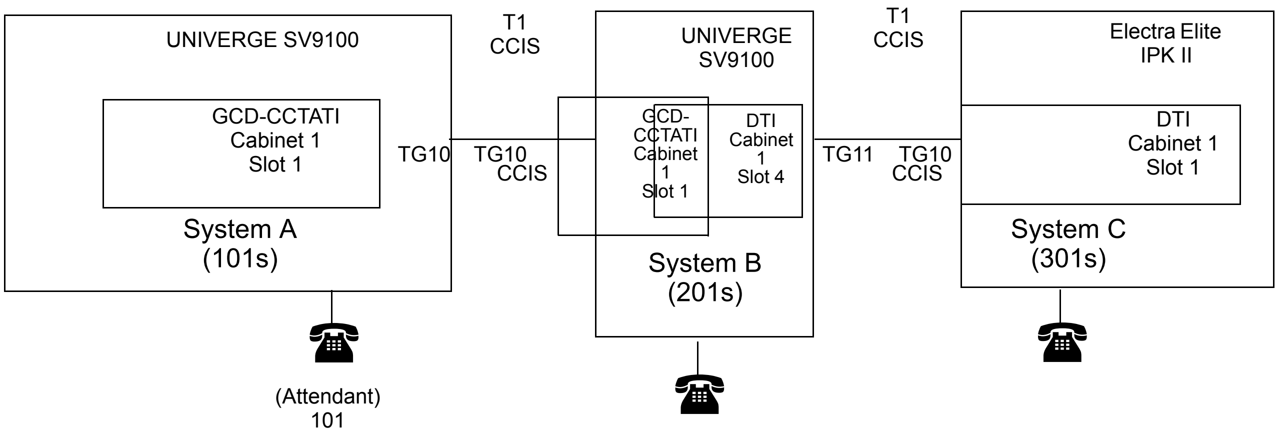
Program/ Item No.	Description/Selection	Assigned Data	Comments
20-17-01	Operator Extension – Operator Extension Number	Up to eight digits Default is 101.	Define extension numbers that are used as operators. Assign only in KTS-to-KTS network.

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 SV9100 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.
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	0-100,101-150,255 0 = Not Set 1~100 = Trunk Group from 14-05 101~150 = Networking 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.

Program/ Item No.	Description/Selection	Assigned Data	Comments
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

System A (101s)	System B (201s)	System C (301s)
	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.

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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All Stations

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

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 SV9100 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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All multiline terminals

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

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 SV9100 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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All Display multiline terminals

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

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

Feature Description

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

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

For more details, refer to the UNIVERGE SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All Stations

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

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 SV9100 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 SV9100 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 ~ 960 = 3101 ~ 3961	Assign up to eight digits for Extension Numbers.

Program/ Item No.	Description/Selection	Assigned Data	Comments
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.	

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 SV9100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All multiline terminals

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

Operating Procedures

To execute at any station programmed for Hot Line:

1. Lift the handset or press Speaker.
   The remote K-CCIS station is called.

Service Conditions

General:

- ❑ Any multiline terminal (a maximum number of 896 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: Ext.101 is in Class 15. All others are in 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 0	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.

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

GPZ-IPLE

Operating Procedures

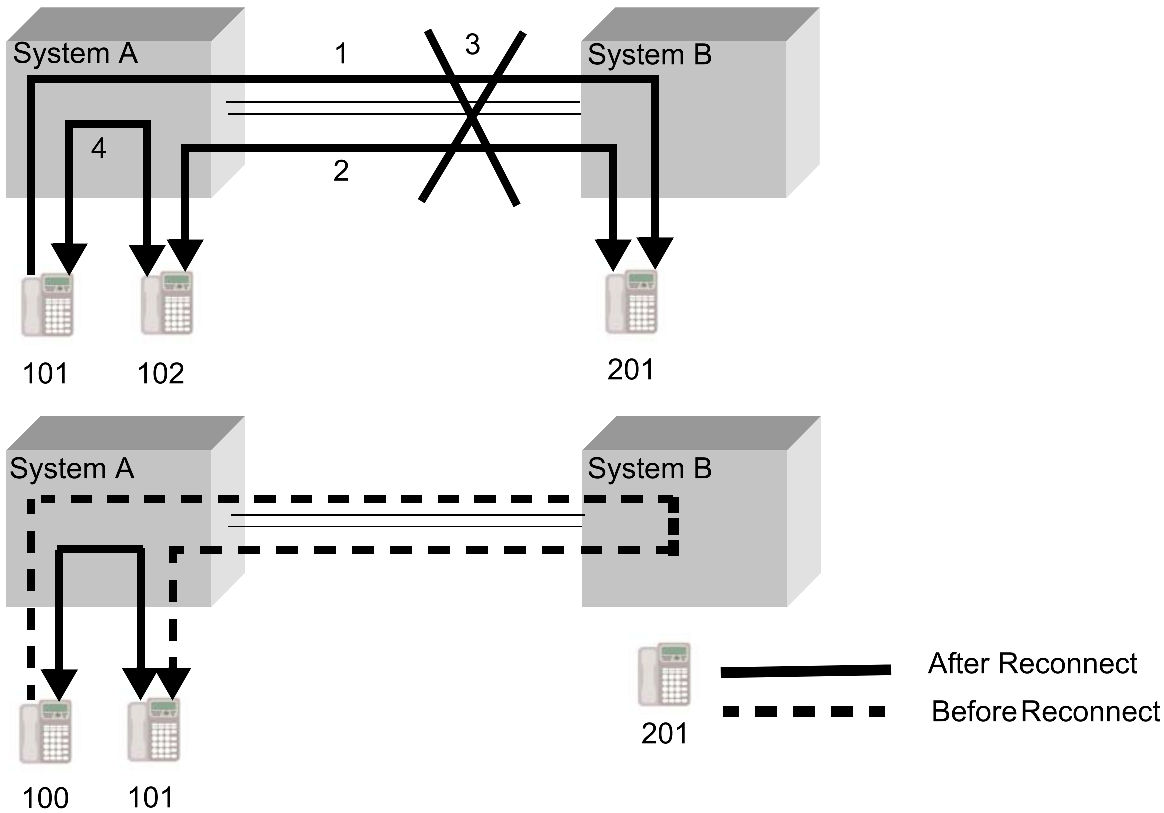
Normal call handling procedures apply.

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.)

Figure 4-1 Link Reconnect for Station Calls

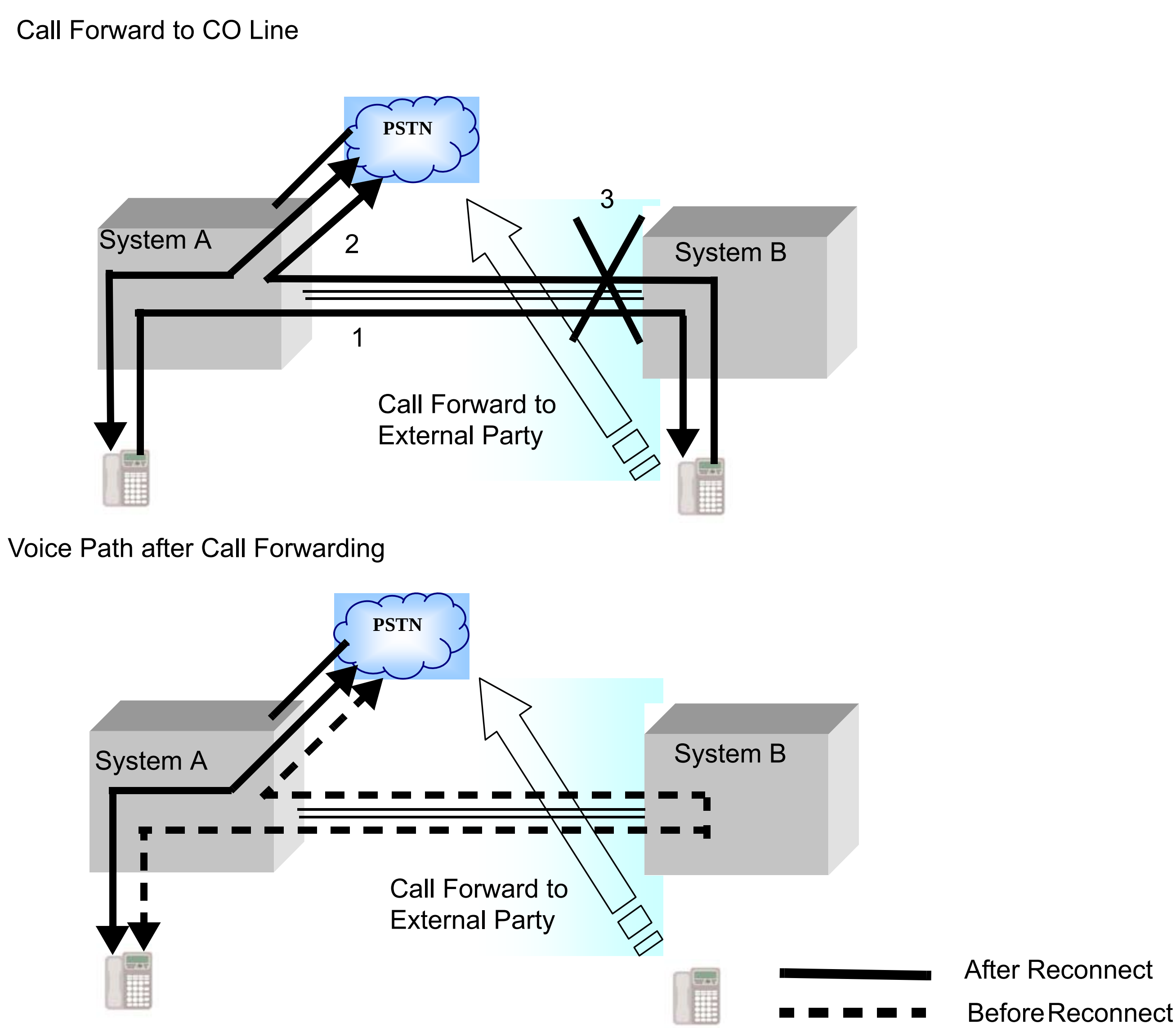


Stations 101, 102, and 201 must be multiline terminals.

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

❑ 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.)

Figure 4-2 Link Reconnect for Trunk Tandem Calls



- ❑ 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.

❑ When Connecting a SV9100 to a NEAX PBX, Link reconnect needs to be turned off in the PBX to the SV9100.

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 10	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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All multiline terminals

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

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.

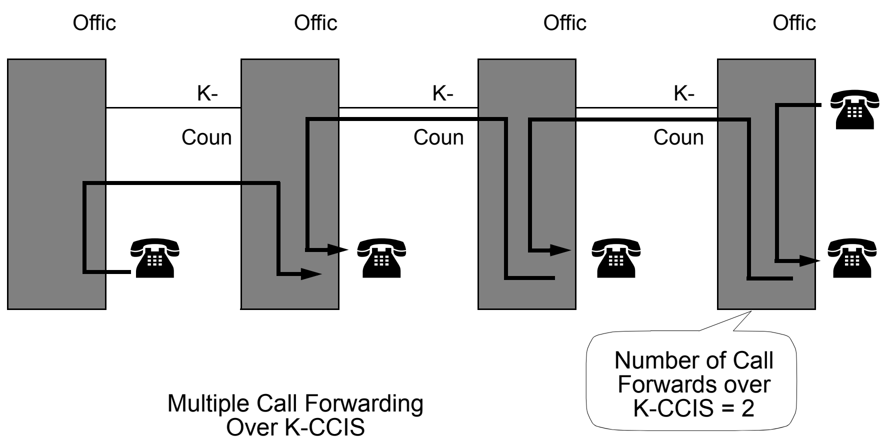
SERVICE CONDITIONS

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



	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).*

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: 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 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.

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 SV9100 Features and Specifications Manual.

SYSTEM AVAILABILITY

Terminal Type:

All multiline terminals

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

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:

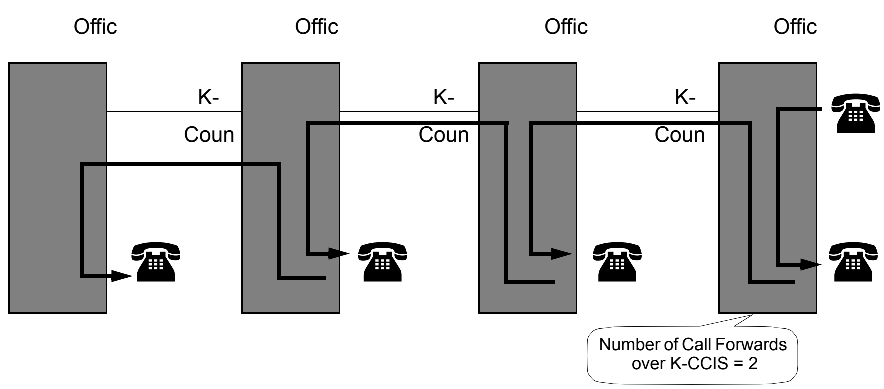
- ❑ Multiple Call Forwarding – Busy/No Answer Calls - K-CCIS can forward a call up to seven times across K-CCIS links (up to seven hops) depending on systems data.
- ❑ Multiple Call Forwarding over a K-CCIS link is combined with Multiple Call Forwarding – All Calls/Busy/No Answer.
- ❑ If the calling 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.

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



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).*

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: 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 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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All multiline terminals

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

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 SV9100 and UNIVERGE SV9100.

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 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.

Program/ Item No.	Description/Selection	Assigned Data	Comments
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.
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.

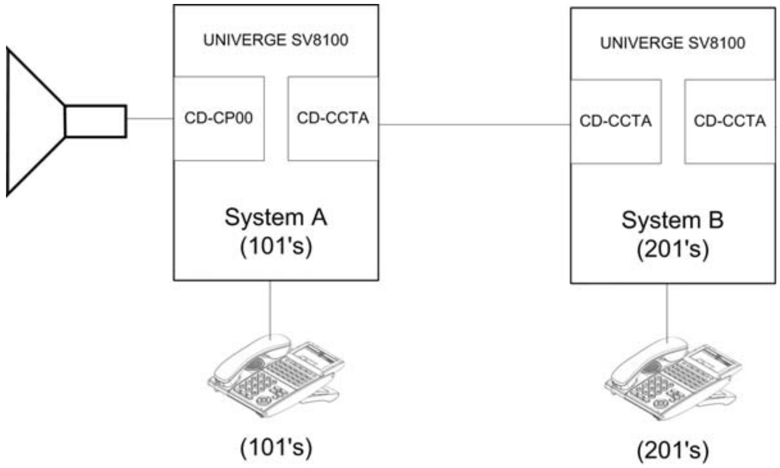
Program/ Item No.	Description/Selection	Assigned Data	Comments
31-06-01	External Speaker Control – Broadcast Splash Tone before Paging (Paging Start Tone)	0 = No Tone 1 = Splash Tone 2 = Chime Tone Default is 2	Enable Splash Tone before paging.
31-06-02	External Speaker Control – Broadcast Splash Tone 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 → 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 51 = 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	Change 703 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.

Program/ Item No.	Description/Selection	Assigned Data	Comments
44-05-01	ARS/F-Route Table – Trunk Group Number	0 = Not Set 1 ~ 100 = Trunk Group from 14-05 101 ~ 150 Networking 255 = Extension Call Default is 0	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 = 511(0~9)

Paging Access for System B = 512(0~9)

System A (101s)	System B (201s)
PRG 11-01-01 51 = 2 digits: F-Route	**PRG 11-01-01** 51 = 2 digits: F-Route
PRG 11-12-20 External Paging = 703 (Page System A)	**PRG 11-12-20** External Paging = 703 (Page System B)
PRG 44-02-01 Table 1 = Dial 511 (Page System A)	**PRG 44-02-01** Table 1 = Dial 512 (Page System B)
PRG 44-02-01 Table 2 = Dial 512 (Page System B)	**PRG 44-02-01** Table 2 = Dial 511 (Page System A)
PRG 44-02-02 Table 1 = F-Route Table 2 = F-Route	**PRG 44-02-02** Table 1 = F-Route Table 2 = F-Route
PRG 44-02-03 Table 1 = F-Route 1 Table 2 = F-Route 2	**PRG 44-02-03** Table 1 = F-Route 1 Table 2 = F-Route 2
PRG 44-05-01 Table 1 = TRK GP 255 Table 2 = TRK GP 10	**PRG 44-05-01** Table 1 = TRK GP 255 Table 2 = TRK GP 10
PRG 44-05-02 Table 1 = Delete Digits 3 Table 2 = Delete Digits 0	**PRG 44-05-02** Table 1 = Delete Digits 3 Table 2 = Delete Digits 0

System A (101s)	System B (201s)
PRG 44-05-03 Table 1 = Add Dial 1 Table 2 = Add Dial 0	**PRG 44-05-03** Table 1 = Add Dial 1 Table 2 = Add Dial 0
PRG 44-05-09 Table 1 = Max Digit 0 Table 2 = Max Digit 4	**PRG 44-05-09** Table 1 = Max Digit 0 Table 2 = Max Digit 4
PRG 44-06-01 Table 1 = Dial 703	**PRG 44-06-01** Table 1 = Dial 703

Programming Example (Internal)

Paging Access for System A = 521

Paging Access for System B = 522

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

System A (101’s)	System B (201’s)
Program 44-06-01 Table 1 = Dial 701	**Program 44-06-01** Table 1 = Dial 701

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

GCD-CCTA

- OR -

GPZ-IPLE

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.
- ❑ The InMail is supported for centralized voice mail in a CCIS network.

Related Feature List

➥ Digital Voice Mail

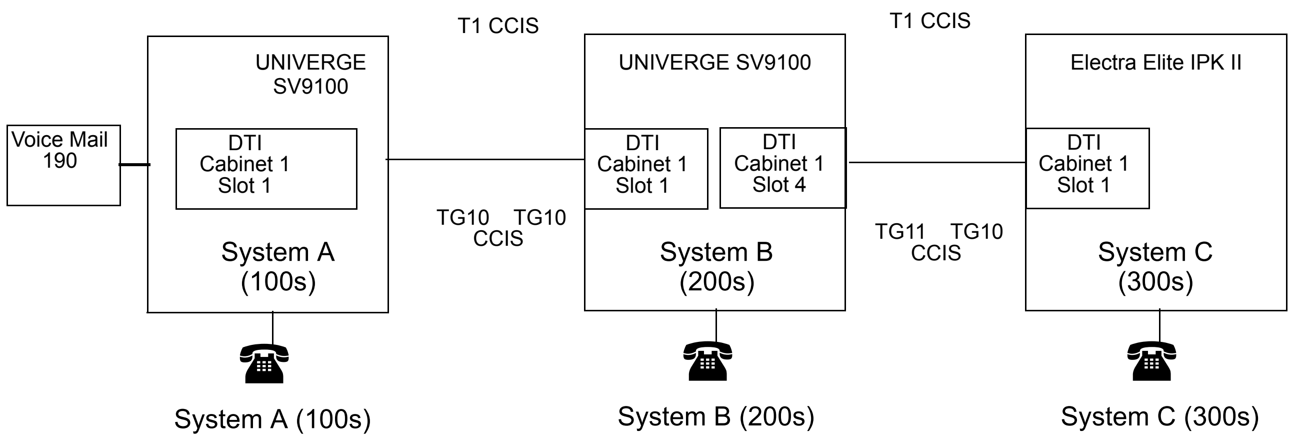
Guide to Feature Programming

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.

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 not assigned. Default is No Setting	Assign the CCIS Centralized Voice Mail Pilot Number for Remote Sites.

Programming Example



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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All multiline terminals

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

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 ~ 960 = 3101 ~ 3961	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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All multiline terminals

Required Components:

GCD-CCTA

- OR -

GPZ-IPLE

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 System Data Programming <US Only>.

Voice Call – K -CCIS

FEATURE DESCRIPTION

This feature provides a voice path, through the K-CCIS network, between a MLT in one office and a MLT 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:

GCD-CCTA

- OR -

GPZ-IPLE

OPERATING PROCEDURES

From one MLT to another MLT:

1. The originating MLT 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 SV9100 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 SV9100 Features and Specifications Manual.

System Availability

Terminal Type:

All Stations

Required Components:

CD-VM00

GCD-CCTA

- OR -

GPZ-IPLE

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.

1. 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.
- ❑ For InMail remote CCIS extensions are not supported in a centralized directory.

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 7-digit station numbers.
- ❑ 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.
- ❑ 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
11-01-01	System Numbering	1 = Service Code 2 = Extension Number 3 = Trunk Access 4 = Special Trunk access 5 = Operator Access 6 = Flexible Routing 7 = Not Used 8 = Networking Access System 9 = Dial Extension Analyze	Defaults for 1X, 2X, and 3X = 2 Extension Number.
11-02-01	Extension Numbering	Assign Station Numbers to Port Numbers	Defaults for Ports 1~960: 101-199 3101-3961
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.

Program/ Item No.	Description/Selection	Assigned Data	Comments
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.
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 Ext.101 is in Class 15. All others 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.

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	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.
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.

Program/ Item No.	Description/Selection	Assigned Data	Comments
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.
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.

SV9100 IP K-CCIS

Chapter 5

This chapter describes the system outline, Hardware installation, and programming procedures for providing **IP K-CCIS** using the GPZ-IPLE on the UNIVERGE SV9100 system.

SECTION 1 SYSTEM OUTLINE

1.1 IP K-CCIS Application using the GPZ-IPLE

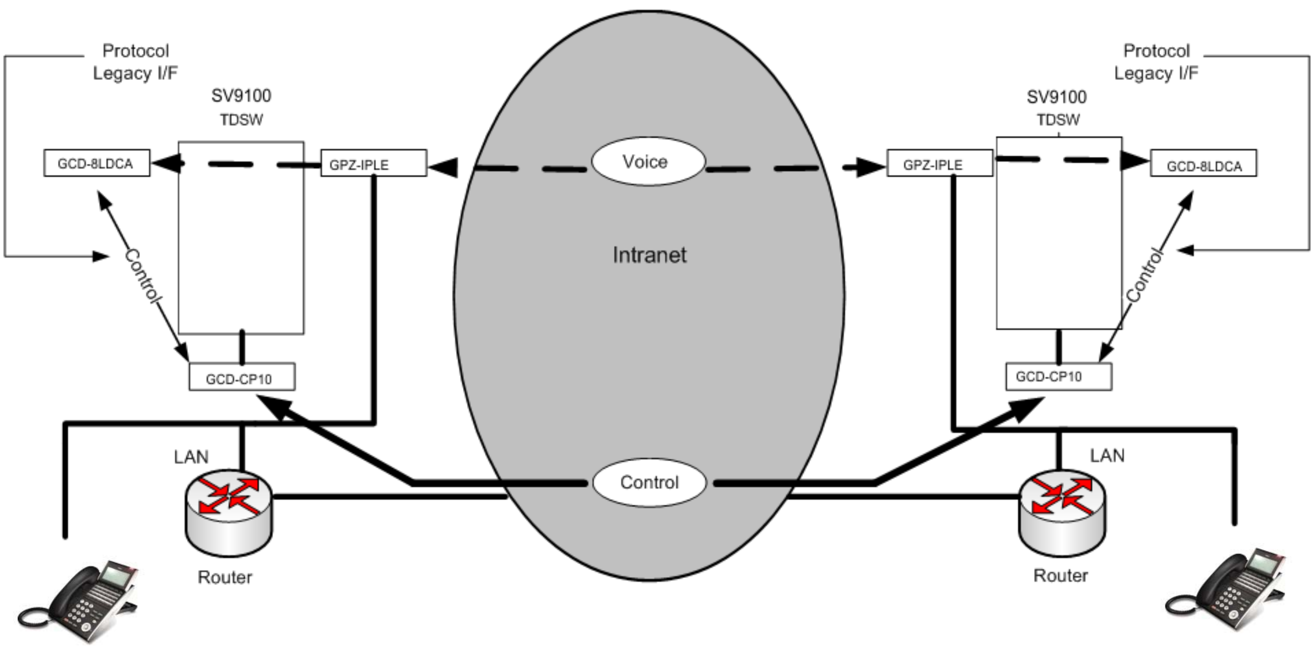
The system uses the GPZ-IPLE to connect multiple systems together over a Data Communication IP Network (Intranet). Key-Common Channel Interoffice Signaling (K-CCIS) is used to provide telephony services between the UNIVERGE SV9100 and another UNIVERGE SV9100 or a NEAX PBX system.

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 SV9100 and UNIVERGE SV9100 and a NEAX IPS, IPX, SV7000, UNIVERGE SV8100 and UNIVERGE SV9300.
- ❒ The SV9100 uses the NEC proprietary CCIS Peer-to-Peer protocol over IP to communicate between system to system.

❒ The GPZ-IPLE is required for connections between IP terminals and IP trunks. A maximum of one GPZ-IPLE can be accommodated per system with a maximum of 256 DSP Resources per system.

Figure 5-1 Peer-to-Peer One IP to TDM



1.2 Description

The GPZ-IPLE 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 GPZ-IPLE board converts IP packet signals. The GPZ-IPLE 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

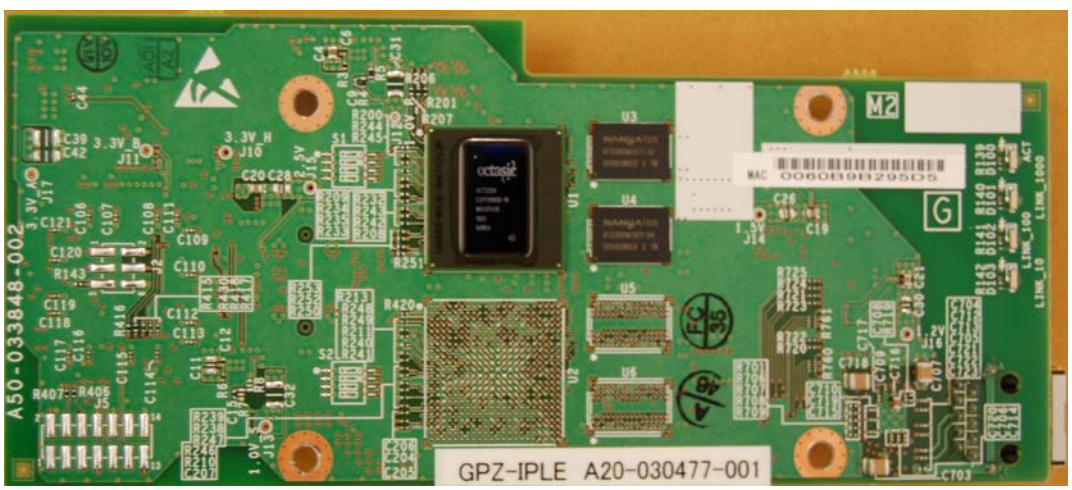
Only voice (RTP/RTCP) processing functions are mounted among VOIP functions on the GPZ-IPLE and all call control functions are handled by the GCD-CP10.

Only one GPZ-IPLE blade can be mounted on the GCD-CP10 at any given time.

The GPZ-IPLE 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 the number of ports licensed in the GCD-CP10. For example, if the GCD-CP10 is installed with a GPZ-IPLE and the GCD-CP10 is licensed for 64 VoIP Channels. The maximum configuration supported by the CCISoIP Application is 64 CCISoIP channels.

Figure 5-2 GPZ-IPLE Daughter Board



1.4 Installation

The GPZ-IPLE board can be mounted only on the GCD-CP10. Only one GPZ-IPLE board at a time can be mounted on the GCD-CP10.

1.4.1 Hot Swap

The GPZ-IPLE is not hot swappable and cannot be removed from the GCD-CP10 without first powering down the chassis and removing the GCD-CP10.

1.4.2 Connectors

The GPZ-IPLE has the following connectors:

- CN1 - RJ-45 Gigabit Ethernet LAN Interface
- CN3 - Connects to the GCD-CP10

1.4.3 Ethernet Status (IPLE)

Four LEDs (two red, one green and one yellow) on the GPZ-IPLE 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 GPZ-IPLE on the GCD-CP10:

WARNING — ***Do not remove or install the GCD-CP10 with the power on.***

1. Turn off the system power.
2. Remove the GCD-CP10, and install the GPZ-IPLE on it.
3. Insert the GCD-CP10 in slot 1 in the Controlling Chassis.
4. Figure 5-3 GCD-CP10 Blade with Daughter Board InstalledConnect the GPZ-IPLE to the CD-RTB or to an external switching hub using an ethernet cable. Refer to the UNIVERGE SV9100 Programming Manual for detailed programming instructions.

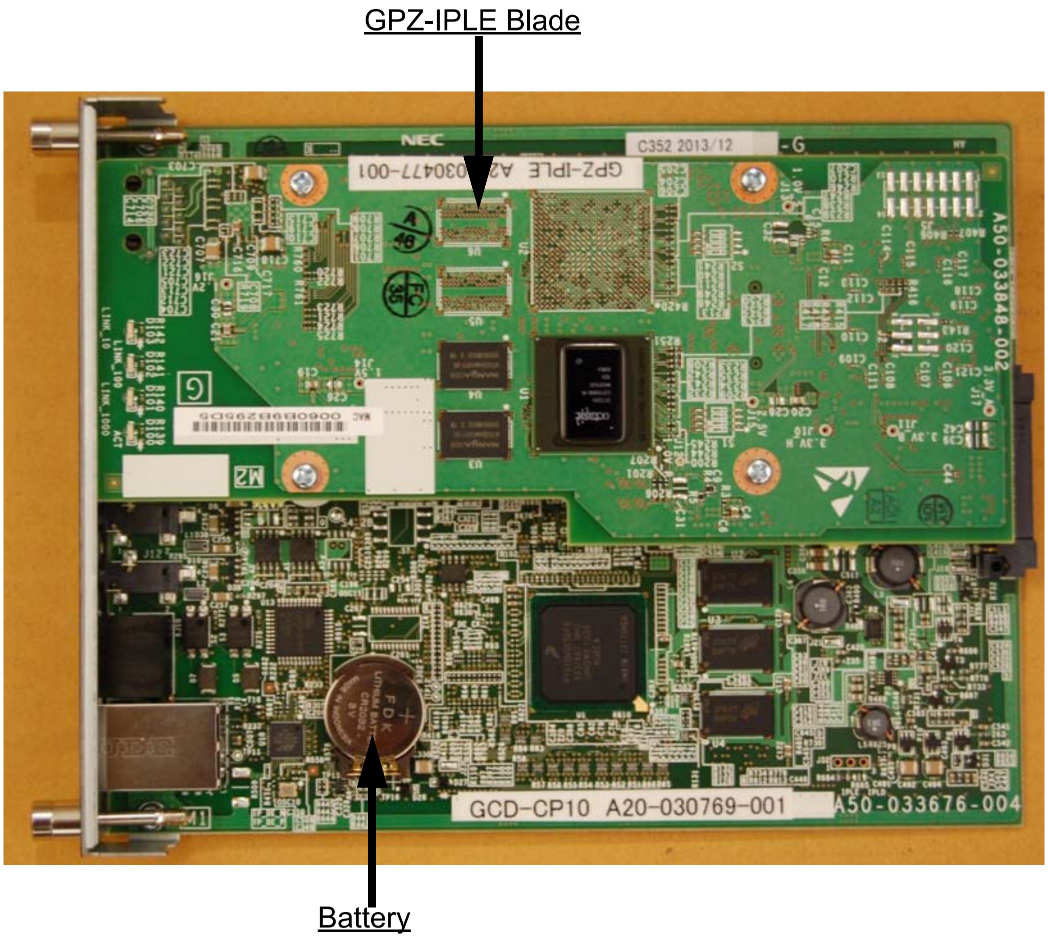
1.4.6 Switch Settings

The daughter board has no switches to set.

1.4.7 LED Indications

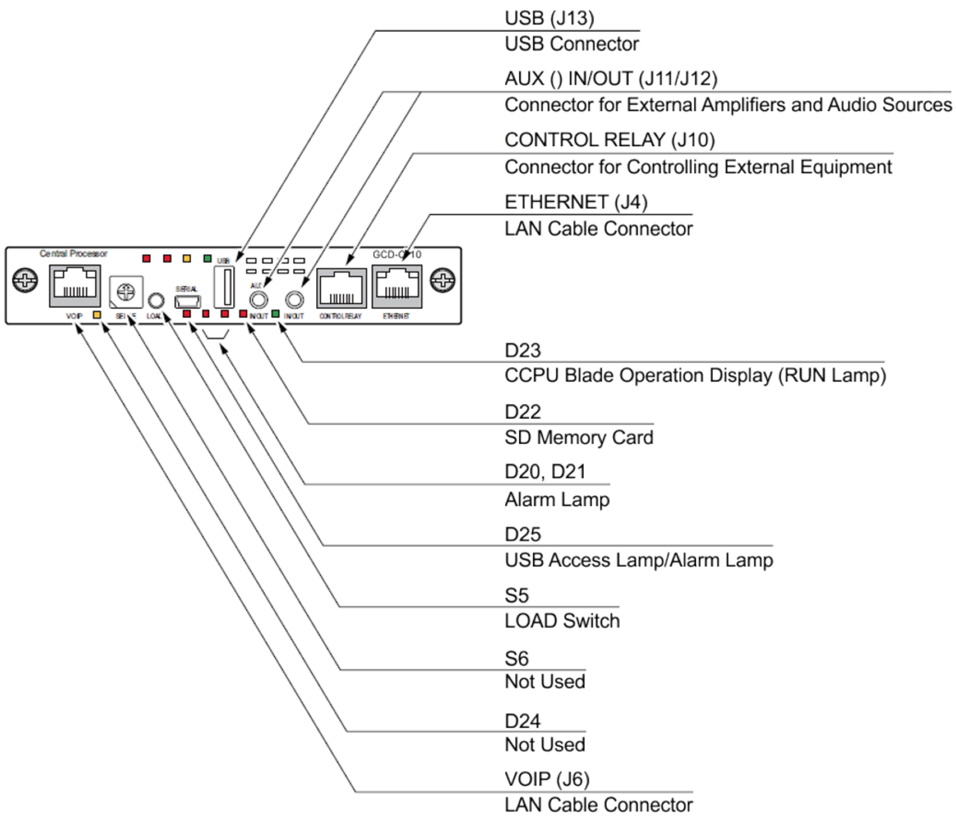
Refer to Figure 5-3 GCD-CP10 Blade with Daughter Board Installed.

Figure 5-3 GCD-CP10 Blade with Daughter Board Installed



See Figure 5-4 GCD-CP10 LED Locations for the location of the LEDs on the blade.

Figure 5-4 GCD-CP10 LED Locations



LED indications for the GPZ-IPLE 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 (LED 10)	10 Base-T link speed indicator	On Red	10 Base-T link up
Link 100 (LED 100)	100 Base-T link speed indicator	On Red	100 Base-T link up
Link 1000 (LED 1G)	1000 Base-T link speed indicator	On Yellow	1000 Base-T link up
ACT (LED A)	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.

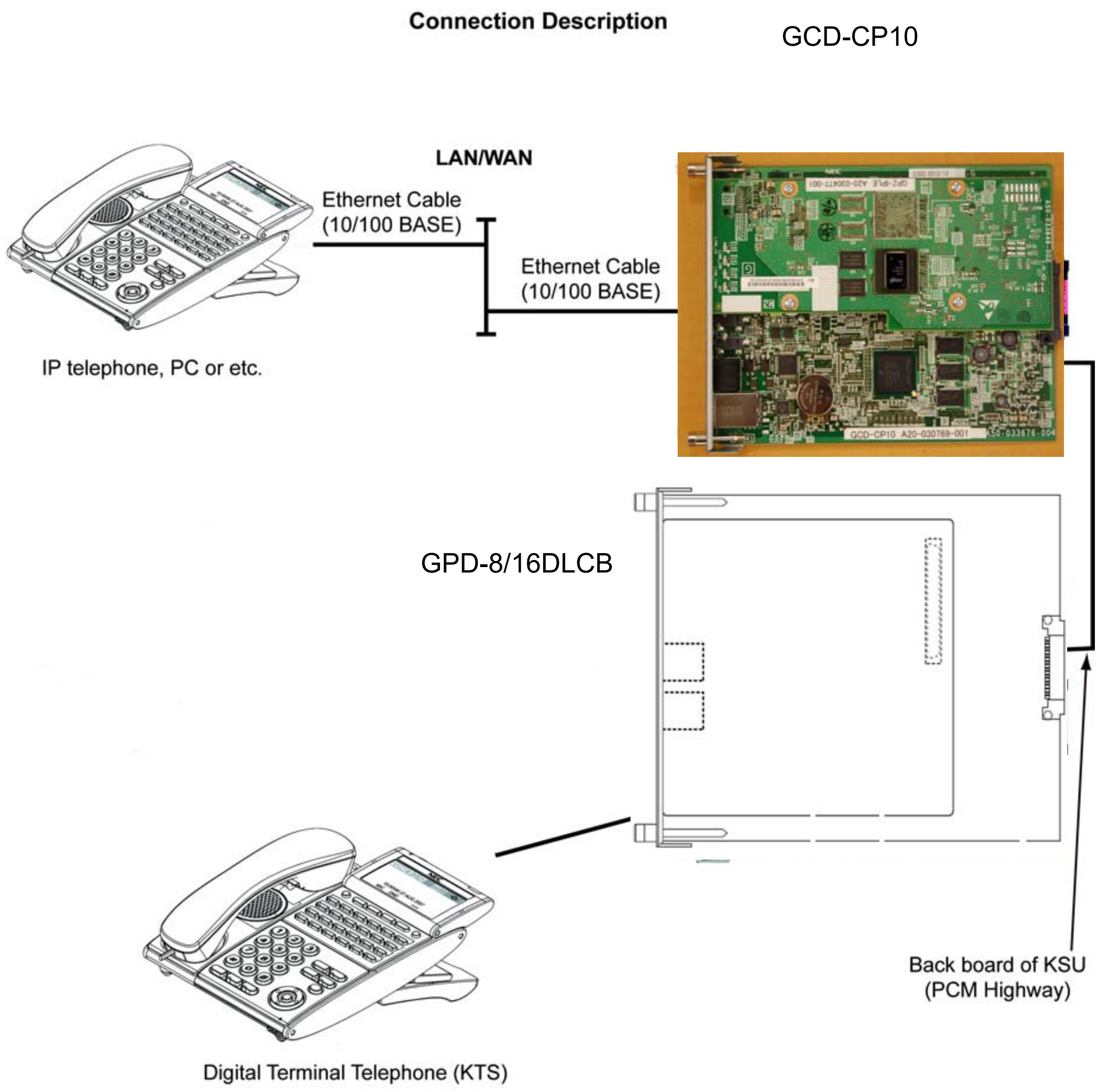
Table 5-2 VOIPDB LED CN1 Transmit/Receive Data Indications

LED	Link Up									
	Auto Negotiation Mode					Force Mode				
	1000Mbps	100Mbps		10Mbps		1000Mbps	100Mbps		10Mbps	
		Half	Full	Half	Full		Half	Full	Half	Full
ACT (LED A)	ON	ON	ON	ON	ON	ON	ON	ON	ON	ON
LINK1000 (LED 1G)	ON	OFF	ON	OFF	ON	ON	OFF	ON	OFF	ON
LINK 100 (LED 100)	ON	OFF	OFF	OFF	OFF	ON	OFF	OFF	OFF	OFF
LINK10 (LED 10)	OFF	ON	ON	OFF	OFF	OFF	ON	ON	ON	ON

1.4.8 Connectors

Figure 5-5 VoIP Connections shows a typical connection layout. Figure 5-6 Connecting a VOIPDB to a Network/PC on page 5-9 illustrates how to connect a VoIP Daughter Board to a Network or PC.

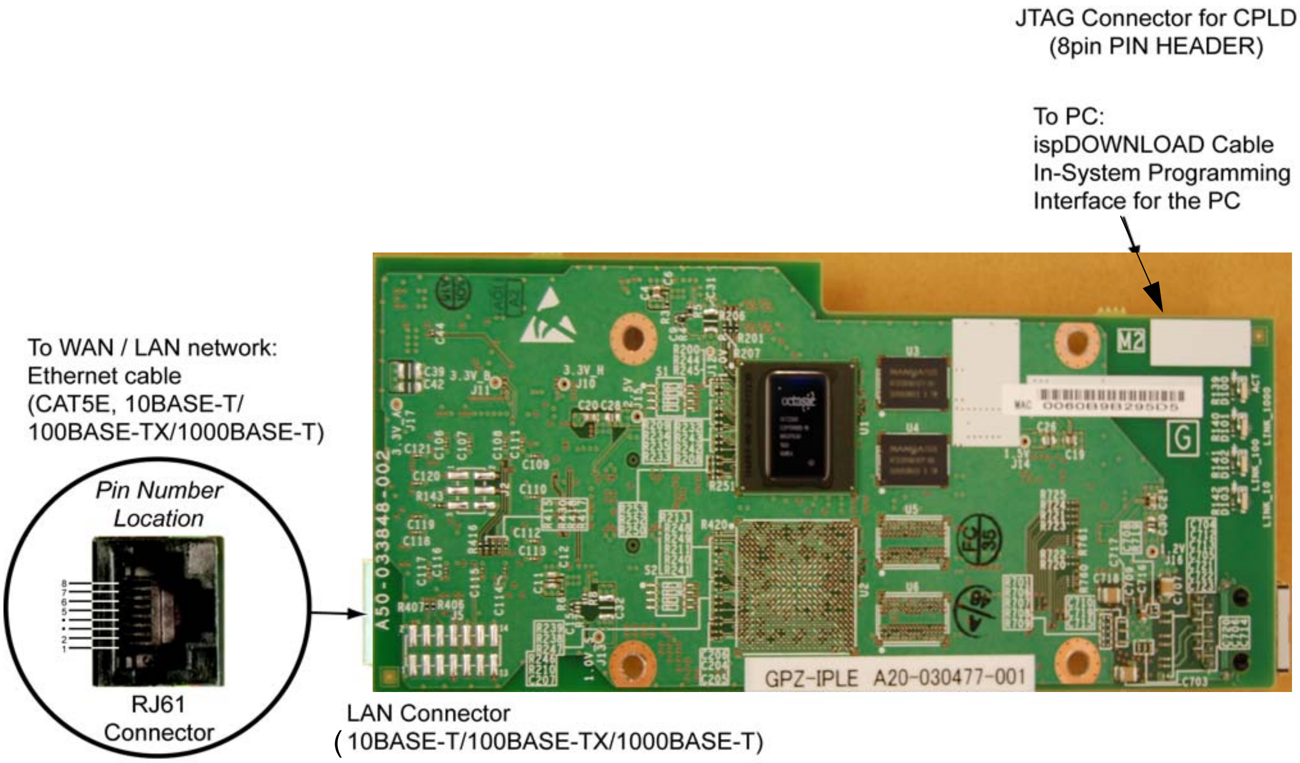
Figure 5-5 VoIP Connections



1.5 IP Addressing

Figure 5-6 Connecting a VOIPDB to a Network/PC shows how the VOIPDB is connected to a network or PC.

Figure 5-6 Connecting a VOIPDB to a Network/PC



One valid IP address must be assigned for all the DSP's that are used in IPLE.

The GPZ-IPLE needs only one IP Address.

IMPORTANT

When assigning the IP address to the IPLE card, the address must be in the same network (subnet). If the CPU is also connected to the network, it requires a separate IP address in a different network (subnet).

When you have an IPLE 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 IPLE NIC.

For example, PC Pro, Web Pro, ACD, etc. terminate to the IPLE NIC card, when installed. Both the IPLE 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, IPLE 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 SV9100 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.

Table 5-5 DSCP Code Points and Binary Values

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.

Table 5-6 ToS Field with Diffserv sBits

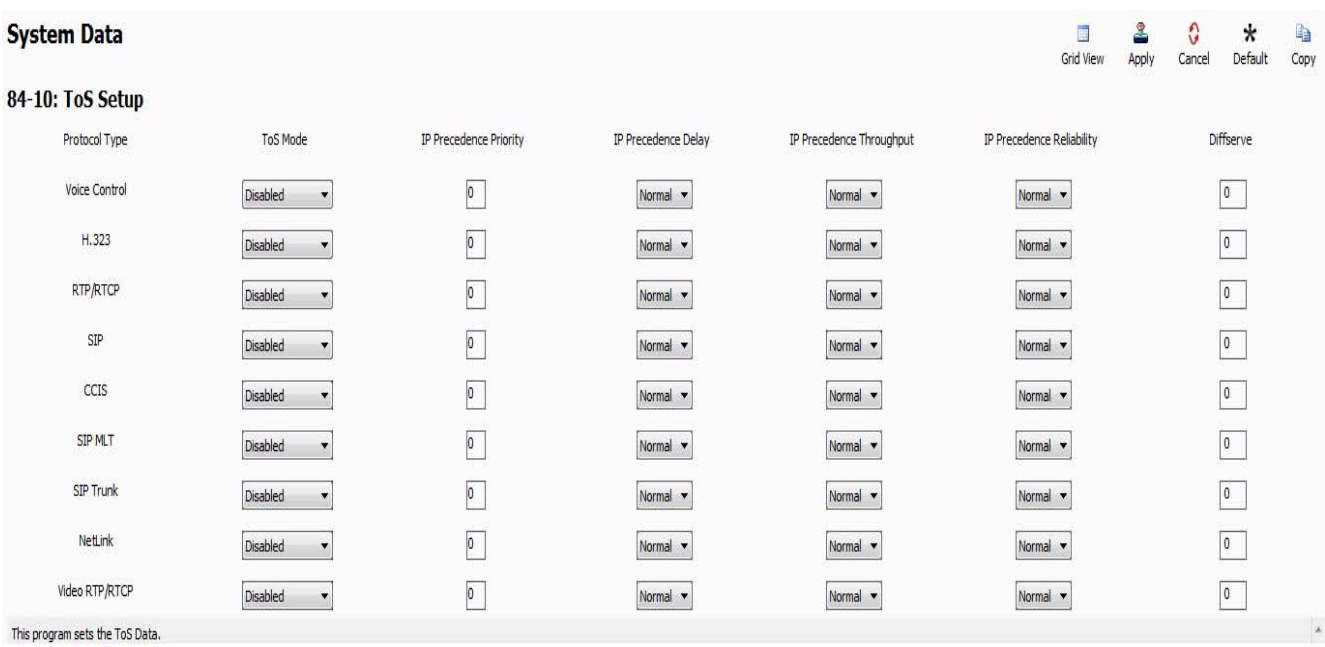
Diffserv	Diffserv	Diffserv	Diffserv	Diffserv	Diffserv	Not Used	Not Use d
1(on) here = value of 32	1(on) here = value of 16	1(on) here = value of 8	1(on) here = value of 4	1(on) here = value of 2	1(on) here = value of 1		

Assignments for the IP Precedence/Diffserv values in the system are submitted in command 84-10.

RTP/RTCP = voice packets

CCIS= signaling packets

Figure 5-7 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

CODEC	Packet Size	PPP	Ethernet
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

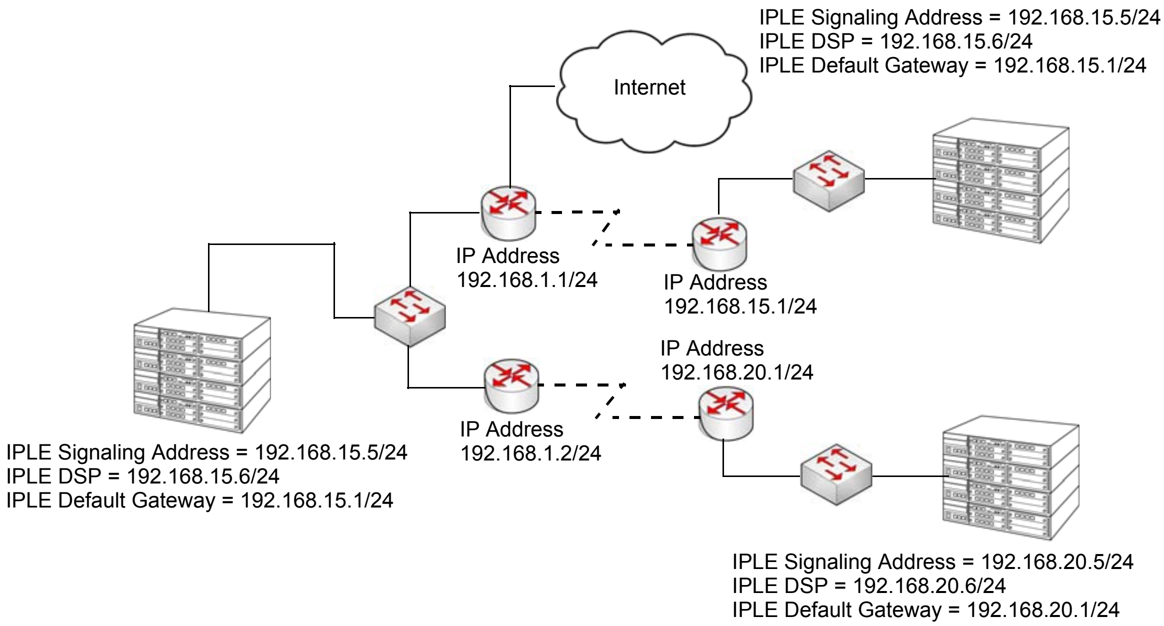
1.5.6 Some Network Considerations

Before adding the UNIVERGE SV9100 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 5-8 Multiple Gateways shows the UNIVERGE SV9100 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.

Figure 5-8 Multiple Gateways



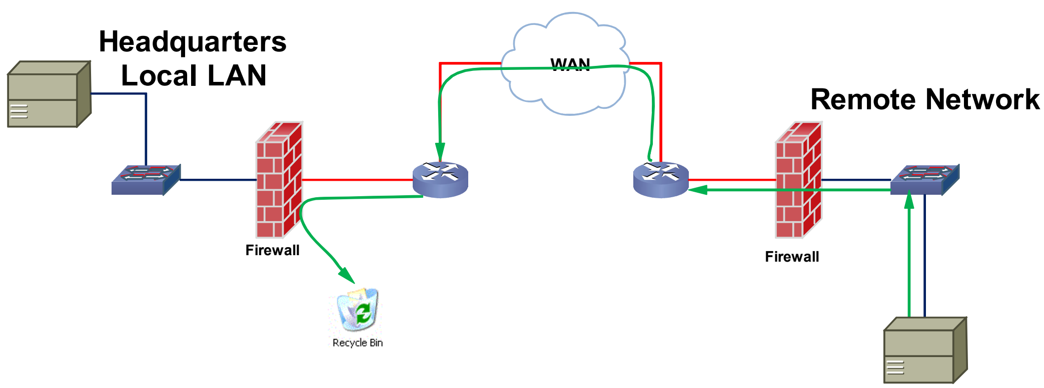
Firewall

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 5-9 Two SV9100 Systems Connected via the WAN shows two SV9100 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 5-9 Two SV9100 Systems Connected via the WAN



The green arrow in Figure 5-9 Two SV9100 Systems Connected via the WAN represents the data packets leaving the REMOTE IPLE card destined for the SV91000 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 SV9100.

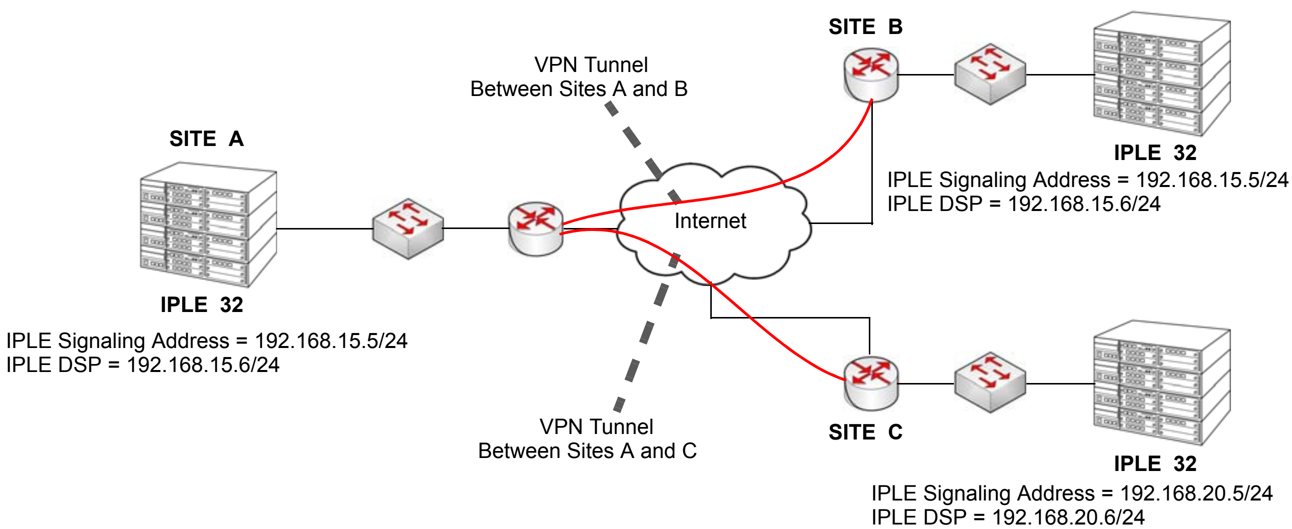
The ports, **57000** and **59000** (TCP) for signaling and the voice ports, are required to be open at each location.

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 5-10 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 SV9100 CHASSIS PROGRAMMING

The following data programs are used when installing the GPZ-IPLE daughter board for UNIVERGE SV9100 IP (K-CCIS).

➨ *If any address or NIC setting is changed, the system must be reset for the changes to take affect.*

2.1 Digital Trunk Assignment

Use these programs to make digital trunk assignments when an IPLE daughter board is installed. .

Table 5-7 Digital Trunk Assignments

Program/ Item No.	Description/Selection	Assigned Data	Comments
10-68-01	IP Trunk Availability - Trunk Type	0 = None 1 = SIP 2 = H.323 3 = CCIS Default is 0	Define the IP Trunk Type
10-68-02	IP Trunk Availability - Start Port	0 ~ 400 Default is 0	Assign the Start Port for IP Trunks.
10-68-03	IP Trunk Availability - Number of Ports	0 ~ 400 Default is 0	Define the number of ports.
10-12-03	GCD-CP10 Network Setup – Default Gateway	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 Default is 0.0.0.0	Assign the Default Gateway for the GCD-CP10. When using the IPLE, the IP Address and Subnet Mask in PRG 10-12-01 and PRG 10-12-02 remain default.
10-12-09	GCD-CP10 Network 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	Assign the IP Address of the IPLE.
10-12-10	GCD-CP10 Network Setup – Subnet Mask	255.255.0.0 Default is 255.255.0.0	Assign the Subnet Mask of the IPLE

Table 5-7 Digital Trunk Assignments (Continued)

Program/ Item No.	Description/Selection	Assigned Data	Comments
10-19-01	VoIP DSP Resource Selection – VoIP DSP Resource Selection	Slot Number 1: DSP Resource Number: 01 ~ 256 0 = Common use for both IP extensions and trunks 1 = IP Extension 2 = SIP Trunk 3 = Networking/CCIS 4 = NetLink 5 = Blocked 6 = Common without Unicast paging 7 = Multicast Paging 8 = Unicast Paging Default: Resource 1 = 1 Resources 2~256 = 0	
10-54-01	License Configuration for Each Package - License Code Assign VoIP Resource Licenses (5103) to GCD-CP10 slot (1). If applying more than 255 licenses to a slot, the licenses must be applied across multiple indexes. For example, assigning 256 VoIP resource licenses (5103) to the CPU slot could be assigned using different methods, as long as the total for the CPU slot is 256. 1. Index 1 has 128 of feature code 5103 and index 2 also has 128 of feature code 5103 for a total of 256. 2. Index 1 has 255 of feature code 5103 and index 2 has 1 of feature code 5103 for a total of 256.	1 ~ 255 Resource Licenses	
14-05-01	Trunk Group	0 ~ 100 Default is 1	Assign Trunk-to-Trunk Groups.

Table 5-7 Digital Trunk Assignments (Continued)

Program/ Item No.	Description/Selection	Assigned Data	Comments
22-02-01	Incoming Call Trunk Setup – Trunk Type	001 ~ 400 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 GPZ-IPLE, daughter board is installed on the GCD-CP10 blade. The GPZ-IPLE daughter board reduces the maximum capacity of trunks in the system.

Table 5-8 VoIP IP Address Assignments

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
84-26-02	IPL Basic Setup – RTP Port Number	0~65534	Default: RTP Port = 10020
84-26-03	IPL Basic Setup – RTCP Port Number (RTP Port Number + 1)	0~65534	Default: RTCP = 10021

2.3 Local Numbering Plan Assignment

Use these programs to assign system, extension and virtual extension numbering for the Local Numbering Plan.

Table 5-9 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 7 = Not Used 8 = Networking System Access 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 Port 1 ~ 99 = 101 ~ 199 Port 100 ~ 960 = 3101 ~ 3961
11-04-01	Virtual Extension Numbering	Assign Station Numbers to Port Numbers	Defaults for Ports Ports 1 ~ 99 = 201 ~ 299 Ports 100 ~ 512 = No setting

2.4 Closed Numbering Plan – Using Closed Number Blocks

Use these programs to assign system numbering and ARS/F-Route dialed digits for a Closed Numbering Plan.

Table 5-10 Closed Numbering Plan Assignments

Program/ Item No.	Description/Selection	Assigned Data	Comments
11-01-01	System Numbering	2 = Extension Number	Defaults for 1X, 2X, 3X = 2 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.

Table 5-10 Closed Numbering Plan Assignments (Continued)

Program/ Item No.	Description/Selection	Assigned Data	Comments
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	0-100,101-150,255 0 = Not Set 1~100 = Trunk Group from PRG 14-05 101~150 = Networking 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

Use these programs to assign system numbering, ARS/F-Route dialed digits and trunk groups as well as other ARS/F-Route table information for an Open Numbering Plan.

Table 5-11 Open Numbering Plan Assignments

Program /Item No.	Description/Selection	Assigned Data	Comments
11-01-01	System Numbering	2 = Extension Number	Defaults for 1X, 2X, 3X = 2 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.
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.

Table 5-11 Open Numbering Plan Assignments (Continued)

Program /Item No.	Description/Selection	Assigned Data	Comments
44-05-01	ARS/F-Route Table – Trunk Group Number	0-100,101-150,255 0 = Not Set 1~100 = Trunk Group from PRG 14-05 101~150 = Networking 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 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.
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

Use these programs to assign CCIS availability, origination and destination point codes for IP K-CCIS.

Table 5-12 IP K-CCIS Assignments

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

Use these programs to assign the CCIPS over IP connection method and the TCP server port number.

Table 5-13 IP K-CCIS (Peer-to-Peer) Assignments

Program/ Item No.	Description/Selection	Assigned Data	Comments
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

Use these programs to assign CCIS over IP CODEC information, including the number of G.711/G.723/G.729 type, number of audio frames, voice activity detection mode, jitter buffers and other CODEC related information.

Table 5-14 CCIS over IP CODEC Information Assignments

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 0	
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 ~ 300 ms Default is 30	
84-21-05	CCIS over IP CODEC Information Basic Setup – G.711 Jitter Buffer Average	0 ~ 300 ms Default is 60	
84-21-06	CCIS over IP CODEC Information Basic Setup – G.711 Jitter Buffer Max	0 ~ 300 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	

Table 5-14 CCIS over IP CODEC Information Assignments (Continued)

Program/ Item No.	Description/Selection	Assigned Data	Comments
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 ~ 300 ms Default is 30	
84-21-10	CCIS over IP CODEC Information Basic Setup – G.729 Jitter Buffer Average	0 ~ 300 ms Default is 60	
84-21-11	CCIS over IP CODEC Information Basic Setup – G.729 Jitter Buffer Max	0 ~ 300 ms Default is 120	
84-21-19	CCIS over IP CODEC Information Basic Setup – 1st Priority of Audio Capability	0 = G.711 PT 2 = G.729 PT 3 = G.722 4 = G.726 Default is 0	
84-21-20	CCIS over IP CODEC Information Basic Setup – 2nd Priority of Audio Capability	0 = G.711 PT 2 = G.729 PT 3 = G.722 PT 4 = G.726 PT Default is 2	
84-21-22	CCIS over IP CODEC Information Basic Setup – Jitter Buffer Mode	1 = Static 2 = Not Used 3 = Self adjusting 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	

Table 5-14 CCIS over IP CODEC Information Assignments (Continued)

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	
84-21-29	CCIS over IP CODEC Information Basic Setup – G.722 Jitter Buffer (min)	0 ~ 300 ms Default is 30	
84-21-30	CCIS over IP CODEC Information Basic Setup – G.722 Jitter Buffer (Average)	0 ~ 300 ms Default is 60	
84-21-31	CCIS over IP CODEC Information Basic Setup – G.722 Jitter Buffer (max)	0 ~ 300 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 ~ 300 ms Default is 30	
84-21-35	CCIS over IP CODEC Information Basic Setup – G.726 Jitter Buffer (Average)	0 ~ 300 ms Default is 60	

Table 5-14 CCIS over IP CODEC Information Assignments (Continued)

Program/ Item No.	Description/Selection	Assigned Data	Comments
84-21-36	CCIS over IP CODEC Information Basic Setup – G.726 Jitter Buffer (max)	0 ~ 300 ms Default is 120	
84-21-37	---Not Used---		

2.9 SIP-MLT CODEC Information Fixed Mode Setup

Use these programs to assign the fixed mode audio capacity and number of audio frames for the SIP MLT..

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 5 = G.722 Default is 1	

Program/ Item No.	Description/Selection	Assigned Data	Comments
84-29-02	SIP-MLT CODEC Information Fixed Mode Setup – Number of Audio Frames	Type: 1 ~ 6 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	

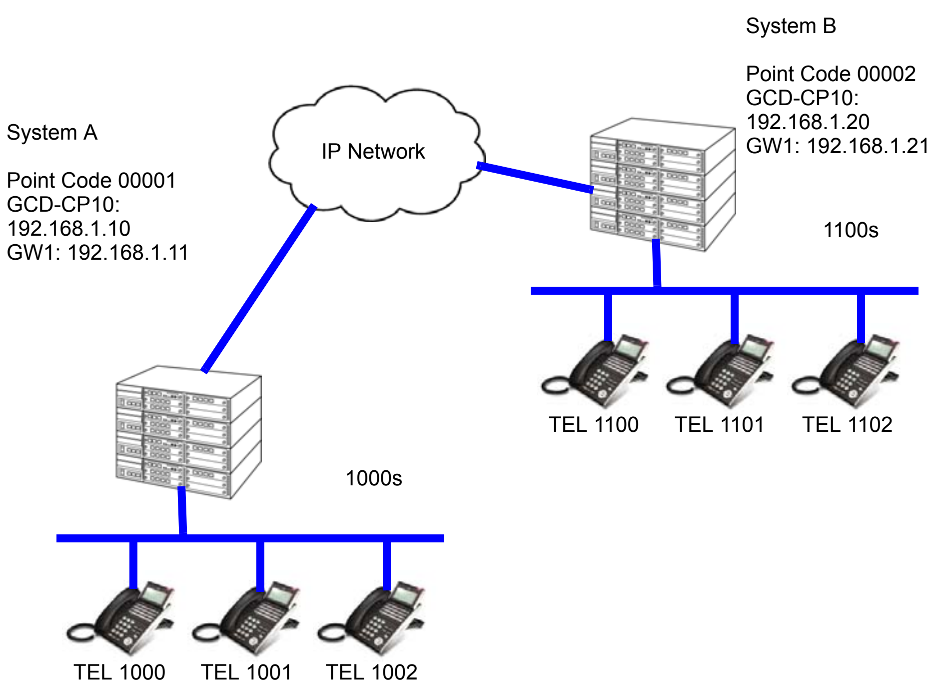
SECTION 3 PROGRAMMING EXAMPLE

3.1 SV9100 IP K-CCIS Programming Example 1

This is an example of UNIVERGE SV9100 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.
- Each system has 16 multiline terminals.
- Each system has eight analog trunks.

Figure 5-11 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)

The following slot assignments were set using program 10-03.

Slot	Card Type During 1st Power On	CARD TYPE AND PORTS			
		System A		System B	
		Card Type	Ports	Card Type	Ports
1	GCD-CP10/GPZ-IPLE	GCD-CP10/GPZ-IPLE	Trunks 5 ~ 28	GCD-CP10/GPZ-IPLE	Trunks 5 ~ 28
2	GCD-8DLCA	GCD-8DLCA	Station 1 ~ 8	GCD-8DLCA	Station 1 ~ 8
3	GCD-8DLCA	GCD-8DLCA	Station 9 ~ 16	GCD-8DLCA	Station 9 ~ 16
4	None	None	N/A	None	N/A
5	GCD-4COTC	GCD-4COTC	Trunks 1 ~ 4	GCD-4COTC	Trunks 1 ~ 4
6	None	None	N/A	None	N/A

3.1.2 Digital Trunk Assignments

This example shows digital trunk assignments made for systems A and B.

System A	System B
Program 10-03-01 Insert GPZ-IPLE on GCD-CP10 in Chassis 1, Slot 1. Verify PRG 10-03-01 (ETU Setup)	Program 10-03-01 Insert GPZ-IPLE on GCD-CP10 in Chassis 1, Slot 1. 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 10-68-01 Assign the Trunk Type as CCIS	Program 10-68-01 Assign the Trunk Type as CCIS
Program 10-68-02 Assign the Start Port as 5	Program 10-68-02 Assign the Start Port as 5
Program 10-68-03 Assign 24 as number of ports	Program 10-68-03 Assign 24 as number of ports
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

This example shows VoIP address assignments made for systems A and B.

System A	System B
Program 10-12-09 IPL IP Address: 192.168.1.10 Program 84-26-01 Slot 1, Assign IP Address GW: 192.168.1.11	Program 10-12-09 IPL IP Address: 192.168.1.20 Program 84-26-01 Slot 1, Assign IP Addresses GW: 192.168.1.20

3.1.4 IP K-CCIS Availability

This example shows IP K-CCIS availability assignments made for systems A and B.

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

This example shows IP K-CCIS assignments made for systems A and B.

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

This example shows IP K-CCIS peer-to-peer assignments made for systems A and B.

System A	System B
Program 50-15-04 Enable/Disable Connection Method 0 = P2P Disable 1 = P2P Enable (Default)	Program 50-15-04 Enable/Disable Connection Method 0 = P2P Disable 1 = P2P Enable (Default)
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

This example shows Local Numbering Plan assignments made for systems A and B.

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 ~ 960: 101 ~ 199 3101 ~ 3199 3200 ~ 3961	Program 11-02-01 Assign Station Numbers to Port Numbers. Defaults for Ports 1 ~ 960: 101 ~ 199 3101 ~ 3199 3200 ~ 3961

3.1.8 Closed Numbering Plan – Using Closed Number Blocks

This example shows Closed Numbering Plan assignments made for systems A and B.

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 ~ 960 101 ~ 199 3101 ~ 3199 3200 ~ 3961	Program 11-02-01 Assign Station Numbers to Port Number. Defaults for Ports 1 ~ 960 101 ~ 199 3101 ~ 3199 3200 ~ 3961
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

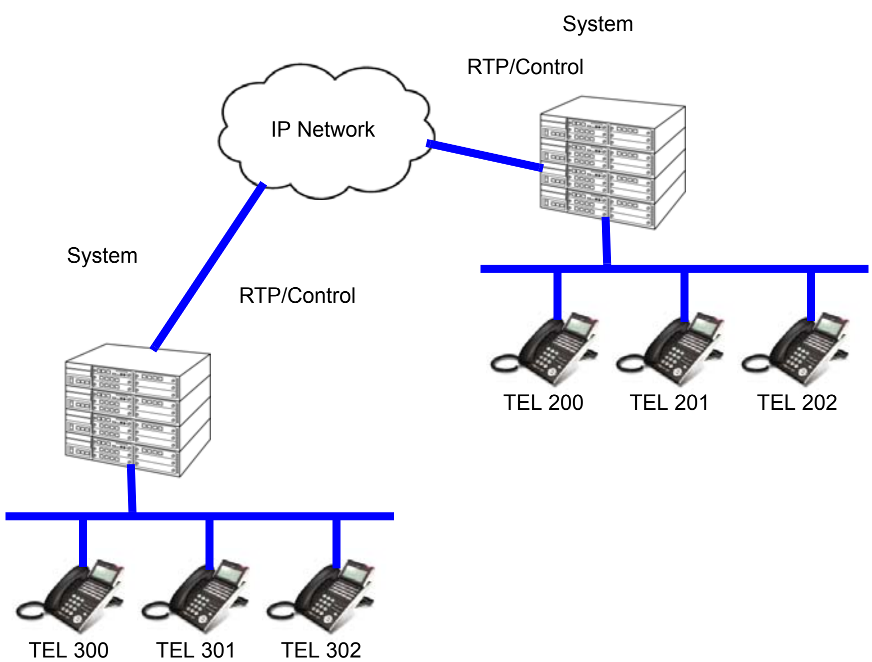
3.1.9 Tandem Connections

This example shows Tandem connections assignments made for systems A and B.

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 5-12 TEL 302 (SIP MLT) makes call to TEL 202 (SIP MLT) via IP K-CCIS illustrates calls between two SIP multiline terminals over IP-CCIS.

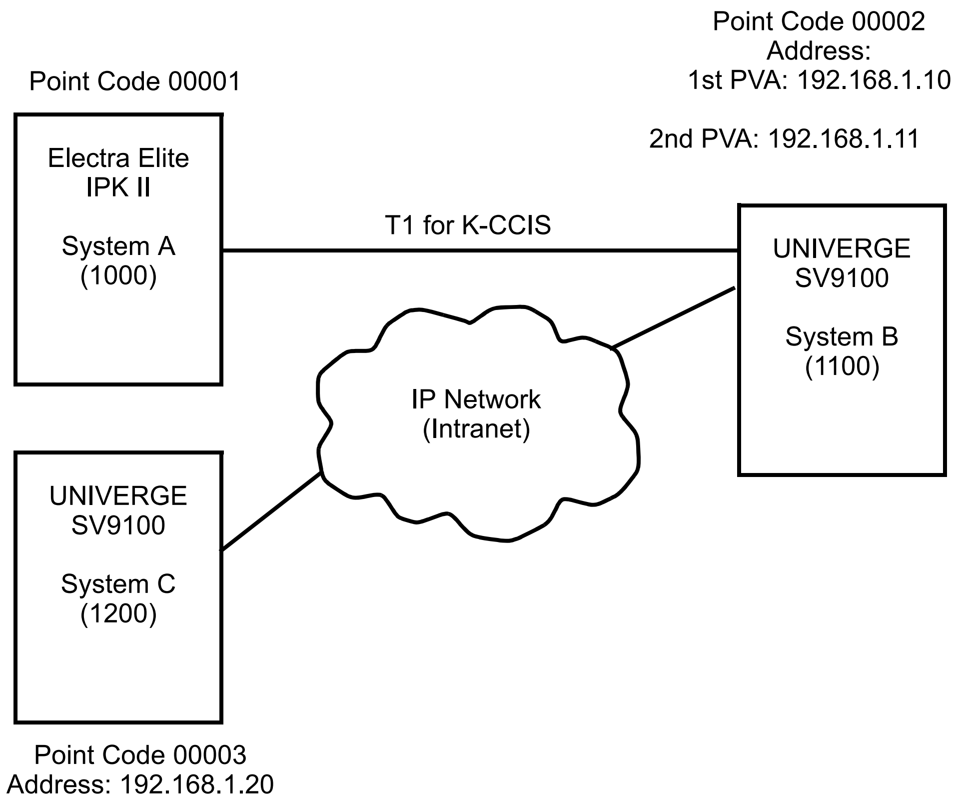
Figure 5-12 TEL 302 (SIP MLT) makes call to TEL 202 (SIP MLT) via IP K-CCIS



3.2 UNIVERGE SV9100 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.

Figure 5-13 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 SV9100 with one CHS2U and one GPZ-IPLE installed on the GCD-CP10 and one GCD-CCTA blade installed.
- System C is a single-cabinet UNIVERGE SV9100 port package with one CHS2U and GPZ-IPLE Daughter Board installed on the GCD-CP10.

- ❒ 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	GCD-CP10 with GPZ-IPLE Blade/Daughter Board	GCD-CP10 with GPZ-IPLE Blade/Daughter Board	Trunks 29 ~ 52
2	GCD-8DLCA Blade	GCD-8DLCA Blade	Station 1 ~ 8
3	GCD-8DLCA Blade	GCD-8DLCA Blade	Station 9 ~ 16
4	None	None	N/A
5	GCD-4COTC Blade	GCD-4COTC Blade	Trunks 1 ~ 4
6	GCD-CCTA Blade	GCD-CCTA Blade	Trunks 5 ~ 28

System C			
Slot	**Card Type During 1st Power On**	**Card Type And Ports**	
		Card Type	**Ports**
1	GCD-CP10 with GPZ-IPLE Blade/Daughter Board	GCD-CP10 with GPZ-IPLE Blade/Daughter Board	Trunks 5 ~ 28
2	GCD-8DLCA Blade	GCD-8DLCA Blade	Station 1 ~ 8
3	GCD-8DLCA Blade	GCD-8DLCA Blade	Station 9 ~ 16
4	None	None	N/A
5	GCD-4COTC Blade	GCD-4COTC Blade	Trunks 1 ~ 4
6	None	None	None

3.2.2 Digital Trunk Assignments

This example shows digital trunk assignments for systems A, B and C.

System A	System B	System C
Program 10-03-01 Notice the Logical Trunk Number	Program 10-03-01 Notice the Logical Trunk Number	Program 10-03-01 Select Slot 1. 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 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 Assign Number of Ports for CCIS for Slot 1 = 24.
Program 14-05-01 Trunks 9 ~ 32 Assign Trunk Group 10	*Program 10-19-01 Assign DSP Resources 001 ~ 024 as CCIS (LK 4).	Program 14-05-01 Trunks 5 ~ 28 Assign Trunk Group 10

System A	System B	System C
Program 22-02-01 Trunks 9 ~ 32 Assign LK 6 (Tie Line)	∗Program 10-40-04 Assign Number of Ports for CCIS for Slot 1 = 24	Program 22-02-01 Trunks 5 ~ 28 Assign LK 6 (Tie Line)
Program 34-01-01 Trunks 9 ~ 32 Assign LK 2 (Wink)	Program 14-05-01 Trunks 5 ~ 28 Assign Trunk Group 11 Trunks 29 ~ 52 Assign Trunk Group 10	Program 34-01-01 Trunks 5 ~ 28 Assign LK 2 (Wink)
	Program 22-02-01 Trunks 5 ~ 52 Assign LK 6 (Tie Line)	
	Program 34-01-01 Trunks 5 ~ 52 Assign LK 2 (Wink)	

∗*For CCISoIP only*

3.2.3 CCH Assignment

This example shows CCH assignments for systems A, B and C.

System A	System B	System C
Program 14-13-01 Select Trunk Group 10 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 Trunks 9 ~ 32 Assign CIC 1 ~ 24	Program 14-13-01 Select Trunk Group 10 Assign CCIS Route ID 1	Program 50-01-01 Enable CCIS
Program 50-01-01 Enable CCIS	Program 14-14-01 Trunks 5 ~ 28 Assign CIC 1 ~ 24	Program 50-02-03 Route ID 9 : 3
Program 50-02-01 Select CCIS Route ID 1 Assign Channel Port 28	Program 50-01-01 Enable CCIS	∗Program 50-03-01 System ID 1 : 2 System ID 2 : 4

System A	System B	System C
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

This section provides port number designations for IP applications.

IP Application	IP Port Numbers	Comments
SIP Trunk		
SIP Trunk Signaling	5060	UDP
SIP Trunk Voice with GPZ-IPLE	10020~10531	UDP
3rd Party SIP		
SIP SLT Signaling	5070	UDP
SIP SLT Voice with GPZ-IPLE	10020~10531	UDP
H.323 Trunk		
H.323 Signaling	1718/1719/1720	TCP
H.323 Trunk Voice with GPZ-IPLE	10020~10531	UDP

IP Application	IP Port Numbers	Comments
SV9100 IP - K-CCIS		
IP K-CCIS Signaling	57000/59000	TCP
IP Trunk Voice with GPZ-IPLE	10020~10531	UDP
SV9100 NetLink		
Primary System Signaling	58000/58001	TCP
Secondary System Signaling	58002	TCP
IP DSP Resource with GPZ-IPLE	10020~10531	UDP
NEC Proprietary SIP (SIP MLT)		
SIP MLT Signaling	5080	UDP
SIP Trunk Voice with GPZ-IPLE	10020~10531	UDP

SECTION 5 CCIS NETWORKING VIA IP (PEER-TO-PEER CONNECTIONS BASIS)

Description

IP-KCCIS supports Peer-to-Peer calls between IP Terminals residing in different offices, without using DSP resources. Two DSP resources in each office/system are consumed for calls between an IP Terminal and an IP Terminal.

The number in each box indicates how many VoIP resources are used		Main System					Remote System				
		TDM Terminal	IP Terminal	CO Analog/ Digital/SIP	CO Conf. IP Terminal	CO Conf. TDM Terminal	TDM Terminal	IP Terminal	CO Analog/ Digital/SIP	CO Conf. IP Terminal	CO Conf. TDM Terminal
Main System	TDM Terminal	0	M: 1	0	M: 1	0	M: 1 R: 1	M: 1	0	M: 1	M: 1 R: 1
	IP Terminal	M: 1	0	M: 1	M: 2	M: 1	R: 1	0	R: 1	M: 2	M: 2 R: 1
Remote System	TDM Terminal	M: 1 R: 1	R: 1	M: 1 R: 1	R: 1	M: 1 R: 1	0	R: 1	0	R: 1	M: 1 R: 1
	IP Terminal	M: 1	0	M: 1	R: 2	M: 1 R: 2	R: 1	0	R: 1	R: 2	M: 1 R: 2

M = Main K-CCIS System
R = Remote K-CCIS System

SECTION 6 SERVICE CONDITIONS

- ❑ For InMail remote CCIS extensions are not supported in a centralized directory.
- ❑ DT800/DT700 terminals are supported for Peer-to-Peer connections via a P2P CCIS call.
- ❑ Standard SIP terminals are not supported for Peer-to-Peer connection.
- ❑ If either Programs15-05-50 or 50-15-04 are set to 0 (Disable) in system A, Peer-to-Peer is disabled for system A and any remote systems when calling system A.
- ❑ When port translation is done through a NAT router, Peer-to-Peer is disabled.
- ❑ When RTP encryption is enabled, Peer-to-Peer is disabled.
- ❑ When connecting to other SV9100s using CCISoIP, Program 84-34-01 (Type 5 CCIS) must be set the same in all systems.
- ❑ SV9100 K-CCIS-IP to another SV9100, for calls from an IP terminal to a TDM terminal/trunk via Peer-to-Peer, the IP Terminal's CODEC must match the CCIS CODEC and the packet size is auto negotiated based on the receiving sides packet size.
- ❑ SV9100 K-CCIS-IP to another SV9100 or to a NEAX PBX (SV9300, SV9500, etc.), for calls from an IP terminal to another IP terminal/trunk via Peer-to-Peer, the IP Terminal’s CODEC must match and the packet size is auto negotiated.
- ❑ SV9100 K-CCIS-IP to a NEAX PBX (SV9300, SV9500, etc.), for calls from an IP terminal to a TDM terminal via Peer-to-Peer, the IP Terminal's CODEC and packet size (Program 84-24-XX) need to match the NEAX PBX CCIS CODEC and packet size settings.
- ❑ For this feature, the GPZ-IPLE is installed on the GCD-CP10. Only one GPZ-IPLE can be installed per system for a maximum of 256 VoIP resources.
- ❑ The audio quality of speech connections depends on the available bandwidth between the GPZ-IPLE 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 GCD-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.
- ❑ If single lines for fax machines are set to Special (PRG 15-03-03), faxing across IP CCIS will always use G.711 CODEC.
- ❑ SV9100 to SV9100, IP terminal to TDM via Peer-to-Peer, the IP Terminal's CODEC must match the CCIS CODEC and the packet size will be auto negotiated based on the receiving sides packet size.

- SV9100 to SV9100 or NEAX PBX (SV9300, SV9500, etc.), IP terminal to IP terminal via Peer-to-Peer, the IP Terminals' CODEC must match and the packet size is auto negotiated.
- SV9100 to NEAX PBX (SV9300, SV9500, etc.), IP terminal to TDM via Peer-to-Peer, the IP Terminal's CODEC and packet size (PRG 84-24-XX) need to match the NEAX PBX CCIS CODEC and packet size settings.
- Any calls across CCISoIP (station to station or stations transferring trunks) that use Quick Transfer to voice mail will require an extra CAP key. The initial call across the CCISoIP link will use the first CAP key and when the digit 8 is pressed, to perform the Quick Transfer, a second CAP key is accessed.
- Verified Account Codes for Toll calls across a CCIS network will not be restricted when a trunk access code is added to the number to allow ARS routing through another K-CCIS T1/IP networked site. This access code (typically a 9) precedes the dialed “1” which is used by the system to identify a LD call. As a result the call is no longer considered LD and the account code will not be required.
- All IP trunks (SIP, CCIS, or H.323) must be contiguous. If any IP trunks are added to a system that already has IP trunks installed, and the next set of trunks is not contiguous, then all IP trunks are moved to a new set of contiguous trunk numbers.

6.1 Restrictions

- Call Forward Busy No Answer must be disabled for a phone to receive a CCIS Call Back request.
- The UNIVERGE SV9100 can send billing information to a billing center office (NEAX2000/2400/SV9300), 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 GPZ-IPLE can support up to 256 VoIP resources dependent on the number of VoIP Channel Licenses installed on the GCD-CP10 which may be configured to support SIP MLT, 3rd Party SIP Stations and Trunks or IP K-CCIS Applications.
- A GCD-CP10 license is required for IP K-CCIS trunks.
- A GCD-CP10 license is required for SIP Clients to include SIP stations and SIP trunks.
- The GPZ-IPLE daughter board may configure VoIP DSP usage criteria with SV9100 Program 10-19-01 DSP Resources. Each GPZ-IPLE 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 GPZ-IPLE daughter board is installed on the GCD-CP10.
  - How many VoIP resources are Active (call in progress) and Reserved (call setup in progress).
  - The MAC Address of the GPZ-IPLE.

- 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.
- When using InMail in a CCIS or Netlink network, 8-digit extensions and mailboxes are not supported.

Book 2 – SV9100 IP Networking

General Information

Chapter 1

SECTION 1 UNIVERGE SV9100 IP NETWORKING

UNIVERGE SV9100 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 SV9100 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 SV9100 telephones in locations that would not normally be supported by multiline telephones.

UNIVERGE SV9100 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 SV9100 VoIP system.

Table 1-1 VoIP Specifications

Category	Feature	Notes
IP Address	DHCP Server	GCD-CP10
	DHCP Client	IP Phone
QoS	802.1p/1q	GCD-CP10
	L3 QoS (ToS)	Diffserv/IP Precedence
Maintenance	HTTP Server	GCD-CP10
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 400 Trunks
IP CCIS	IP CCIS Trunks	Maximum 400 Trunks

IP Networking

Chapter 2

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 SV9100 CCIS Network and SIP trunks. These methods are explained below.

Section 2 UNIVERGE SV9100 IP K-CCIS Network SV9100

The SV9100 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 SV9100 IP K-CCIS Network allows many offices to connect their UNIVERGE SV9100 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 SV9100 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.

SV9100 IP K-CCIS is explained in detail in the UNIVERGE SV9100 Key-Common Channel Interoffice Signaling (K-CCIS) section. Please refer to this for a complete description and installation instructions.

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 GPZ-IPLE daughter board in the UNIVERGE SV9100 system, external line ports are allotted in accordance with the number of Licensed ports for the particular IP Application.

The UNIVERGE SV9100 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 SV9100 Programming Manual. A basic list of the programming commands required for F-Route is shown in the example below.

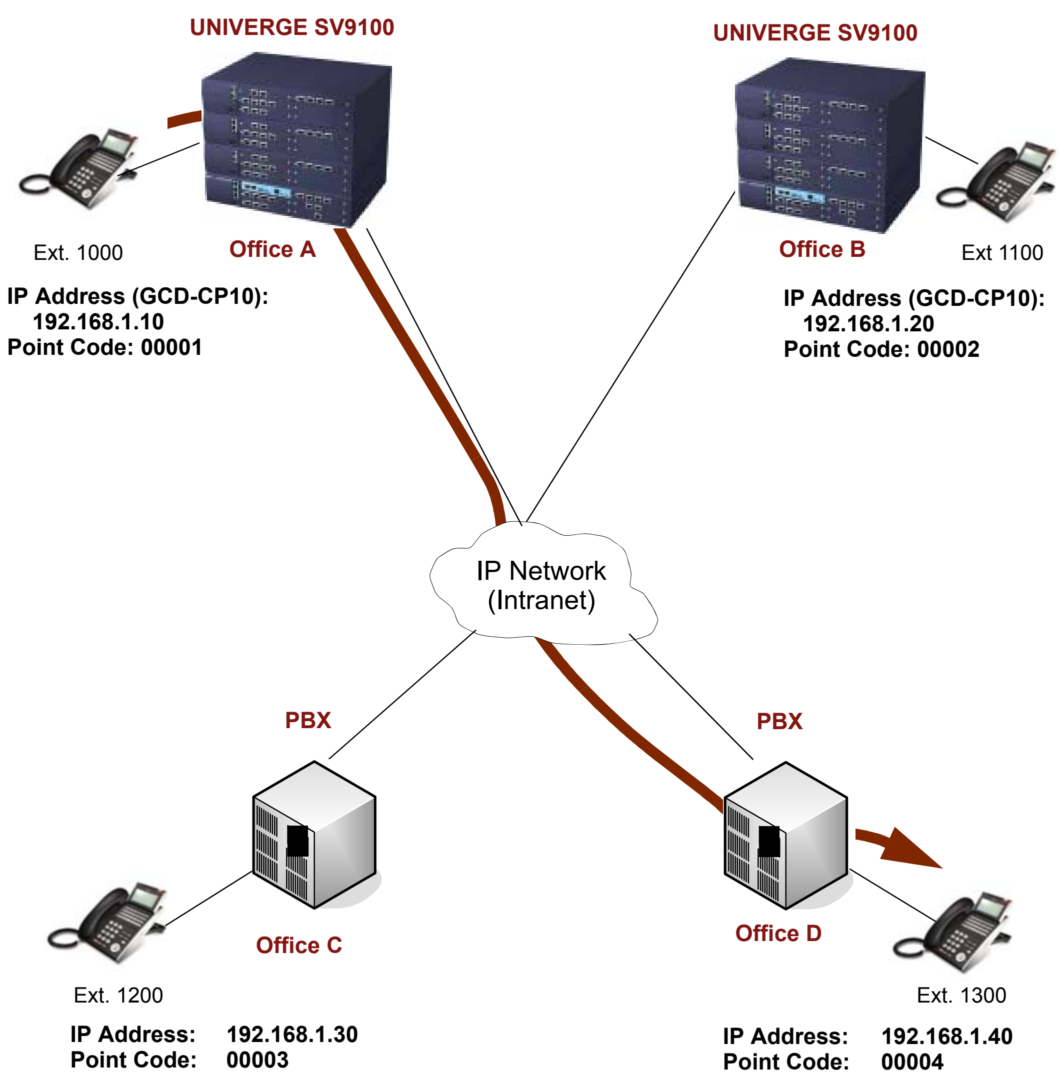
SECTION 4 EXAMPLE CONFIGURATIONS

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.

Figure 2-1 Example IP Network Configuration

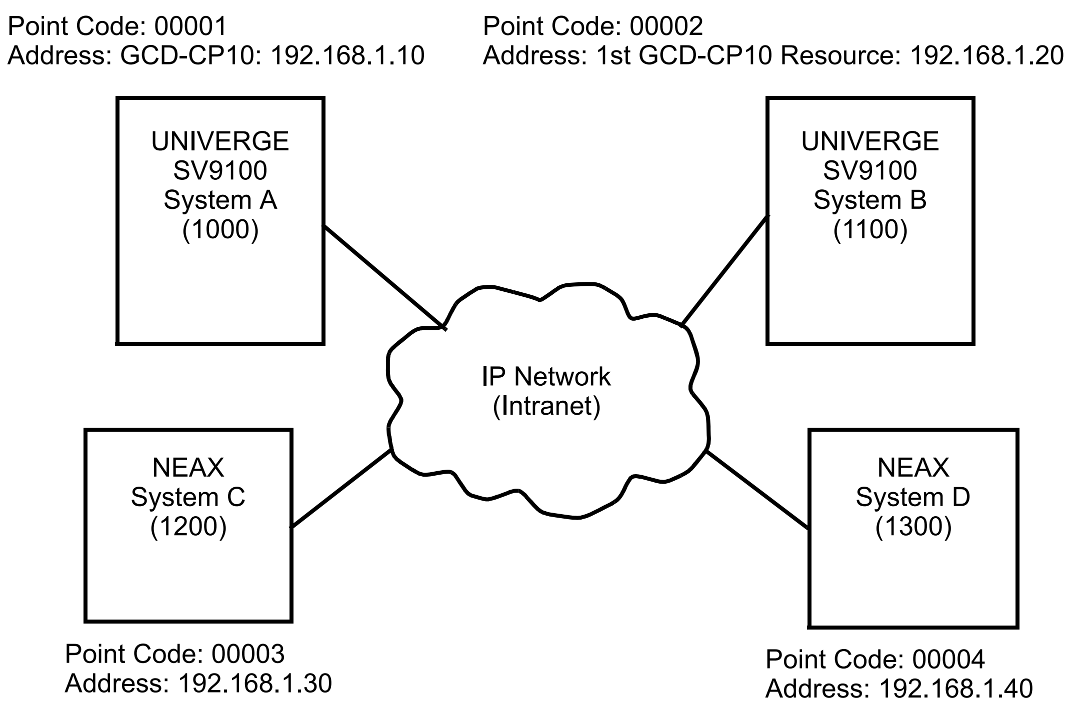


4.2 UNIVERGE SV9100 IP K-CCIS Programming Example 1

This is an example of UNIVERGE SV9100 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-xx.
- Each system has 16 multiline terminals.
- Each system has eight analog Trunks.

Figure 2-2 Programming Example 1



➨ *Systems C and D are NEAX systems. Refer to the applicable system manuals for programming details.*

NOTE

It is assumed that the systems were defaulted (first power on) with the following blades installed as described below.:

4.2.1 Card Interface Slot Assignment (PRG 10-03 ETU Setup)

The following table provides information for assigning the blade interface slots.

Slot	Card Type During 1st Power On	Card Type and Ports			
		System A		System B	
		Card Type	Ports	Card Type	Ports
1	GCD-CP10/GPZ-IPLE	GCD-CP10/GPZ-IPLE	Trunks 5 ~28	GCD-CP10/GPZ-IPLE	Trunks 5~28
2	GCD-8DLCA	GCD-8DLCA	Station 1~8	GCD-8DLCA	Station 1~8
3	GCD-8DLCA	GCD-8DLCA	Station 9~16	GCD-8DLCA	Station 9~16
4	None	None	N/A	None	N/A
5	GCD-4COTC	GCD-4COTC	Trunks 1~4	GCD-4COTC	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 Insert GPZ-IPLE on GCD-CP10 in Chassis 1, Slot 1. Verify PRG 10-03-01 (ETU Setup).	**Program 10-03-01** Insert GPZ-IPLE on GCD-CP10 in Chassis 1, Slot 1. 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 10-68 Assign the Trunk Type as CCIS with the Start Port as 5 and Number of Port 24.	**Program 10-68** Assign the Trunk Type as CCIS with the Start Port as 5 and Number of Port 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 (GPZ-IPLE) addresses.

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

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)
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	**PRG 50-06-02 CCIS Feature Availability - Centralized Day/Night Switching** 0 = Disable (default) 1 = Enable
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	**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
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	**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

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.	**PRG 50-08-01 CCIS Centralized BLF Sending Group Assignment – Destination Point Code** Select Group ID (1~8) + Destination Point Code. Default not assigned.
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	**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
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.	**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.
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	**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
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	**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

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 Port 1 ~ 99 = 101 ~ 199 Port 100 ~ 960 = 3101 ~ 3961
11-04-01	Virtual Extension Numbering	Assign Station Numbers to Port Numbers	Defaults for Ports Ports 1 ~ 99 = 201 ~ 299 Ports 100 ~ 512 = No setting

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, 3X = 2 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.
44-05-01	ARS/F-Route Table – Trunk Group Number	0 = No Setting 1 ~ 100 = Trunk Group from PRG14-05 101 ~ 150 = Networking 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 SV9100 CCISoIP network it may be necessary to configure the item below. This allows the sending UNIVERGE SV9100 to detect DTMF tones and convert them to data, then regenerate the DTMF tones at the receiving side.

➲ **84-34-01** **VoIPDB Setup DTMF Relay Mode (05-CCIS over IP)**
When using G.729, NEC recommends setting this item to (1) = RFC2833

General IP Configuration

Chapter 3

SECTION 1 INTRODUCTION

The UNIVERGE SV9100 system uses IP for various applications, including:

- ❑ System Programming
- ❑ Voice Over IP
- ❑ Switching

This section describes the procedure for connecting the UNIVERGE SV9100 system to an existing data network and configuring TCP/IP. This is the first step in implementing VoIP and other IP applications.

SECTION 2 NETWORK ADDRESSING OVERVIEW

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

2.1 IP Address

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

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

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

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

2.2 Subnet Mask

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

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

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

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

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

2.3 DHCP

Dynamic Host Configuration Protocol (DHCP) assigns a dynamic IP address. Network control may be easier with DHCP as there is no need to assign and program individual IP addresses for the LAN equipment. To use a dynamic IP address, a DHCP server must be provided. The UNIVERGE SV9100 system GCD-CP10 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 SV9100 system programming.

Note that the GCD-CP10 blade must always have a static IP address. This address is set in Program 10-12-01 : GCD-CP10 Network Setup – IP Address (default: 192.168.0.10).

SECTION 3 CONFIGURATION EXAMPLES

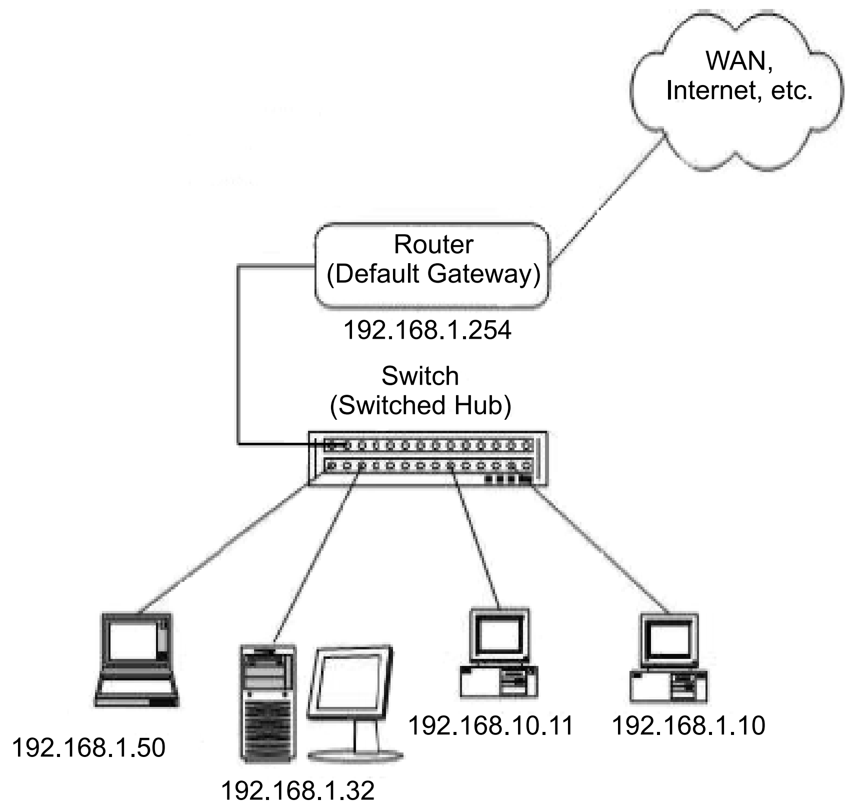
The following configuration examples illustrate a typical network configuration for an existing network that uses a static address and a typical configuration for a new network that uses a dynamic address.

3.1 Example Configuration 1 - Existing Network with Static Addressing

Figure 3-1 Example Configuration 1 - Existing Network with Static IP Address on page 3-3 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 SV9100 is added to the existing data network. The Network Administrator (or IT Manager) should provide the following:

- IP Address (for the GCD-CP10 blade)
- IP Addresses (for the GPZ-IPLE daughter board)
- Subnet Mask
- Default Gateway
- A spare switch/hub port

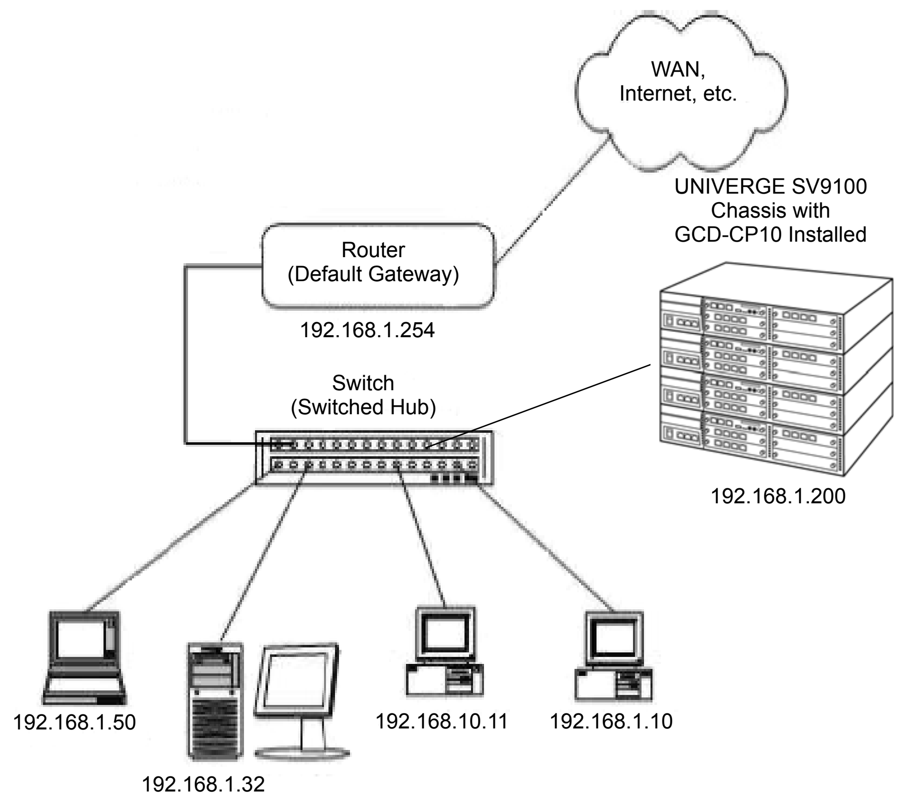
First, program the UNIVERGE SV9100:

- 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 GCD-CP10 blade Ethernet Port to the switch/hub port, using a standard Cat-5 patch cable. The UNIVERGE SV9100 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 SV9100 Chassis to the Network.

Figure 3-2 Example Configuration 1 - Adding the UNIVERGE SV9100 Chassis to the Network



3.2 Example Configuration 2 - New Network with Dynamic Addressing

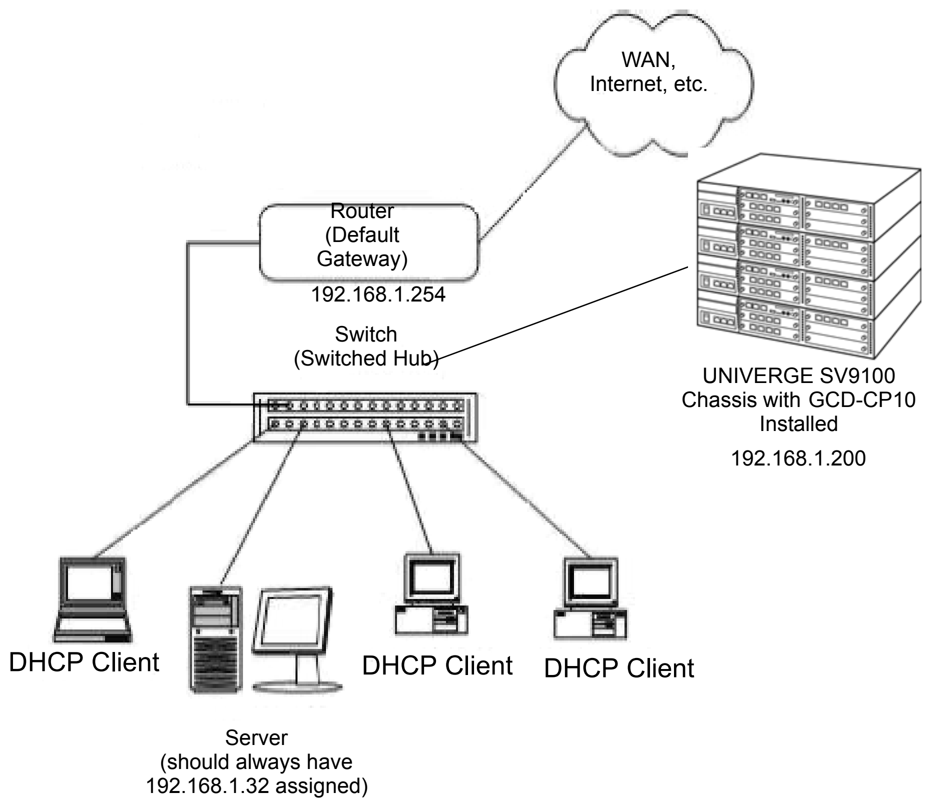
Figure 3-3 Example Configuration 2 - New Network with Dynamic Addressing on page 3-6 shows a typical network configuration using Dynamic IP Addressing, and the UNIVERGE SV9100 Internal DHCP server. In most cases, the customer would use an external DHCP server (for example on a Windows Server) or static addressing (as illustrated in Figure 3-2 Example Configuration 1 - Adding the UNIVERGE SV9100 Chassis to the Network on page 3-5). However, if the UNIVERGE SV9100 is to be installed on a new network the Network Administrator may want to use the UNIVERGE SV9100 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:

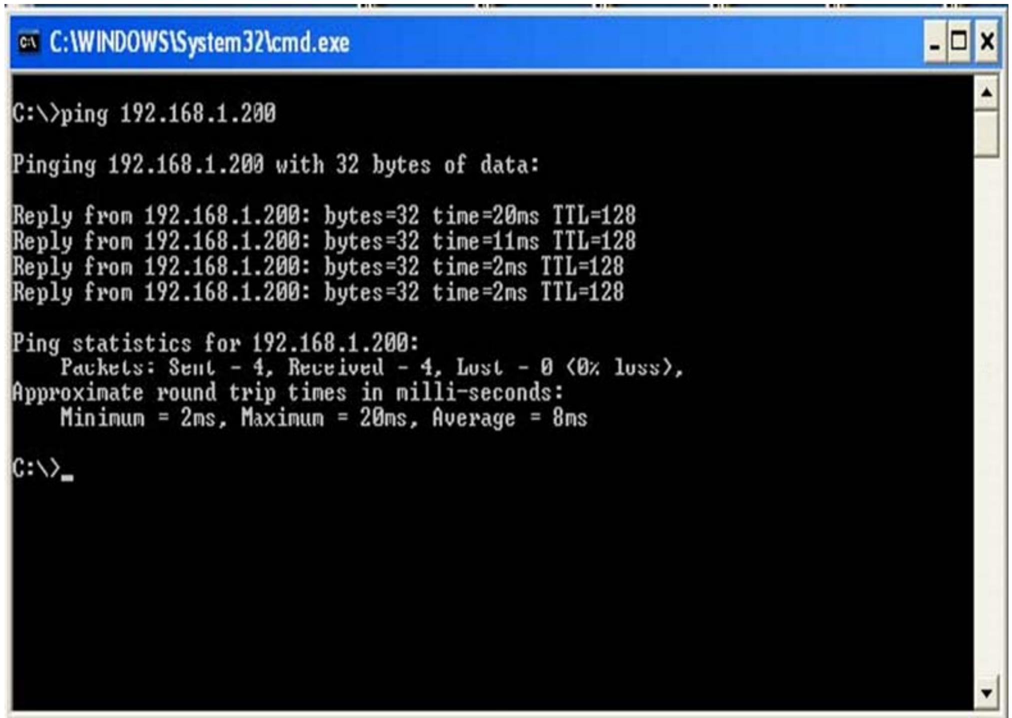
- IP Address (for the GCD-CP10 blade)
- IP Addresses (for the GPZ-IPLE)
- Subnet Mask
- Default Gateway
- Range(s) of IP Addresses to assign
- List of permanent IP addresses, with corresponding MAC Addresses
- A spare switch/hub port

Figure 3-3 Example Configuration 2 - New Network with Dynamic Addressing



Now connect the UNIVERGE SV9100 GPZ-IPLE Ethernet Port to the switch/hub port, using a standard CAT-5 patch cable. The UNIVERGE SV9100 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.

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.

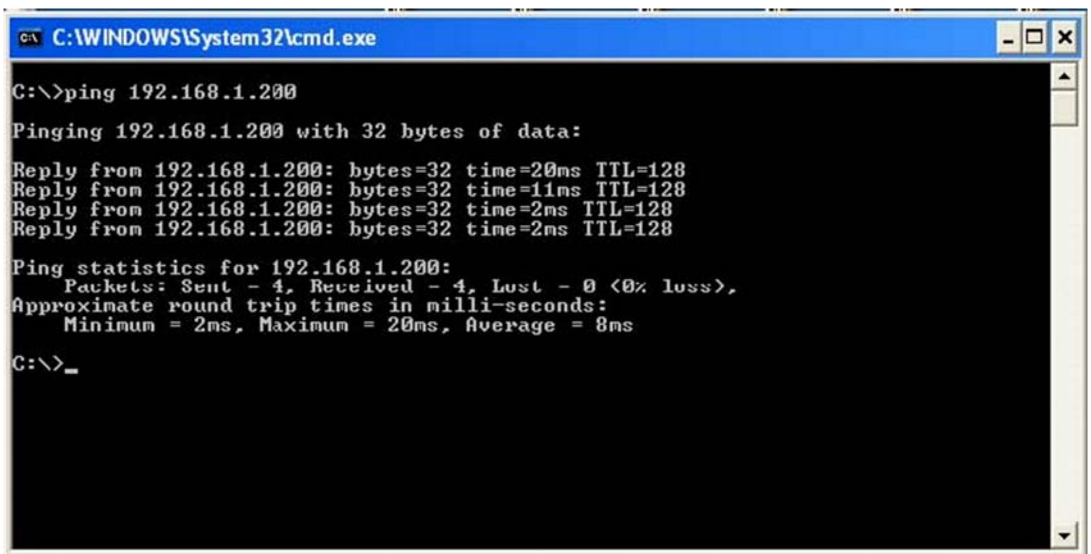
SECTION 4 TESTING THE UNIVERGE SV9100 NETWORK CONNECTION

To test the UNIVERGE SV9100 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 SV9100 system has replied to the Ping request – this indicates that the UNIVERGE SV9100 system is correctly connected to the network.

Figure 3-5 Testing the Network Connection



Programming

Chapter 4

SECTION 1 BEFORE YOU START PROGRAMMING

This chapter provides you with detailed information about the UNIVERGE SV9100 program blocks that may be required to connect the SV9100 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.

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
----	**----**	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 USB Drive is required. Insert the USB Drive into the GCD-CP10 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 4 Using Keys to Move Around in the Programs

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.

Table 4-1 Keys for Entering Data (Continued)

Keys for Entering Data	
Use this key...	**When you want to...**
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 preprogrammed 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.
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

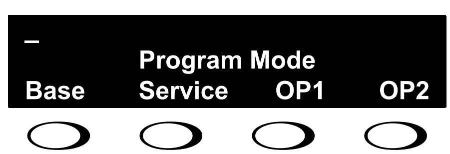
<table>
<tr><th>Use this keypad digit . . .</th><th>When you want to. . .</th></tr>
<tr><td>1</td><td>Enter characters:
1 @ [¥] ^ _ ` { | } ∅ ♦ Á À Â Ã Ç É Ê ì ó</td></tr>
<tr><td>2</td><td>Enter characters: A-C, a-c, 2.</td></tr>
<tr><td>3</td><td>Enter characters: D-F, d-f, 3.</td></tr>
</table>

Table 4-2 Keys for Entering Names

Use this keypad digit . . .	When you want to. . .
4	Enter characters: **G-I, g-i, 4**.
5	Enter characters: **J-L, j-l, 5**.
6	Enter characters: **M-O, m-o, 6**.
7	Enter characters: **P-S, p-s, 7**.
8	Enter characters: **T-V, t-v, 8**.
9	Enter characters: **W-Z, w-z, 9**.
0	Enter characters: 0 ! “ **#** $ % & ’ ( ) ô Õ ú ä ö ü α ε θ
✱	Enter characters: **✱** + , - . / : ; < = > ? π Σ σ Ω □ ¢ £
#	**#** = Accepts an entry (only required if two letters on the same key are needed – ex: TOM). Pressing **#** again = Space. (In system programming mode, use the right arrow Softkey instead to accept and/or add a space.)
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 SV9100 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.



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

Softkey Display Prompts	
If you press this Softkey . . .	**The system will. . .**
back	Go back one step in the program display. You can press VOLUME ▲ or VOLUME ▼ to scroll forward or backward through a list of programs.
↑	Scroll down through the available programs.
↓	Scroll up through the available programs.
select	Select the currently displayed program.
←	Move the cursor to the left.
→	Move the cursor to the right.
-1	Move back through the available program options.
+1	Move forward through the available program options.

Section 8 Programs

This sections describes the programs used to connect the SV9100 to a data network and to configure the VoIP functions.

Program 10 : System Configuration Setup

10-03 : ETU Setup

Level:
IN

Description

Use **Program 10-03 : ETU Setup** to setup and confirm the Basic Configuration data for each 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.

IMPORTANT

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**	0~960	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 3 = Bluetooth Cordless Handset 6 = PGD(2)-U13 (Paging) 7 = PGD(2)-U13 (Tone Ringer) 8 = PGD(2)-U13 (Door Box) 9 = PGD(2)-U13 (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 (1~960) 6 = PGD(2)-U13 (Paging) (1~8) 7 = PGD(2)-U13 (for Tone Ringer) (1~8) 8 = PGD(2)-U13 (for Door Box) (1~8) 9 = PGD(2)-U13 (for ACI) (1~96) 10 = DSS (1~32) 11 = --- Not Used ---	0
03	**--- Not Used ---**		

B-Channel 2			
Item No.	**Item**	**Input Data**	**Default**
06	**Terminal Type (B2)**	0 = Not set 6 = PGD(2)-U13 (Paging) 7 = PGD(2)-U13 (Tone Ringer) 8 = PGD(2)-U13 (Door Box) 9 = PGD(2)-U13 (ACI) 12 = APR(B2 Mode)	0
07	**Logical Port Number (B2)**	0 = Not set 6 = PGD(2)-U13 (Ext. Speaker) 7 = PGD(2)-U13 (Paging/Tone Ringer) (1~8) 8 = PGD(2)-U13 (for Door Box) = (1~8) 9 = PGD(2)-U13 (ACI) = (1~96) 12 = APR (for B2 mode) (193~512)	0
08	**Multiline Telephone Type**	0 = DT3** 1 = DT4**	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 4 = Not Used 5 = BCA	0

B-Channel 2			
Item No.	**Item**	**Input Data**	**Default**
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**	0~960	0
03	**Transmit Gain Level (S-Level)**	1~57 (-15.5 +12.5dB)	32 (0dB)
04	**Receive Gain Level (R-Level)**	1~57 (-15.5 +12.5dB)	32 (0dB)

For COTC Unit Setup

Physical Port Number	1~8

Item No.	Item	Input Data	Default
01	**Logical Port Number**	0~400	0

For ODTB PKG Setup

Physical Port Number	01~04

Item No.	Item	Input Data	Default
01	**Logical Port Number**	0~400	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 = 1~400 (LD Trunk) 1 = 1~960 (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:1~400 2 = S-Point:1~960 3 = NW Mode (Leased Line): 1~256 4 = NW Mode (Interconnected Line): 1~256 5 = NW Mode (Interconnected Line, Fixed Layer1=NT): 1~256 6 = S-Point (Leased Line): 1~960	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 (1~400)	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	**S-Point Connection Bus Mode**	0 = Extended Passive Bus 1 = Short Passive Bus	0
07	**S-Point DID Digit**	Range 0 ~ 4	0
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	1
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	**NW Mode Networking System**	Range 0 ~ 50	0

Item No	Item	Input Data	Default
14	**S-Point Service Protocol**	0 = Keypad Facility 1 = Specified Protocol	0
15	**S-Point Call Busy Mode**	0 = Alerting Message 1 = Disconnect Message	0
17	**ISDN Line Ringback Tone** If Telco does not provide ringback tone, SV9100 can if set to 1:Enable.	0 = Disable 1 = Enable	0
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	0
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	0
22	**--- Not Used ---**		
23	**Straight/Cross Wiring**	0 = Auto 1 = Manual (Cross) 2 = Manual (Straight)	0
24	**S-Point Power Feeding**	0 = Disable 1 = Enable	0
25	**S-Point DID Digit**	0 = Disable 1 = Enable	0

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 1~400	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	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	**--- 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	**--- Not Used ---**		
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	0
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	0
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.	0~400	0
02	**T1 Signal Format Selection**	0 = D4 (12 Multi Frame) 1 = ESF (24 Multi Frame)	0
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 IPLE PKG Setup

Physical Port Number	001~256

Item No.	Item	Input Data	Default
01	**VOIP Type**	0 = Not Used 1 = Not Used 2 = Not Used 3 = IPLE	None (will be determined when Installed)
02	**Number of Channels**	0 ~ 256 (Read Only)	0
03	**Number of Voice Channels**	0 ~ 256 (Read Only)	0

For VM00 PKG Setup

Physical Port Number	01~16

Item No.	Item	Input Data	Default
01	**Logical Port Number**	0~480	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.	0~400	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 = Internal MOH(IC) 1 = External MOH 2 = Service Tone 3 = VMDB	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)-U13 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**	1~57 (-15.5 ~ +12.5dB)	32 (0dB)	

Conditions

None

Feature Cross Reference

- ➥ Analog Communications Interface (ACI)
- ➥ Background Music
- ➥ Music on Hold

Program 10 : System Configuration Setup

10-12 : GCD-CP10 Network Setup

Level:
SA

Description

Use **Program 10-12 :** GCD-CP10 **Network Setup** to setup the IP Address, Subnet-Mask, and Default Gateway addresses.

Input Data

Item No.	Item	Input Data	Default	Description
01	IP Address	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	192.168.0.10	Set for GCD-CP10.
02	Subnet Mask	128.0.0.0 192.0.0.0 224.0.0.0 240.0.0.0 248.0.0.0 252.0.0.0 254.0.0.0 255.0.0.0 255.128.0.0 255.192.0.0 255.224.0.0 255.240.0.0 255.248.0.0 255.252.0.0 255.254.0.0 255.255.0.0 255.255.128.0 255.255.192.0 255.255.224.0 255.255.240.0 255.255.248.0 255.255.252.0 255.255.254.0 255.255.255.0 255.255.255.128 255.255.255.192 255.255.255.224 255.255.255.240 255.255.255.248 255.255.255.252 255.255.255.254 255.255.255.255	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 ~ 191.255.255.254 192.0.0.1 ~ 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 5 = 1Gbps, Full Duplex			0	NIC Auto Negotiate (GCD-CP10)
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 IPLE.
10	**Subnet Mask**	128.0.0.0	192.0.0.0	224.0.0.0	255.255.0.0	Set for IPLE.
		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			

Input Data (Continued)

Item No.	Item	Input Data	Default	Description
11	**NIC Setup**	0 = Auto Detect 1 = 100Mbps, Full-Duplex 3 = 10Mbps, Full-Duplex 5 = 1 Gbps, Full-Duplex	0	Set for IPLE.
13	**DNS Primary Address**	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0	Set for adding a function for DNS.
14	**DNS Secondary Address**			
15	**DNS Port**	0 ~ 65535	53	
17	**IPL NIC Port Setting**	0 = MDI 1 = MDI-X	0	

Conditions

None

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 GCD-CP10 blade.

Input Data

Item No.	Item	Input Data	Default	Description
01	**DHCP Server Mode**	0 = Disable 1 = Enable	0	Enable/Disable the built-in DHCP Server.
02	**Lease Time**	Days 0~255	0 day	Lease Time of the IP address to a client. ➨ *Press Transfer to increment to the next setting data.*
		Hour 0~23	0 hour	
		Minutes 0~59	30 minutes	
05	**Last DHCP Data**	0 = Disable 1 = Enable	1	If 10-13-01 is enabled, Enable/ Disable DHCP resource.

Conditions

- The System must be reset in order for these changes to take affect.

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	Description
01	**The range of the IP address to lease**	Minimum: 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	172.16.0.100	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.
		Maximum: 0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.254.255.254 192.0.0.1 ~ 223.255.255.254	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	1~960

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

Level:
SA

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
08	**Download Protocol** Set Download Protocol used for AutoConfig (for DT800/ DT700 Series).	Code number 0~255	43 (Fixed)
		Sub code number	163
		1 = FTP 2 = HTTP	1

Input Data (Continued)

Item No.	Item	Input Data	Default
09	**Encryption Information** Set an Encryption Information used for AutoConfig (for DT800/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
17	**SIP Server (Domain Name)**	Code number 0~255	120 (Fixed)
		Maximum 20 character strings	No setting

Input Data (Continued)

Item No.	Item	Input Data	Default
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
28	**Config File Name**	Code Number 0~255	43 (Fixed)
		Subcode Number 0~255	152 (Fixed)
		Up to 15 characters	No setting

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	01~256

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 = Networking/CCIS 4 = Use for NetLink 5 = Blocked 6 = Common without Unicast paging 7 = Multicast paging 8 = Unicast paging	Resource 1 = 1 Resource 2~256 = 0

Conditions

None

Feature Cross Reference

➥ None

Program 10 : System Configuration Setup

10-54 : License Configuration for Each Package

Level:
IN

Description

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

Input Data

Slot Number	1~24

License Index Number	1~32

Item No.	Item	Read Data
01	**License Code**	0000~9999
02	**License Quantity**	0~255

Conditions

❍ If applying more than 255 licenses to a slot the licenses must be applied across multiple indexes. For example assigning 256 VoIP resource licenses (5103) to the CPU slot could be assigned using different methods as long as the total for the CPU slot is 256:

1. Index 1 has 128 of feature code 5103 and index 2 also has 128 of feature code 5103 for a total of 256.
2. Index 1 has 255 of feature code 5103 and index 2 has 1 of feature code 5103 for a total of 256.

❍ When using IP devices IP Resource licenses (5103) must be assigned to the CPU Slot (1) for them to be available for use. If this is not done, IP related features will not work.

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**	1 = H.323 2 = SIP 3 = None 4 = DT800/DT700	3	Viewing Only – No changes permitted	
02	**Terminal MAC Address**	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.	
07	**Using IP Address**	0.0.0.0~255.255.255.255	0.0.0.0	Informational Only registered IP Phones	15-05-01
09	**Call Procedure Port**	0 ~ 65535	0		
11	**DT800/DT700 C/CTR Port**	0 ~ 65535	0		
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
16	**Authentication Password**	Up to 24 characters	None	Assign the authentication password for SIP single line telephones.	15-05-01

Item No.	Item	Input Data	Default	Description	Related Program
18	**IP Duplication Allowed Group**	0 = Disable 1 = Enable	0	When this program is 1 = Enable, the duplication of an IP address is allowed at the time of SIP/DT700 terminal registration.	10-03-09 15-05-22
19	**Side Option Information**	0 = No Option 1 = 8LK Unit 2 = 16LK Unit 3 = 24ADM 4 = EHS	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 3 = Not Used 4 = BCA	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
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

Item No.	Item	Input Data	Default	Description	Related Program
26	**D800/DT700 Terminal Type**	0 = Not Set 1 = ITL-**E-1D/IP-*E-1 2 = ITL-**D-1D/ITL-12BT1D/ITL- 12PA-1D [without 8LKI(LCD)-L] 3 = ITL-**D-1D/ITL-12BT-1D/ITL-12PA-1D [with 8LKI(LCD)-L] 4 = ITL-320C-1 5 = Softphone 6 = CTI 7 = AGW 8 = IP3AT-8WV 9 = Not Used 10 = ITL-**DG-3 11 = ITL-**CG-3 12 = ITL-2CR-1 13 = ITZ-**D-1D/ITZ-**PD-1D/ITZ-**pA-1D/ITZ-**DG/ITZ-**LDG 14 = ITZ-*CG 15 = ITZ-*DE 16 = ITZ-*LDE	0		
27	**Personal ID Index**	0~960	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		
30	**DTMF play during conversation at Receive Extension**	0 = Do Not Play 1 = Play	0		

Item No.	Item	Input Data	Default	Description	Related Program
31	**Alarm Tone during conversation** (RTP packet loss alarm)	0 = Not Ringing 1 = Ringing	1		
32	**Key Calling**	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).	
35	**Encryption Mode**	0 = Off 1 = On	0		
36	**DT800/DT700 Firmware Version**	00.00.00.00~FF.FF.FF.FF	00.00.00.00		
37	**DT800/DT700 Large LED Illumination Setup**	2 = Red 3 = Green 4 = Blue 5 = Yellow 6 = Purple 7 = Light Blue 8 = White 9 = Rotation	2	Sets LED color for internal Intercom call. In DT800/ DT700 local terminal setting menu, illumination setting must be 'Automatic', otherwise the terminal will ignore PRG 14-01-35, PRG 15-05-37 and PRG 15-23 settings.	
38	**Paging Protocol Mode**	0 = Multicast 1 = Unicast 2 = Auto	0	Sets the protocol mode for the Paging function.	
39	**CTI Override Mode**	0 = Disable 1 = Enable	0	Sets the override function against the terminal that is controlled by the CTI.	
40	**Calling Name Display Info via Trunk for Standard SIP**	0 = Both name and number 1 = Name only 2 = Number only 3 = None	0	Sets the incoming calling name display type on a standard SIP terminal. Trunk name is the first priority and abbreviated (SPD) name is second priority.	
41	**Time Zone (Hour)**	0~24 (-12~+12)	12	Sets the time difference from the system time set in Program 10-01. Input hour(s) based on this Program.	

Item No.	Item	Input Data	Default	Description	Related Program
43	**Video Mode**	0 = Disable 1 = Enable	0	This Program is used to select the video function with the standard SIP terminal. If the standard SIP terminal supports the video function, the SV9100 transfers the video CODEC in SDP information.	
44	**Using Standard SIP Display for CPN**	0 = Disable 1 = Enable	0	This Program is used to Enable or Disable the system to send INVITE from tag Display attribute which is sent from a standard SIP terminal as CPN to ISDN, and if there is no Display attribute from standard SIP terminal, the system will not refer to either PRG 21-12-01 or 21-13-01 and no CPN will be sent.	
45	**NAT Plug & Play**	0 = Disable 1 = Enable	0	Select sending RTP port number to remote router. (0) uses result from negotiation result, (1) from received RTP packet. Effective only when 10-46-14 is to NAT Mode.	10-46-14
46	**Door Phone Number (Read Only)**	0 = Not assigned 1 ~ 8 = Door Phone Number	0	Indicates automatically assigned IP Door Phone Number after system registers the Door Phone port. System assigns the number not to duplicate with the Door Phone connected to PGD(2)-U13. (Read Only)	10-03 (DLCA) PGD
47	**Registration Expiration Timer for NAT**	0 = Disable 60 ~ 65535(sec)	180	On a per station basis, this setting defines the SIP registration expiry timer. If this value is set to 0, for a NAPT terminal, the value in PRG 84-23- 01 is applied.	10-46-14
48	**Subscribe Expire Timer for NAT**	0 = Disable 60 ~ 65535(sec)	180	On a per station basis, this setting defines the SIP Subscribe expiry timer. If this value is set to 0, for a NAPT terminal, the value in PRG 84-23-02 is applied.	10-46-14

Item No.	Item	Input Data	Default	Description	Related Program
49	**Receiving SIP INFO**	0 = Disable 1 = Allowed any time 2 = Allowed while RTP is not available	1	Select whether or not system can receive DTMF from standard SIP phone via SIP INFO message. There are two receive types. 1='Allowed any time' can receive a SIP INFO message from a standard SIP phone as a dial information any time. '2=Allowed while RTP is not available' can receive a SIP INFO message before establishing RTP connection.	
50	**Peer to Peer Mode**	0 = Disable 1 = Enable	On	On a per station basis enables P2P.	

Conditions

- 15-05-04 – Nickname must be unique in the system.
- 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

84-09 : 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	0 = Disable (Off) 1 = Enable (On)	0
02	VLAN ID	1~4094	0
03	VLAN Priority	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

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 = SIP MLT 9 = SIP Trunk 10 = Net-Link 11 = Video RTP/RTCP

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

84-21 : CCIS over IP CODEC Information Basic Setup

Level:
IN

Description

Use **Program 84-21 : CCIS over IP CODEC Information Basic Setup** to set the CODEC parameters of the GPZ-IPLE.

Input Data

Item No.	Item	Input Data	Default
01	**Number of G.711 Audio Frames**	1~4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms	3
02	**G.711 Type**	0 = A-law 1 = μ-law	0
03	**G.711 Voice Activity Detection Mode**	0 = Disable 1 = Enable	0
04	**G.711 Jitter Buffer (min)**	0~300ms	30
05	**G.711 Jitter Buffer (average)**	0~300ms	60
06	**G.711 Jitter Buffer (max)**	0~300ms	120
07	**G.729 Audio Frame Number**	1~6 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms 5 = 50ms 6 = 60ms	3
08	**G.729 Voice Activity Detection Mode**	0 = Disable 1 = Enable	0
09	**G.729 Jitter Buffer (min)**	0~300ms	30
10	**G729 Jitter Buffer (average)**	0~300ms	60
11	**G729 Jitter Buffer (max)**	0~300ms	120

Input Data (Continued)

Item No.	Item	Input Data	Default
19	**1st Priority of Audio Capability**	0 = G.711 PT 2 = G.729 PT 3 = G.722 4 = G.726	0
20	**2nd Priority of Audio Capability**	0 = G.711 PT 2 = G.729 PT 3 = G.722 PT 4 = G.726 PT	2
22	**Jitter Buffer Mode**	1 = Static 3 = Self Adjusting	3
23	**Voice Activity Detection Threshold**	0 ~ 30 (-20dB ~ +10dB) 0 = -20dB (-50dBm) 1 = -19dBm (-49dBm) : 20 = 0dBm (-30dBm) : 29 = +9dBm (-21dBm) 30 = +10dBm (-20dBm)	20
27	**G.722 Audio Frame Number**	1~4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms	3
29	**G.722 Jitter Buffer (min)**	0~300ms	30
30	**G.722 Jitter Buffer (average)**	0~300ms	60
31	**G.722 Jitter Buffer (max)**	0~300ms	120
32	**G.726 Audio Frame Number**	1~4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms	3
33	**G.726 Voice Activity Detection Mode**	0 = Disabled 1 = Enabled	0
34	**G.726 Jitter Buffer (min)**	0~300ms	30
35	**G.726 Jitter Buffer (average)**	0~300ms	60
36	**G.726 Jitter Buffer (max)**	0~300ms	120
43	**RTP Filter** To avoid incorrect voice pass connection, this Program checks the sending side address from received RTP packet at VoIPDB.	0 = Disable 1 = Enable 2 = Enable (include SSRC)	1

Conditions

None

Feature Cross Reference

➥ Voice Over Internet Protocol (VoIP)

Program 84 : Hardware Setup for VoIP

84-33: Fax Over IP Setup

Level:
IN

Description

Use **Program 84-33 : Fax Over IP Setup** to set up the parameters of the Fax Over IP function.

Index 1

Type	1 = H.323 Trunk 2 = Networking 3 = SIP Trunk 4 = SIP Extension 5 = CCIS over IP 6 = NetLink

Input Data

Item No.	Item	Input Data	Default	Profile 1	Profile 2
01	**FAX Relay Mode**	0 = Disable 1 = Enable 2 = Each Port Mode	0		
02	**T.38 Protocol Mode**	1 = R/U 2 = U/R 3 = RTP 4 = UDPTL	1		
04	**Jitter Buffer (max)**	0 ~ 300	160		
05	**T.38 RTP Format Payload Number**	96 ~ 127	100		
06	**T.38 Fax Maximum Speed**	0 = V.27ter, 4800bps 1 = V.29, 9600bps 2 = V.17, 14400bps	2		
07	**T.38 Data Error Correction Mode**	0 = Redundancy 1 = FEC	0		
08	**T.38 Error protection depth for Signaling**	0 ~ 2	0		
09	**T.38 Error protection depth for Data**	0 ~ 2	0		

Input Data

Item No.	Item	Input Data	Default	Profile 1	Profile 2
10	**T.38 TCF Method**	1 = VOIPDB 2 = G3FE	1		
11	**T.38 ECM (Error Correction Mode)**	0 = Disable 1 = Enable	1		
12	**FAX CODEC**	1 = G.711 a-law 2 = G.711 u-law 3 = G.726	1		
13	**Payload Size**	1 ~ 4 (10ms base)	2		
14	**Jitter Buffer Mode**	1 = Static 2 = Self adjusting	1		
15	**Minimum Jitter Buffer**	0 ~ 300	80		
16	**Average Jitter Buffer**	0 ~ 300	120		
17	**Maximum Jitter Buffer**	0 ~ 300	160		
18	**FAX RTP Payload Type**	97 ~ 127	103		
19	**FAX over IP Type**	1 = Type 1 2 = Type 2	1		

Conditions

None

Feature Cross Reference

- ❏ Voice Over Internet Protocol (VoIP)

Program 90 : Maintenance Program

90-23 : Deleting Registration of IP Telephones

Level:
IN

Description

NOTE

This program is available only via telephone programming, Web 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**	00.00~15.15
03	**VoIPDB Version**	DEV/PR/REL-00.00 00.00.00.00~ FF.FF.FF.FF
04	**DSP Project Number**	00000000~ FFFFFFFF
05	**Vocallo F/W Version**	00.00.00.00~ FF.FF.FF.FF
06	**OCT1010ID Version**	00.00.00.00~ FF.FF.FF.FF

Conditions

- These Programs are ‘Read Only.’

Feature Cross Reference

None

Network Design Considerations

Chapter 5

SECTION 1 INTRODUCTION

This chapter explains some issues that should be considered when planning a UNIVERGE SV9100 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 SV9100 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 SV9100 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.

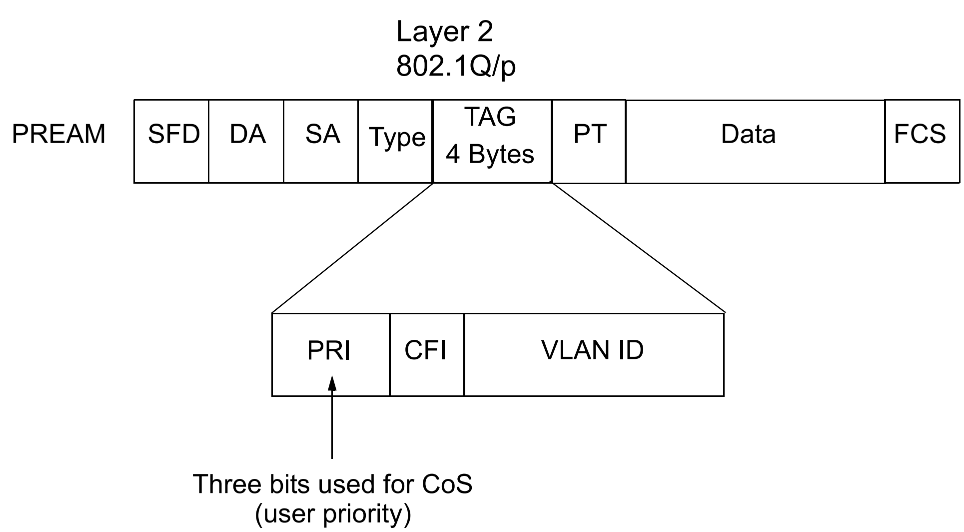
Therefore, the user can implement a Quality of Service mechanism on the IP network. At its most basic form, this denotes certain IP packets as being more important than others. So they would tell this 64Kbps line that IP packets with voice deserve a higher priority than those without voice. This allows 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 SV9100 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.

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 5-8.)

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 SV9100 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 SV9100 device to be open, or to place the SV9100 outside of the firewall.

Applications	Rx Port	UNIVERGE SV9100 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	84-26-02
Realtime Transport Control Protocol (RTCP)	10021	84-26-03

4.3 Virtual Private Network (VPN) Tunnelling

A Virtual Private Network is a private data network that maintains privacy through using a tunneling protocol and security procedures. Allowing for remote networks (including VoIP devices), which reside behind NATs and/or Firewalls to communicate freely with each other. In UNIVERGE SV9100 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.

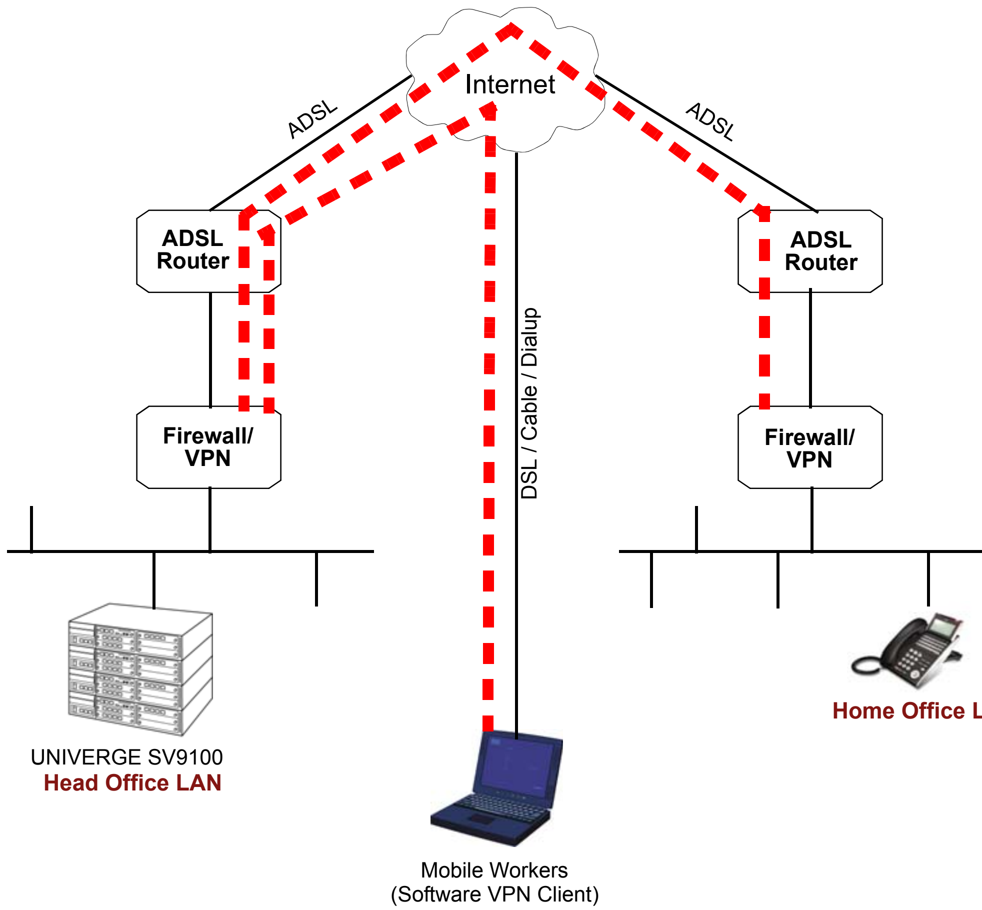
Companies today are exploring the use of VPN for a variety of connectivity solutions, such as:

- **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 SV9100 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 SV9100 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 SV9100 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 SV9100 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[2] score of 4.2 but uses a large bandwidth for transmission. This CODEC is not commonly used due to the bandwidth required, although it can be acceptable in LAN environment (i.e., IP Phones connected over a 100Mbps LAN).

- **G.722**
  G.722 is an ITU standard CODEC that provides 7kHz wideband audio at data rates from 48 to 64kbps. This is useful in a fixed network Voice Over IP applications, where the required bandwidth is typically not prohibitive, and offers a significant improvement in speech quality over older narrowband CODECS such as G.711, without an excessive increase in implementation complexity.

2. 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.

- **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 SV9100 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 SV9100 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 SV9100, 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 SV9100 programming. To do this, we tell the UNIVERGE SV9100 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 SV9100 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[3] 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.

DSP resources are provided by the GPZ-IPLE It can be difficult to work out how many DSP resources are required in an UNIVERGE SV9100 system, because:

- ❑ not all IP Extensions/trunks are used at the same time
- ❑ peer-to-peer calls do not use a DSP resource

GPZ-IPLE IP Addressing

The GPZ-IPLE requires two IP Addresses, one for Signaling (PRG 10-12-09), and one for the DSP Resources (PRG 84-26-01).

➨ *When assigning the IP addresses to the GPZ-IPLE 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 GPZ-IPLE 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 GPZ-IPLE NIC. For example, PC PRO, Web Pro, and ACD all terminate to the GPZ-IPLE NIC card when installed. Both the GPZ-IPLE 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, GPZ-IPLE or CPU, to route outside of its own network.*

The following chart shows the minimum and maximum number of IP addresses used with different GPZ-IPLE card configurations.

Card	Minimum IP Addresses	Maximum IP Addresses	Notes
GPZ-IPLE	1	2	The number of DSP channels depends on the VOIP license loaded to GCD-CP10 up to 256.

3. TDM = Time Division Multiplexing - traditional circuit-based telephony

To calculate the maximum DSPs required:

The manual calculations listed below are used in the UNIVERGE SV9100.

Non-Peer-to-Peer Mode (Program15-05-50=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) 4
```

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 5-4.

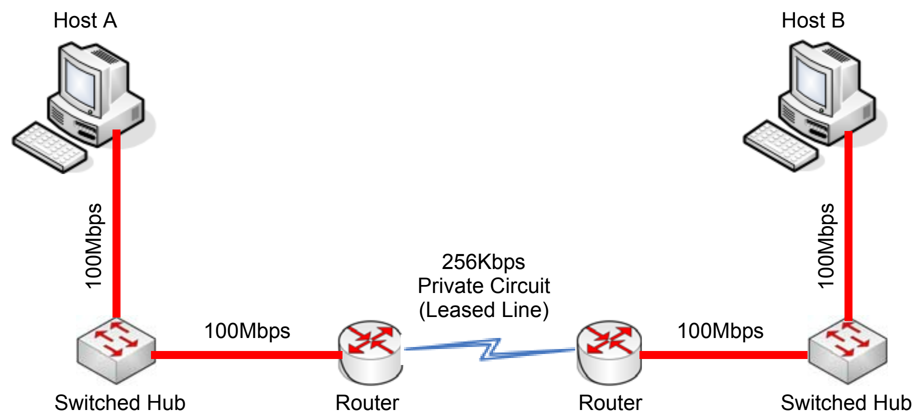
7.1 Prioritization

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.

4. 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.

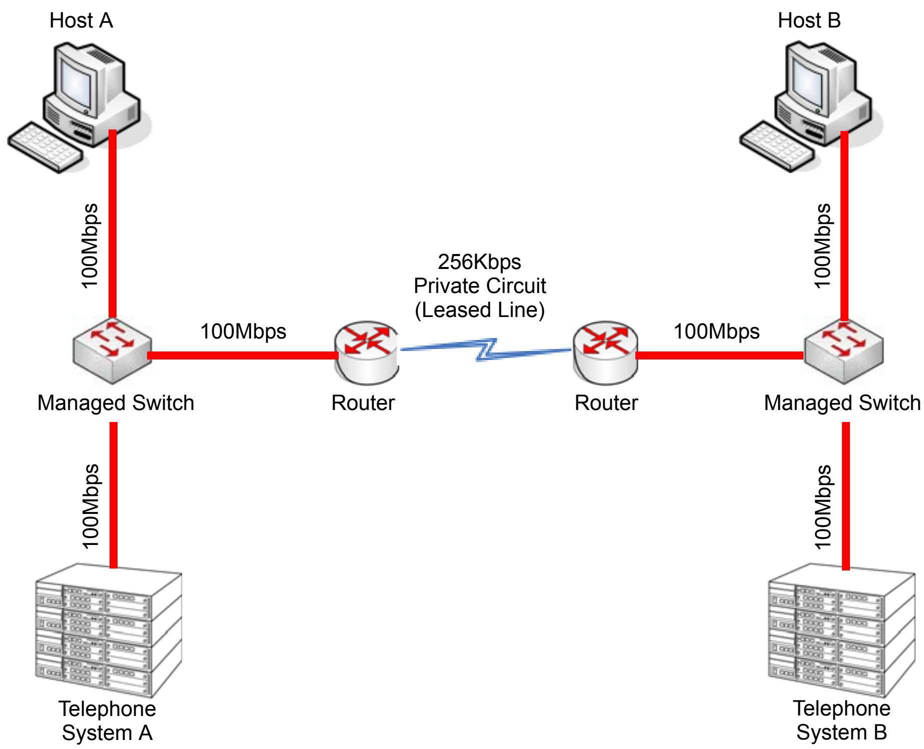
For most data applications this packet loss/delay is not critical.For example, a delay of one to five seconds to transmit an email is imperceptible. When VoIP is implemented, this loss/delay has a massive impact on the voice quality. The resulting gaps in speech, distortion and delay are unacceptable 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[5] 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.

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

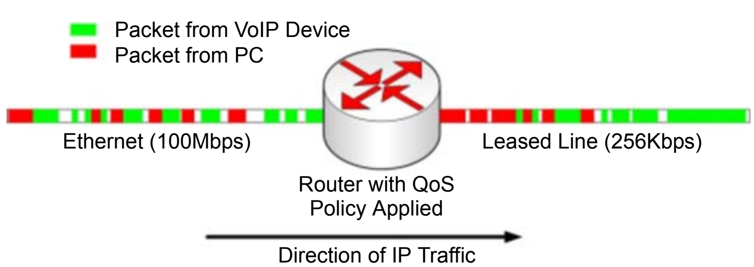
Figure 5-4 Voice and Data Network Implementation shows how a voice and data network can be implemented.

Figure 5-4 Voice and Data Network Implementation



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.

Figure 5-6 Protocol Structure for Layer 2 QoS

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

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

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 SV9100 in hex. When a frame has the EtherType equal to SV9100, 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.

Figure 5-7 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 (0xSV9100)
```

Figure 5-7 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.

Figure 5-8 Layer 3 QoS Example

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

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

Type of Service Field (Diffserv)

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

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

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

Listed below are the fields used in Figure 5-8 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
0	Normal Cost
1	Low Cost

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 5-23) 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.

Figure 5-9 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
```

Figure 5-9 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		
28	011100		AF32 (Assured Forwarding)

Table 5-3 IP Precedence and Diffserv Values Comparison (Continued)

DSCP Decimal	DSCP Binary	IP Precedence	Description
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		
56	111000	7	Class Selector 7
57	111001		
58	111010		

Table 5-3 IP Precedence and Diffserv Values Comparison (Continued)

DSCP Decimal	DSCP Binary	IP Precedence	Description
59	111011		
60	111100		
61	111101		
62	111110		
63	111111		

7.7 Programming QoS in the UNIVERGE SV9100 System

7.7.1 Marking Voice Traffic - Program 84-10-XX

Before programming the UNIVERGE SV9100 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 SV9100 Voice Protocols

The UNIVERGE SV9100 system supports the following types of VoIP traffic (refer to the Input Data section of Program 84-10 : ToS Setup on page 5-32).

Table 5-4 Voice Protocols

Number	Protocol Type	Description
1	DRS	Not Used
2	Protims	Not Used
3	Voice Control (H.245)	Communication from GPZ-IPLE to GPZ-IPLE
4	H.323	Communication from (TIE Lines) GPZ-IPLE to GPZ-IPLE -- or -- Communication from GPZ-IPLE to Network
5	RTP/RTCP	Voice (RTP) and Call Quality Data (RTCP)
6	SIP	Communication from GPZ-IPLE to Standard SIP device.
7	CCISoIP	CCIS Signaling Messages
8	SIP MLT	Communication from GPZ-IPLE to SIP Multiline Terminal.
9	SIP Trunks	Communication from GPZ-IPLE to Network
10	NetLink	Communication from GPZ-IPLE to GPZ-IPLE

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 5-32). The choices are:

Table 5-5 DIffserv Configuration

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

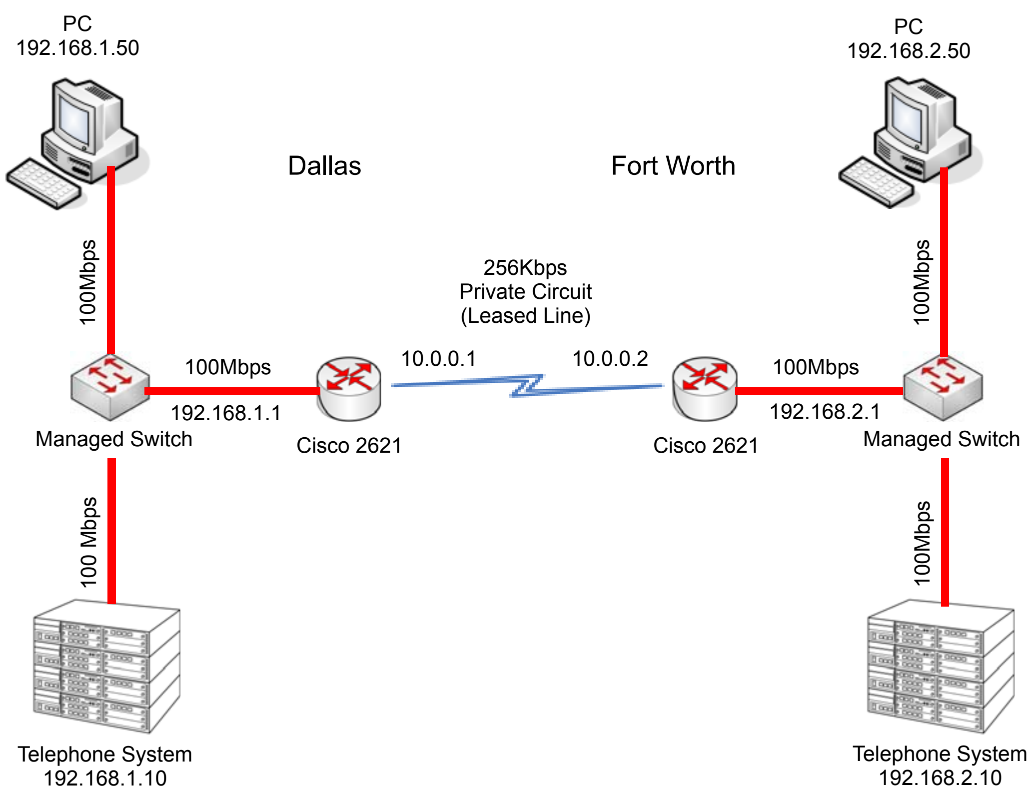
Figure 5-10 Common Network with Cisco Router shows a typical network scenario and an example of a Cisco router configuration.

➨ *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.*

IMPORTANT

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

Figure 5-10 Common Network with Cisco Router



Refer to Table 5-6 Cisco Router Configuration Example for configuration information about the Cisco 2621 router. A description of key commands follows.

Table 5-6 Cisco Router Configuration Example

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

Configuration Example Explanation:

1. Defines a Class Map called VoIPClass.
2. Matches any packets that have the ToS field set to IP Precedence 5 / DSCP 40 and assigns them to VoIPClass.
3. Defines a Policy Map called VoIPPolicy.
4. Creates a Class called VoIPClass and assigns this to the VoIPPolicy.
5. Allocates 50Kbps of bandwidth to the VoIPClass.

6 & 7. Determines that any data that does not match VoIPClass should be processed using the “fair-queue” method (i.e., No Prioritization).

8. Determines the amount of bandwidth available on the Serial interface – essential for the QoS calculations.
9. Applies the VoIP Policy to any packets that exit the serial interface. This means that data being received (input) does not use this policy.

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 = SIP MLT 9 = SIP Trunk 10 = NetLink 11 = Video RTP/RTCP

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

Item No.	Item	Input Data	Default	Description
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

The following lists the port numbers for supported IP applications.

Table 5-7 Port Designations for IP Applications

IP Application	IP Port Numbers	Comments
SIP Trunk		
SIP Trunk Signaling	5060	UDP
SIP Trunk Voice with GPZ-IPLE	10020~10531	UDP
3rd Party SIP		
SIP SLT Signaling	5070	UDP
SIP SLT Voice with GPZ-IPLE	10020~10531	UDP
H.323 Trunk		
H.323 Signaling	1718/1719/1720	TCP
H.323 Trunk Voice with GPZ-IPLE	10020~10536	UDP
SV9100 IP - K-CCIS		
IP K-CCIS Signaling	57000/5900	TCP

Table 5-7 Port Designations for IP Applications (Continued)

IP Application	IP Port Numbers	Comments
IP Trunk Voice with GPZ-IPLE	10020~10531	UDP
SV9100 NetLink		
Primary System Signaling	58000/58001	TCP
Secondary System Signaling	58002	TCP
IP DSP Resource with GPZ-IPLE	10020~10531	UDP
NEC Proprietary SIP (SIP MLT)		
SIP MLT Signaling	5080	UDP
SIP Trunk Voice with GPZ-IPLE	10020~10531	UDP

SIP Trunking

Chapter 6

SECTION 1 DESCRIPTION

The UNIVERGE SV9100 IP Trunk SIP 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.

With the SV9100 you can have two SIP Profiles allowing you to connect to two different SIP Carries, or allow you to have a SIP System Interconnection and connection to a SIP Carrier.

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.

The GPZ-IPLE Daughter Board interface can provide IP trunks and Tie Lines that can operate in the following modes:

- COI
- COID
- DID
- TLI
- DTI

Depending on the requirements and resource allocation in the LAN/WAN/Internet, the GPZ-IPLE - SIP can be configured to use any of the following voice compressions:

- G.711 u/A-Law – Highest Bandwidth
- G.729 (a) – Most often used
- The LAN/WAN or Internet connection is provided by a 10 Base-T/100 Base-TX Ethernet.

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

Conditions

- ❍ The option to set the SIP trunk Codec to G711 or G729 Fixed is supported in Program 84-13-28.
- ❍ A maximum of 400 IP Trunks are supported in the SV9100.
- ❍ The SV9100 supports G.711 or T.38 for FAX.
- ❍ The SV9100 does not support fallback to G.711 from G.729/G.726 for data (FAX) calls.
- ❍ A transferred call can not use T.38 at the transferred destination.
- ❍ SIP trunks are assigned in increments of four.
- ❍ Calling Party Name is not provided for outgoing calls on SIP trunks.
- ❍ With SV9100 software and GPZ-IPLE daughter board installed, half duplex connections are not supported. For troubleshooting purposes, a managed switch capable of port mirroring is required to capture packet data from the SV9100 IPLE Ethernet port.
- ❍ The SV9100 software enables multiple SIP trunk carriers to be utilized when NetLink is configured. Refer to Chapter 10 SV9100 NetLink for additional details.
- ❍ Each SIP Profile will require a different SIP Listen Port assigned in Program 84-14-06.
- ❍ Each SIP Profile will have to connect using one route because the SV9100 can have only one Gateway.
- ❍ If SIP Trunks are programmed in a NetLink Secondary system, only SIP Profile 1 can be utilized.
- ❍ The SV9100 can only use one DNS Server, two SIP Carriers that require their own DNS Server can't be used.
- ❍ VBD is supported on the GPZ-IPLE
- ❍ VBD on SIP trunk is only supported on SLT terminals.
- ❍ VBD supports G.711/G.729 (PCMU/PCMA) CODEC.
- ❍ The selection of (FAX or VBD) mode is on a per terminal basis.

  ❍ ***The FAX connection may not work properly when in VBD Mode.***

- ❍ VBD supports one way switching from the Voice session to VBD. VBD to Voice session is not supported. When VBD session ends, the session is closed.
- ❍ The VBD feature is not dependent on Carrier Type (Program 10-29-14).
- ❍ VBD is only supported on analog terminals and SIP trunks within the same system.

- VoIPDB cancels the VAD and Echo canceller automatically when changed into the VBD CODEC.
- RTP forwarding (Program 10-26-02) is not running if VAD is enabled.
- When using VAD on SDP (**Version 7000 or higher**) the setting is effective for G.711 and G.729 CODEC types.
  - *The FAX connection may not work properly when in VBD Mode.*
- VBD supports one way switching from the Voice session to VBD. VBD to Voice session is not supported. When VBD session ends, the session is closed.
- The VBD feature is not dependent on Carrier Type (Program 10-29-14).
- VBD is only supported on analog terminals and SIP trunks within the same system.
- VoIPDB cancels the VAD and Echo canceller automatically when changed into the VBD CODEC.
- RTP forwarding (Program 10-26-02) is not running if VAD is enabled.
- When using VAD on SDP the setting is effective for G.711 and G.729 CODEC types.

Default Settings

None

System Availability

Terminals

All Multiline Terminals

Required Component(s)

- GPZ-IPLE
- R2 Enhancement License (0412)
- System Port License (0300)
- VoIP Resource License (5103)
- IP Trunk License (5001)

SECTION 2 SIP TRUNK E.164 SUPPORT

Description

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

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

Number dialed = **00441202223344**

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

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

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

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

This feature uses the following SIP header fields:

Request-URI

To

From

P-Asserted Identity

P-Preferred Identity

Conditions

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

Default Settings

Disabled

System Availability

Terminals

All Multiline Terminals

Trunks

IP SIP

Required Component(s)

- GCD-CP10
- GPZ-IPLE
- R2 Enhancement License (0412)
- System Port License (0300)
- VoIP Resource License (5103)
- IP Trunk License (5001)

SECTION 3 SIP TRUNK E.164 CLIP ENHANCEMENT

Description

With the SIP Trunk E.164 CLIP Enhancement enabled, when an incoming SIP call from an external ITSP is presented at the system with a **+** in the From header field as the international access code, it is recognized and displayed as an international call at the terminal display and also logged in the terminals incoming caller history, allowing any outbound calls made from a multiline terminals caller history possible using this numbering scheme.

This presentation can be a requirement of certain SIP ITSPs (Internet Telephony Service Providers) so it is necessary the PBX can handle these calls and modify any SIP messages to the correct format accordingly.

Conditions

- E.164 Enhancement is applied for the SIP trunk interface.
- Outgoing call from caller history of incoming calls is only possible from multiline terminals.
- Netlink systems deployed in multiple countries using this feature may not work correctly because the system will not know which international code should be added at each node.

Default Settings

Disabled

System Availability

Terminals

All Multiline Terminals

Trunks

IP SIP

Required Component(s)

- ❍ GCD-CP10
- ❍ GPZ-IPLE
- ❍ R2 Enhancement License (0412)
- ❍ System Port License (0300)
- ❍ VoIP Resource License (5103)
- ❍ IP Trunk License (5001)

SECTION 4 VIDEO SUPPORT OVER SIP TRUNKS

Description

The SV9100 can support video calling over SIP interconnection trunks. The IP Trunk license (5001/5103), IP Station licenses (5111), R2 Enhancement license (0412) and Video MCU license (0042) are required.

Conditions

- ❍ Calls over SIP Interconnection while in P2P mode cannot be put on hold.
- ❍ Calls over SIP Interconnection while in P2P mode cannot be transferred, i.e. an internal call cannot be transferred to a SIP Interconnection trunk.
- ❍ A video call cannot be changed to a voice call. A voice call cannot be

changed to a video call.

- A video caller cannot use CTI/OAI at the same time as the CTI/OAI feature needs P2P to be set to off.
- When the video interconnection using a SIP trunk is configured, other SIP connections, such as a SIP carrier connection is not supported in the same system.
- Video capability in the initial invite message is required for the Video Terminal.
- When using an MCU, the SV9100 requires the Carrier Type Setting (Program 10-29-14) to be set to 0 = Standard.
- When using an MCU, the same video capability must be set between the MCU and the Video SIP Terminal.

Default Settings

This feature is not enabled at default.

System Availability

Terminals

- Polycom HDX4003
- Polycom VVX1500D

Required Component(s)

GCD-CP10 with GPZ-IPLE installed.

Related Features

None

Section 5 Guide to Feature 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.

IP Trunk – (SIP) Session Initiation Protocol:

Program Number	Program Name/Description	Input Data	Default	Level 1	Level 2	Level 3
10-12-03	**GCD-CP10 Network Setup – Default Gateway** Assign the default gateway IP address for the GCD-CP10.	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	✓		
10-12-05	**GCD-CP10 Network Setup – NIC Interface** Set up the NIC Interface.	0 = Auto Detect 1 = 100Mbps, Full Duplex 2 = 100Mbps, Half Duplex 3 = 10Mbps, Full Duplex 4 = 10Mbps, Half Duplex 5 = 1Gbps, Full Duplex	0	✓		
10-12-07	**GCD-CP10 Network Setup – NAPT Router IP Address (Default Gateway [WAN])** Define the IP Address of the WAN side of the router. ➨ *Only used when Program 10-20-21 is enabled.*	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		✓	
10-12-08	**GCD-CP10 Network Setup – ICMP Redirect** When receiving ICMP redirect messages, this determines if the IP Routing Table updates automatically or not.	0= (Enable) 1= (Disable)	0	✓		
10-12-09	**GCD-CP10 Network Setup – IP Address** Set IP address for GPZ-IPLE.	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	✓		
10-12-10	**GCD-CP10 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	255.255.0.0		✓	

Program Number	Program Name/Description	Input Data	Default	Level		
				1	2	3
10-12-11	**GCD-CP10 Network Setup – NIC Setup** Define the LAN interface Speed and Mode of the VoIP Application supported. ➨ *IPLE daughter board does not support half duplex connection.*	0 = Auto Detect 1 = 100Mbps, Full Duplex 3 = 10Mbps, Full Duplex 5 = 1Gbps, Full Duplex	0	✓		
10-19-01	**VoIP DSP Resource Selection** Select type of GPZ-IPLE DSP Resource. This program setting has no affect on the terminal/ trunk port assignment or usage.	0 = Common use for both IP extensions and trunks 1 = IP Extension 2 = SIP Trunk 3 = CCIS/Networking 4 = Use for NetLink 5 = Blocked 6 = Common without Unicast Paging 7 = Multicast Paging 8 = Unicast Paging	0	✓		
10-23-01	**SIP System Interconnection Setup – System Interconnection** Determine if the system is interconnected to another system.	0 = No (Disable) 1 = Yes (Enable)	0	✓		
10-23-02	**SIP System Interconnection Setup – IP Address** Define the IP Address of the SIP System Interconnection.	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	✓		
10-23-04	**SIP System Interconnection Setup – Dial Number** Define Dial Number for the SIP System Interconnection.	Maximum of 12 digits.	No Setting	✓		
10-23-06	**SIP System Interconnection Setup – SIP Profile** Assign the Interconnection to a SIP Profile.	1 = Profile 1 2 = Profile 2	1	✓		
10-28-01	**SIP System Information Setup – Domain Name** Define the Domain name. This information is generally provided by the SIP carrier.	Maximum of 64digits. (Profile 1 ~ Profile 2)	No Setting	✓		
10-28-02	**SIP System Information Setup – Host Name** Define the Domain name. This information is generally provided by the SIP carrier.	Maximum of 48 digits. (Profile 1 ~ Profile 2)	No Setting	✓		
10-28-03	**SIP System Information Setup – Transport Protocol** Define the Transport type. This option is always set to UDP.	0 = UDP 1 = TCP (Profile 1 ~ Profile 2)	0	✓		

Program Number	Program Name/Description	Input Data	Default	Level 1	Level 2	Level 3
10-28-05	**SIP System Information Setup – Domain Assignment** 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, then the domain is: L0.cbeyond.net and the host name is SIPconnect-sca.	0 = IP Address 1 = Domain Name (Profile 1 ~ Profile 2)	0	✓		
10-28-06	**SIP System Information Setup – IP Trunk Port Binding** Enable/Disable IP Trunk Port binding.	0 = Disable 1 = Enable (Profile 1 ~ Profile 2)	0	✓		
10-29-01	**SIP Server Information Setup – Default Proxy (Outbound)** 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 an SIP Server and the SIP Server is removed or not used, the entries in Program 10-29-xx must be set back to their default settings. Even if Program 10-29-01 is set to 0 (off), the UNIVERGE SV9100 still checks the settings in the remaining 10-29 programs.*	0 = Off 1 = On (Profile 1 ~ Profile 2)	0	✓		
10-29-02	**SIP Server Information Setup – Default Proxy (Inbound)** Define the Default Proxy (Inbound).	0 = Off 1 = On (Profile 1 ~ Profile 2)	0	✓		
10-29-03	**SIP Server Information Setup – Default Proxy IP Address** 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.	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 (Profile 1 ~ Profile 2)	0.0.0.0	✓		
10-29-04	**SIP Server Information Setup – Default Proxy Port Number** Define the Proxy Port Number.	0 ~ 65535 (Profile 1 ~ Profile 2)	5060	✓		
10-29-05	**SIP Server Information Setup – Registrar Mode** This program is used to Enable/Disable the ability to Register to the ITSP. ➨ *In Registration mode (IP or Domain Name) this MUST be enabled.*	0 = None 1 = Manual (Profile 1 ~ Profile 2)	0	✓		

Program Number	Program Name/Description	Input Data	Default	Level		
				1	2	3
10-29-06	**SIP Server Information Setup – Registrar IP Address** Define the Registrar IP Address. The carrier may provide an IP address. In most cases, a domain name is used so this entry is left at the default.	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 (Profile 1 ~ Profile 2)	0.0.0.0	✓		
10-29-07	**SIP Server Information Setup – Registrar Port Number** Define the Registrar Port Numbers.	0 ~ 65535 (Profile 1 ~ Profile 2)	5060	✓		
10-29-11	**SIP Server Information Setup – Registrar Domain Name** Define the Registrar Domain Name (normally provided by the SIP carrier). For example: mysipserver.sipprovider.com	Maximum of 128 characters. (Profile 1 ~ Profile 2)	No Setting	✓		
10-29-12	**SIP Server Information Setup – Domain Name** Define the domain name of the SIP PROXY Server provided by the SIP Carrier. For example: if the SIP Proxy server address is proxy.sipprovider.com, you would assign **sipprovider.com** in this program. ➨ *If no SIP Proxy address is provided, use the SIP Registration address as the proxy address.*	Maximum of 64 characters. (Profile 1 ~ Profile 2)	No Setting	✓		
10-29-13	**SIP Server Information Setup – Proxy Host Name** Assign the Proxy Host Name of the SIP PROXY Server provided by the SIP Carrier. For example: if the SIP Proxy server address is proxy.sipprovider.com, you would assign **proxy** in the program. ➨ *If no SIP Proxy address is provided, use the SIP Registration address as the proxy address.*	Maximum of 48 characters. (Profile 1 ~ Profile 2)	No Setting	✓		
10-29-14	**SIP Server Information Setup – SIP Carrier Choice** Define the SIP Carrier Choice. ➨ *Selecting Carrier B automatically sets Program 10-29-16 to on (1). Program 10-29-16 MUST be set to off for incoming calls to route using the lowest available trunk port.* ➨ *Each certified vendor my use a different carrier type. Visit NTAC website (http://www.necntac.com) to verify the proper setting per vendor.*	0 ~ 26 1 ~ 26 = Carrier Type A ~ Carrier Type Z (Profile 1 ~ Profile 2)	0	✓		

Program Number	Program Name/Description	Input Data	Default	Level 1	Level 2	Level 3
10-29-15	**SIP Server Information Setup – Registration Expiry (Expire) Time** This program defines the SIP Trunk Registration timer. This timer is negotiated between the Carrier and the SV9100 during the registration process. The Carrier will make the final decision on the value to be used which means the value specified in this program may be ignored. ➨ *When half of this timer expires, the SV9100 will re-register itself with the Carrier.*	120 ~ 65535 seconds (Profile 1 ~ Profile 2)	3600	✓		
10-29-20	**SIP Server Information Setup – Authentication Trial** Define the number of times the SV9100 will attempt to authenticate before timing out and not completing the registration process. ➨ *Recommend changing this to at least 2 in case the first attempt did not succeed due to network problems or any other issues.*	0 ~ 9 0 = No Authentication (Profile 1 ~ Profile 2)	1 (1 Authentication Attempt)	✓		
10-29-21	**SIP Server Information Setup – NAT Router** If the SV9100 is connecting to the SIP Carrier using NAT Translations, this setting must be enabled.	0 = Disabled 1 = Enabled (Profile 1 ~ Profile 2)	0	✓		
10-36-01	**SIP Trunk Registration Information Setup – Registration** Enable/Disable the SIP trunk registration.	0 = Disable 1 = Enable (Profile 1 ~ Profile 2)	0	✓		
10-36-02	**SIP Trunk Registration Information Setup – User ID** Assign the SIP User ID provided by your SIP Carrier. In most cases this is your 10 digit main billing number. This field is also used to send outbound caller ID information when not programmed on a per station or per trunk basis. ➨ *For non-registration and SIP Tie Lines to another system, you **MUST** have a USER ID entered.*	Maximum of 32 characters. (Profile 1 ~ Profile 2)	No Setting	✓		
10-36-03	**SIP Trunk Registration Information Setup – Authentication User ID** Define the Authentication USER ID for the SIP Trunk.	Maximum of 64 characters. (Profile 1 ~ Profile 2)	No Setting	✓		
10-36-04	**SIP Trunk Registration Information Setup – Authentication Password** Define the Authentication Password for the SIP Trunk.	Maximum of 32 characters. (Profile 1 ~ Profile 2)	No Setting		✓	
10-37-01	**UPnP Setup – UPnP Mode** Enable/Disable UPnP.	0 = Disable 1 = Enable	0	✓		

Program Number	Program Name/Description	Input Data	Default	Level		
				1	2	3
10-37-02	**UPnP Setup – Retry Time** Define the retry time for UPnP.	0, 60 ~ 3600 (1 ~ 59 cannot be input)	60	✓		
10-54-01	**License Configuration for Each Package – License Code** Assign VoIP Resource Licenses (5103) to GCD-CP10 slot (1).	1 ~ 255 Resource Licenses	0	✓		
10-67-01	**SIP Stack Configuration Setup – DNS Mode** Select whether DNS mode is enabled.	0 = Disable 1 = Enable	0	✓		
10-67-02	**SIP Stack Configuration Setup – DNS IP Address** Set the IP Address of DNS Server. The SV9100 can only use one DNS Server, two SIP Carriers requiring their own DNS Server can't be used.	XXX.XXX.XXX.XXX	0.0.0.0	✓		
10-67-03	**SIP Stack Configuration Setup – DNS Port** Assign the DNS port.	0 ~ 65535	53		✓	
10-67-04	**SIP Stack Configuration Setup – DNS Source Port** Assign the DNS source port.	0 ~ 65535	53		✓	
10-68-01	**IP Trunk Availability – Trunk Type** Assign the trunk type as (1) SIP.	0 = None 1 = SIP 2 = H.323 3 = CCIS (NetLink System ID 0 ~ 50)	0	✓		
10-68-02	**IP Trunk Availability – Start Port** Assign the Start Port for your SIP trunks.	0 ~ 400 (NetLink System ID 0 ~ 50)	0	✓		
10-68-03	**IP Trunk Availability – Number of Port** Assign the number of SIP port trunks.	0 ~ 400 (NetLink System ID 0 ~ 50)	0	✓		
11-01-01	**System Numbering – Service Code** Set the system internal (Intercom) numbering plan. The numbering plan assigns the first and second digits dialed and affects the digits an extension user must dial to access other extensions and features, such as service codes and trunk codes. If the default numbering plan does not meet the site requirements, use this program to tailor the system numbering to the site.		Refer to the Programming Manual for default values.	✓		
14-01-24	**Basic Trunk Data Setup – Trunk-to-Trunk Outgoing Caller ID through Mode** Enable/Disable the ability to send the original Caller ID through when the call is Forward Off-Premise.	0 = Disable (No) 1 = Enable (Yes)	0		✓	
14-02-01	**Analog Trunk Data setup – Signaling Type (DP/DTMF)** Set the outgoing signaling type for the tie trunk.	0 = Dial Pulse (10 PPS) 1 = Dial Pulse (20 PPS) 2 = DTMF	2			✓

Program Number	Program Name/Description	Input Data	Default	Level		
				1	2	3
14-05-01	**Trunk Group – Trunk Group Number** Assign trunks to trunk groups.	Trunk Port 1 ~ 400 = Priority 1 ~ 400	Default = Trunks 1 ~ 400 assigned to trunk group 1 with priorities equal to the trunk number. Trunk 1 = Priority 1 Trunk 400 = Priority 400.	✓		
14-18-05	**IP Trunk Data Setup – SIP Profile (SIP Trunk)** Assign each SIP Trunk to either Profile 1 or Profile 2.	Profile 1 Profile 2	Profile 1	✓		
15-03-18	**Single Line Telephone Basic Data Setup – Select Special Terminal** Used for selecting Special terminal type (FAX or Modem). This setting influences how data is transmitted via SIP trunk. ➨ *Program 15-03-03 must be set to 1 (Special) to use this feature.*	Type: 0 = FAX 1 = Modem	0		✓	
21-17-01	**IP Trunk (SIP) Calling Party Number Setup for Trunk** Assign 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 Program 21-17 and Program 21-18/ 21-19, the system uses the entry in Program 21-18/21-19.	0 ~ 9, ✱, # Maximum of 16 digits	No Setting	✓		
21-19-01	**IP Trunk (SIP) Calling Party Number Setup for Extension** 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 Program 21-18/Program 21-19, the system uses the data in Program 21-18/Program 21-19.	0 ~ 9, ✱, # Maximum of 16 digits (Profile 1 ~ Profile 2)	No Setting	✓		

Program Number	Program Name/Description	Input Data	Default	Level		
				1	2	3
22-02-01	**Incoming Call Trunk Setup** Assign the incoming trunk type for each trunk.	Trunks 1 ~ 400 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	0	✓		
44-02-01	**Dial Analysis Table for ARS/F-Route Access – Dial** Set the Dial digits for the Pre-Transaction Table for selecting ARS/F-Route (eight digits max: 1 ~ 9, 0 ✱ #, @). To enter a wild card/don't care digit, press Line Key 1 to enter an @.	Maximum of eight digits. (Use line key 1 for a 'Don't Care' digit, @)	No Setting		✓	
44-02-02	**Dial Analysis Table for ARS/F-Route Access – Service Type** Set the Service Type (0 ~ 3) for the Pre-Transaction Table for selecting ARS/F-Route.	0 = No Setting (None) 1 = Extension Call (Own) 2 = ARS/F-Route Table (F-Route) 3 = Dial Extension Analyze Table (Option)	0		✓	
44-02-03	**Dial Analysis Table for ARS/F-Route Access – Additional Data** If a Service Type is selected in Program 44-02-02, set the additional data, if required, for the Pre-Transaction Table for selecting ARS/F-Route (24 digits max: 1 ~ 9, 0 ✱ #, @). To enter a wild card/don't care digit, press Line Key 1 to enter an @.	1 = Delete Digit = 0 ~ 255 (255: Delete All Digits) 2 = 0 ~ 500 (0 = No Setting) 3 = Dial Extension Analyze Table Number = 0 ~ 4 (0 = No Setting)	0		✓	
44-05-01	**ARS/F-Route Table – Trunk Group Number** Assign the trunk group to be used by the ARS/F-Route Table.	0-100,101-150,255 0 = Not Set 1~100 = Trunk Group from PRG14-05 101~150 = Networking 255 = Extension Call Default is 0	0		✓	
44-05-09	**ARS/F-Route Table – Maximum Digit** Input the maximum number of digits to send when using the F-Route.	0 ~ 24	0		✓	
84-10-01	**ToS Setup – ToS Mode** When Input Data is set to 1, Protocol 7 is invalid. When Data is set to 2, Protocols 2 ~ 6 are invalid.	0 = Disable (Invalid) 1 = IP Precedence 2 = Diffserv	0	✓		

Program Number	Program Name/Description	Input Data	Default	Level		
				1	2	3
84-10-02	**ToS Setup – Priority, IP Precedence** 1 = Router queuing priority.	0 ~ 7 0 = Low 7 = High	0	✓		
84-10-03	**ToS Setup – Low Delay** 1 = Optimize for low delay routing.	0 ~ 1 0 = Normal Delay, Low Delay	0	✓		
84-10-04	**ToS Setup – Wideband (Throughout)** 1 = Optimize for high bandwidth routing.	0 ~ 1 0 = Normal Throughput 1 = High Throughput	0	✓		
84-10-05	**ToS Setup – High Reliability** 1 = Optimize for reliability routing.	0 ~ 1 0 = Normal Reliability 1 = Low Reliability	0	✓		
84-10-07	**ToS Setup – Priority (D.S.C.P. - Differentiated Services Code Point)** DSCP (Differentiated Services Code Point).	0 ~ 63	0	✓		
84-13-01	**SIP Trunk CODEC Information Basic Setup – Number of G.711 Audio Frames** Set the G.711 Audio Frame Number.	1 ~ 4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms (Profile 1 ~ Profile 2)	2	✓		
84-13-02	**SIP Trunk CODEC Information Basic Setup – G.711 Voice Activity Detection Mode** Enable/Disable the G.711 VAD Detection Mode.	0 = Disable 1 = Enable (Profile 1 ~ Profile 2)	0	✓		
84-13-03	**SIP Trunk CODEC Information Basic Setup – G.711 Type** Define the G.711 type.	0 = A-law 1 = μ-law (Profile 1 ~ Profile 2)	0	✓		
84-13-04	**SIP Trunk CODEC Information Basic Setup – G.711 Jitter Buffer (min)** Set the minimum G.711 Jitter Buffer.	0 ~ 300ms (Profile 1 ~ Profile 2)	20	✓		
84-13-05	**SIP Trunk CODEC Information Basic Setup – G.711 Jitter Buffer (Average)** Set the average G.711 Jitter Buffer.	0 ~ 300ms (Profile 1 ~ Profile 2)	40	✓		
84-13-06	**SIP Trunk CODEC Information Basic Setup – G.711 Jitter Buffer (max)** Set the maximum G.711 Jitter Buffer.	0 ~ 300ms (Profile 1 ~ Profile 2)	80	✓		

Program Number	Program Name/Description	Input Data	Default	Level 1	Level 2	Level 3
84-13-07	**SIP Trunk CODEC Information Basic Setup – Number of G.729 Audio Frames** Set the G.729 Audio Frame Number.	1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms 5 = 50ms 6 = 60ms (Profile 1 ~ Profile 2)	2	✓		
84-13-08	**SIP Trunk CODEC Information Basic Setup – G.729 Voice Activity Detection Mode** Enable/Disable the G.729 VAD Detection Mode.	0 = Disable 1 = Enable (Profile 1 ~ Profile 2)	0	✓		
84-13-09	**SIP Trunk CODEC Information Basic Setup – G.729 Jitter Buffer (min)** Set the minimum G.729 Jitter Buffer.	0 ~ 300ms (Profile 1 ~ Profile 2)	20	✓		
84-13-10	**SIP Trunk CODEC Information Basic Setup – G.729 Jitter Buffer (Average)** Set the average G.729 Jitter Buffer.	0 ~ 300ms (Profile 1 ~ Profile 2)	40	✓		
84-13-11	**SIP Trunk CODEC Information Basic Setup – G.729 Jitter Buffer (max)** Set the maximum G.729 Jitter Buffer.	0 ~ 300ms (Profile 1 ~ Profile 2)	80	✓		
84-13-17	**SIP Trunk CODEC Information Basic Setup – Jitter Buffer Mode** Set the Jitter Buffer Mode.	1 = Static 2 = Self Adjusting	3	✓		
84-13-18	**SIP Trunk CODEC Information Basic Setup – VAD Threshold** Set the VAD (Voice Activity Detection) threshold.	0 ~ 30 = -19dB ~ +10dB 1 =-19dB (-49dBm) : 2 =0dB (-30dBm) : 29 =9dBm (-21dBm) 30 =0dBm (-20dBm) (Profile 1 ~ Profile 2)	20	✓		
84-13-28	**SIP Trunk CODEC Information Basic Setup – Audio Capability Priority** Define the Codec Priority.	0 = G.711_PT 2 = G.729_PT 3 = G.722_PT 4 = G.726_PT 6 = G711_Fix 7 = G729_Fix (Profile 1 ~ Profile 2)	0	✓		

Program Number	Program Name/Description	Input Data	Default	Level		
				1	2	3
84-13-33	**SIP Trunk CODEC Information Basic Setup – Number of G.722 Audio Frames** Define the number of G.722 Audio Frames.	1 ~ 4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms (Profile 1 ~ Profile 2)	3		✓	
84-13-35	**SIP Trunk CODEC Information Basic Setup – G.722 Jitter Buffer (min)** Define the minimum level for the G.722 jitter buffer.	0 ~ 300ms (Profile 1 ~ Profile 2)	30		✓	
84-13-36	**SIP Trunk CODEC Information Basic Setup – G.722 Jitter Buffer (average)** Define the average level for the G.722 Jitter Buffer.	0 ~ 300ms (Profile 1 ~ Profile 2)	60		✓	
84-13-37	**SIP Trunk CODEC Information Basic Setup – G.722 Jitter Buffer (max)** Define the Max level for the G.722 Jitter buffer.	0 ~ 300ms (Profile 1 ~ Profile 2)	120		✓	
84-13-38	**SIP Trunk CODEC Information Basic Setup – Number of G.726 Audio Frames** Define the number of G.726 audio frames.	1 ~ 4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms (Profile 1 ~ Profile 2)	3		✓	
84-13-39	**SIP Trunk CODEC Information Basic Setup – G.726 VAD Mode** Enable/Disable the VAD mode for G.726.	0 = Disable 1 = Enable (Profile 1 ~ Profile 2)	0		✓	
84-13-40	**SIP Trunk CODEC Information Basic Setup – G.726 Jitter Buffer (min)** Define the minimum level for the G.726 jitter buffer.	0 ~ 300ms (Profile 1 ~ Profile 2)	30		✓	
84-13-41	**SIP Trunk CODEC Information Basic Setup – G.726 Jitter Buffer (average)** Define the average level for the G.726 jitter buffer.	0 ~ 300ms (Profile 1 ~ Profile 2)	60		✓	
84-13-42	**SIP Trunk CODEC Information Basic Setup – G.726 Jitter Buffer (max)** Define the max level for the G.726 jitter buffer.	0 ~ 300ms (Profile 1 ~ Profile 2)	120		✓	
84-13-49	**SIP Trunk CODEC Information Basic Setup – RTP Filter** To avoid incorrect voice pass connection, this program checks the sending side address from received RTP packet at VoIPDB.	0 = Disable 1 = Enable (Profile 1 ~ Profile 2)	0		✓	

Program Number	Program Name/Description	Input Data	Default	Level		
				1	2	3
84-13-65	**SIP Trunk CODEC Information Basic Setup – VAD Negotiation on SDP**	Used to determine the VAD method for setting VAD information on SDP. This is effective when VAD is enabled on each CODEC. ➨ *Supports G.711 and G.729.*	0 = Disable 1 = Enable (default = 0)		✓	
84-13-66	**SIP Trunk CODEC Information Basic Setup – Voice Band Data (VBD)**	Used to Disable or Enable VBD. ➨ *Program 15-03-03 must be set to 1 (Special) to use this feature.*	0 = Disable 1 = Enable (default = 0)		✓	
84-13-67	**SIP Trunk CODEC Information Basic Setup – VBD Payload Type**	Used to specify the payload type for VBD.	96~127 (default = 97)		✓	
84-14-06	**SIP Trunk Basic Information Setup – SIP Trunk Port Number** Set the SIP UA (User Authorized) Trunk port number (Receiving Transport for UNIVERGE SV9100 SIP). ➨ *Each SIP Profile will need to have a different SIP Listen Port.*	1 ~ 65535 (Profile 1 ~ Profile 2)	Profile 1 = 5060 Profile 2 = 5062	✓		
84-14-07	**SIP Trunk Basic Information Setup – Session Timer Value** The Session Timer, when enabled, will allow periodic refreshing of the SIP session using the SIP Re-Invite message. The refreshing will allow both the SV9100 and the Carrier to determine if the session is still active. The periodic refresh can be used to release calls that are not active anymore when the SIP BYE message may have been lost. ➨ *When this timer is set to 0, the session timer is NOT included in the SIP Invite message.*	1 ~ 65535 seconds (Profile 1 ~ Profile 2)	0	✓		
84-14-08	**SIP Trunk Basic Information Setup – Minimum Session Timer Value** This timer controls the minimum accepted value of the Session timer that the SV9100 will allow. If the Session timer from the Carrier is lower than the value defined here, the SV9100 will deny the call.	1 ~ 65535 seconds (Profile 1 ~ Profile 2)	1800	✓		
84-14-09	**SIP Trunk Basic Information Setup – Called Party Information** This Program is used for inbound calls only. When set to a 0 (Request-URI), the SV9100 will look into the SIP "Request-URI" field to route the inbound call. When set to a 1 (To Header), the SV9100 will look into the SIP "To" field to route the inbound call.	0 = Request URI 1 = To Header (Profile 1 ~ Profile 2)	0	✓		

Program Number	Program Name/Description	Input Data	Default	Level		
				1	2	3
84-14-10	**SIP Trunk Basic Information Setup – URL Type** Define the URL type for SIP trunks.	0 = SIP-URL 1 = TEL-URL	0	✓		
84-14-11	**SIP Trunk Basic Information Setup – URL/To HeaderSetting Information** When set to a 0 (Proxy Server Domain), the SV9100 will use the proxy settings in Programs 10-29-12 and 10-29-13 within the SIP Request-URI and To headers. If neither of these programs are assigned, the value in program 10-12-11 is used. When set to a 1 (SIP UA Domain), the SV9100 will use the domain settings in Program 10-28-02 within the SIP Request-URI and To headers.	0 = Proxy Server Domain 1 = SIP UA Domain (Profile 1 ~ Profile 2)	0			
84-14-15	**SIP Trunk Basic Information Setup – 100rel Settings** This program specifies if the 100rel message is included or not. When set to a 0 (Use Default Settings), the 100rel will be included in the initial SIP Invite and any provisional 1XX responses (excluding the 100 trying). When set to a 1 (Use opposite settings), the 100rel will NOT be included in the initial SIP Invite and provisional 1XX responses (excluding the 100 trying)	0 = Use Default Settings (100rel included) 1 = Use Opposite Settings (100rel not included) (Profile 1 ~ Profile 2)	0			
84-26-01	**IPL Basic Setup – IP Address** Assign the IP address for each DSP on the GPZ-IPLE.	xxx.xxx.xxx.xxx	172.16.0.20	✓		
84-26-02	**IPL Basic Setup – RTP Port Number** Assign the RTP port number to be used for each DSP on the GPZ-IPLE. ➨ *Only even numbered ports are supported.*	0 ~ 65534	10020	✓		
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.	0 ~ 65534	10021	✓		
90-10-01	**System Alarm Setup – Alarm Type** Define if Alarms are Minor, Major, or Not Set.	0 = Not Set 1 = Major Alarm 2 = Minor Alarm	No Setting	✓		

SIP Trunk E.164 Support

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

SIP Trunk E.164 CLIP Enhancement

Program Number	Program Name/Description	Input Data	Default	Level 1	Level 2	Level 3
10-02-01	**Location Setup – Country Code** Enter the country code.	Dial (maximum of four digits) 0 ~ 9, *, #	1	✓		
10-02-02	**Location Setup – International Access Codes** Enter the international access code.	Dial (maximum of four digits) 0 ~ 9, *, #	No Setting	✓		
10-02-03	**Location Setup – Other Area Access Code** Enter the other area access code.	Dial (maximum of two digits) 0 ~ 9, *, #	9	✓		

Program Number	Program Name/Description	Input Data	Default	Level 1	Level 2	Level 3
84-14-13	**SIP Trunk Basic Information Setup – Incoming/ Outgoing SIP Trunk for E.164** When this data is set to 1, then for any outbound SIP calls a **+** is added as a prefix to the Request-URI, To and From header fields of the SIP message. When it is set to 2 then if the dialed international access code matches the value in Program 10-02-02 this value is removed from the number dialed and the **+** added as a prefix to the Request-URI, To and From header fields of the SIP Message.	0 = Off 1 = Mode 1 2 = Mode 2 3 = Mode 3 (Profile 1 ~ Profile 2)	0	✓		
84-14-16	**SIP Trunk Basic Information Setup – SIP Trunk SIP-URI E.164 Incoming Mode** When this data is set to 1, then for any inbound SIP calls that include a **+** and a country code not defined in Program 10-02-01, delete the **+** and add the International Access Code in Program 10-02-02. If the country code is a match then delete both the **+** and country code but do not add the International Access code. When it is set to 2, then for any inbound SIP calls that include a **+** and a country code not defined in Program 10-02-01, delete the **+** and add the International Access Code in Program 10-02-02. If the country code is a match then delete both the **+** and country code and add the Caller ID Edit Code from Program 10-02-03.	0 = Disable 1 = Mode 1 2 = Mode 2 (Profile 1 ~ Profile 2)	0	✓		

Video Support over SIP Trunks

Program Number		Input Data	Default	Level 1	Level 2	Level 3
10-12-01	**GCD-CP10 Network Setup – IP Address** It is recommended to set this program to **0.0.0.0**. All connections to the system are made through the IPLE.	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	192.168.0.10		✓	
10-12-03	**GCD-CP10 Network Setup – Default Gateway** Assign the default gateway IP address for the GCD-CP10.	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	✓		
10-12-08	**GCD-CP10 Network Setup – ICMP Redirect** When receiving ICMP redirect messages, this determines if the IP Routing Table updates automatically or not.	0= (Enable) 1= (Disable)	0		✓	

Program Number		Input Data	Default	Level		
				1	2	3
10-12-09	**GCD-CP10 Network Setup – IP Address** Set IP address for IPLE. ➨ *The IP Address assigned in Program 10-12-01 cannot start with the same leading digits as the IP Address assigned here.*	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	172.16.0.10	✓		
10-12-10	**GCD-CP10 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	255.255.0.0	✓		
10-12-11	**GCD-CP10 Network Setup – NIC Setup** Define the LAN interface Speed and Mode of the VoIP Application supported. ➨ *IPLE daughter board does not support half duplex connection.*	0 = Auto Detect 1 = 100Mbps, Full Duplex 3 = 10Mbps, Full Duplex 5 = 1Gbps, Full Duplex	0	✓		
10-23-01	**SIP System Interconnection Setup – System Interconnection** Determine if the system is interconnected to another system. ➨ *For the SIP System Interconnection set to* ***1*** *(Yes).*	0 = No (Disable) 1 = Yes (Enable)	0	✓		
10-23-02	**SIP System Interconnection Setup – IP Address** Define the IP Address for the SIP System Interconnection.	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	✓		
10-23-04	**SIP System Interconnection Setup – Dial Number** Define the Dial Number for the SIP System.	Maximum of 12 digits (0 ~ 9)	No Setting	✓		
10-23-06	**SIP System Interconnection Setup – SIP Profile** Assign the Interconnection to a SIP Profile.	1 = Profile 1 2 = Profile 2	1	✓		

Program Number		Input Data	Default	Level 1	Level 2	Level 3
10-36-02	**SIP Trunk Registration Information Setup – User ID** Assign the SIP User ID provided by your SIP Carrier. In most cases this is your 10 digit main billing number. This field is also used to send outbound caller ID information when not programmed on a per station or per trunk basis. ➨ *For non-registration and SIP Tie Lines to another system, you **MUST** have a USER ID entered.*	Maximum of 32 characters. (Profile 1 ~ Profile 2)	No Setting	✓		
10-54-01	**License Configuration for Each Package – License Code** Assign VoIP Resource Licenses (5103) to GCD-CP10 slot (1)	1 ~ 255 Resource Licenses	0	✓		
11-01-01	**System Numbering – Service Code** Assign the 1st and 2nd dial digit to F-Route for Remote System Extension Numbering.		Refer to the Programming Manual for default values.	✓		
11-02-01	**Extension Numbering – Dial (Up to 8 Digits)** Assign extension numbers to Extension ports.	Maximum of eight digits.	Port 1 ~ 99 = 101 ~ 199 Port 100 ~ 960 = 3101 ~ 3961		✓	
14-05-01	**Trunk Group – Trunk Group Number** Assign SIP Trunks to same Trunk Group.	Trunk Port 1 ~ 400 = Priority 1 ~ 400	Default = Trunks 1 ~ 400 assigned to trunk group 1 with priorities equal to the trunk number. Trunk 1 = Priority 1 Trunk 400 = Priority 400.	✓		
14-18-03	**IP Trunk Data Setup – P2P Mode** Enable/Disable P2P Mode. ➨ *For Video Call via the System Interconnection set to **1** (Enable).*	0 = Disable 1 = Enable	0	✓		
14-18-04	**IP Trunk Data Setup – Video Mode** Enable/Disable Video Mode. ➨ *For Video Call via the System Interconnection set to **1** (Enable).*	0 = Disable 1 = Enable	0	✓		
14-18-05	**IP Trunk Data Setup – SIP Profile (SIP Trunk)** Assign each SIP Trunk to either Profile 1 or Profile 2.	Profile 1 Profile 2	Profile 1	✓		
15-05-43	**IP Telephone Terminal Basic Data Setup – Video Mode** Enable/Disable Video Mode for Standard SIP terminals. ➨ *For Video Call via the System Interconnection set to **1** (Enable).*	0 = Disable 1 = Enable	0	✓		

Program Number		Input Data	Default	Level 1	Level 2	Level 3
15-05-50	**IP Telephone Terminal Basic Data Setup – Peer to Peer Mode** On a per station basis enable or disable Peer to Peer mode. ➨ *For Video Call to function set to 1 (Enable).*	0 = Disable 1 = Enable	1	✓		
20-08-13	**Class of Service Options (Outgoing Call Service) – ISDN CLIP** Determine if the ISDN calling line identity presentation and screening indicators should be allowed.	0 = Off 1 = On	0		✓	
21-17-01	**IP Trunk (SIP) Calling Party Number Setup for Trunk** Assign 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 Program 21-17 and Program 21-18/ 21-19, the system uses the entry in Program 21-18/21-19.	Maximum of 16 digits (0 ~ 9, *****, **#**)	No Setting		✓	
21-19-01	**IP Trunk (SIP) Calling Party Number Setup for Extension** 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 Program 21-18/Program 21-19, the system uses the data in Program 21-18/Program 21-19.	Maximum of 16 digits (0 ~ 9, *****, **#**) (Profile 1 ~ Profile 2)	No Setting		✓	
22-02-01	**Incoming Call Trunk Setup** Assign the incoming trunk type for each trunk. ➨ *For the SIP System Interconnection, set each trunk to* ***5*** *(E&M Tie Line).*	Trunks 1 ~ 400 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	0	✓		
44-02-01	**Dial Analysis Table for ARS/F-Route Access – Dial** Set the Dial digits for the Pre-Transaction Table for selecting ARS/F-Route (eight digits maximum: **1~9**, **0** * **#**, **@**). ➨ *To enter a wild card/don't care digit, press Line Key 1 to enter an* ***@***.	Maximum of eight digits.	No Setting	✓		

Program Number		Input Data	Default	Level		
				1	2	3
44-02-02	**Dial Analysis Table for ARS/F-Route Access – Service Type** Set the Service Type (0~3) for the Pre-Transaction Table for selecting ARS/F-Route. ➨ *For the SIP System Interconnection, set each Dial Digit to **2** (F-Route).*	0 = No Setting (None) 1 = Extension Call (Own) 2 = ARS/F-Route Table (F-Route) 3 = Dial Extension Analyze Table (Option)	0	✓		
44-02-03	**Dial Analysis Table for ARS/F-Route Access – Additional Data** This is the F-Route Table set in Program 44-05.	1 = Delete Digit = 0 ~ 255 (255: Delete All Digits) 2 = 0 ~ 500 (0 = No Setting) 3 = Dial Extension Analyze Table Number = 0 ~ 4 (0 = No Setting)	0	✓		
44-05-01	**ARS/F-Route Table – Trunk Group Number** Assign the Trunk Group to be used by the F-Route Table. ➨ *This is the Trunk Group assigned to the SIP System Interconnection trunks in Program 14-05-01.*	0 ~ 100, 255 0 = No Setting 255 = Extension Call	0	✓		
44-05-09	**ARS/F-Route Table – Maximum Digit** Input the maximum number of digits to send when using the F-Route.	0 ~ 24	0		✓	
84-26-01	**IPL Basic Setup – IP Address** Assign the IP address for each DSP on the IPLE.	xxx.xxx.xxx.xxx	172.16.0.20		✓	

Operation

SIP Trunk E.164 Support

To make a call using E.164 number format:

1. Lift the handset or press **Speaker**.
2. Dial **00441202223344#**.

   ¸ *The system automatically modifies the required header fields of the SIP INVITE message using the configuration settings in the table below before forwarding to the ITSP.*

Table 6-1 SIP INVITE Header Fields

Program 84-14-13	Program 10-02-01	Program 10-02-02	Description Calling Party Number = 441509555123 Called Party Number = 00441202223344
0	44	–	Request-URI: Invite sip: 00441202223344@172.16.18.100 SIP/2.0 To header: To:sip:00441202223344@172.16.18.100 From header: From<sip:441509555123@172.16.0.10>
	No Setting	–	Request-URI: Invite sip: 00441202223344@172.16.18.100 SIP/2.0 To header: To:sip:00441202223344@172.16.18.100 From header: From<sip:441509555123@172.16.0.10>
1	44	–	Request-URI: Invite sip:+4400441202223344@172.16.18.100 SIP/2.0 To header: To:sip:+4400441202223344@172.16.18.100 From header: From<sip:+44441509555123@172.16.0.10>
	No Setting	–	Request-URI: Invite sip:+00441202223344@172.16.18.100 SIP/2.0 To header: To:sip:+00441202223344@172.16.18.100 From header: From<sip:+441509555123@172.16.0.10>
2	–	00	Request-URI: Invite sip:+441202223344@172.16.18.100 SIP/2.0 To header: To:<sip:+441202223344@172.16.18.100> From header: From<sip:441509555123@172.16.0.10> P-Asserted-Identity: P-Asserted-Identity441509555123@172.16.0.10> P-Preferred-Identity: P-Preferred-Identity441509555123@172.16.0.10>
	–	No Setting	No Function

SIP Trunk E.164 CLIP Enhancement

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

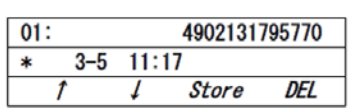
Table 6-2 Delete + from Incoming SIP INVITE

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

<Example Output>

Incoming call from: +4902131795770

Displayed in terminal incoming caller history as:



Original

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

Table 6-3 Delete + and Country Code from Incoming SIP INVITE

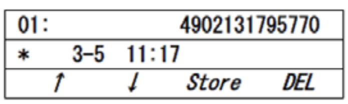
Program 84-14-16	Program 84-14-13	Description
1: Mode 1	1: On	With a SIP INVITE for incoming calls. When a **+** is presented as the international access code along with a country code that **DOES NOT** match the value in Program 10-02-01, then delete the **+** and add the international access code value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

<Example Output>

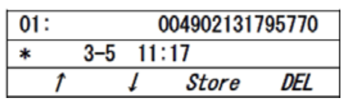
Incoming call from: +4902131795770

Program 10-02-02 = 00

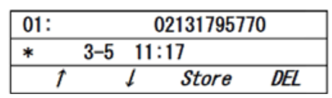
Displayed in terminal incoming caller history as:



Original



Program 10-02-01 = 0



Program 10-02-01 = 49

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

Table 6-4 Delete + and Country Code from Incoming SIP INVITE

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

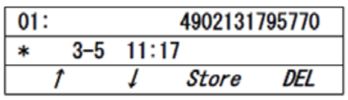
<Example Output>

Incoming call from: +4902131795770

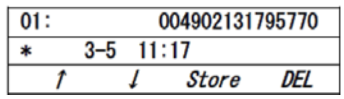
Program 10-02-02 = 00

Program 10-02-03 = 9

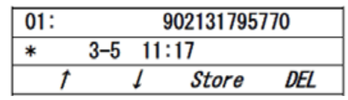
Displayed in terminal incoming caller history as:



Original



Program 10-02-01 = 0

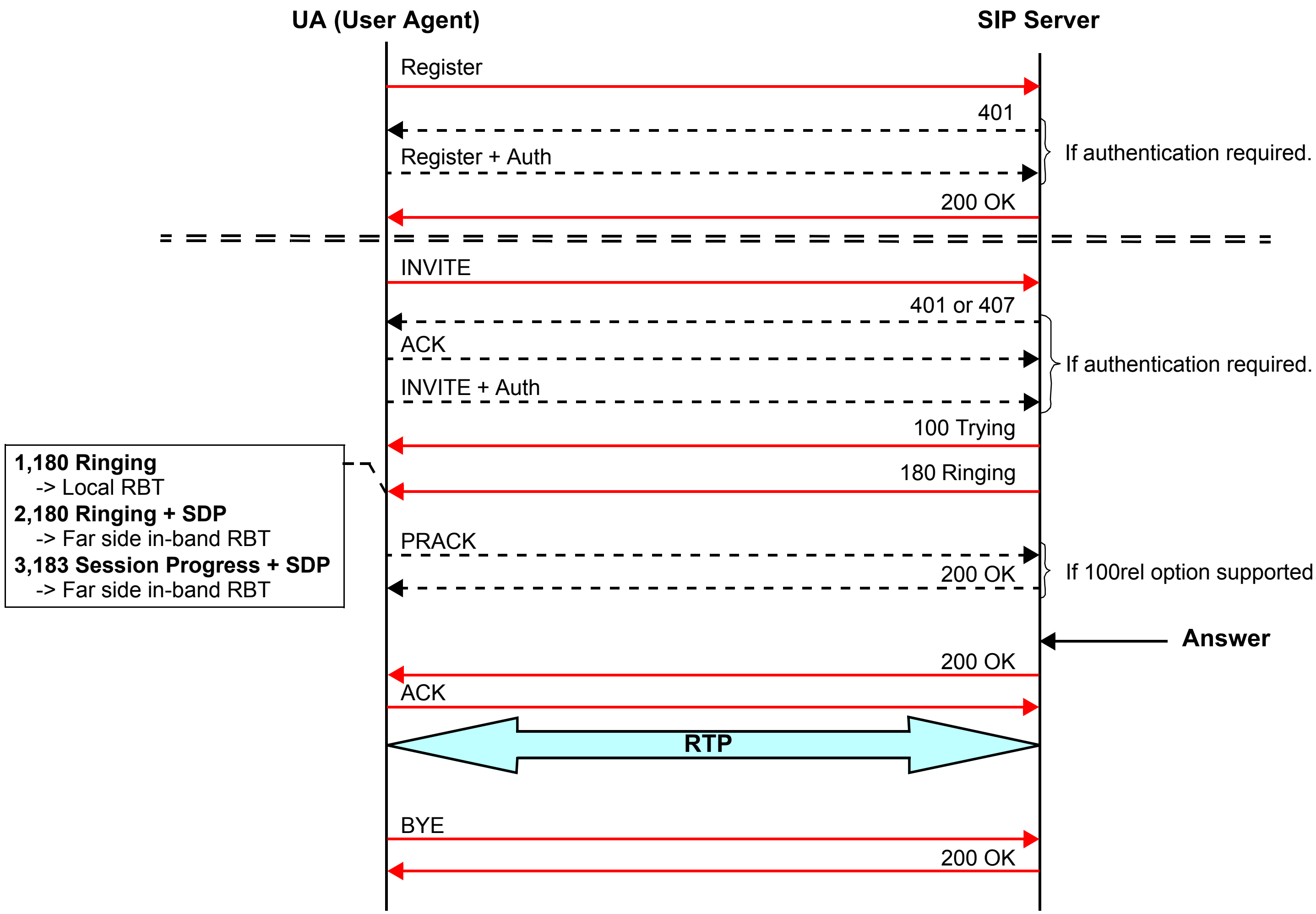


Program 10-02-01 = 49

Making an outgoing call from history of incoming calls:

1. From an idle multiline terminal.
2. Press soft key **List**.
3. Press soft key **CID**.
4. Press **Speaker**.

Table 6-5 SIP Packet Sequence



H.323 Trunking

Chapter 7

Section 1 Introduction

H.323 is an International Telecommunication Union (ITU) standard for Packet Based Multimedia Communication Systems. The UNIVERGE SV9100 can use H.323 to connect to another UNIVERGE SV9100 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 SV9100 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 SV9100 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 GPZ-IPLE 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 GPZ-IPLE – 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 SV9100 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

After installing IPLE (VOIP Daughter Board) in UNIVERGE SV9100 System, VOIPDB trunk ports need to be defined in 10-68-01.

By default the VOIPDB trunk ports are defined as None. When using H.323, you need to change the trunk ports to H.323 in 10-68-01.

2.2.1 H.323 Setup

Use the following programs to assign the information for H.323.

Table 7-1 H.323 Setup Assignments

Program/ Item No.	Description/ Selection	Assigned Data	Comments
10-68-01	IP Trunk Availability – Trunk Type	0: None 1: SIP 2: H.323 3: CCIS	Assign the H.323 Trunk Availability
10-68-02	IP Trunk Availability – Start Port	Range: 0 ~ 400 Default is 0	Assign the start port number of H.323 Trunks
10-68-03	IP Trunk Availability – Number of Ports	Range: 0 ~ 400 Default is 0	Assign the Number of H.323 Trunks
14-05-01	Trunk Group – Trunk Group Number	0~100 Priority = 1 ~ 400 Default is Group 1	Default priorities for trunks 1~400 is 1~400. 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

Use the following programs to assign VoIPDB network configuration information.

Table 7-2 VoIPDB Network Configuration Assignments

Program/ Item No.	Description/ Selection	Assigned Data	Comments
10-68-01	IP Trunk Availability – Trunk Type	0: None 1: SIP 2: H.323 3: CCIS	Assign the H.323 Trunk Availability
10-68-02	IP Trunk Availability – Start Port	Range: 0 ~ 400 Default is 0	Assign the start port number of H.323 Trunks
10-68-03	IP Trunk Availability – Number of Ports	Range: 0 ~ 400 Default is 0	Assign the Number of H.323 Trunks

2.2.3 VoIPDB (DSP) Basic Setup

Use the following programs to assign VoIPDB (DSP) basic setup information.

Table 7-3 VoIPDB (DSP) Basic Assignments

Program/ Item No.	Description/ Selection	Assigned Data	Comments
84-26-01	IPL Basic Setup – IP Address	Default = 172.16.0.20	When installing GPZ-IPLE in UNIVERGE SV9100 system, only 1 IP address is assigned.
84-26-02	IPL Basic Setup – RTP Port Number	0~65534 Default = 10020	Assign the RTP Port Number.
84-26-03	IPL Basic Setup – RTCP Port Number (RTCP Port Number + 1)	0~65534 Default = 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 SV9100 has a built-in H.323 gatekeeper and also supports connection to an external gatekeeper. In most cases, the SV9100 internal Gatekeeper is sufficient. External gatekeepers are used usually only in very large H.323 networks.

To configure the UNIVERGE SV9100 to use its internal Gatekeeper, set Program 10-17-01 to Manual, and then set the GCD-CP10 IP address (Program10-12-09) as the gatekeeper address in Program 10-17-02.

The UNIVERGE SV9100 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

Use the following programs to assign H.323 GateKeeper setup information.

Table 7-4 H.323 GateKeeper Setup Assignments

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.

Table 7-4 H.323 GateKeeper Setup Assignments (Continued)

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

Use the following programs to assign H.323 alias address setup information.

Table 7-5 H.323 Alias Address Setup Assignments

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 SV9100 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 SV9100 Features and Specifications Manual.

2.4.1 H.323 Call Routing

Use the following programs to assign H.323 call routing.

Table 7-6 H.323 Call Routing Assignments

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.
10-23-05	Keep Alive Mode for SIP	0 = Disable 1 = Enable (default = 0)	
10-23-06	SIP Profile	1 = Profile 1 2 = Profile 2 (default = 1)	Only used when Trunk Type is defined as (SIP) in PRG 10-68-01.

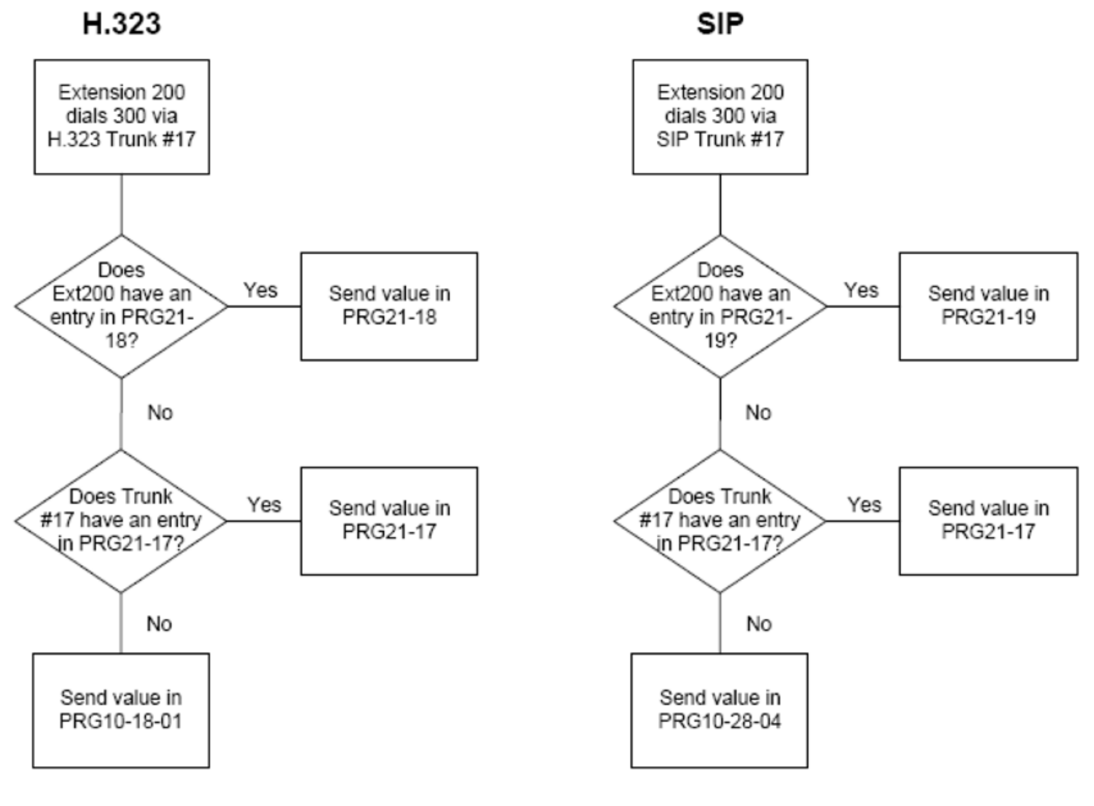
2.4.2 Caller ID (H.323)

Use the following programs to assign H.323 caller ID.

Table 7-7 H.323 Caller ID Assignments

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 1	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 1	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)

Use the following programs to assign CODEC selections for H.323.

Table 7-8 IP Networking (H.323) CODEC Assignments

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 (SV9100) Default =3	
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 0	
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)	0~300ms Default is 30	
84-01-08	H.323 Trunk Basic Information Setup – G.729 Jitter Buffer (average)	0~300ms Default is 60	
84-01-09	H.323 Trunk Basic Information Setup – G.729 Jitter Buffer (max)	0~300ms Default is 120	
84-01-15	H.323 Trunk Basic Information Setup – Jitter Buffer Mode	1 = Fixed 3 = Self adjusting Default is 3	
84-01-16	H.323 Trunk Basic Information Setup – G.711 Jitter Buffer(min)	0~300ms Default is 30	

Table 7-8 IP Networking (H.323) CODEC Assignments (Continued)

Program/ Item No.	Description/ Selection	Assigned Data	Comments
84-01-17	H.323 Trunk Basic Information Setup – G.711 Jitter Buffer (average)	0~300ms Default is 60	
84-01-18	H.323 Trunk Basic Information Setup – G.711 Jitter Buffer (max)	0~300ms Default is 120	
84-01-22	Voice Activity Detection 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	
84-01-33	H.323 Trunk Basic Information Setup – Priority CODEC setting Priority of voice encoding method.	0~3 0 = G.711 2 = G.729 3 = G.722 Default is 0	Priority of voice encoding method.
84-01-34	---Not Used---		
84-01-35	---Not Used---		
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	
84-01-65	H.323 Trunk Basic Information Setup – G.722 Jitter Buffer (min)	0~300ms Default is 30	
84-01-66	H.323 Trunk Basic Information Setup – G.722 Jitter Buffer (average)	0~300ms Default is 60	

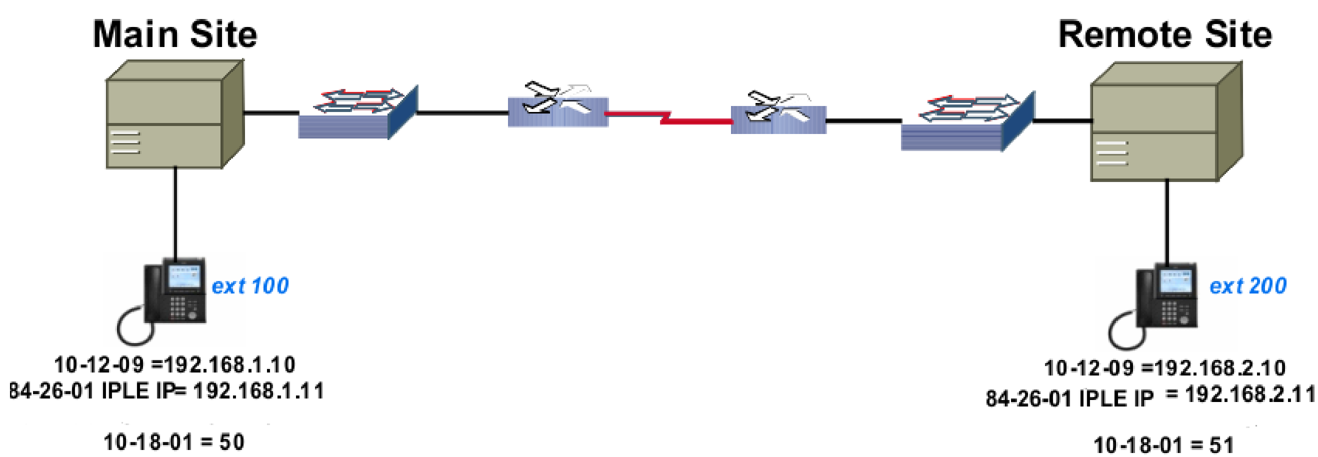
Table 7-8 IP Networking (H.323) CODEC Assignments (Continued)

Program/ Item No.	Description/ Selection	Assigned Data	Comments
84-01-67	H.323 Trunk Basic Information Setup – G.722 Jitter Buffer (max)	0~300ms Default is 120	
84-01-68	RTP Filter	0 = Disable 1 = Enable 2 = Enable (SSRC) Default is 1	

2.6 H.323 TIE Line Programming Example

Figure 7-1 H.323 TIE Line Programming Example illustrates how to program a TIE line for H.323.

Figure 7-1 H.323 TIE Line Programming Example



The following programs are assigned in this 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	

Program	Main Site (Office A)	Remote Site (Office B)
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
PRG 84-26-01 (DSP)	192.168.1.11	192.168.2.11
H.323 Settings		
PRG 10-68-01	Enable H.323 Trunk Availability	Enable H.323 Trunk Availability
PRG 10-68-02	Define the Start Port Number	Define the Start Port Number
PRG 10-68-03	Assign the number of Ports (4 Ports)	Assign the number of Ports (4 Ports)
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

* GCD-CP10 reset is required after making changes to this Programs.

IP Multiline Station (SIP)

Chapter 8

SECTION 1 INTRODUCTION[1]

The UNIVERGE SV9100 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 DT800/DT700 IP Multiline Telephones (ITL/ITZ) support IP extensions.

- ITL-2E-1 (BK) TEL
- ITL-6DE-1 (BK) TEL
- ITL-8LD-1 (BK) TEL / ITL-8LD-1 (WH) TEL
- ITL-12D-1 (BK) TEL / ITL-12D-1 (WH) TEL
- ITL-12PA-1 (BK) TEL
- ITL-24D-1 (BK) TEL / ITL-24D-1 (WH) TEL
- ITL-32D-1 (BK) TEL / ITL-32D-1 (WH) TEL
- ITL-320C-1 (BK) TEL
- ITL-12CG-3 (BK) TEL
- ITL-12DG-3 (BK) TEL
- ITZ-12D-3 (BK) TEL/ ITZ-12D-3 (WH) TEL
- ITZ-24D-3 (BK) TEL/ ITZ-24D-3 (WH) TEL
- ITZ-12CG-3 (BK) TEL/ ITZ-12CG-3 (WH) TEL
- ITZ-24CG-3 (BK) TEL
- ITZ-8LD-3 (BK) TEL/ ITZ-8LD-3 (WH) TEL
- ITZ-8LDG-3 (BK) TEL/ ITZ-8LDG-3 (WH) TEL
- ITZ-12DG-3 (BK) TEL/ ITZ-12DG-3 (WH) TEL
- ITZ-24DG-3 (BK) TEL/ ITZ-24DG-3 (WH) TEL

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

Section 2 IP to TDM Conversion

When an IP telephone calls a DT300/DT400 multiline telephone, single line telephone or trunk, the speech must be converted from IP to TDM (Time Division Multiplexing) technology. The GPZ-IPLE daughter board provides this function.

Each GPZ-IPLE has a number of DSP resources; each can convert a speech channel from IP to TDM and vice versa.

It is possible for DT800/DT700 IP Phones to talk directly to other DT800/DT700 IP Phones without using a GPZ-IPLE resource. For more information, refer to Section 6 Peer-to-Peer on page 8-7.

2.1 DT800/DT700 IP Multiline Telephones (ITL/ITZ)

The IP multiline telephone operates in the same way as a DT400 (DTZ)/DT300 (DTL) digital multiline telephone. It has all features and flexibility you expect from a DT400/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 GCD-8DLCA/GCD-16DLCA.

Figure 8-1 DT800/DT700 IP Telephone (ITL)

2.2 Conditions

When using DT800/DT700 IP phones, it is not recommended to assign the following features to a large number of phones (100 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 (DT800/DT700) multiline telephones and digital (DT400/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 SV9100 system.

No programming is required on the UNIVERGE SV9100 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

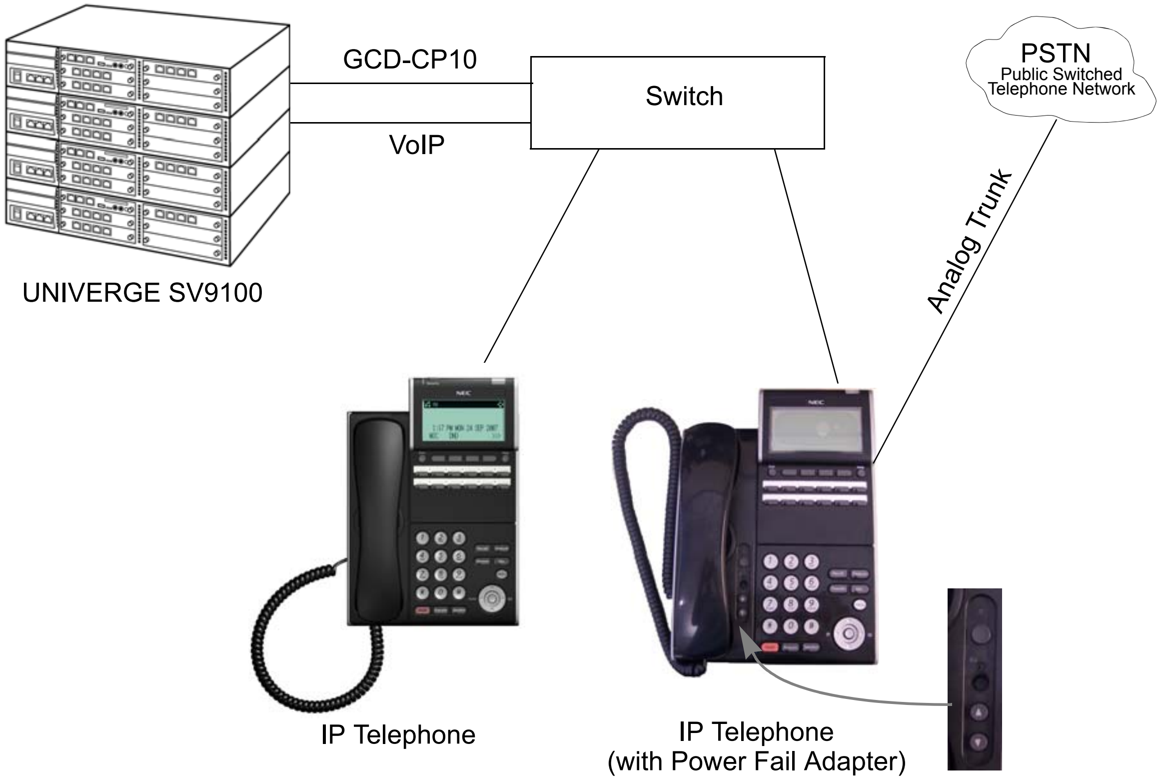
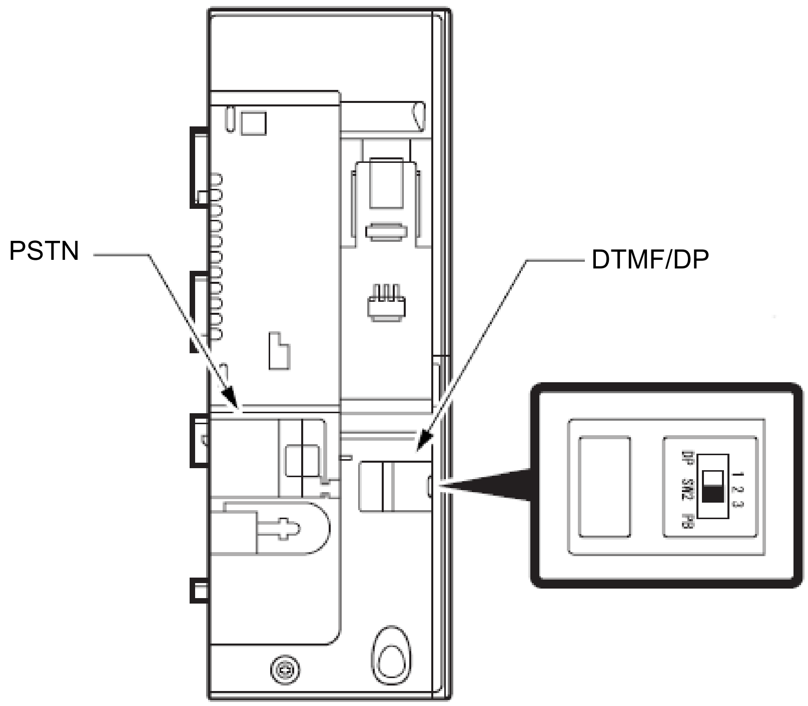


Figure 8-3 IP Telephone Connection



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 SV9100 system (e.g., data network failure), the IP telephone attempts to reconnect. If this fails, press the button on the top of the adapter. Refer to Figure 8-2 Power Fail Adapter Connection on page 8-4. This puts the IP telephone in analog mode. The telephone display shows *LINE -> PSTN*.

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

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

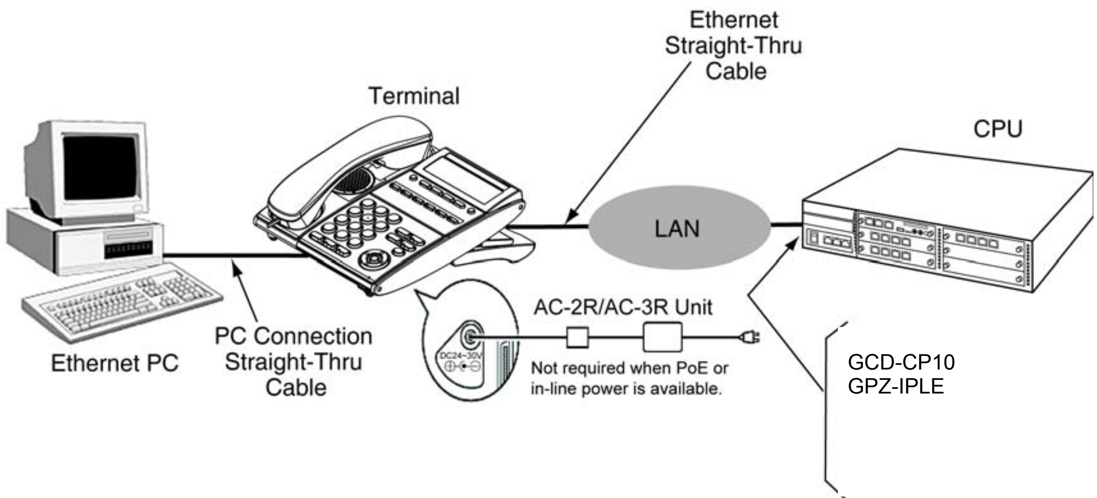
➨ *Handsfree (Speaker) mode is not supported on calls made to or from the Power Fail Adapter. The handset must always be used.*

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

SECTION 4 LAN CONNECTION

As illustrated in Figure 8-4 Typical Network IP Connection, the IP telephone has two RJ-45 connections on the back side marked PC and LAN. This allows the IP telephone and a PC to share one cable run and switch port.

Figure 8-4 Typical Network IP Connection



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

❑ Use a different cable and complete the following steps:
   ❍ Leave the PC connected to the LAN.
   ❍ Patch a switch port to a new cable run.
   ❍ Connect a CAT 5 straight-through cable from the wall outlet to the LAN port on the IP telephone.

❑ Share the existing cable and complete the following steps:
   ❍ Unplug the cable from the PC network card (NIC).
   ❍ Connect that cable to the LAN port on the IP telephone.
   ❍ Connect a new straight-through patch lead from the PC NIC to the PC port on the IP telephone.

SECTION 5 PROVIDING POWER

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

5.1 Local Power

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

➨ *Use only the NEC supplied power supply.*

5.2 802.3af Power Over Ethernet (PoE)

A 802.3af PoE switch is a data switch that also provides power over the spare pairs. The switch can be used with any device (not just IP phones) and detects whether or not power is required. As all phones receive their power from one device, it is easy to protect the IP phones from loss of power (by connecting the PoE switch to a UPS).

SECTION 6 PEER-TO-PEER

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

If a DT800/DT700 IP multiline telephone or trunk line is required, a DSP resource is needed and a GPZ-IPLE 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 GPZ-IPLE. 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 SV9100 chassis must still have a registered GPZ-IPLE installed in the system.

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

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

Section 7 Miscellaneous

7.1 System Tones and Ring Tones

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

7.2 Music on Hold

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

Section 8 Configuration Examples

8.1 IP Addressing

When using a GPZ-IPLE, only 1 IP Address is needed for all DSP's. It should be the same subnet of the IP defined in 10-12-09.

The following chart shows the minimum and maximum number of IP addresses used with different GPZ-IPLE card configurations.

Card	Minimum IP Addresses	Maximum IP Addresses	Notes
GPZ-IPLE	1	2	The number of DSP channels depends on the VOIP license loaded to GCD-CP10 up to 256.

:

IMPORTANT

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

When an GPZ-IPLE 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 GPZ-IPLE NIC. E.g. PCPro, Web Pro, ACD, etc. all terminate to the GPZ-IPLE NIC card when installed.

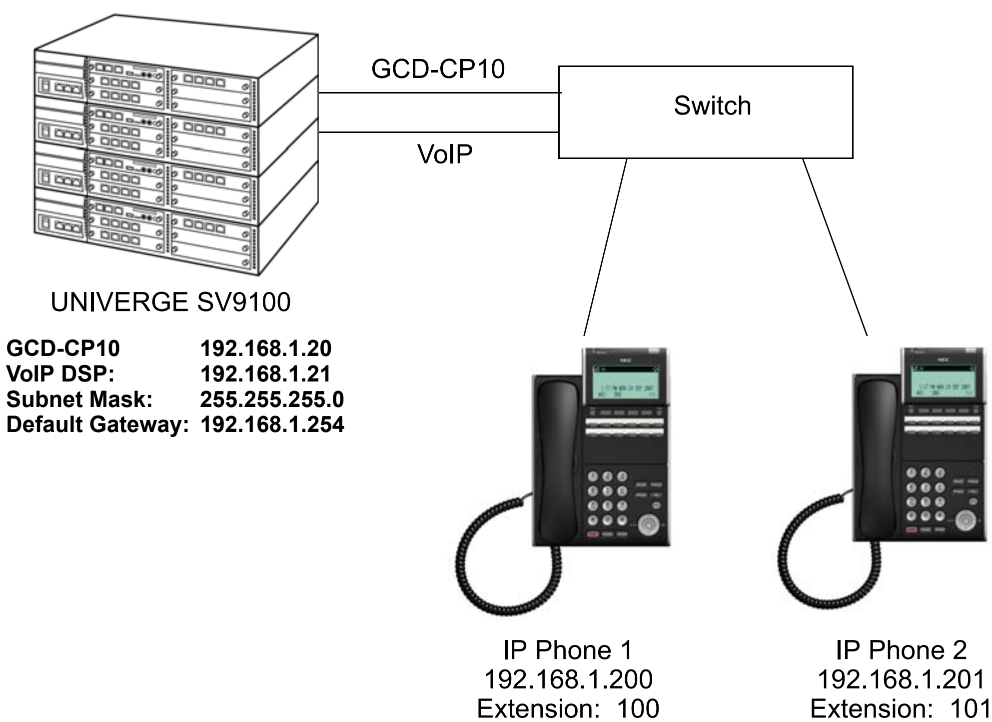
The GPZ-IPLE and the CPU NIC share the same gateway assignment. The default gateway command in Program10-12-03 is used by both NICs allowing only one device, or CPU, to route outside of its own network.

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

8.2 Example Configuration 1 - Static IP Addressing, One LAN

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

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



Programming - GCD-CP10:	
10-12-01 : GCD-CP10 Network Setup - IP Address (for GCD-CP10) 10-12-02 : GCD-CP10 Network Setup - Subnet M ask 10-12-03 : GCD-CP10 Network Setup - Default Gateway 10-12-09 : GCD-CP10 Network Setup - IP Address (This assignment is for GCD-CP10 when the GPZ-IPLE daughter board is installed.)	0.0.0.0 255.255.255.0 192.168.1.254 192.168.1.20
Programming - GPZ-IPLE :	
84-26-01 : IPL Basic Setup - IP Address (Slot # - DSP)	192.168.1.21
Programming - IP Phones:	
DHCP Mode IP Address Subnet Mask Default Gateway 1st Server Address 1st Server Port	Disabled 192.168.1.200 255.255.255.0 192.168.1.254 192.168.1.20 5080

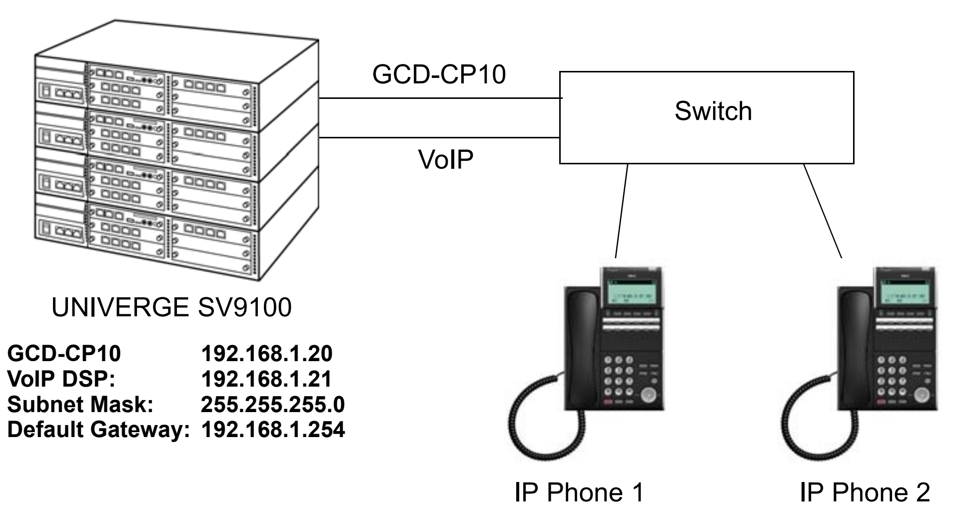
8.3 Example Configuration 2 - Dynamic IP Addressing, One LAN

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

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

In this case, additional programming would be required. Refer to Chapter 3 General IP Configuration.

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

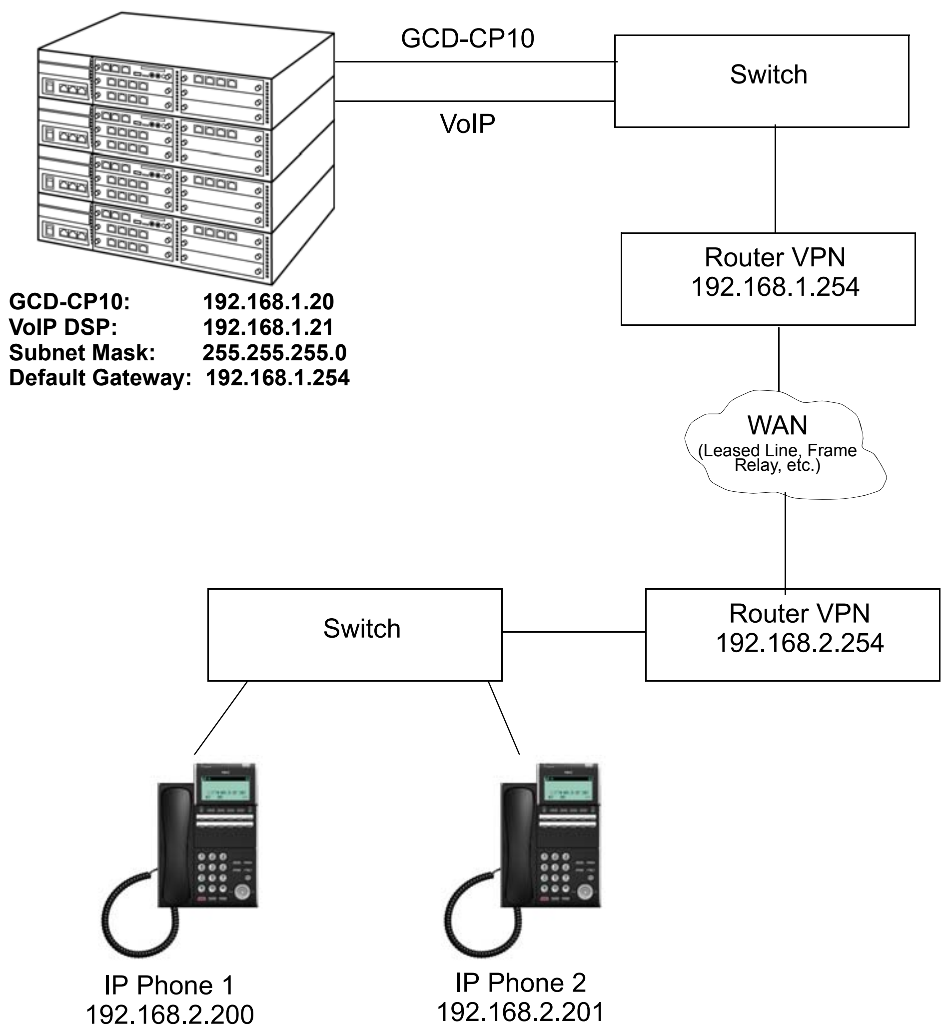


Programming - GCD-CP10:	
10-12-01 : GCD-CP10 Network Setup - IP Address (for GCD-CP10)	0.0.0.0
10-12-10 : GCD-CP10 Network Setup - Subnet Mask	255.255.255.0
10-12-03 : GCD-CP10 Network Setup - Default Gateway	192.168.1.254
10-12-09 : GCD-CP10 Network Setup - IP Address	192.168.1.20
(This assignment is for GCD-CP10 when the GPZ-IPLE daughter board is installed.)	On
10-13-01 : In-DHCP Server Setup - DHCP Server Mode	Enable
10-14-01 : Managed Network Setup	192.168.1.200 Min
10-14-02: Managed Network Setup	192.168.1.250 Max
10-16-16: SIP Server Address	192.168.1.20
10-16-27: SIP Server Port	5080
Programming - GPZ-IPLE:	
84-26-01 : IPL Basic Setup - IP Address	192.168.1.21
Programming - IP Phones:	
DHCP Mode (Ext. 200):	Enabled

8.4 Example Configuration 3 - Static IP Addressing, Routed WAN

This example shows IP Phones connected to an UNIVERGE SV9100 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 - GCD-CP10:	
10-12-01: GCD-CP10 Network Setup - IP Address (for GCD-CP10) 10-12-10: GCD-CP10 Network Setup - Subnet Mask 10-12-03: GCD-CP10 Network Setup - Default Gateway 10-12-09: GCD-CP10 Network Setup - IP Address (This assignment is for GCD-CP10 when the GPZ-IPLE daughter board is installed.)	0.0.0.0 255.255.255.0 198.168.1.254 192.168.1.20
Programming - GPZ-IPLE:	
84-26-01: IPL Basic Setup - IP Address (Slot # - DSP)	192.168.1.21
Programming - IP Phones	
DHCP Mode: IP Address: Subnet Mask: 1st Server Address 1st Server Port	Disabled 192.168.2.200 255.255.255.0 198.168.1.20 5080

SECTION 9 IP PHONE PROGRAMMING INTERFACE

This section describes how to access the programming interface for IP phones. To access the User Menu follow the steps listed below.

1. Using a DT800/DT700 telephone, press the **Menu** button to enter program mode. The IP User Menu is displayed.
2. On the IP User Menu, select **Config (0)** for the IP Phone. Settings are listed in Table 8-1 IP Phone Programming Options User Menu.

Table 8-1 IP Phone Programming Options User Menu

Programming Option	Default
UserName	ADMIN
Password	6633222

SECTION 10 DHCP SERVER CONFIGURATION

It is possible to use either an external DHCP server (e.g., Windows Server) or the UNIVERGE SV9100 internal DHCP server. With IP Phones, either of these options requires the DHCP server to be configured to supply the IP terminal options.

If using the internal DHCP server, enable the DHCP server. Refer to 8.3 Example Configuration 2 - Dynamic IP Addressing, One LAN on page 8-10.

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

SECTION 11 AUTO CONFIG FOR IP TERMINALS SV9100

11.1 Required Equipment

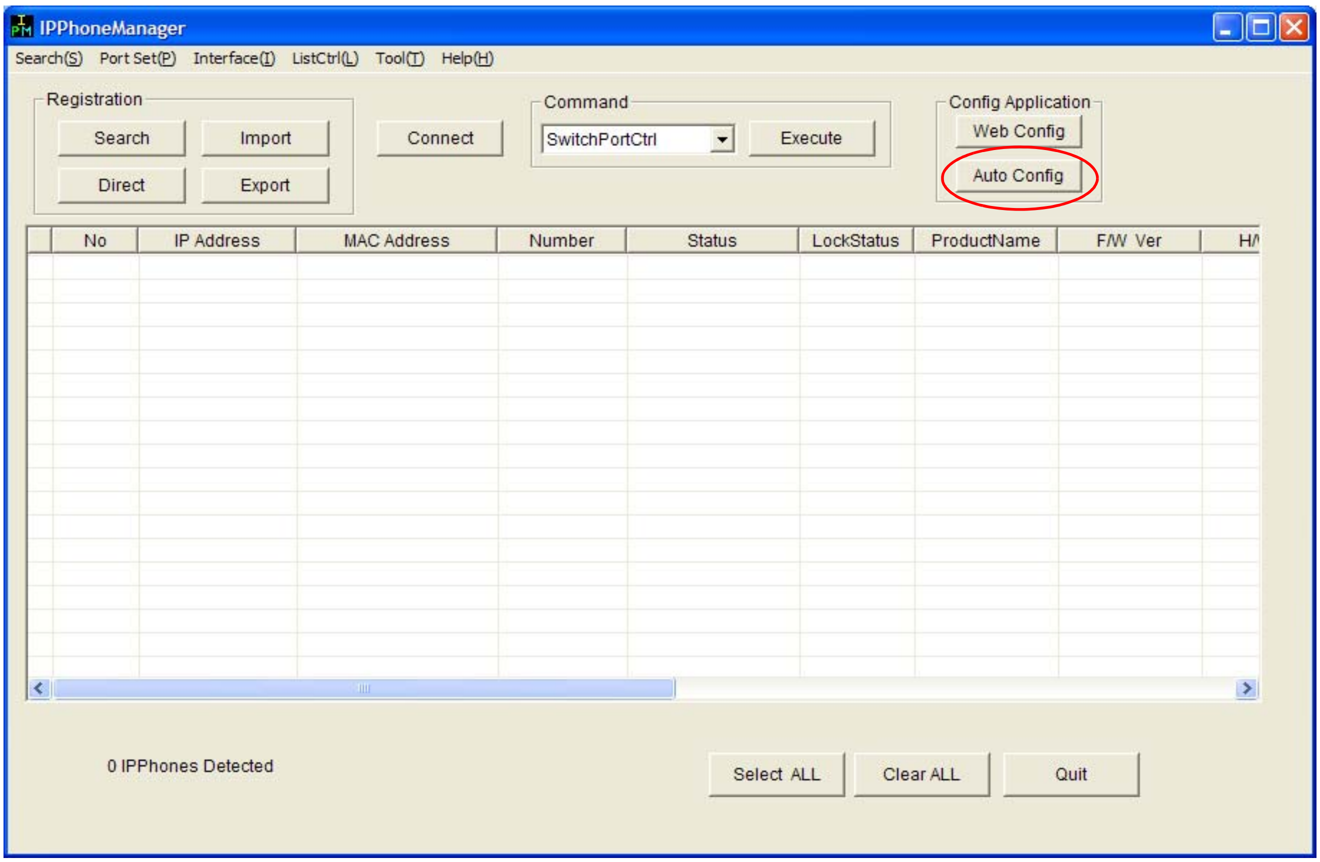
- IP Phone Manager and software to create the config file for the IP terminals. Software is available for download free on the NEC website.
- FTP server and free software available on the web.
- DHCP Server supporting ability to define the following:
    - Vendor Class
    - Option Codes

11.2 Building Config File

When building a config file, follow the steps below to launch the phone manager and create a file.

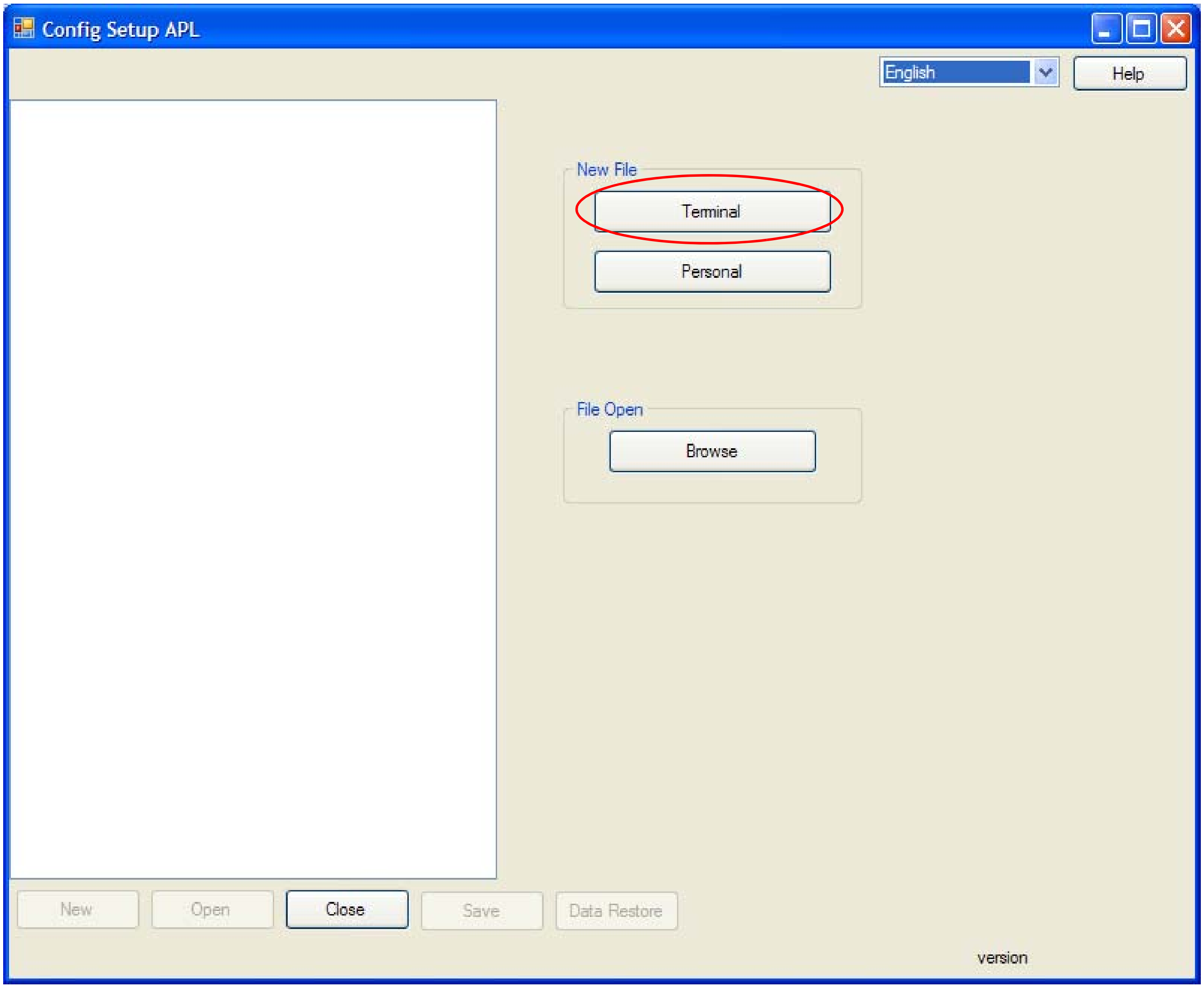
1. Launch the IP Phone Manager software.
2. Once the software is launched, click **Auto Config**.

Figure 8-8 IP Phone Manager



3. Click **Terminal**.

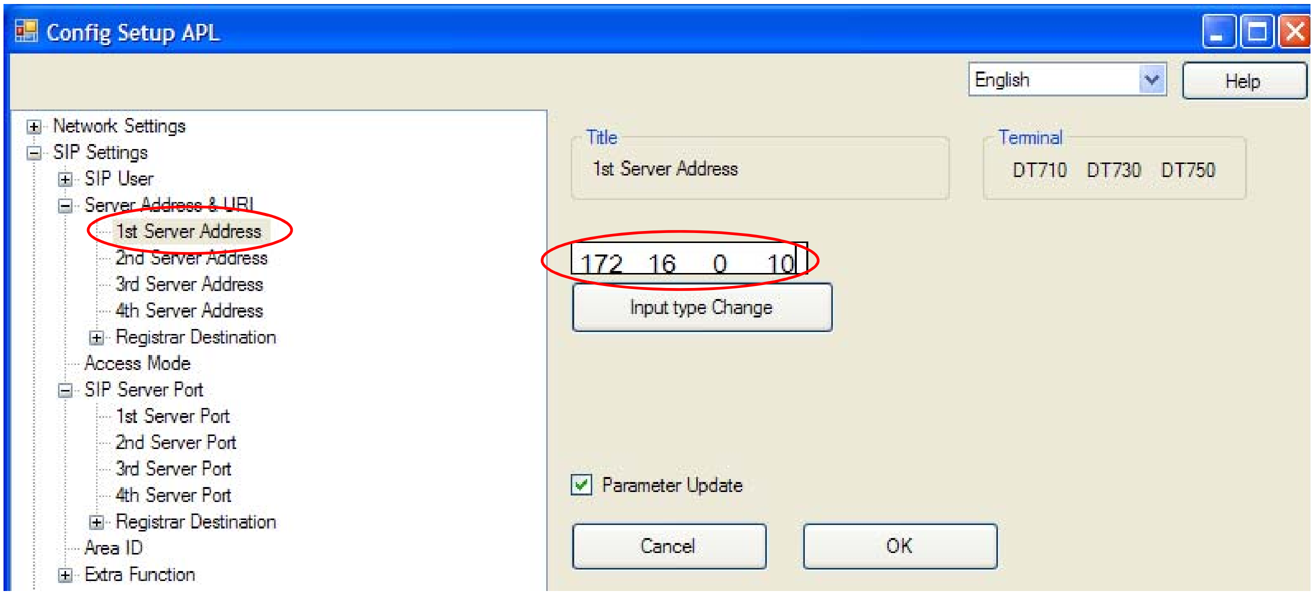
Figure 8-9 Config Setup APL



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

6. Click **OK**.

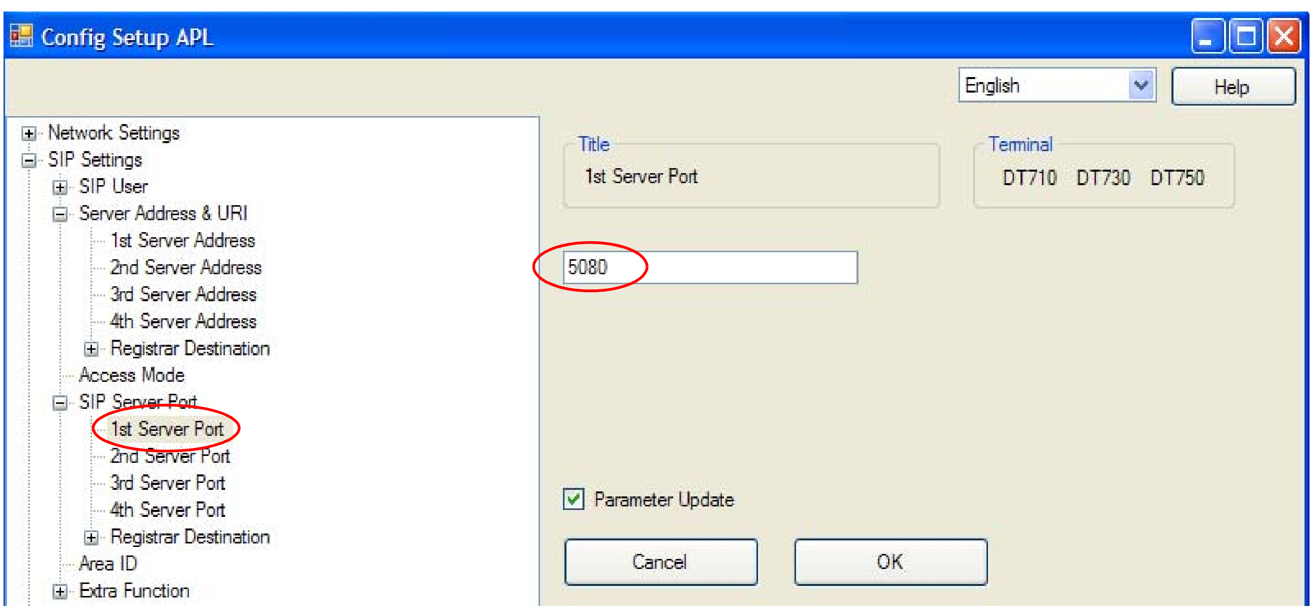
Figure 8-10 Assign IP Address



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

8. Click **OK**.

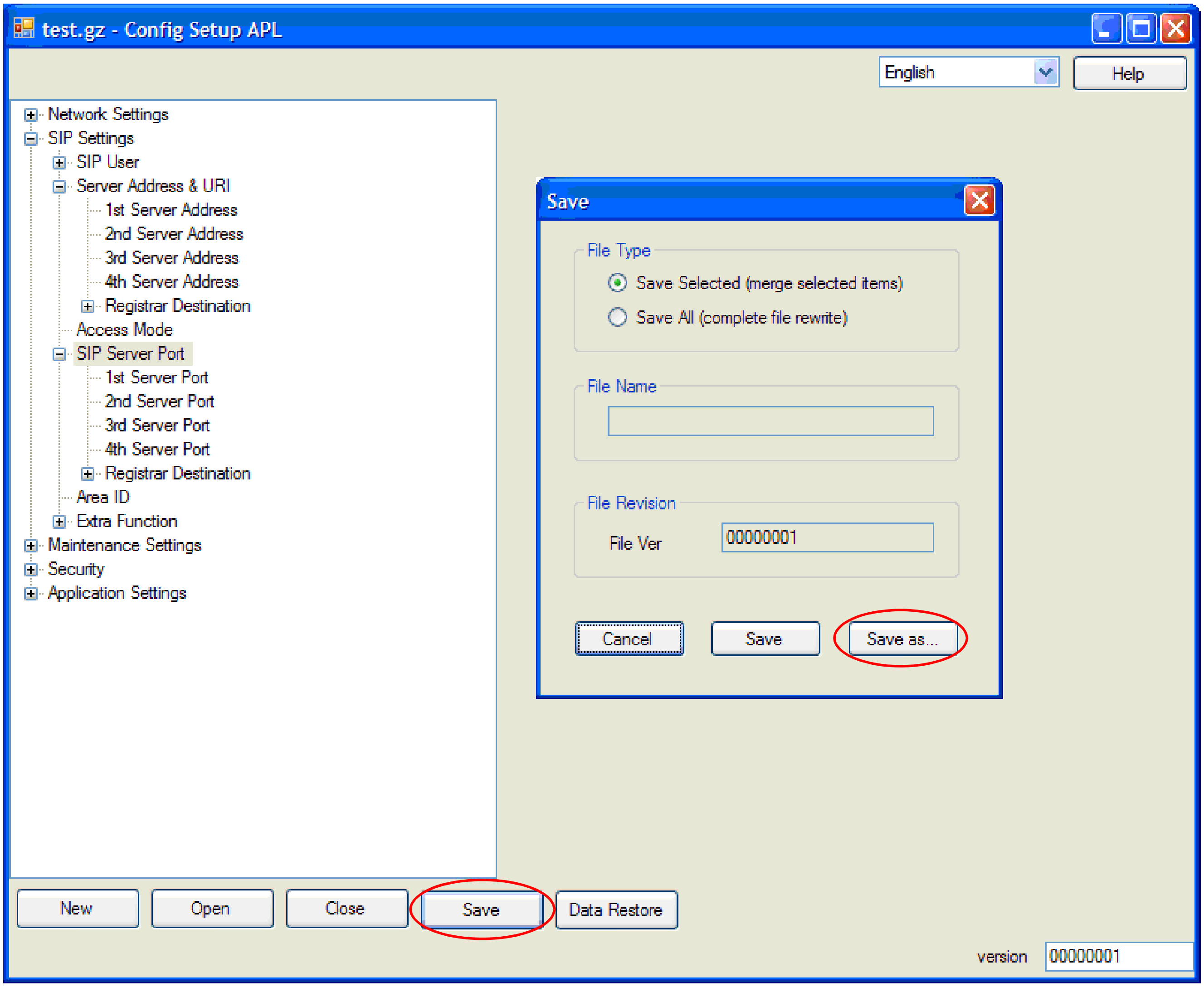
Figure 8-11 Network Settings - 1st Server Port



9. After the changes are made, click **Save**.

10. When the Save window opens, click **Save as...**

Figure 8-12 Save As/Name File Window



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

    Example: To name the file test, enter **test.gz**

12. Place this file in the FTP server.

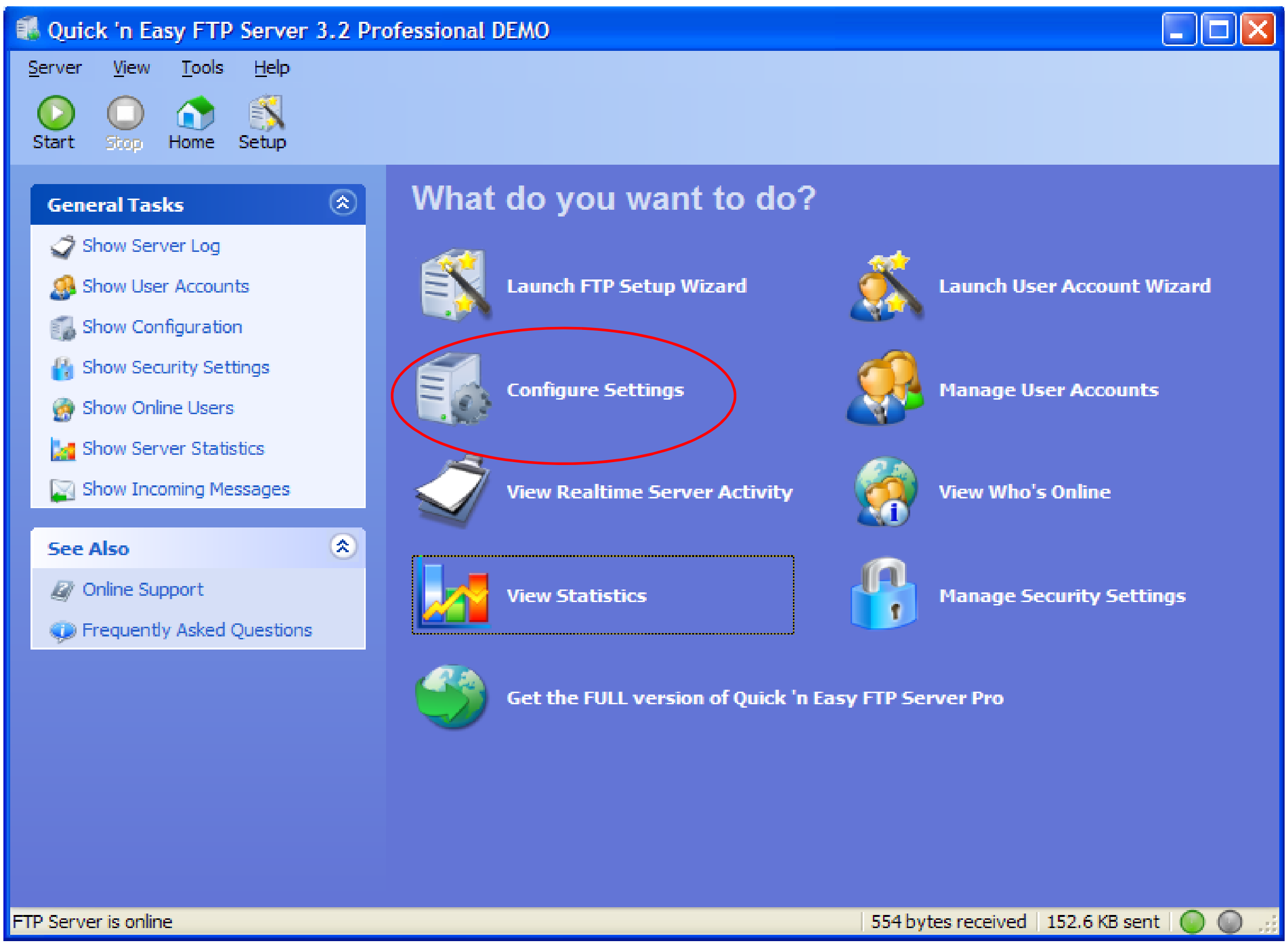
11.3 Configuring an FTP Server

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

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

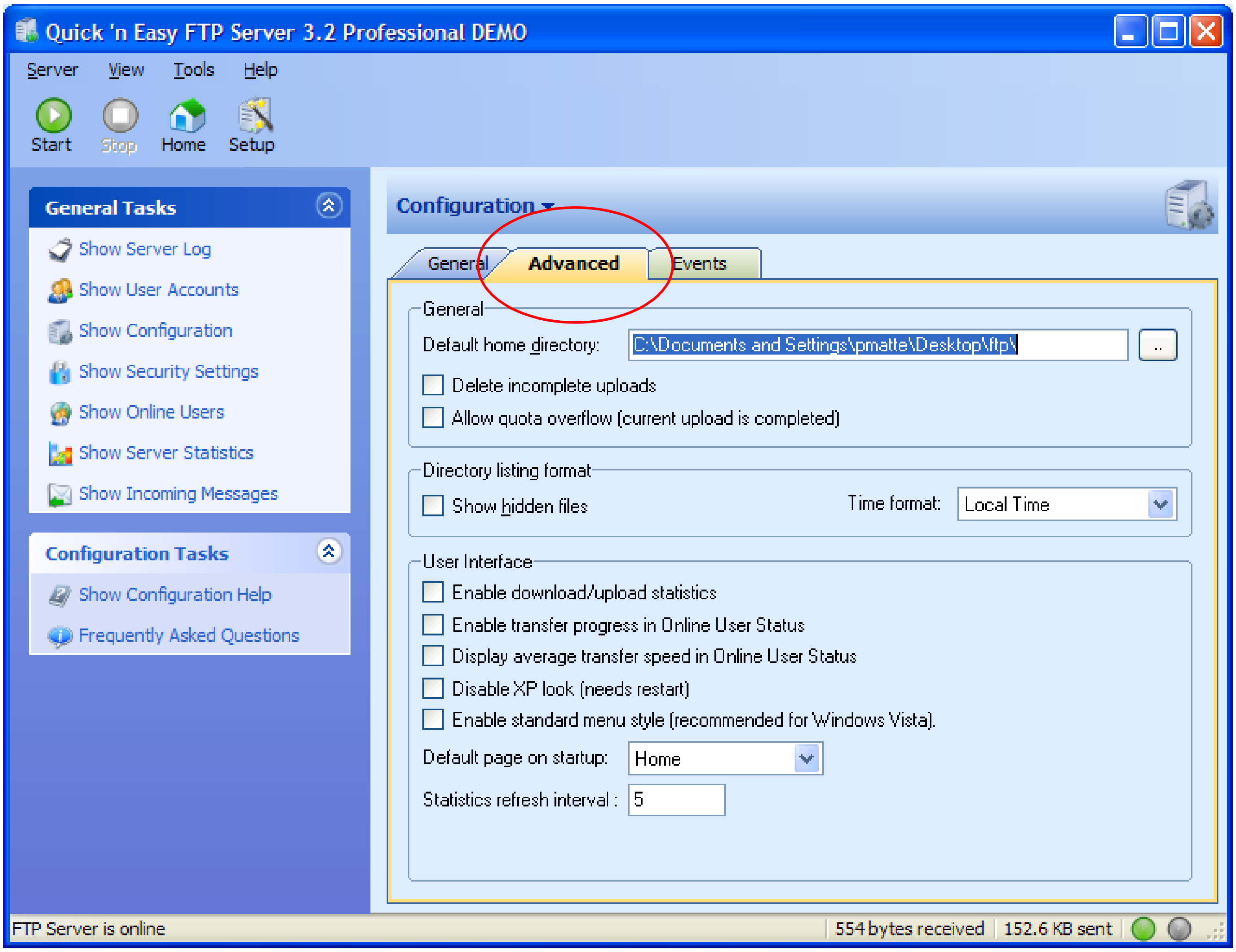
1. Click **Configure Settings**.

Figure 8-13 Quick 'n Easy FTP Server



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

Figure 8-14 Quick 'n Easy FTP Server Configuration



3. Place the file (for example: test.gz) in the Default home directory.

4. After the file is loaded to the proper directory, click **Start** to start the FTP server.

5. At this point the FTP server can be minimized to run in the background.

11.4 DHCP Server Setup Windows Server

This section provides instructions for defining vendor classes and setting

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 = NECDT800/DT700
4. Description = auto config
5. In the same window there is a section that shows ID, Binary, and ASCII. Click under ASCII.
6. Enter **NECDT800/DT700**. 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 **NECDT800/DT700**.
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 NECDT800/DT700.
3. Place a check mark next to 141 FTP Address. Down below assign the IP address of the FTP server. Then click **Apply**.
4. Place a check mark next to 151 auto config file name. Enter the name of the config file created using IP Phone Manager. Then click **Apply**.
5. Place a check mark next to 163 download protocol. Down below change the HEX address to be 0x1.
6. Click on **Apply** and **OK**.

SECTION 12 IP TELEPHONE REGISTRATION AND DELETION

When an IP Phone connects to the UNIVERGE SV9100 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 GPZ-IPLE
- ❑ Program 11-02-01 Extension Numbering
- ❑ Configure a System IP Phone and connect to the LAN

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

To delete a telephone registration:

Via Telephone Programming:

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

Via Web Pro:

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

SECTION 13 SYSTEM IP PHONES AND ANALOG TRUNKS

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

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

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

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

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

SECTION 14 FIRMWARE UPGRADE PROCEDURE

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

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

14.1 Manually Upgrading Firmware

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

To manually upgrade the firmware:

1. Install and configure an FTP/TFTP server.
2. Copy the firmware file **itlisip(e,s,v).tgz** to the default FTP/TFTP directory.
3. To enter Programming Mode, press the **Help** button on the IP Phone.
4. To enter Maintenance Mode, press **0** (Config) and **#3** (Maintenance).
5. To access the Download menu, press **1**.
6. Enter the FTP/TFTP server IP address in Option 2 - Download Address.
7. To enter the protocol, press **#3** (Protocol FTP or TFTP).

8. To select download by file, press option **1** (Download File).
9. To boot the program, press **3** (Boot Program).
10. Press the softkey.

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

14.2 Checking the Firmware Version

To check the IP Phone firmware version:

1. Press and hold the **Help** button on the System IP Phone.
2. To access information, press **8** (System Information).
3. To display the firmware version, press the **Up** softkey.

14.3 Upgrading Automatically

This procedure causes all IP Phones to attempt firmware upgrade the next time they connect to the GCD-CP10. This can make the upgrade procedure easier, as it is not necessary to visit every telephone to perform the upgrade.

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

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

To enable automatic upgrade:

When the IP phone boots up and connects to the SV9100 it receives download information from the system that includes firmware information. When the SV9100 reports a version that is different than the version the IP phone is currently utilizing, the IP phone will initiate the upgrade procedure.

IMPORTANT

Below is an example of this setup.

This is just an example, you must enter your own local information.


System Data
84-07 : Firmware Download Setup

01 - Server Mode	TFTP
02 - File Server IP Address	172.16.0.23
03 - Login Name	
04 - Password	

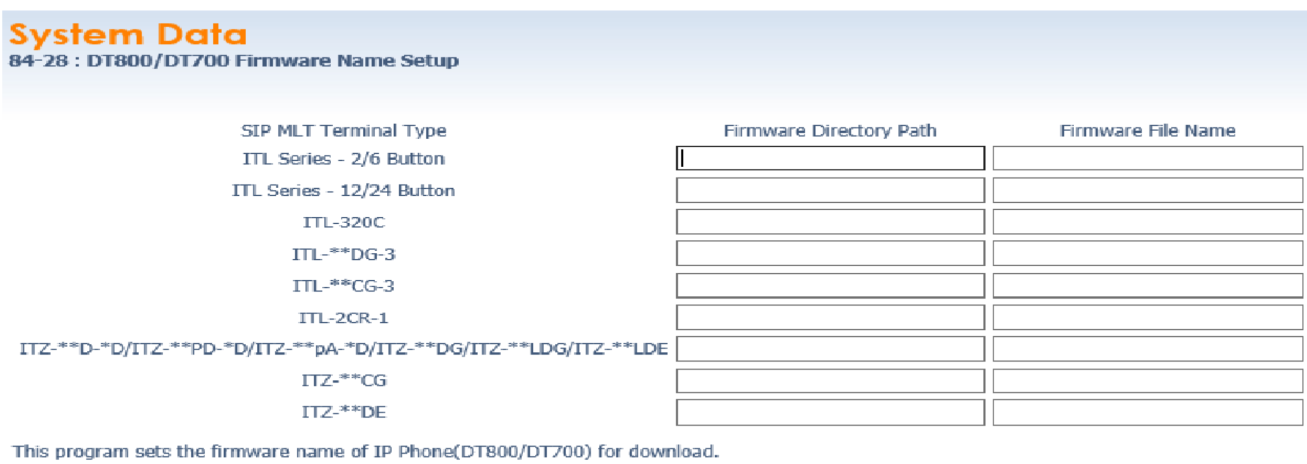
Value should be within the following ranges:

0.0.0.0-126.255.255.254
128.0.0.1-191.255.255.254
192.0.0.1-223.255.255.254

Use Program 84-07: Firmware Download Setup to specify the file server detials for downloading data for IP phones.

Enter the IP Address of the TFTP server that the IP Phone Firmware is stored on.

If using FTP, change the server mode to FTP and enter the proper IP Address and login name/ password.

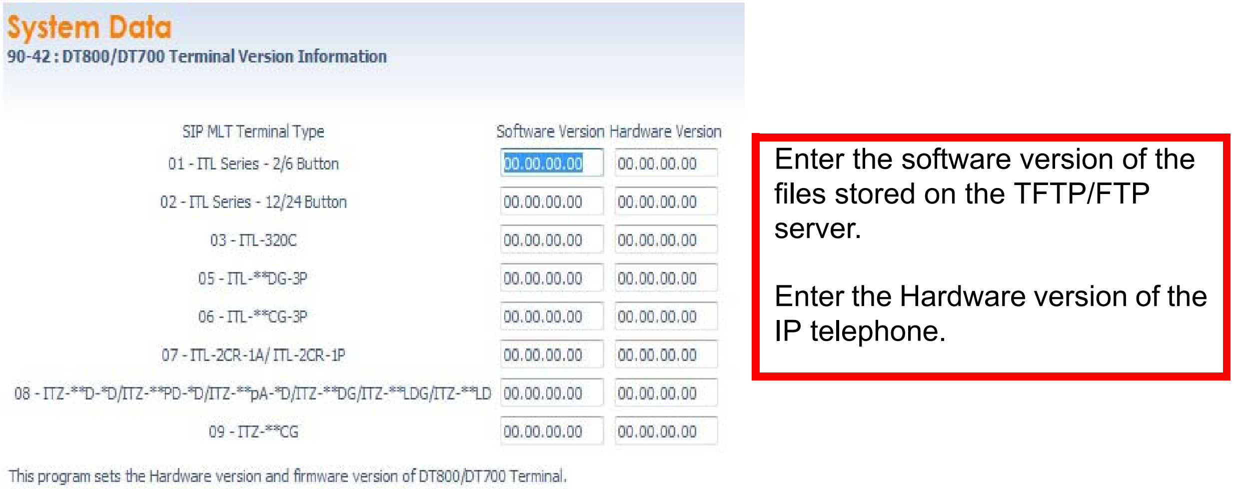

System Data
84-28 : DT800/DT700 Firmware Name Setup

SIP MLT Terminal Type	Firmware Directory Path	Firmware File Name
ITL Series - 2/6 Button		
ITL Series - 12/24 Button		
ITL-320C		
ITL-**DG-3		
ITL-**CG-3		
ITL-2CR-1		
ITZ-**D-*D/ITZ-**PD-*D/ITZ-**pA-*D/ITZ-**DG/ITZ-**LDG/ITZ-**LDE		
ITZ-**CG		
ITZ-**DE		

This program sets the firmware name of IP Phone(DT800/DT700) for download.

Enter the full directory pathof the exact location where the firmware is stored on to the FTP/ TFTP server.

Enter the filenames for the appropriate style of telephone.



SECTION 15 IP STATION (SIP MULTILINE TELEPHONE)

15.1 IP Phone Registration Modes

The SIP MLT supports three different registration modes.

- Plug and Play
- Automatic Login
- Manual Login

Plug and Play Registration

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

Automatic Login

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

Manual Login

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

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

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

Multiple Login

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 and uses UC Suite Applications with the Enhancement bundle controlling the IP Multiline, three different ports are used in the system.

Encryption

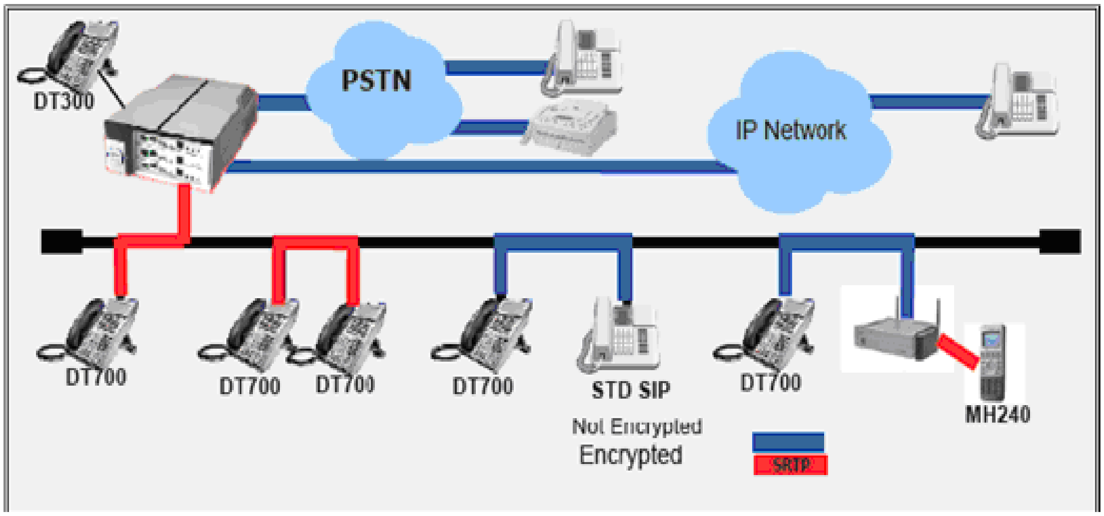
The SV9100 Supports AES 128-bit encryption between DT800/DT700 terminals and the GPZ-IPLE. This feature requires the SV9100 ENCRYPTION LIC which is a system license. Once installed, any of the DT800/DT700 IP terminals can use the encryption feature.

Source	Destination	SRTP	Comment
DT800/DT700	SDT SIP (P2P)	N	
DT800/DT700	STD SIP (Non P2P)	S	DT800/DT700 VoIPDB Encryption between DT800/DT700 and VoIPDB
DT800/DT700	DT800/DT700	S	
DT800/DT700	PSTN	S	DT800/DT700 VoIPDB Encryption between DT800/DT700 and VoIPDB
DT800/DT700	IP Network (SIP/H323/ CCIS)	S	DT800/DT700 VoIPDB Encryption between DT800/DT700 and VoIPDB

Source	Destination	SRTP	Comment
DT800/DT700 Encryption On	DT800/DT700 Encryption Off	N	

S = Supported N = Not Supported

Figure 8-15 DT800/DT700 Encryption



Conditions

- ❍ Encryption is not supported on DT800/DT700 series phones that are connected via NAPT.
- ❍ Encryption is not supported on DT800/DT700 series phones that are registered to a secondary NetLink system.
- ❍ DT800/DT700 series phones that are registered to a primary NetLink can fail over to a secondary system regardless of encryption settings.
- ❍ If the encryption feature is enabled in terminal programming but not licensed, the terminal display will no longer display "Invalid Info", however the encryption feature will not work until licensed.

15.2 General IP Configuration

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

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

- One way delay must not exceed 100ms.
- Round trip delay must not exceed 200ms.
- Packet loss must not exceed 1%.
- Data switches must be manageable.
- Routers must provide QOS.
- There must be adequate bandwidth for estimated VoIP traffic. Refer to Section 15.5 Bandwidth on page 8-44.

Depending on how QOS policies are built in the network, assignments may be needed in both the CPU and IP terminal. The UNIVERGE SV9100 supports the flagging of packets at layer 2 (VLAN tagging 802.1Q/802.1P) and at layer 3 levels.

15.3 VLANs

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

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

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

Currently the IPLE 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/IPLE is plugged into a data switch supporting VLANS, the VLAN tag must be removed before passing the frame onto the CPU/IPLE.

Tagging Voice and Data Packets

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

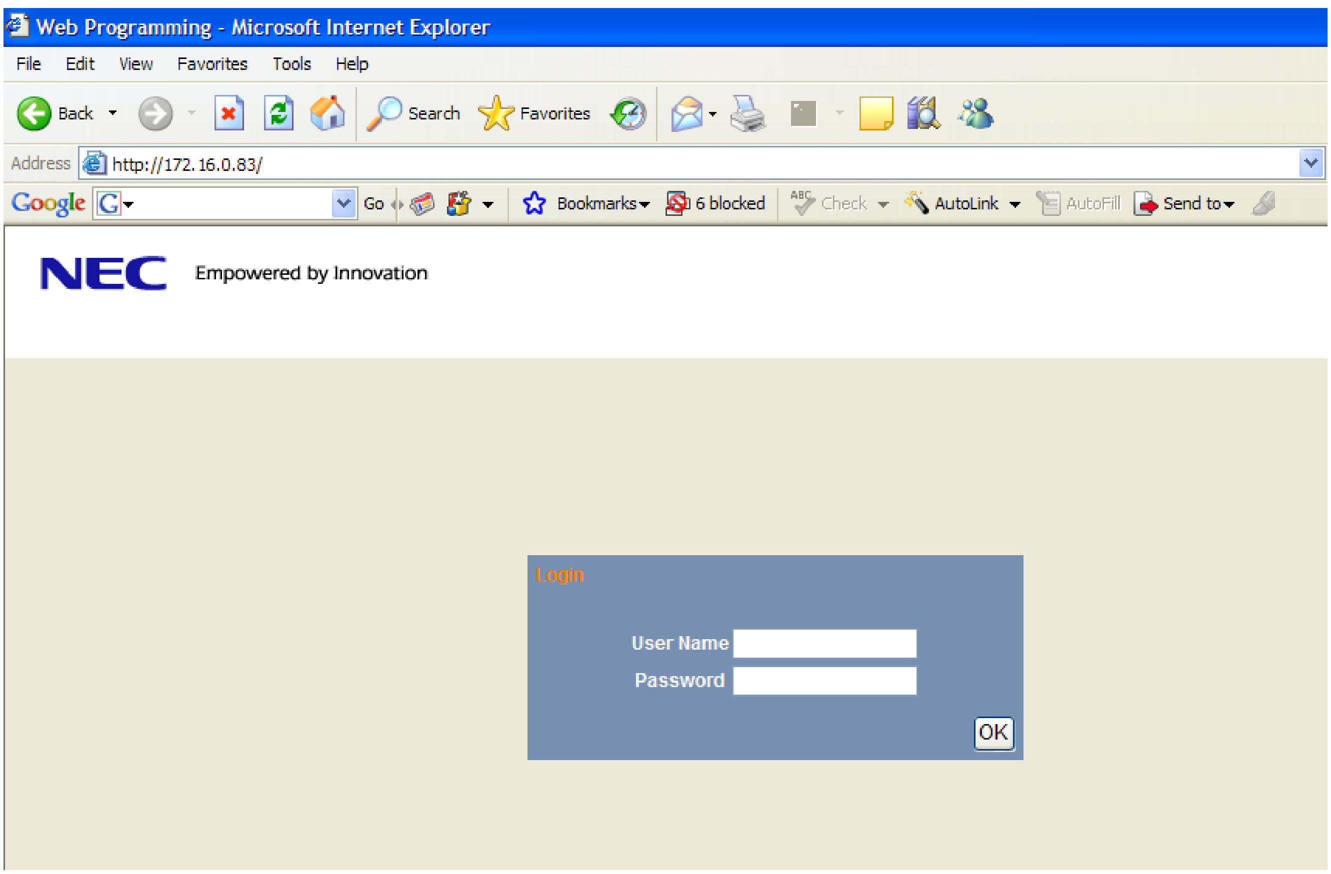
The following procedures describe two methods for tagging the voice packets and the data packets separately, using the PC, or using the phone keypad.

15.3.1 Logging In on the PC

Follow these steps to log into a PC.

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

Figure 8-16 Log In to IP Phone

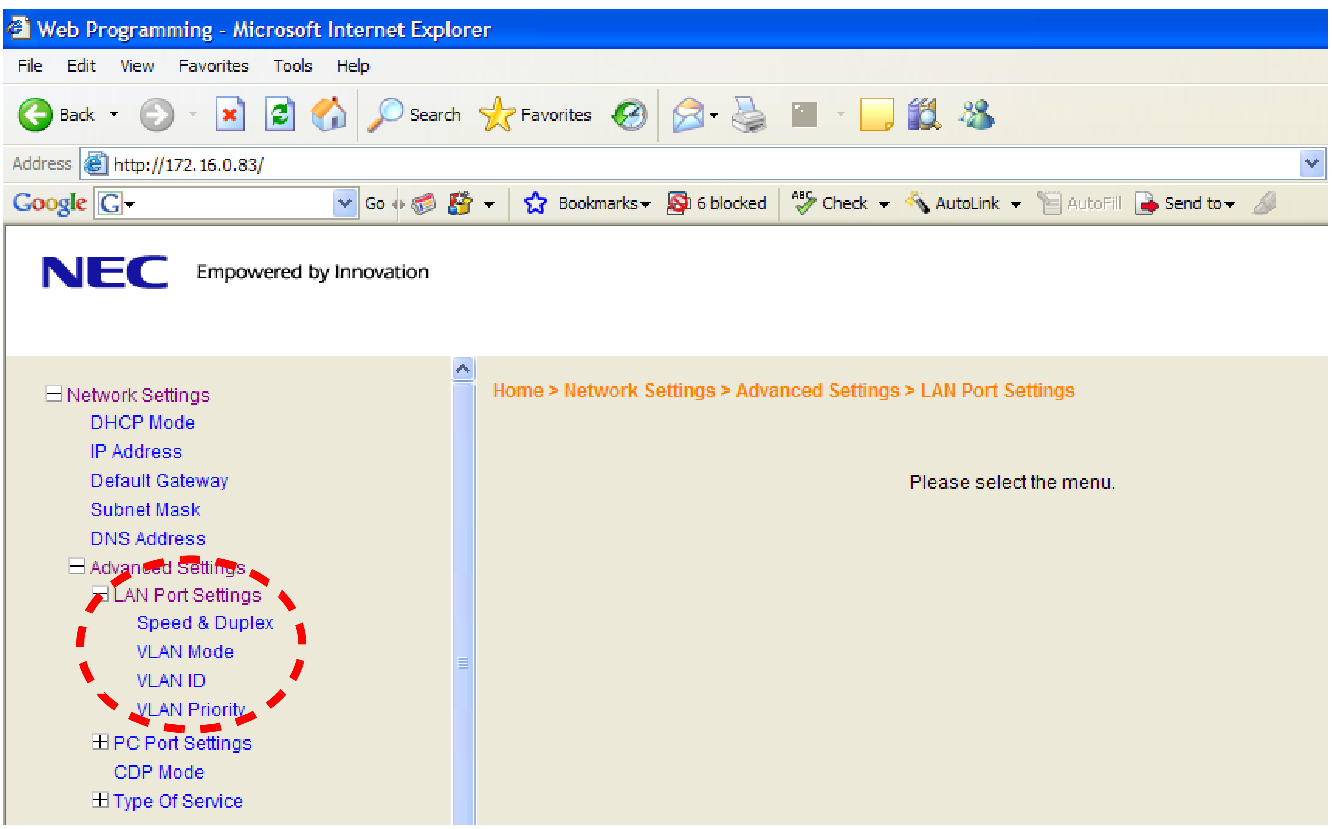


15.3.2 Tagging Voice Packets Using IP Phone

Follow these steps to program for voice packet tagging using an IP telephone.

1. Log in. Refer to Section 15.3.1 Logging In on the PC on page 8-30.
2. To apply a tag to the voice packets only, go to **Network Settings>Advanced Settings>LAN port settings**.
3. Access the following three menus to select options for LAN Port Settings:
   - VLAN Mode
   - VLAN ID
   - VLAN Priority

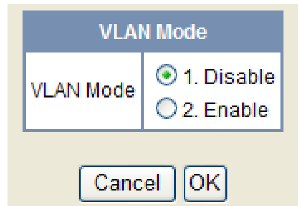
Figure 8-17 LAN Port Settings Window



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

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

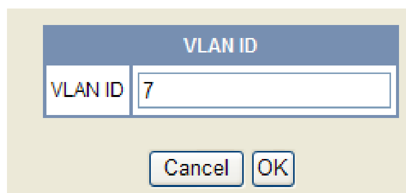
Figure 8-18 VLAN Mode



6. VLAN ID allows an entry of 1~4094 for the VLAN ID. VLAN Mode must be enabled for this entry to be valid.

   Enter the VLAN ID and click **OK**.

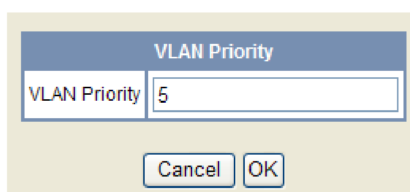
Figure 8-19 VLAN ID



7. VLAN Priority allows an entry of 0~7 for the VLAN Priority. VLAN mode must be enabled for this entry to be valid.

   Enter the required priority, and click **OK**.

Figure 8-20 VLAN Priority



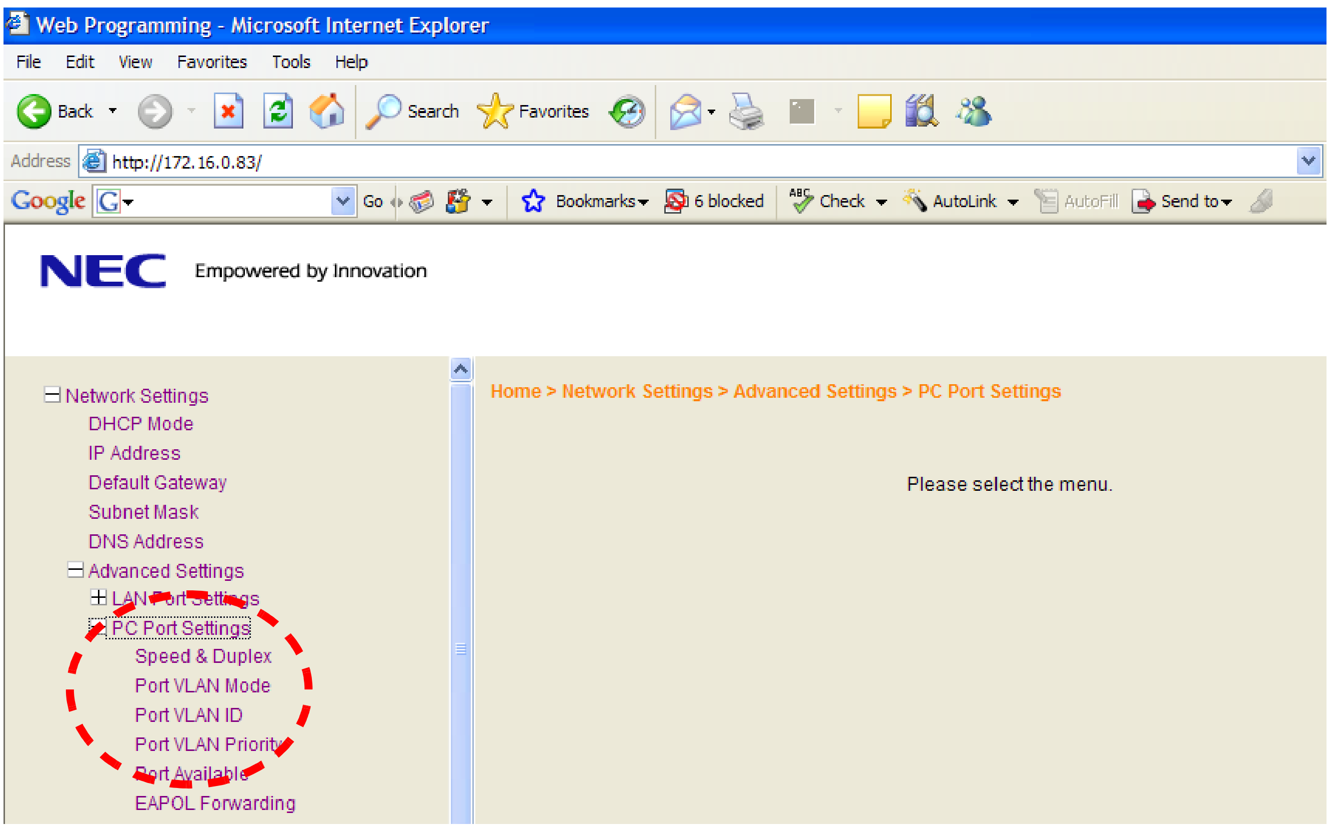
15.3.3 Tagging Data Packets Using IP Phone

Follow these steps program for data packet tagging using an IP telephone.

1. While logged in to the IP address of the phone on the PC, go to **Network Settings>Advanced Settings>PC Port Settings**. Refer to Section 15.3.1 Logging In on the PC on page 8-30.

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

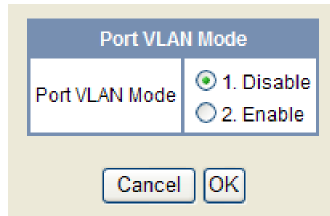
Figure 8-21 PC Port Settings Window



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

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

Figure 8-22 Port VLAN Mode

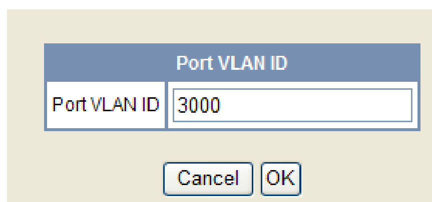


➨ *The remaining data packets settings for VLAN on the PC Port are the same as those for the voice packets.*

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

   Enter the VLAN ID, and click **OK**.

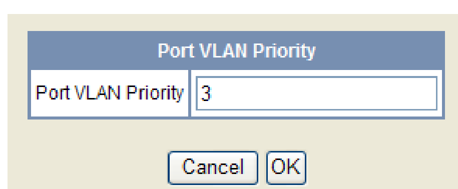
Figure 8-23 Port VLAN ID



6. VLAN Priority allows an entry of 0~7 for the VLAN Priority. VLAN mode must be enabled for this entry to be valid.

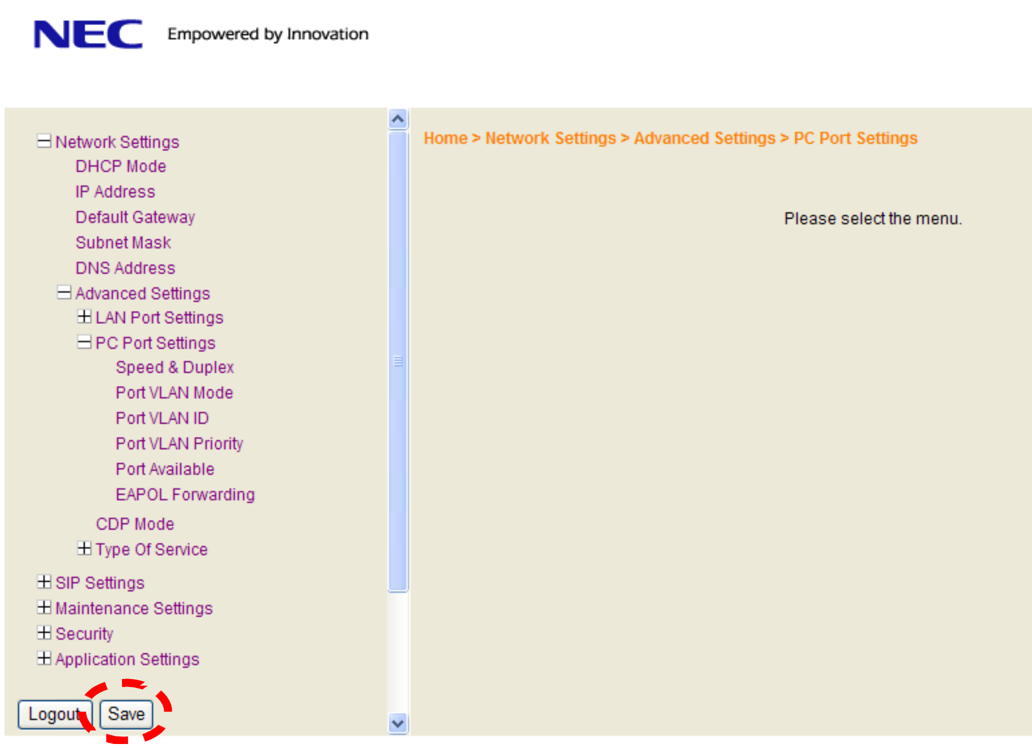
   Enter the required priority, and click **OK**.

Figure 8-24 Port VLAN Priority



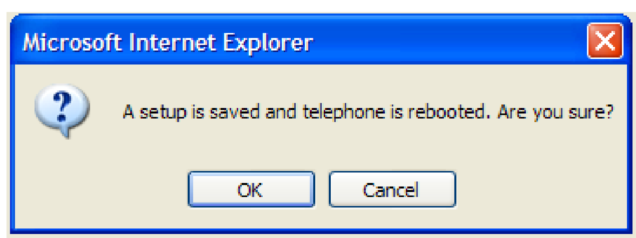
7. Click **Save**.

Figure 8-25 Save Network Settings



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

Figure 8-26 Save Confirmation Window



15.3.4 Entering VLAN Settings by Phone (Voice Packets Only)

Follow these steps to enter VLAN settings using a telephone.

1. Log in. Refer to Section 15.3.1 Logging In on the PC on page 8-30.
2. Press and hold the phone **Menu** button until the screen changes.
3. Press **0** on the dial pad to access Configuration.
4. Press the **OK** soft key. No login or password is associated with the IP terminal in the default state.
5. Press **1** on the dial pad for Network Settings.
6. Press **6** on the dial pad for Advanced Settings
7. Press **1** on the dial pad for LAN Port Settings (VLAN for the voice packets only)
8. Press **2** on the dial pad for VLAN Mode
9. Press **1** or **2** to disable/enable the VLAN for the voice packets. Press the OK soft key after the setting is changed.
10. Press **3** on the dial pad for VLAN ID
11. Enter a valid VLAN ID of 1~4094. Press the **OK** soft key after the setting is changed.
12. Press **4** on the dial pad for VLAN Priority
13. Enter the VLAN priority of 0~7. Press the **OK** soft key after the setting is changed.
14. If no more changes are made, press the **Exit** soft key three times. Then press the **Save** soft key, and the phone reboots.

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

Follow these steps to enter the VLAN setting for PC port using a telephone.

1. Log in. Refer to Section 15.3.1 Logging In on the PC on page 8-30.
2. Access the main menu.
3. Press **6** on the dial pad for Advanced Settings.
4. Press **2** on the dial pad for PC Port Settings.
5. Press **2** on the dial pad for Port VLAN Mode.
6. Press **1** or **2** to disable/enable the VLAN for the data packets. Press the **OK** soft key after the setting is changed.
7. Press **3** on the dial pad for Port VLAN ID.

8. Enter a valid VLAN ID of 1~4094. Press the **OK** soft key after the setting is changed.

9. Press **4** on the dial pad for Port VLAN Priority.

10. Enter the VLAN priority of 0~7. Press the **OK** soft key after the setting is changed.

11. If no more changes are made, press the **Exit** soft key three times. Then press the **Save** soft key, and the phone reboots.

15.4 ToS Settings (Layer 3 QoS)

The marking of packets at layer 3 is done by marking the ToS byte in the IP header of the voice packet. The UNIVERGE SV9100 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)
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)

DSCP Code Points	Binary Values	Names
110000	48	Class Selector 6 (CS6)
111000	56	Class Selector 7 (CS7)

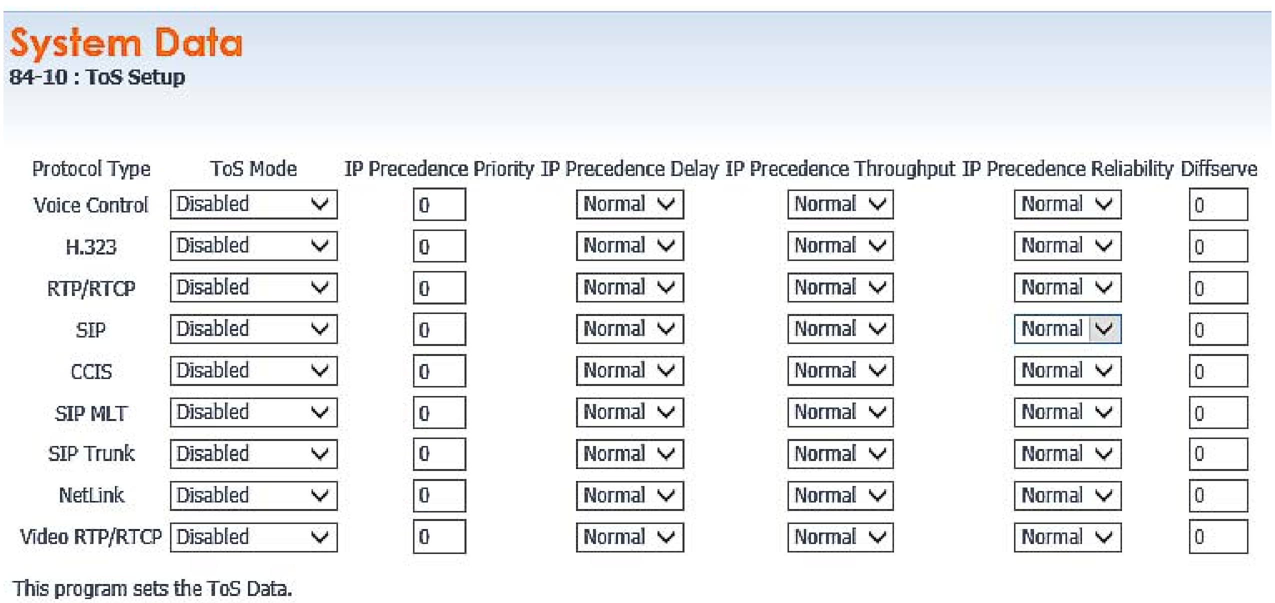
The following table shows the 8 bit TOS field and the associated Diffserv bits.

Diffserv	Diffserv	Diffserv	Diffserv	Diffserv	Diffserv	Not Used	Not Used
1(on) here = value of | 32	1(on) here = value of 1 6	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 IPLE card. This does not affect the packets sent from the IP terminals.

Figure 8-27 84-10: ToS Setup



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

❍ **System wide.** If all IP phones use the same ToS value, this can be assigned in commands 84-23-06 and 84-23-12. When an IP phone registers with the CPU, it looks for settings in these commands. If these are found, they override any previous individual settings.

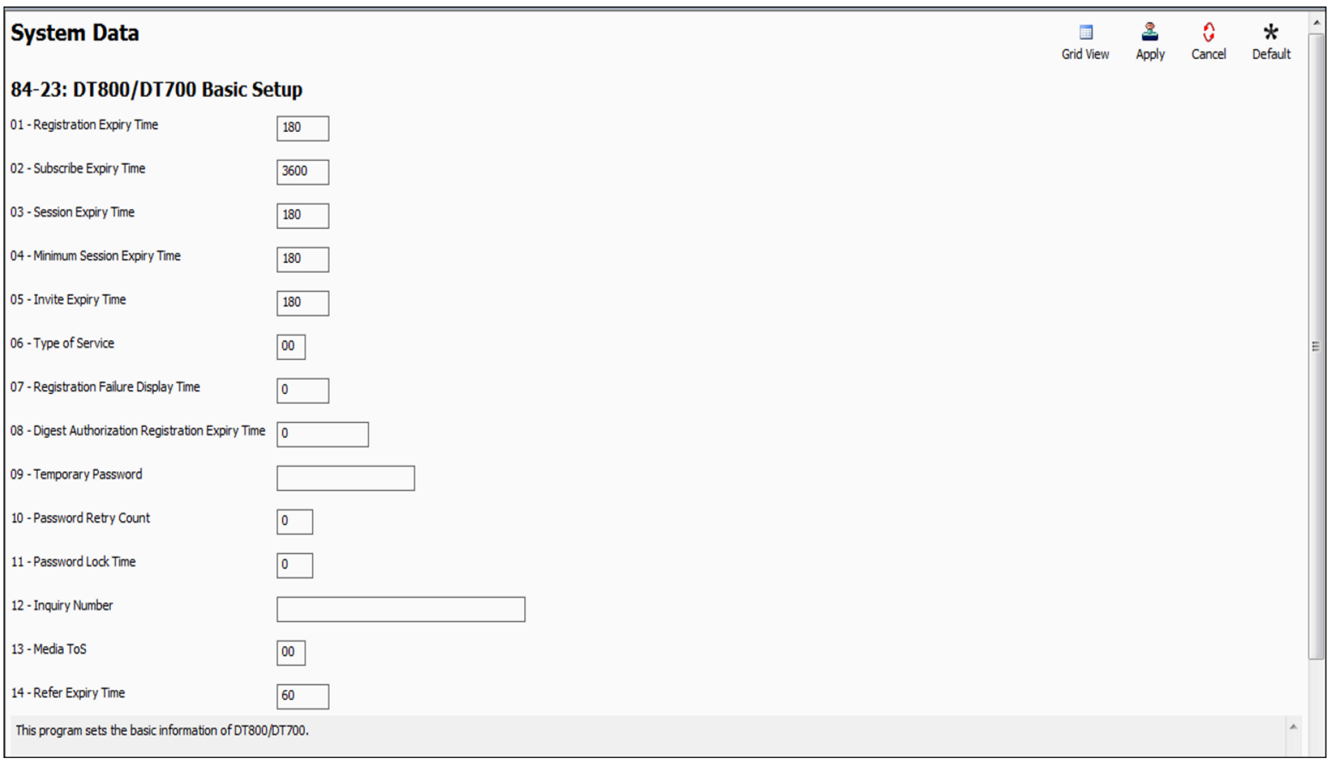
❍ **Individual.** If different IP phones require different ToS assignments, due to the network configuration, these assignments must be set at each individual station

Command 84-23 requires a Hexadecimal representation of the 8 bit ToS field. For example, to assign the signaling packets an IP precedence value of 4 and the voice packets an IP precedence value of 5, it would be as follows. Refer to Figure 8-28 SIP DT800/DT700 Basic Setup.

❍ 80 in Hex is 10000000 - This represents the signaling packets leaving the IP phone

❍ A0 in Hex is 10100000 - This represents the voice packets leaving the IP phone

Figure 8-28 SIP DT800/DT700 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

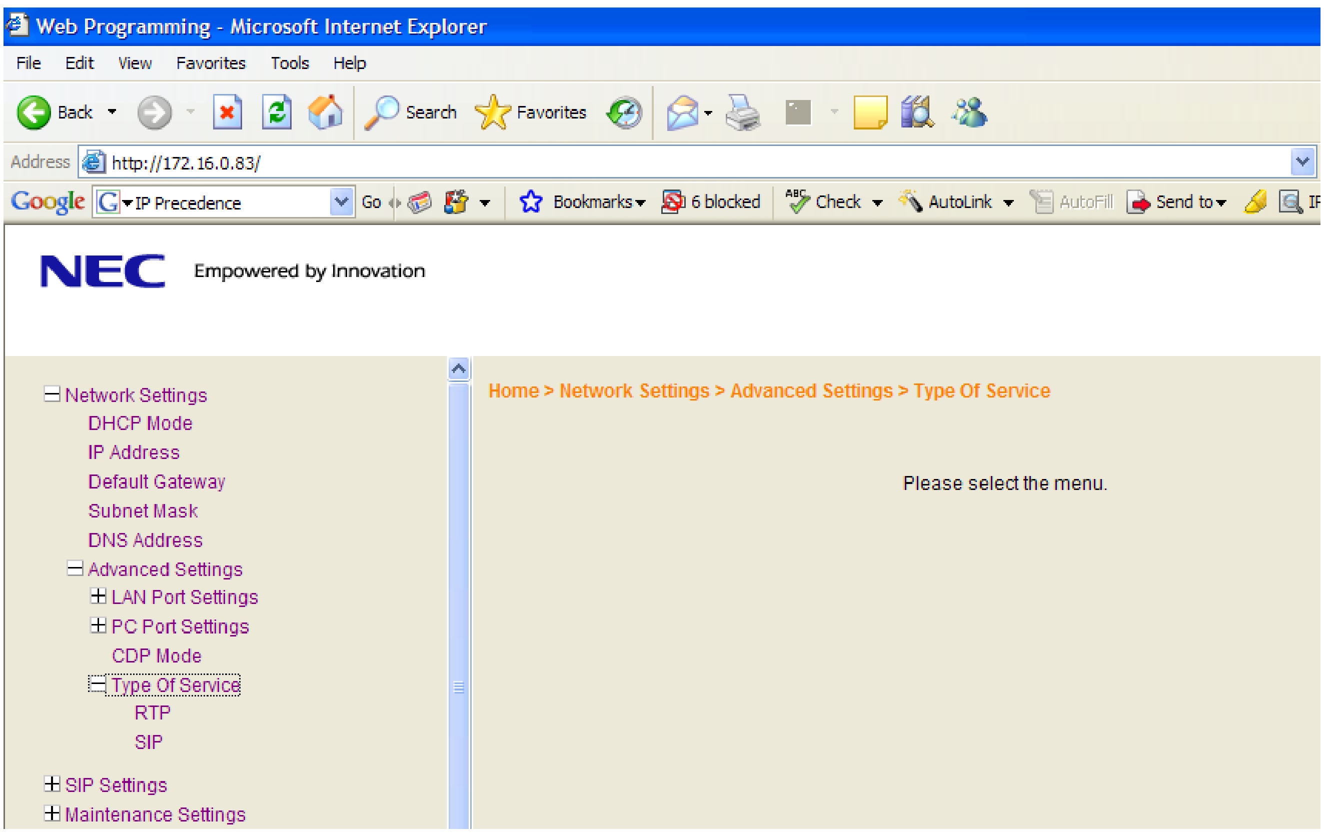
- By the web browser
- By configuration mode through the dial pad

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 15.3.1 Logging In on the PC on page 8-30.
2. Go to **Network Settings>Advanced Settings>Type Of Service**.

Figure 8-29 Type of Service Window



3. There are two choices: RTP and SIP. RTP = voice packets and SIP = signaling packets.

   Select each field and assign the appropriate value. Then select **OK**.

   These fields are also looking for a Hexadecimal value as with command 84-23. Refer to Table 8-2 Common IP Precedence/Diffserv Values and Hexadecimal Equivalent on page 8-41.

Access the following menus to select options:

- ❐ RTP - Voice Packets
- ❐ SIP - Signalling Packets

Figure 8-30 RTP - Voice Packets

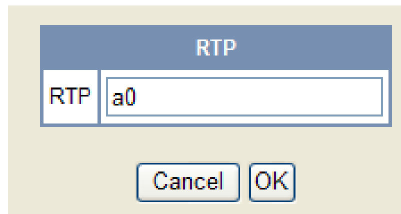
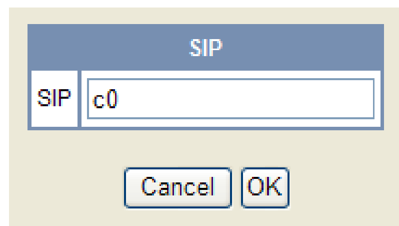


Figure 8-31 SIP - Signalling Packets



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

Assign Values on IP Terminal

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

1. Press and hold the **Menu** button until display changes.
2. Press **0** on the dial pad for Configuration.
3. At a default state there is no login or password associated to the IP terminal. Press the **OK** soft key.
4. Press **1** on the dial pad for Network Settings.
5. Press **6** on the dial pad for Advanced Settings.
6. Press **4** on the dial pad for Type of Service.
7. There are two options (1) is RTP and (2) is SIP.
8. Press **1** on the dial pad for RTP (voice packets), enter the hexadecimal value, and then press the **OK** soft key.

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

10. If no more changes are to be made, press the Exit soft key three times, and then press the **Save** soft key. The phone reboots.

15.5 Bandwidth

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

- Layer 2 media
- CODEC
- Packet Size
- RTP Header Compression
- Voice Activity Detection (VAD)
- Number of simultaneous calls
- Possibly add encryption after research.

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

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

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

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

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

- VoIP packet without RTP header compression

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

- VoIP packet with RTP header 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

15.6 Some Network Considerations

When adding the SV9100 to a customers network there are many things to be aware of. Before implementation a detailed network diagram of the existing network must be obtained from the customer. This diagram may provide you with information about possible network conditions that can prevent or hinder the VoIP equipment from functioning correctly.

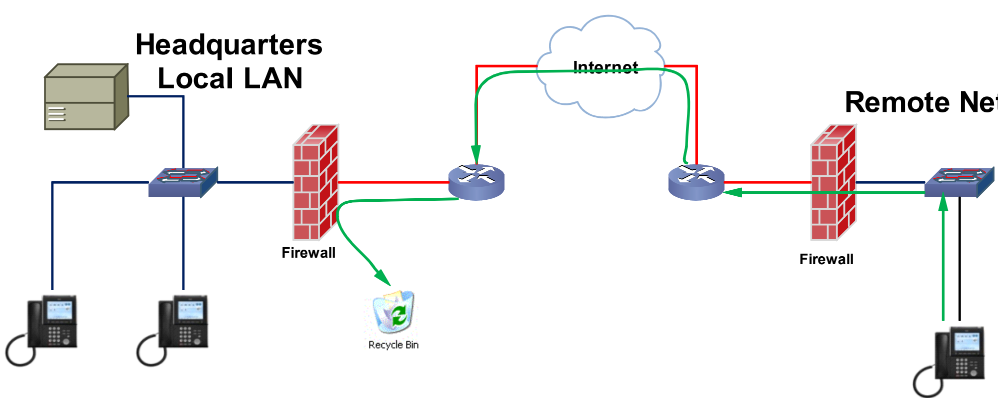
Firewalls

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

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

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

Figure 8-32 SV9100 Network Example No. 1



The green arrow in the diagram above represents the data packets leaving the IP phone destined for the SV91000 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 SV9100.

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

- **IPLE UDP Ports : 10020~10531**

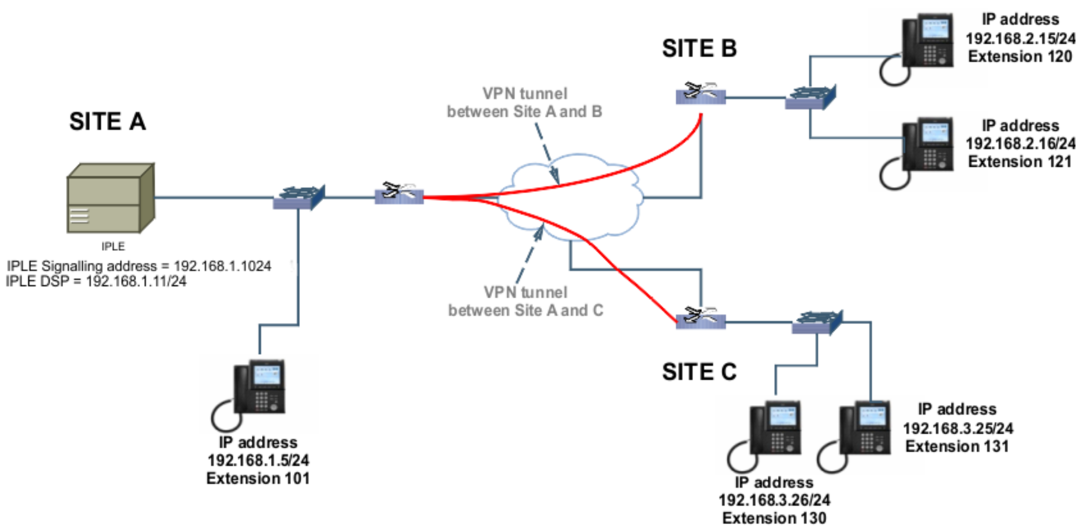
The ports that need to be opened on the Remote network are **5060** (UDP) for signaling and ports **3462** and **3463** for voice (UDP).

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 SV9100. The downside to disabling Peer-to-Peer is using more DSPs and consumption of additional bandwidth at the MAIN location.

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

Figure 8-33 SV9100 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 (IPLE).
- 192.168.1.10 (IPLE) 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 (IPLE). In the 200 OK message is the SDP field reporting the IP address of 192.168.3.26 (ext 130).
- 192.168.1.10 (IPLE) 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 (IPLE).
- 192.168.1.10 (IPLE) 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 GPZ-IPLE card. This also affects IP phones at the REMOTE locations calling other IP phones at the same location. Without Peer-to-Peer enabled the voice path must route to the MAIN location and then back to the REMOTE instead of directly between the two stations on the REMOTE network. This forces the use of additional bandwidth on the MAIN, and REMOTE locations. Peer-to-Peer is disabled in command 15-05-50 per IP Phone.

15.7 Guide to Feature Programming

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
10-12-03	GCD-CP10 Network Setup – Default Gateway	IPLE 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	GCD-CP10 Network Setup – IP Address	Assign Layer 3 IP Address to the IPLE 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	GCD-CP10 Network Setup – Subnet Mask	Assign Subnet Mask to the IPLE connected to CCPU.	Default is 255.255.0.0	X		
84-26-01	IPL Basic Setup – IP Address	Assign an IP Address to IPLE	Default Value: 172.16.0.10	X		
84-26-02	IPL Basic Setup – RTP Port Number		Range: 0 ~ 65534 Default Values: RTP Port = 10020			X
84-26-03	IPL Basic Setup – RTCP Port Number (RTCP Port Number +1)		Range: 0 ~ 65534 Default Values: RTCP Port +1 = 10021			X

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
10-46-01	DT800/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	DT800/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	DT800/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
10-46-07	DT800/DT700 Server Information Setup – Encryption Mode	Enable or disable encryption mode.	0 = Off 1 = On Default is 0		X	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
10-46-08	DT800/DT700 Server Information Setup – Encryption Type	Assign the encryption type.	0 = Mode 1 Default is 0		X	
10-46-09	DT800/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	DT800/DT700 Server Information Setup – Start Port	With Automatic logon the starting port number for automatic port allocation.	Range = (1 ~ 960) Default = 1		X	
15-05-01	IP Telephone Terminal Basic Data Setup – Terminal Type	Type of IP terminal registered with the specified extension number.	1 = H.323 2 = SIP 3 = None 4 = DT800/DT700			X
15-05-02	IP Telephone Terminal Basic Data Setup – IP Phone Fixed Port Assignment	Allow association of a MAC Address to an extension. When the IP phone sends a register message to the CPU the CPU responds back with the extension number associated to the MAC address.	00.00.00.00.00.00~ FF.FF.FF.FF.FF.FF Default is 00.00.00.00.00.00		X	
15-05-07	IP Telephone Terminal Basic Data Setup – Using IP Address	IP address the IP Terminal is using for the specified extension number.	0.0.0.0~ 255.255.255.255. Default is 0.0.0.0			X
15-05-15	IP Telephone Terminal Basic Data Setup –CODEC Type	Assign CODEC type for IP Terminal. If SIP SLT, use Program 84-19. If SIP MLT, use Program 84-24.	1 = Type 1 2 = Type 2 3 = Type 3 4 = Type 4 5 = Type 5 1 = Default is 1		X	
15-05-19	IP Telephone Terminal Basic Data Setup – Side Option Information	Read Only CM showing type of Line Key unit installed on the 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 telephone.	0 = No Option 1 = ADA 2 = BHA 4 = BCA Default is 0 **READ ONLY**			X

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
15-05-21	IP Telephone Terminal Basic Data Setup –Handset Option Information	Read Only CM showing type of Handset installed on the 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 telephone.	0 = No Setting 1~32 = DSS Console Number Default is 0 **READ ONLY**			X
15-05-23	IP Telephone Terminal Basic Data Setup –Handset Option Additional Information		0 = No Setting 1-16 = Terminal equipment number (TEN) of Bluetooth Cordless Handset (BCH) Default is 0			X
15-05-24	IP Telephone Terminal Basic Data Setup –Protection Service	When enabled allows SIP Multi-Line phones to use the Security button located at top of the SIP MLT display. When disabled, the Security key has no effect.	0 = Not Used 1 = Used Default is 0		X	
15-05-26	IP Telephone Terminal Basic Data Setup – DT800/ DT700 Terminal Type	Assign type of SIP MLT terminal connected.	0 = No Setting (default) 1 = ITL-**E-1D/IP-*E-1 2 = ITL-**D-1D/ITL 12BT-1D/ ITL-12PA-1D (without 8LKI(LCD)-L) 3 = ITL-**D-1D/ITL-12BT-1D/ITL-12PA-1D (with 8LKI(LCD)-L) 4 = ITL-320C-1 5 = SoftPhone 6 = CTI 7 = AGW 8 = IP3NA-8WV 9 = Not Used 10 = ITL-**DG-3 11 = ITL-**CG-3 12 = ITL-2CR-1 13 = ITZ-**D-1D/ITZ-**PD-1D/ITZ-**pA-1D/ ITZ-**DG/ITZ-**LDG 14 = ITZ-*CG 15 = ITZ-*DE 16 = ITZ-*LDE			X

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
15-05-27	IP Telephone Terminal Basic Data Setup –Personal ID Index	For SIP Multiline phone using Manual/Auto registration. Assign each phone a unique personal index. When complete go to command 84-22 and set the user name and password.	0 = Not Set 1-960 = Set Default is 0		X	
15-05-28	IP Telephone Terminal Basic Data Setup –Additional Information Setup		0 = Disable 1 = Enable Default is 0		X	
15-05-29	IP Telephone Terminal Basic Data Setup –Terminal WAN Side IP Address	Future use with NAT	0.0.0.0 ~ 255.255.255.255 Default is 0.0.0.0		X	
15-05-30	IP Telephone Terminal Basic Data Setup –DTMF Play During Conversation at Receive Extension		0 = Disable 1 = Enable Default is 0			X
15-05-31	IP Telephone Terminal Basic Data Setup – Alarm Tone During Conversation (RTP packet loss alarm)		0 = Disable 1 = Enable Default is 1			X
15-05-32	IP Telephone Terminal Basic Data Setup – Key Pad Talkie		0 = Disable 1 = Enable Default is 0			X
15-05-33	IP Telephone Terminal Basic Data Setup – LAN Side IP Address of Terminal		0 = Disable 1 = Enable Default is 0.0.0.0 **READ ONLY**			X
15-05-34	IP Telephone Terminal Basic Data Setup – Terminal Touch Panel On/Off	Whether the touch screen used on ITL-320C-( ) (BK) TEL can be used (1) or cannot be used (0).	0 = Off 1 = On Default is 1			X

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
84-10-XX	ToS Setup	Assignments deal with setting of the Layer 3 IP Header ToS field as it leaves the VoIPDB unit. Specify the protocol to assign the ToS field for, and select the populated field to conform to either IP Precedence or Differentiated Services. When setting IP Precedence, assign Priority, Delay, Throughput, and Reliability in Programs 84-10-01/02/03/04/05. When setting DiffServ, only assign the DSCP in Program 84-10-07.	Protocol Type 1 = Not Used 2 = Not Used 3 = Voice Control 4 = H.323 5 = RTP/RTCP 6 = SIP 7 = CCISoIP 8 = SIP MLT 9 = SIP Trunk 10 = NetLink 11 = Video RTP/RTCP		X	
84-22-01	DT800/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	DT800/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	DT800/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	DT800/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	DT800/DT700 Multiline Logon Information Setup – Nickname		Assign up to 32 Alpha/ Numeric Characters Default is No Setting		X	
84-23-01	DT800/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	DT800/DT700 Multiline Basic Information Setup – Subscribe Expire Timer	At half the value of this timer the IP terminal sends another Subscribe message to the CPU.	Range: 60~65535 Sec. Default is 3600		X	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
84-23-03	DT800/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	DT800/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	DT800/DT700 Multiline Basic Information Setup – Invite Expire Timer	When INVITE message received from SIP MLT does not contain Expires header, the CPU uses this value for timeout of outgoing call. E.g. The SIP MLT hears RBT for duration of this timer and then is disconnected.	Range: 0~65535 Sec. Default is 180		X	
84-23-06	DT800/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	DT800/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	DT800/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	DT800/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	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
84-23-11	DT800/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	DT800/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	DT800/DT700 Multiline Basic Information Setup – Media Type of Service	Assign the IP terminals ***MEDIA TOS*** values.	Range: 0x00 ~ 0xFF (0~9, A~F) Default is 00		X	
84-23-14	DT800/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	DT800/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	DT800/DT700 Multiline CODEC Basic Information Setup – G.711 Voice Activity Detection	Enable/Disable VAD for G.711	0 = Disable 1 = Enable Default is 0		X	
84-24-03	DT800/DT700 Multiline CODEC Basic Information Setup – G.711 Type	μ-law used in N.A.	0 = A-law 1 = μ-law Default is 0		X	
84-24-04	DT800/DT700 Multiline CODEC Basic Information Setup – G.711 Jitter Buffer Minimum	Minimum value of the dynamic jitter buffer.	Range: 0 ~ 300ms Default is 20		X	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
84-24-05	DT800/DT700 Multiline CODEC Basic Information Setup – G.711 Jitter Buffer Average	Average value of the dynamic jitter buffer.	Range: 0 ~ 300ms Default is 40		X	
84-24-06	DT800/DT700 Multiline CODEC Basic Information Setup – G.711 Jitter Buffer Maximum	Maximum value of the dynamic jitter buffer.	Range: 0 ~ 300ms Default is 80		X	
84-24-07	DT800/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 5 = 50ms 6 = 60ms Default is 2		X	
84-24-08	DT800/DT700 Multiline CODEC Basic Information Setup – G.729 Voice Activity Detection	Enable/Disable VAD for G.729	0 = Disable 1 = Enable Default is 0		X	
84-24-09	DT800/DT700 Multiline CODEC Basic Information Setup – G.729 Jitter Buffer Minimum	Minimum value of the dynamic jitter buffer.	0~300ms Default is 20		X	
84-24-10	DT800/DT700 Multiline CODEC Basic Information Setup – G.729 Jitter Buffer Average	Average value of the dynamic jitter buffer.	0~300ms Default is 40		X	
84-24-11	DT800/DT700 Multiline CODEC Basic Information Setup – G.729 Jitter Buffer Maximum	Maximum value of the dynamic jitter buffer.	0~300ms Default is 80		X	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
84-24-17	DT800/DT700 Multiline CODEC Basic Information Setup – Jitter Buffer Mode		1 = Static 2 = Self adjusting 3 = Adaptive immediate Default is 2		X	
84-24-18	DT800/DT700 Multiline CODEC Basic Information Setup - Voice Activity Detection Threshold		0 ~ 30 Default is 20			
84-24-23	DT800/DT700 Multiline CODEC Basic Information Setup - Echo Canceller Non-linear Processing Noise		40 ~ 70 dBm Default is 70dBm			X
84-24-25	DT800/DT700 Multiline CODEC Basic Information Setup – Echo Canceller 4W DET		0 = Disable 1 = Enable Default is 1			X
84-24-28	DT800/DT700 Multiline CODEC Basic Information Setup – Audio Capability Priority	This assigns the CODEC to be used.	0~3 0 = G.711_PT 2 = G.729_PT 3 = G.722_PT Default is 0		X	
84-24-32	DT800/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 3		X	
84-24-33	--Not Used--					
84-24-34	DT800/DT700 Multiline CODEC Basic Information Setup – G.722 Jitter Buffer Minimum	Minimum value of the dynamic jitter buffer.	Range: 0 ~ 300ms Default is 30		X	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
84-24-35	DT800/DT700 Multiline CODEC Basic Information Setup – G.722 Jitter Buffer Average	Average value of the dynamic jitter buffer.	Range: 0 ~ 300ms Default is 60		X	
84-24-36	DT800/DT700 Multiline CODEC Basic Information Setup – G.722 Jitter Buffer Maximum	Maximum value of the dynamic jitter buffer.	Range: 0 ~ 300ms Default is 120		X	
84-28-01	DT800/DT700 Multiline Firmware Name Setup – Firmware Directory	Maximum 64 characters.	Default is No Setting		X	
84-28-02	DT800/DT700 Multiline Firmware Name Setup – Firmware File Name	Maximum 30 characters.	Default is No Setting		X	

15.8 SIP MLT Quick Startup Guide

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

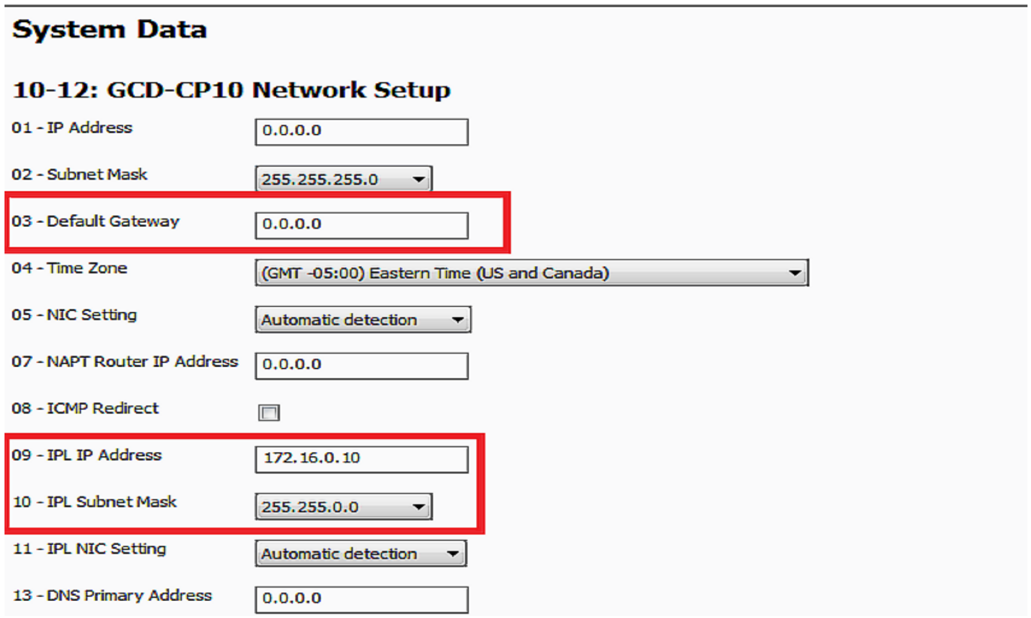
- Plug and Play
- Automatic Registration
- Manual Registration

15.8.1 Plug and Play

1. Program 10-12

   Assign the IPLE 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 8-34 System Data 10-12: GCD-CP10 Network Setup



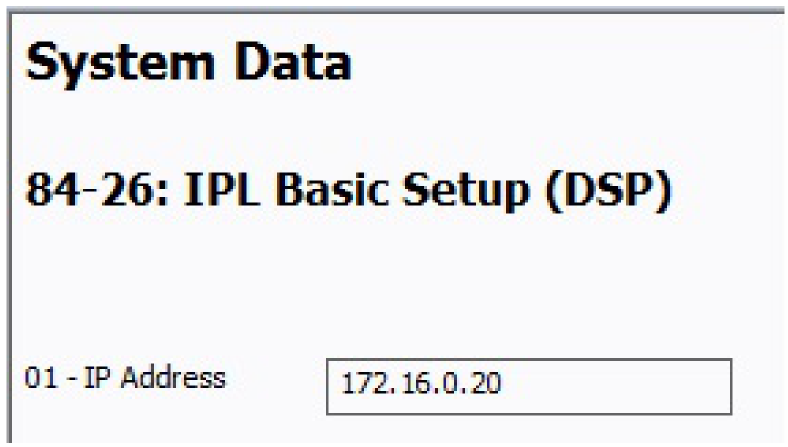
2. Program 84-26

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 to the IPLE. An IPLE provides 256 voice paths from IP to TDM & vice versa with appropriate licenses.

Figure 8-35 System Data 84-26: IPLE Basic Setup (DSP)



3. Program 11-02

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

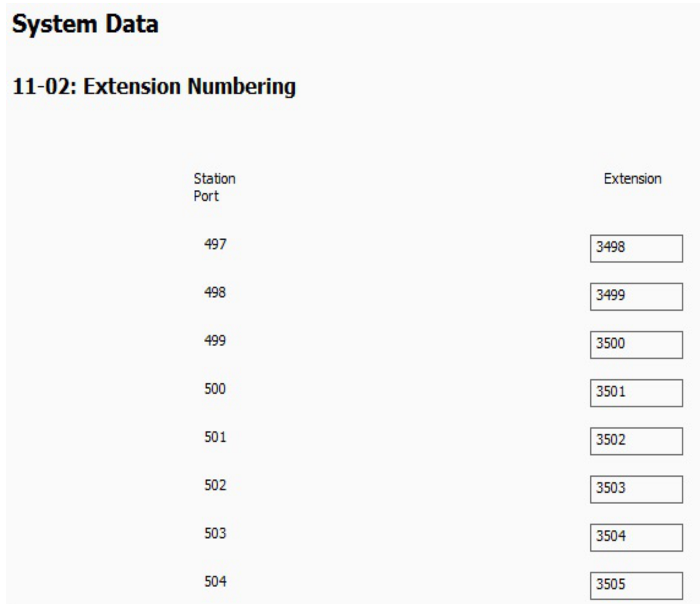
Physical Station ports are assigned automatically from lowest number ascending as cards are added to the system.

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 SV9100. 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. For example, in the image to the left, if port 504 (Extension 5203) 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 8-36 System Data 11-02: Extension Numbering



System Data

11-02: Extension Numbering

Station Port	Extension
497	3498
498	3499
499	3500
500	3501
501	3502
502	3503
503	3504
504	3505

4. This step is optional.

   To enable Key data and other station feature programming (before IP Phone is brought online) the extensions must be identified as IP Phones. Once checked in the IP Phone List in PCPro, the extensions are available for selection in Program 15 and other station related

Programs.

Figure 8-37 PCPro Programming Unregistered Phones

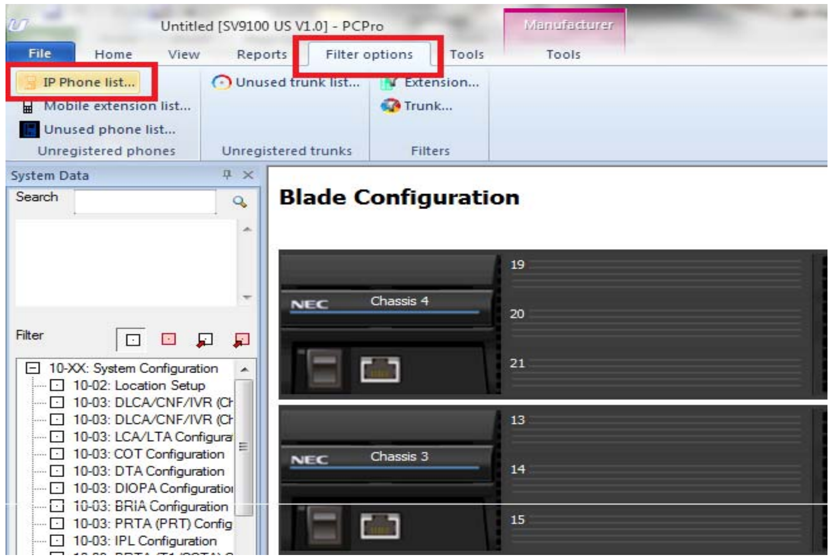
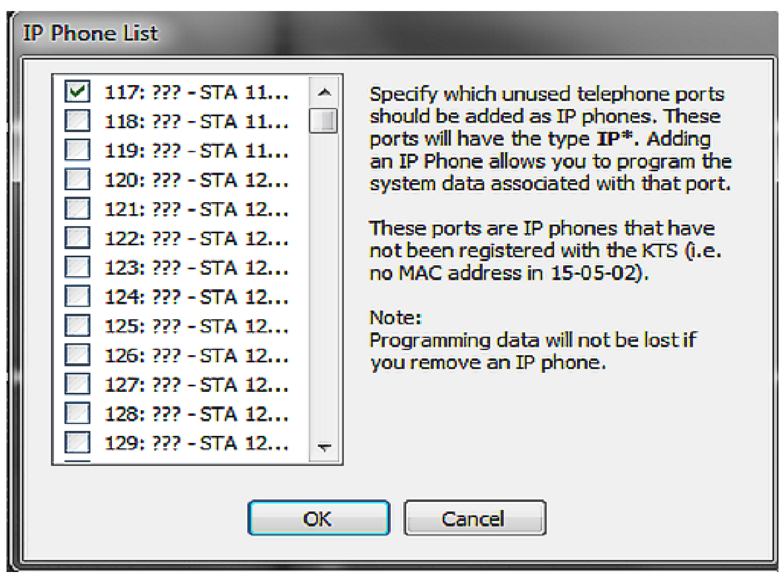


Figure 8-38 IP Phone List



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.

NOTE

- *The station does not require an Ethernet connection to enter 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:

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

     You are prompted with a User Name and Password. The defaults are:
     User Name: **ADMIN**
     Password: **6633222**

     The user name should already be entered in the terminal.
  2. *Press **Set** soft key to step down to the password field.*
  3. *After you enter the password, press **OK** soft key.*

Network Settings

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

SIP Settings

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

15.8.2 Automatic Registration

Follow these steps for 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 8-39 PCPro Programming Unregistered Phones

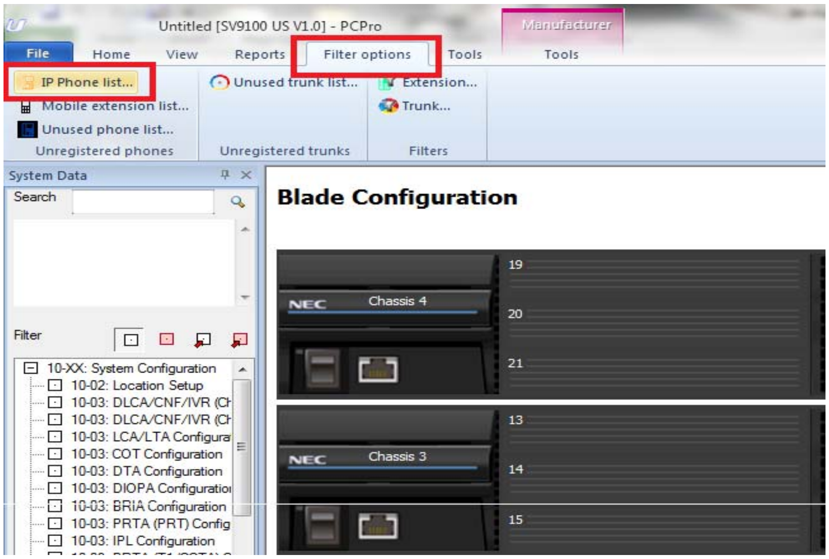
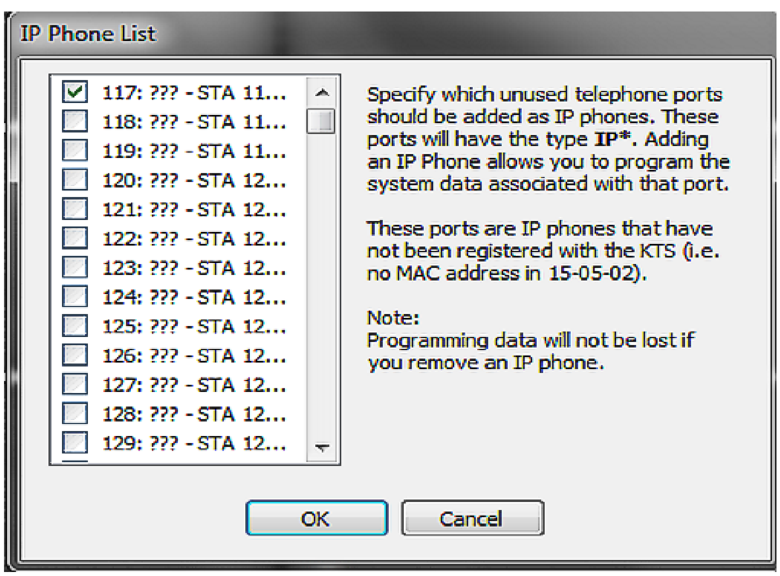


Figure 8-40 IP Phone List



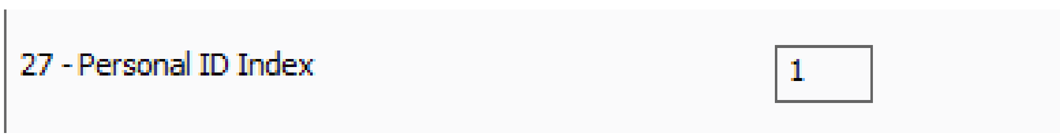
5. Program 10-46
   Change Program 10-46-01 to **Automatic**.

Figure 8-41 Automatic Registration Setting

6. Program 15-05-27
   Each IP phone requires a unique personal ID index. Valid settings are 1 ~ 960.

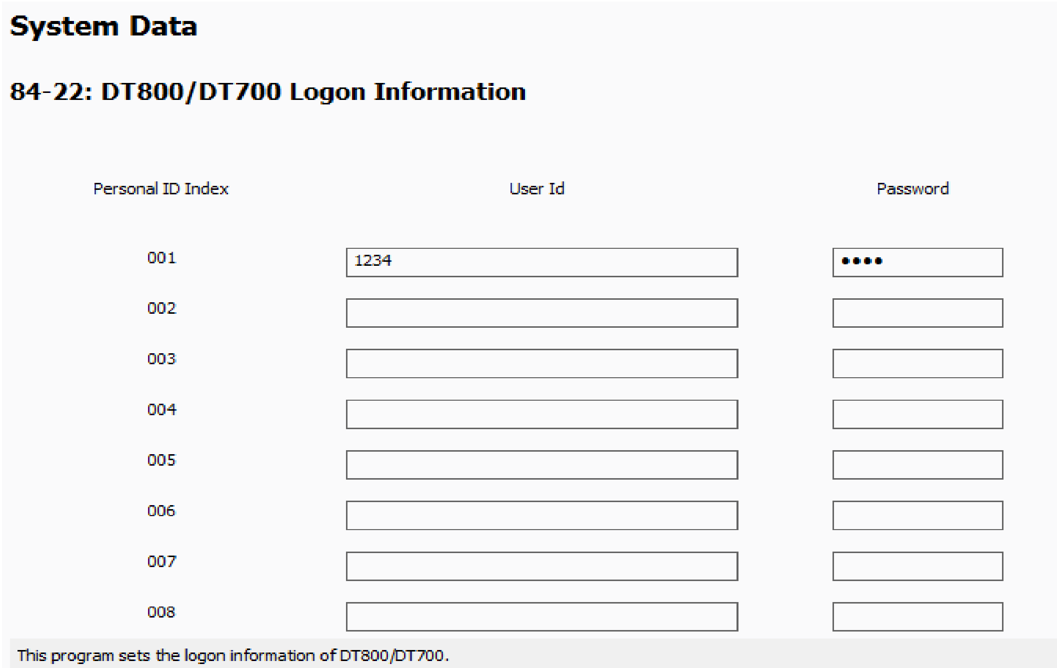
Figure 8-42 Automatic Registration Basic Setup

Figure 8-43 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 8-44 Automatic Registration User Name and Password Assignment



System Data

84-22: DT800/DT700 Logon Information

Personal ID Index	User Id	Password
001	1234	••••
002		
003		
004		
005		
006		
007		
008		

This program sets the logon information of DT800/DT700.

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.

NOTE

- ***The station does not require an Ethernet connection to enter 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:

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

     You are prompted with a User Name and Password. The defaults are:
     User Name: **ADMIN**
     Password: **6633222**

     The user name should already be entered in the terminal.
  2. *Press **Set** soft key to step down to the password field.*
  3. *After you enter the password, press the **OK** soft key.*

Network Settings

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

SIP Settings

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

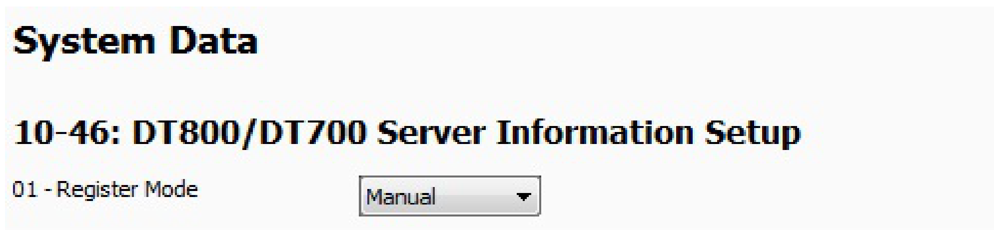
15.8.3 Manual Registration

Follow these steps for manual registration.

Steps 1~4 are the same as for Section 15.8.2 Automatic Registration on page 8-66.

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

Figure 8-45 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.

NOTE

- ***The station does not require an Ethernet connection to enter 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:

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

     You are prompted with a User Name and Password. The defaults are:
     User Name: **ADMIN**
     Password: **6633222**

     The user name should already be entered in the terminal.
  2. *Press **Set** soft key to step down to the password field.*
  3. *After you enter the password, press the **OK** soft key.*

Network Settings

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

SIP Settings

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

15.9 IP Phone Relocation

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

IP Phone Relocation Override

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

Conditions

- Multiple IP Phones cannot use the same user ID and the same password at the same time.
- When a user is using multiple IP Phones at the same time, the user ID and password must be different for each phone.
- When a user is using SoftPhone (CTI mode) and controlling the IP Phone by this SoftPhone, the user ID and password should be different for the SoftPhone and IP Phone.
- An IP Phone (IP Phone and Soft phone) with DSS console cannot override another IP Phone.
- An IP Phone (IP Phone and Soft phone) with DSS console cannot be overridden from another IP Phone.
- 2-button Phones cannot use the IP Phone Relocation feature.
- The login ID and Password are programmed in Program 15-05-27 and Program 84-22.
- IP Phone Relocation can be used only in Manual Registration Mode.
- The system sees terminal types 1 (Economy), 2 (Value), 3 (Desi-Less), 4 (Sophisticated), and 5 (Softphone) as the same terminal type.
- When using Multiple Login, the same Personal ID index can be assigned to an ITL/Softphone and a CTI (Desktop).
- 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.

- ❍ Override is not supported in a SV9100 system that had a 3rd Party CTI connection to the CPU (i.e., UC Suites 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.
- ❍ 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.

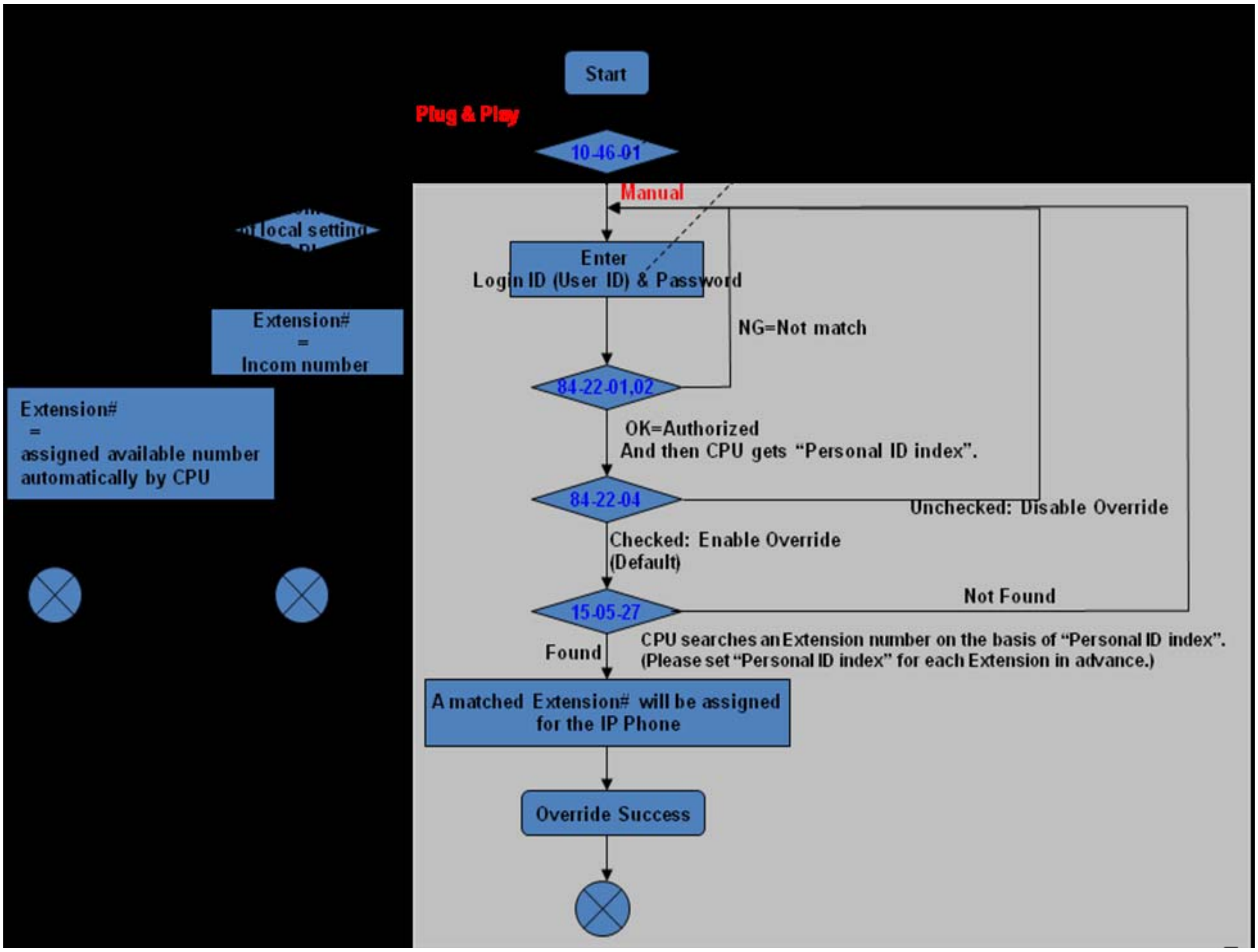
Table 8-3 IP Phone Relocation

Program/Item No.	Description/ Selection	Assigned Data	Comments
10-46-01	DT800/DT700 Server Information Setup – Register Mode	0 = Normal 1 = Auto 2 = Manual Default is 0	Set up the information of the SIP Multiline (DT800/DT700 series) Server. This PRG is a system-wide setting.
15-05-27	IP Telephone Basic Setup – Personal ID Index	0~960 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	DT800/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	DT800/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	DT800/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 8-46 IP Phone Relocation Flow Chart



IP Single Line Telephone (SIP)

Chapter 9

SECTION 1 INTRODUCTION

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

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

With the GPZ-IPLE up to 256 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 GPZ-IPLE DSP Resource. This is a required component of SIP implementation in the UNIVERGE SV9100. The UNIVERGE SV9100 GPZ-IPLE 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 GPZ-IPLE is installed and assigned. The GPZ-IPLE supports IP signaling for up to 256 (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) GPZ-IPLE can be installed per UNIVERGE SV9100 System, supporting the maximum of 896 IP stations.

The UNIVERGE SV9100 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. A/μ–Law – High bandwidth requirement usually used on Local Area Networks.
- G.722 This CODEC is useful in fixed network, Voice over IP applications, where the required bandwidth is typically not prohibitive.
- G.726 is an ITU-T ADPCM speech-coded standard covering the transmission of

voice at rates of 16-, 24-, 32-, and 40Kbps.

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

- Draft-Ietf-sipping-service-examples-09.txt.
- Draft-ietf-sipping-service-examples- (Transfer - Attended) 15.txt
- IETF RFC is defined as: Internet Engineering Task Force (RFC) Request for Comments.
- The SIP Station feature set supports the Message Waiting Indication (MWI) based on RFC3842.
- When out of band DTMF is used, via RFC2833. The IPLE supports an out of band DTMF payload of 96 ~ 127.
- If a user on a standard SIP phone is talking to another station via voice announce (receiving station has not pressed speaker or lifted the handset) and the SIP phone presses transfer or hold, the call will be terminated. A standard SIP call cannot be placed on hold or transferred until the other party answers.
- SIP INFO works independent from other DTMF methods such as RFC2833. This means SIP Terminals should send DTMF information by a single method otherwise the system will receive both separately causing double digits.
- The system has the ability to receive DTMF information in SIP INFO messages sent by Standard SIP Terminals. This allows the SIP Terminal to initiate features during a ringing state such as CAMP ON and Message Waiting. SIP Terminal must support this feature and have it enabled.
- When PRG 15-05-49 is set to 2: Allowed while RTP is not available, SIP INFO will be received while RTP is not established. In-band method such as RFC2833 will be used once voice path is established.
- When PRG 15-05-49 is set to 1: Allowed any time, SIP INFO will be received whenever they arrive.
- The SV9100 supports NAT for Standard SIP Terminals.

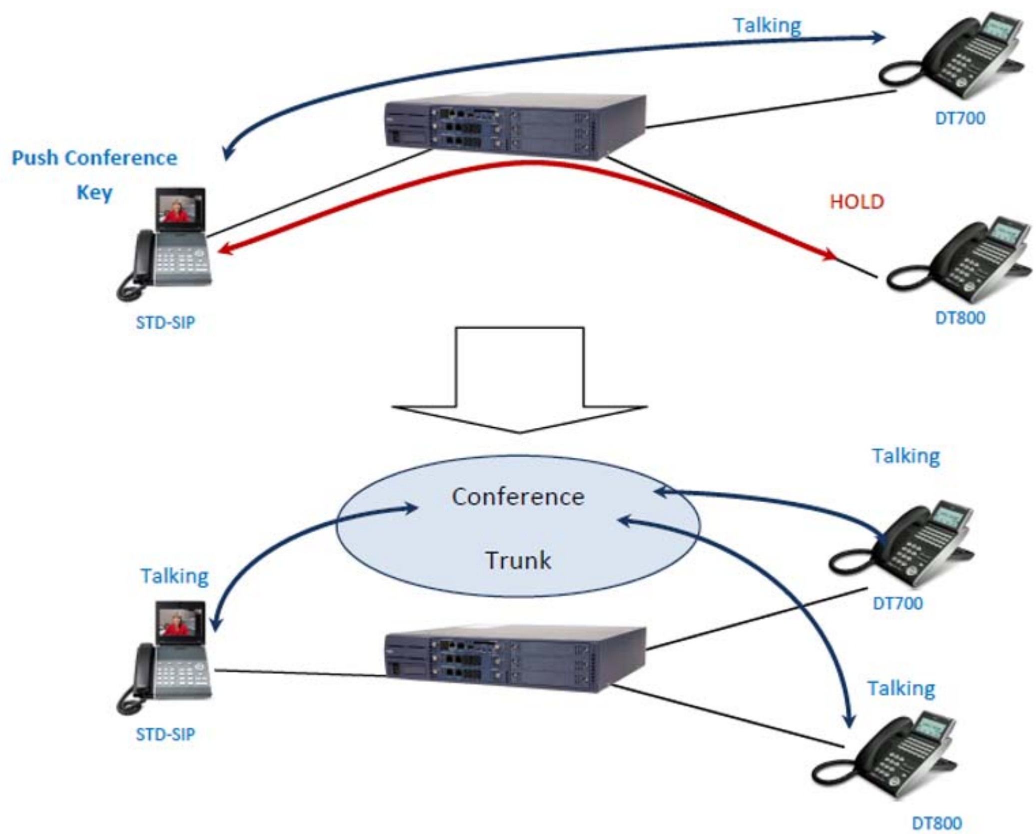
Enhancements

- With Version 2000 (2.00 or higher) Standard SIP Conference is supported.

Section 2 STD SIP Conference

Description

With SV9100 software 2.00 or higher Standard SIP Terminals can initiate a conference call.



Conditions

- ❏ PRG 20-13-08 must be enabled for the Class of Service the Standard SIP Terminal is in.
- ❏ A DSP Resource is needed for each Standard SIP or DT800/700 that is in the conference.

Member 1	Member 2	Member 3	Necessary Number of DSP
STD- SIP	DT400	DT400	1
STD- SIP	STD SIP	DT300	2
STD- SIP	DT800	STD-SIP	3
: :	: :	: :	: :

- Video calls are not supported.
- The following conference features are Not Supported with Standard SIP:
  - Meet Me Conference
  - Barge in to Conference
  - Split between the parties in conference
  - Transfer a call into a conference

Default Setting

None

System Availability

Terminals

- Standard SIP Terminal

Required Components

- GCD-CP10
- GPZ-IPLE
- System Port License (0300)
- VoIP Resource License (5103)
- IP Terminal License (5111)

Related Features

- ➥ IP Single Line Telephone (SIP)
- ➥ Conference

Programming

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
20-13-08	**Class of Service Options (Supplementary Service) Conference**	Turn off or on an extension user ability to initiate a Conference.	0 = Off 1 = On Default = 1	X		

Operation

<u>Polycom VVX 500</u>

To establish a conference:

1. Establish an intercom or trunk call.
2. Press the “Hold” softkey to hold the first call.
3. Establish a second call (intercom or trunk), which is to be added in the conference.
4. When called party answers, press the “Join” softkey to start the conference.
5. Repeat steps 2-4 to add more parties to the conference.

-OR-

1. Establish an intercom or trunk call.
2. Press the “Conference” softkey to hold the first call.
3. Establish second call (intercom or trunk), which is to be added in the conference.
4. When called party answers, press the “Conference” softkey to start the conference.
5. Repeat steps 2-4 to add more parties to the conference.

<u>NEC ITX-1DE-1W</u>

1. Establish an intercom or trunk call.
2. Press the “Hold key” to hold first call.
3. Establish a second call (intercom or trunk), which is to be added in the conference.
4. When call party answers, press the “CONF” key to start the conference.
5. Repeat steps 2-4 to add more parties to the conference.

SECTION 3 STD SIP TRANSFER–UNATTENDED

Description

Any standard SIP terminal can perform an Unattended (Blind/Unsupervised) transfer.

Conditions

- Program 10-05-50 (Peer-to-Peer Mode) must be disabled for the Unattended Transfer to be performed.
- A SIP terminal must receive the re-Invite message of Session Timer in a state of Unattended transfer.
- When the transferred destination terminal is busy, unanswered or the extension number in the Refer-To header is wrong or out of service, the call is sent back to the original terminal.
- If the standard SIP phone is placed on hold/park, from another extension, this call cannot be transferred until the station that placed the call on hold/park retrieves the call. A unattended transfer can only be completed while both parties are in a talking state.
- An Unattended Transfer can only be performed to the following locations:
    - Extension Number
    - Department Group Pilot Number
    - ACD Group Pilot Number
    - Operator Access
    - Trunk/Alternate trunk access code
    - F-Route Access
    - Network Access
- Quick transfer to Voice Mail is not supported when using Unattended Transfer.

Default Setting

None

System Availability

Terminals

- Standard SIP Terminal

NEC

Required Components

- ❑ GCD-CP10
- ❑ GPZ-IPLE

Related Features

- ➥ IP Single Line Telephone (SIP)
- ➥ Transfer

SECTION 4 NAT MODE FOR STANDARD SIP TELEPHONE

Description

The SV9100 supports NAT for Standard SIP Terminals.

Conditions

- ❑ When PRG 10-33-05 NAT mode for SIP phone is set to 1 (Enable), the P2P mode for SIP Phone becomes always Off, regardless of PRG 15-05-50 setting.
- ❑ Standard SIP Video call feature which uses P2P mode cannot be established in the same system, since the P2P mode is disabled by enabling PRG 10-33-05.
- ❑ When connecting multiple SIP Phones via NAT, PRG 15-05-18 has to be set to admit registration of multiple SIP Phones which are using the same IP address. For example, if you had a STD SIP Terminal that had two lines registering with the same IP Address, you would need to flag PRG 15-05-18 for both Extension numbers.
- ❑ In the router/firewall that the SV9100 resides behind port forwarding is required. Port forwarding at the SIP Terminal end is not required as long as PRG 15-05-45 (Plug and Play) is enabled. The ports that must be forwarded to the SV9100 are as follows:
  - ❍ UDP Port 5070 MUST be forwarded to the IP Address assigned in PRG 10-12-09.
  - ❍ UDP Ports 10020 ~10083, UDP Ports 10020~10147 and UDP Ports 10020~10275 (GPZ-IPLE) **MUST** be forwarded to the IP Address(s) assigned in PRG 84-26-01.
- ❑ When PRG 15-05-45 is set to “1” the manual table setting for port forwarding may not be required on the remote side router, but the router must support the NAT function itself. If PRG 15-05-45 is set to “0” port forwarding at the Remote side router is required. This feature requires the installation of GPZ-IPLE.

Default Setting

None

System Availability

Terminals

- Standard SIP Terminal

Required Components

- GCD-CP10
- GPZ-IPLE

Related Features

- IP Single Line Telephone (SIP)
- Transfer

Programming

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
10-12-03	**GCD-CP10 Network Setup- Default Gateway**	Assign the default gateway IP Address for the CPU.	0.0.0.0~ 126.255.255.254 128.0.0.1~ 191.255.255.254 192.0.0.1~ 223.255.255.254 (default = 0.0.0.0)	X		
10-12-07	**GCD-CP10 Network Setup - NAPT Router IP Address (Default Gateway [WAN]**	Define the IP 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)	X		
10-12-08	**GCD-CP10 Network Setup – ICMP Redirect**	When receiving ICMP redirect messages, this determines if the IP Routing Table updates automatically or not.	0= (Enable) 1= (Disable) (default = 0)		X	
10-12-09	**GCD-CP10 Network Setup – IP Address**	Assign the IP Address for the VoIPDB. If a VoIPDB is installed in the system it is recommended to set PRG 10-12-01 to 0.0.0.0 and all connections to the system will be made through the VoIPDB.	0.0.0.0~ 126.255.255.254 128.0.0.1~ 191.255.255.254 192.0.0.1~ 223.255.255.254 (default = 172.16.0.10)	X		

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
10-12-10	**GCD-CP10 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)	X		
10-12-11	**GCD-CP10 Network Setup – NIC Setup**	Define the LAN interface Speed and Mode of the VoIP Application supported.	*0 = Auto Detect* *1 = 100Mbps, Full Duplex* *2= 10Mbps, Full Duplex* *3 = 1Gbps, Full Duplex* *(default = 0)*	X		
15-05-50	**IP Phone Basic Setup - Peer to Peer Mode**	Enable or Disable Peer-to-Peer mode for SIP Phone. *When PRG 10-33-05 NAT mode is set 1 = Enable, the P2Pmode for SIP Phone is always set (Off) automatically regardless of this program setting.*	*0 = No (Disable)* *1 = Yes (Enable)* *(default = 0)*		X	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
10-33-02	**SIP Registar/Proxy Information Basic Setup - Authentication Mode**	When connecting a standard SIP terminal via NAT, this opion must be enabled to prohibit illegal SIP phone registration.	*0 = Disable* *1 = Enable* *(default = 0)*	X		
10-33-05	**SIP Registar/Proxy Information Basic Setup – NAT Mode**	When system controls remote SIP phone via NAT router, set this program to 1 = Enable.	0 = Disable 1 = Enable (default = 0)	X		
10-37-01	**UPnP Setup – UPnP Mode**	Enable/Disable UPnP.	0 = Disable 1 = Enable (default = 0)		X	
10-58-01	**Network Address**	This program sets the IP or Network address for phones that are not to be routed through the NAPT translations. For example, if a system had multiple NAPT phones and another site, with multiple IP phones connected via a VPN connection, you would not want the phones connected over the VPN to use the NAPT feature. The network address (or single IP phone addresses) of the Remote location would be entered here. This is for the IP Phones at this location to not use the NAPT feature.	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)		X	

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
10-58-02	**Network Address – Subnet Mask**	This program sets the netmask for the IP Addresses assigned in Program 10-58-01.	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)		X	
15-05-16	**IP Telephone Terminal Basic Data Setup – Authentication Password**	Assign the authentication password for SIP single line telephones.	Up to 24 characters (default not assigned)	X		

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
15-05-45	**IP Telephone Terminal Basic Data Setup – NAT Plug and Play**	This program is valid when Program 10-46-14 is On (NAT feature activated). Select sending RTP port number to remote Router, use from negotiation result (0) or received RTP packet (1). SV9100 uses this program to decide a destination port of RTP transmitting packets from IPLE to a remote IP terminal. If “0:OFF” is selected, the destination port of RTP transmitting packets will be a SIP/SDP negotiation result.(same behavior as before). If you chose “1:ON”, the destination port of RTP transmitting packet will be the same port of a source port of a receiving RTP packet on IPLE. IPLE is required.	0 = Disable 1 = Enable (default = 0)	X		
15-05-47	**IP Telephone Terminal Basic Data Setup – Registration Expire Timer for NAT**	On a per station basis, this setting defines the SIP registration expiry timer. If this value is set to 0, for a NAPT terminal, the value in Program 84-23-01 is applied.	0 = Disable 60~65535 (sec) (default = 180)		X	
15-05-48	**IP Telephone Terminal Basic Data Setup – Subscribe Expire Timer for NAT**	On a per station basis, this setting defines the SIP Subscribe expiry timer. If this value is set to 0, for a NAPT terminal, the value in Program 84-23-02 is applied.	0 = Disable 60~65535 (sec) (default = 180)		X	
84-20-01	**SIP Extension Basic Information Setup – Registrar/Proxy Port**	Define SIP station Proxy Port.	1~65535 (default = 5070)	X		
84-26-01	**IPL Basic Setup – IP Address**	Assign the IP address for each DSP on the IPLE.	xxx.xxx.xxx.xxx Defaults: Slot 1 = 172.16.0.20	X		
84-26-02	**IPL Basic Setup – RTP Port Number**	Assign the RTP port number to be used for each DSP on the IPLE. Only even numbered ports are supported.	0~65534 Defaults: RTP Port = 10020		X	
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.	0~65534 Defaults: RTCP Port = 10021		X	

SECTION 5 PROGRAMMING

5.1 Configure VoIPDB Networking Information

The VoIPDB should be connected to the same IP Subnet as the GCD-CP10. It also requires a valid IP address. Use these programs to assign VoIPDB network related information.

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

Program Number	Program Name	Description/Comments	Assigned Data	1	2	3
10-12-03	GCD-CP10 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	GCD-CP10 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	GCD-CP10 Network Setup –Subnet Mask	Assign the Subnet Mask of the VoIPDB.	255.255.0.0 Default is 255.255.0.0	X		
10-68-01	IP Trunk Availability – Trunk Type	Assign the H.323 Trunk Availability.	0 = None (Default) 1 = SIP 2 = H.323 3 = CCIS	X		
10-68-02	IP Trunk Availability – Start Port	Assign the start port number of H.323 Trunks.	Range: 0~400 Default is 0	X		
10-68-03	IP Trunk Availability – Number of Ports	Assign the Number of H.323 Trunks	Range: 0~400 Default is 0	X		

5.1.1 VoIPDB (DSP) Basic Setup

Use these programs to make basic VoIPDB assignments.

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
84-26-01	IPL Basic Setup – IP Address		Default: Slot 1 = 172.16.0.20	X		
84-26-02	IPL Basic Setup – RTP Port Number	Assign the RTP Port Number.	0~65534 Default for 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 is 10021		X	

5.1.2 VoIP ToS Setup

Use this programs to define ToS.

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	

5.1.3 SIP Peer-to-Peer

Use these programs to make SIP peer-to-peer assignments.

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
15-05-50	IP Phone Basic Setup - Peer to Peer Mode	Disabling 15-05-50 results in SIP MLT Station-to-SIP MLT Station calls using a DSP resource.	0 = Off 1 = On Default is 1		X	

5.1.4 IP Extension Numbering

Use these programs to assign IP extension numbering.

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
11-01-01	System Numbering	Refer to the SV9100 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 1~300 = 200~499 Ports 301~960 = 5000~5659	X		

5.1.5 SIP Extension CODEC Information

Use these programs assign 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 0			
84-19-04	SIP Extension CODEC Information Basic Setup – G.711 Jitter Buffer (min)	Define G.711 Jitter Buffer minimum accepted value.	0~300ms Default is 20			
84-19-05	SIP Extension CODEC Information Basic Setup – G.711 Jitter Buffer (Average)	Define G.711 Jitter Buffer setting.	0~300ms Default is 40			

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
84-19-06	SIP Extension CODEC Information Basic Setup – G.711 Jitter Buffer (max)	Define G.711 Jitter Buffer maximum accepted value.	0~300ms Default is 80			
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.	0~300ms Default is 20			
84-19-10	SIP Extension CODEC Information Basic Setup – G.729 Jitter Buffer (average)	Define G.729 Jitter Buffer setting.	0~300ms Default is 40			
84-19-11	SIP Extension CODEC Information Basic Setup – G.729 Jitter Buffer (max)	Define G.729 Jitter Buffer maximum accepted value.	0~300ms Default is 80			
84-19-17	SIP Extension CODEC Information Basic Setup – Jitter Buffer Mode	Define the Jitter Buffer mode – supported Static or Immediate.	1 = Static 2 = Not Used 3 = Self Adjusting Default is 3			
84-19-18	SIP Extension CODEC Information Basic Setup –VAD Threshold	Define the VAD Threshold.	0~30 Default is 20			
84-19-28	SIP Extension CODEC Information Basic Setup – Audio Capability Priority	Define Audio capability priority.	0 = G.711_PT 2 = G.729_PT 3 = G.722 4 = G.726 5 = Not Used Default is 0			

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
84-19-33	SIP Extension CODEC Information Basic Setup – Number of G.722 Audio Frames		1~4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms Default is 3			
84-19-35	SIP Extension CODEC Information Basic Setup – G.722 Jitter Buffer (min)		0~300ms Default is 30			
84-19-36	SIP Extension CODEC Information Basic Setup – G.722 Jitter Buffer (Average)		0~300ms Default is 60			
84-19-37	SIP Extension CODEC Information Basic Setup – G.722 Jitter Buffer (max)		0~300ms 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			
84-19-40	SIP Extension CODEC Information Basic Setup – G.726 Jitter Buffer (min)		0~300ms Default is 30			
84-19-41	SIP Extension CODEC Information Basic Setup – G.726 Jitter Buffer (Average)		0~300ms Default is 60			
84-19-42	SIP Extension CODEC Information Basic Setup – G.726 Jitter Buffer (max)		0~300ms 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---					

5.1.6 SIP Extension Basic Information Setup

Use these programs to assign basic SIP extension information.

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			
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			

5.1.7 IP Phone Configuration

Use these programs assign IP phone configuration information.

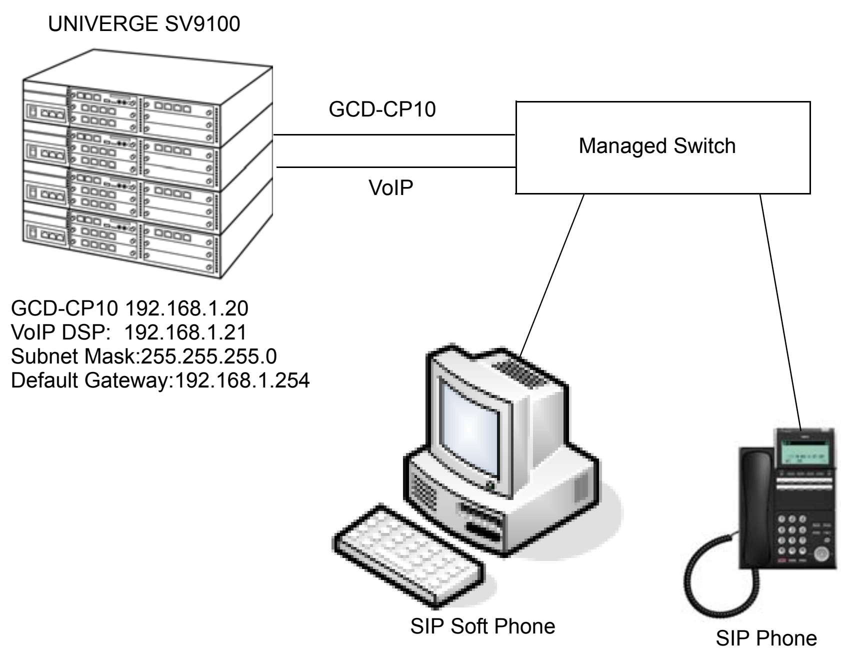
Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
15-05-01	IP Telephone Terminal Basic Data Setup – Terminal Type	Review the type protocol support by the IP Phone. Viewing Only – No changes permitted.	1 = H.323 2 = SIP 3 = None 4 = DT800/DT700 Default = 3			

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
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			
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 = Disable 1 = Enable Default is 0			
15-05-43	Video Mode	This Program is used to select the video function with the standard SIP terminal. If the standard SIP terminal supports the video function, the SV9100 transfers the video CODEC in SDP information.	0 = Disable 1 = Enable Default is 0			

Program Number	Program Name	Description/ Comments	Assigned Data	1	2	3
15-05-49	IP Telephone Terminal Basic Setup - Receiving SIP INFO	Determines whether SIP INFO messages are received by the system.	0 = Disable 1 = Allowed any time 2 = Allowed while RTP is not available Default = 1			

5.2 SIP Phone Example

The following example show a SIP phone attached to an SV9100 system.



SV9100 NetLink

Chapter 10

SECTION 1 INTRODUCTION

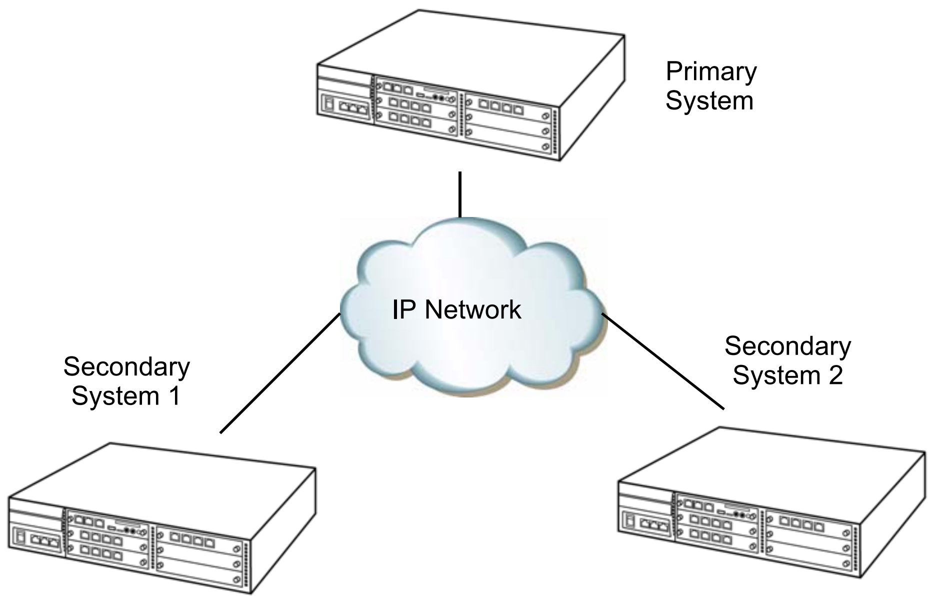
NetLink Networking provides a seamless connection, using an IP network, to join multiple SV9100 communication servers into what would appear to be a single communications server. With a unified numbering plan, users can access any extension in the network as if they were in the same location.

With NetLink Networking, multiple locations are linked, allowing only one operator and a shared voice mail in the network. Up to 50 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 SV9100 programming, and it takes over the Primary System database allowing the other linked nodes to continue functioning.

Figure 10-1 NetLink Fail-Over Example



SECTION 2 SV9100 REQUIREMENTS

2.1 Hardware

Each SV9100 Communications Server needs the following items:

- GCD-CP10
- GPZ-IPLE

Install the station/trunk blades as required for your configuration.

- When installing a Secondary System in a NetLink network, and the Secondary System has GCD-LCA blades, the GCD-LCA blades will come online and get assigned ports **before** any GCD-DLCA blades.

Invalid data is displayed in the LCD of the terminal if Program 51-01 is enabled.

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 in the network.

2.2 Capacity

- A maximum of 50 nodes (connected sites).
- 960 Extensions/400 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.

2.3 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.

2.4 Protocol

TCP/IP IPv4 (IPv6 available in a future release)

2.5 Determining System ID

NetLink allows connecting a maximum of 50 communication servers. To identify the communication server, each communication server must have one unique number between 1 and 50.

2.6 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.

2.7 Software Information

All communication servers in the NetLink network should be at the same SV9100 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
- Type of VoIP installed: IPLE

SECTION 3 INSTALLATION

3.1 NetLink Connection

When assigning the IP addresses to the IPLE 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 IPLE 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 IPLE NIC.

For example, PC Pro, Web Pro, and ACD terminate to the IPLE NIC card, when installed. Both the IPLE 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, IPLE 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 SV9100 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.

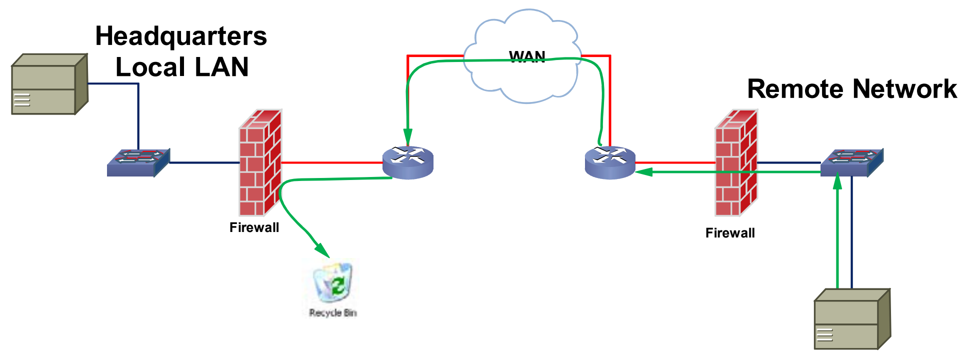
Firewalls

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 SV9100 Systems Connected Via the WAN shows two SV9100 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 SV9100 Systems Connected Via the WAN



The green arrow in Figure 10-2 Two SV9100 Systems Connected Via the WAN represents the data packets leaving the REMOTE IPLE card destined for the SV91000 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 SV9100.

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 IPLE ports are installed.

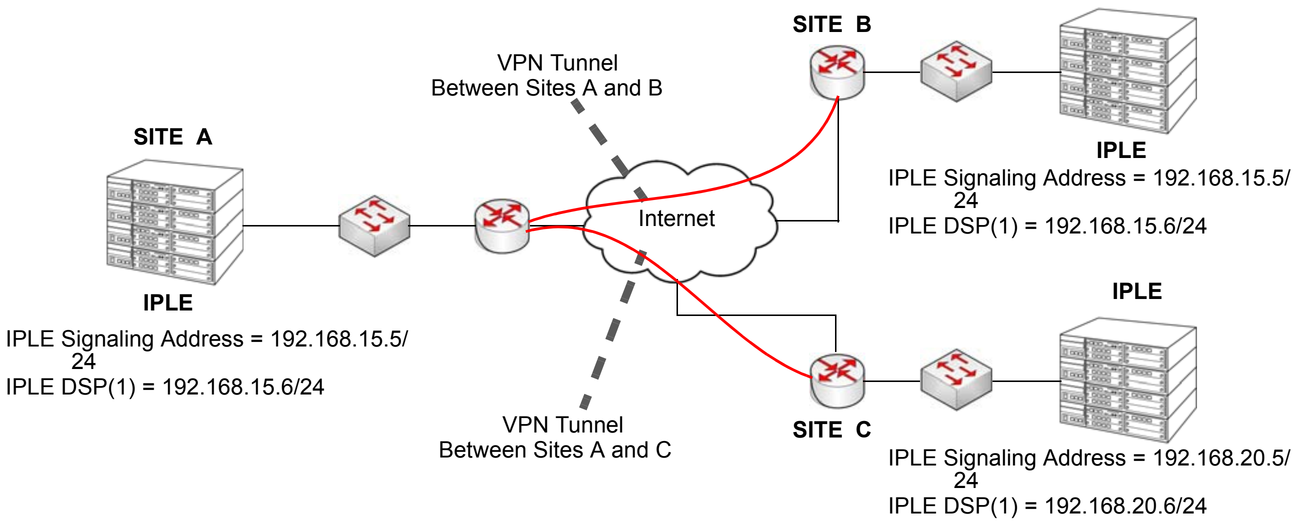
- IPLE 256 Open UDP Ports **10020~10531**

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.

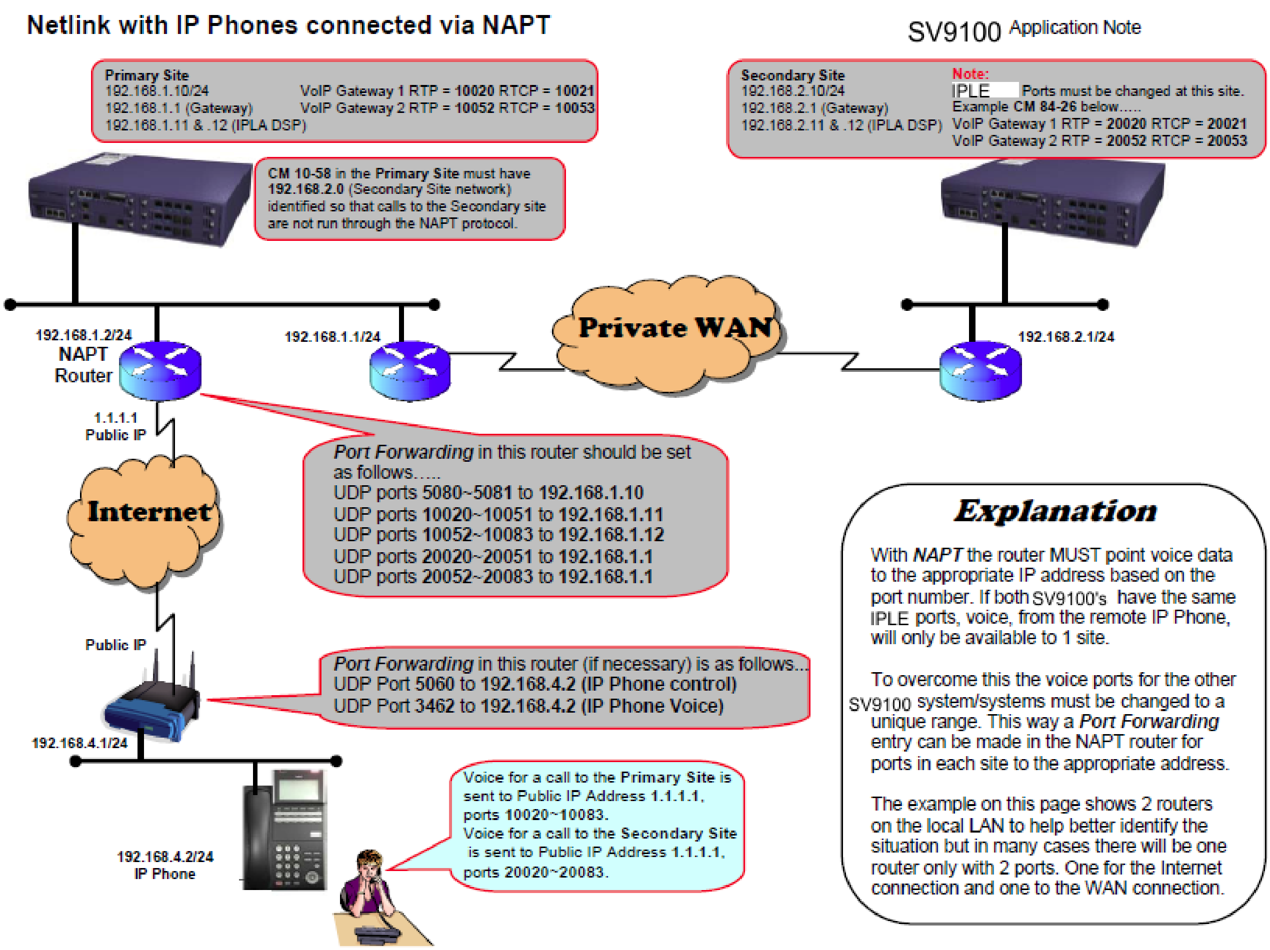
Figure 10-3 VPN Network



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 Limitations

3.2.1 Features

Synchronous Ringing via NetLink is not supported.

3.2.2 IP Terminal

All DT800/DT700 terminals and softphones must register to the primary system.

All DT800/DT700 terminals can register to either the primary or a secondary system. Softphones and wireless LAN phones must still register to the primary system.

The SV9100 Encryption feature is only supported in the primary system.

3.2.3 IP Trunks

When configuring SIP Trunks on Netlink Secondary system, only SIP Profile One is supported.

3.2.4 General limitations

- SV8100 CPU is not supported in a Netlink environment with SV9100 CPU.
- The Primary System (Main Site) requires the appropriate NetLink licenses dependant upon the number of nodes in the NetLink network.
- Up to 50 Nodes can be supported in a NetLink network.
- When the attendant's telephone exists on a secondary system, alarm information cannot be displayed on the attendant's telephone.
- When Fail Over occurs between the Primary System and two or more Secondary Systems, the attendant telephone displays the System ID of the system that went into Fail-Over last.
- 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 GCD-CP10 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 GCD-CP10 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.
- Duplicate license information in the Secondary System is available for only 28 days.

- After seven days or 28 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 the specified duration based on software version above.)
- 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 External MOH is assigned, the AUX1 or AUX2 on the front of the GCD-CP10 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].
- A DT800/DT700 may be registered to either the primary or a secondary NetLink system. When NetLink is turned on, the system refers to programs 51-17-01 and 51-17-02 for DT800/DT700 registrar information.
- With both the Primary Site and Secondary Sites can have their own local MOH source connected to the AUX1 or AUX2 on the front of the GCD-CP10.

- With the T-1 CCTA blade is supported in the primary system and/or secondary systems.
- When using Mobile Extension in a NetLink Network, the ISDN/PRI must be used in the Primary System.
- InMail is supported in a NetLink network; however, replication should be scheduled for non-peak hours of operation.
- InMail is supported for centralized voice mail in a KTS to KTS CCIS network.
- It is possible to set Register ID for trunks that belong to that specific system. For example, a Register ID set in the Primary system cannot be assigned to a trunk in the Secondary system. The allocation of the trunk and Register ID of Program14-12 must be in the same system.
- Assigned SIP trunk ports will be cleared after establishing NetLink.
- The following programs no longer replicate and can be set on a per system basis: PRG10-23, 10-37, 21-19, 84-13, and 84-14.
- When using multiple SIP trunk carriers, programs 10-28, 10-29, 10-30, and 10-23 will have to be set at each system via WebPro or PCPro (does not replicate). Program 10-68 can be accessed via system ID.
- When using multiple SIP trunk carriers, Program 51-19 must be enabled for extensions on a secondary system to use CPN information from program 21-19.
- When using multiple SIP trunk carriers, SIP trunk CODEC setup must be done from each system (does not replicate).
- When using multiple SIP trunk carriers, SIP trunk basic setup must be done from each system (does not replicate).
- Registered SIP trunks can be utilized by any system in the NetLink network, as long as trunk route programming allows it.
- When using InMail in a CCIS or Netlink network, 8-digit extensions and mailboxes are not supported.
- 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.
- 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 (DT800/ 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 (DT800/ DT700)	P:1	0	P:1	P:1	S1:1	S1:1	S2:1	S2:1
Secondary System 1	TDM Terminal	P:1 S1:1	S1:1	P:1 S1:1	P:1 S1:1	0	0	P1:1 S2:1	S1:1 S2:1
Secondary System 2	TDM Terminal	P:1 S2:1	S2:1	P:1 S2:1	P:1 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.
- The SV9100 can recognize what system that each DT800/DT700 extension(s) are connected to and provide an ARS COS based on the System (System ID) when using NetLink.
- When NetLink is enabled synchronous ringing (PRG 14-02-17) automatically is disabled. Synchronous ringing is not supported in a NetLink environment.

Restrictions:

- 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 GCD-LCA blades installed, the GCD-LCA blades will come online and assign ports before any GCD-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. SV9100 NetLink connections are only supported within the same countries/areas. For example, the SV9100 can be connected via NetLink between the US and Canada, however the SV9100 **cannot** be connected via NetLink with systems 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

3.3 DSP

(PB Receiver, Dial Tone Detector, Caller ID Detector, Caller ID Sender, MF Receiver).

- ❍ A DSP is used on the system GCD-CP10 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).

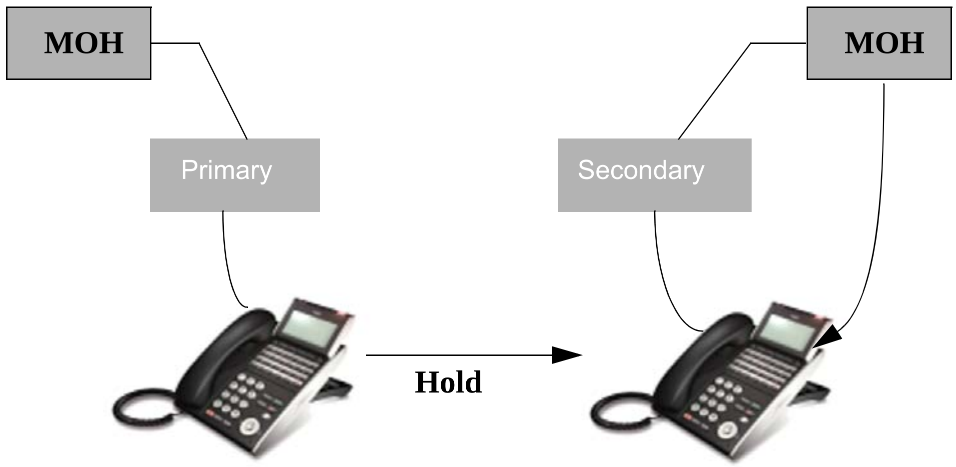
- ❍ All communication servers follow the DSP Setting in Program 10-09 in the Primary System.

3.4 E911

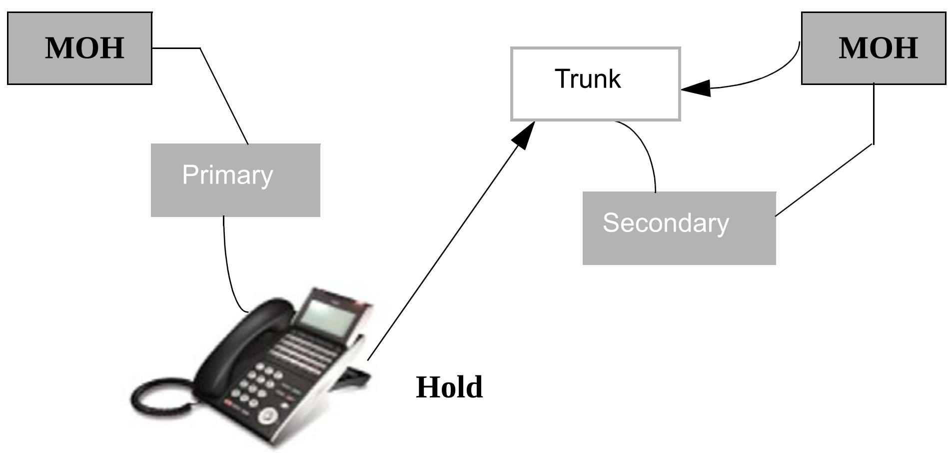
A trunk is required for each communication server for 911 calls and the ARS routing table must be set up to seize the proper trunk. If no DSP resource is available, one of the active DSPs is dropped and the communication server uses that DSP for the 911 call.

3.5 Music on Hold

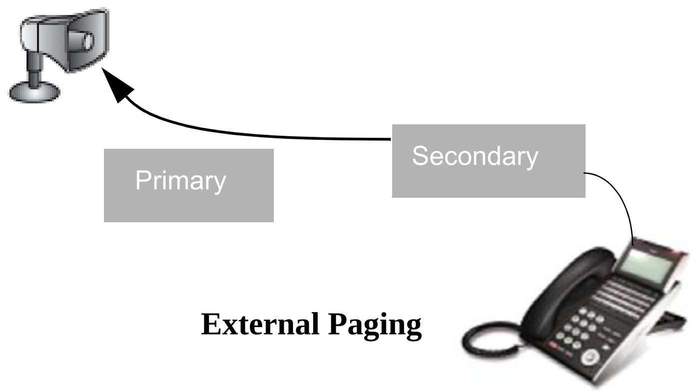
- ❍ When External MOH is assigned, the AUX1 or AUX2 on the front of the GCD-CP10 can only be utilized at the Primary Site. All Secondary sites must provide the External MOH input via an ACI Input [PGD(2)-U10 ADP].
- ❍ Both the Primary Site and Secondary Sites can have their own MOH source connected to the CN8 or CN9 on the front of the GCD-CP10. 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 terminal is placed on hold, MOH comes from the communications server in which the terminal resides.



- *External Paging uses an output on the GCD-CP10 of the Primary System.*
- *A PGD(2)-U10 ADP must be used if External Paging is required in the Secondary Systems.*

3.6 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.

3.7 Conference Block

- The number of conference blocks in NetLink is the same as with a stand-alone communication server.

3.8 Voice Mail

- The InMail must be installed in the Primary System.
- Only one Voice Mail or InMail can be installed in NetLink network.
- Voice Mail can be installed in the Primary or Secondary System.

- For InMail remote CCIS extensions are not supported in a centralized directory.

3.9 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 (Continued)

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 GCD-CP10, then all Secondary Systems also must get their music off of the GCD-CP10.
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	
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	

Table 10-3 Supported Features in Secondary System (Continued)

Feature Name	Supported in Secondary System	Comments
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	
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	
Facsimile CO Branch Connection	Yes	
Flash	Yes	

Table 10-3 Supported Features in Secondary System (Continued)

Feature Name	Supported in Secondary System	Comments
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	
IP Multiline Station (SIP)		
IP Single Line Telephone (SIP)	Yes	All IP Terminals connect to the Primary System.
IP Trunk - (SIP) Session Initiation Protocol	Conditions	Only SIP Profile One is supported in Secondary systems.
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.
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	

Table 10-3 Supported Features in Secondary System (Continued)

Feature Name	Supported in Secondary System	Comments
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.
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	

Table 10-3 Supported Features in Secondary System (Continued)

Feature Name	Supported in Secondary System	Comments
Programming from a Multiline Terminal	Yes	
Pulse to Tone Conversion	Yes	
Redial Function	Yes	
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	
Softkeys	Yes	
Speed Dial – System/Group/Station	Yes	
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.
SV9100 Communications Analyst Enterprise	Conditions	The SV9100 Communications Analyst Server must connect to the Primary System.

Table 10-3 Supported Features in Secondary System (Continued)

Feature Name	Supported in Secondary System	Comments
SV9100 UC Suite Applications	Conditions	The SV9100 UC Suite must connect to the Primary System.
SV9100 Internal Router	Yes	
SV9100 PoE Gigabit Switch	Yes	
SV9100/SV9300 Terminals	Yes	
Synchronous Ringing	No	
Tandem Ringing	Yes	
Tandem Trunking (Unsupervised Conference)	Yes	
TAPI Compatibility	Yes	
Tone Override	Yes	
Traffic Reports	No	
Transfer	Yes	
Trunk Group Routing	Yes	
Trunk Groups	Yes	
Trunk Queuing/Camp-On	Yes	
UCB (Unified Communications for Business)	No	UCB is not supported with the SV9100.
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	
InMail	Yes	
Voice Mail Integration (Analog)		
Voice Mail Message Indication on Line Keys	Yes	
Voice Over	Yes	
Voice Response System (VRS)	Yes	

Table 10-3 Supported Features in Secondary System (Continued)

Feature Name	Supported in Secondary System	Comments
Volume Controls	Yes	
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 GCD-CP10 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 : GCD-CP10 Network Setup – VoIP Daughter Board IP Address**

  Set the LAN IP address VoIP daughter board. The IP number increases by one for each increase in the slot number.

- **10-12-10 : GCD-CP10 Network Setup – VoIP Daughter Board Subnet Mask**

  Define the subnet mask for the channel VoIP daughter board. 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 SV9100 in the NetLink network. Each SV9100 must have a unique number in the network.

  When this option is set to 0, NetLink is disabled.

  ➨ *The SV9100 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 SV9100 from in this list. When no Primary SV9100 is seen or Fail-Over occurs, the Node List is referred 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 SV9100). Once the communications server connects to the Primary SV9100, this setting is updated by the Primary SV9100 when Program 51-13-01 is enabled. This allows any new or changed SV9100s 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 SV9100. This setting is needed to use the Primary SV9100 Automatic Integration Feature (Program 51-06-01). If the secondary flag is set in Program 51-01-03 secondary SV9100s connect with this IP address.

- **51-05-01 : Timer Settings for NetLink – Keep Alive Interval**

  Set the interval (1~3600) for the secondary SV9100 to send the Keep Alive packet to the Primary SV9100 to confirm communication.

- **51-05-02 : Timer Settings for NetLink – Keep Alive Response Waiting Time**

  The secondary SV9100 waits this time (0, 5~10800) for response to the Keep Alive packet to confirm communication with Primary SV9100. If no response is received, Fail-Over occurs. A setting of 0 disables this and the SV9100 waits indefinitely for a response from the Primary SV9100. 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 SV9100 waits when searching for a new Primary SV9100 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 SV9100 is reviewing priority levels for a new higher-priority Primary SV9100.

- **51-05-05 : Timer Settings for NetLink – Top Priority Primary Detection Packet Sending Interval**

  When the current Primary SV9100 is not the Top Priority Primary SV9100, the SV9100 sends packets at this interval (1-3600) to check if a higher priority SV9100 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 SV9100 searches for the new Primary SV9100 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 SV9100 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 SV9100s may be established. When the connection is recovered, with this option enabled (1), the NetLink feature is automatically reestablished around the top priority Primary SV9100.

❍ **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 SV9100s are reset. This option determines if the secondary SV9100 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 SV9100 Time and Date through SV9100 programming. Extension users can also dial Service Code 728 to change the Time if allowed by an extension Class of Service.

❍ **51-01-02 : NetLink System Settings – Primary Candidate Order**

When the Primary SV9100 is turned off or disconnected from the network, this value (1~50) is used to select the new Primary SV9100 with the Fail-Over feature. The smaller the number, the higher the priority for the SV9100. When the value for two SV9100s is the same, lower NetLink System ID (51-01-01) is selected as the Primary SV9100.

❍ **51-01-03 : NetLink System Settings – Secondary System**

The link between SV9100s is established based on this setting (0, 1). When enabled, the SV9100 connects with the top priority Primary SV9100 (address set in Program 51-04-01). If the SV9100 is not found within the defined time (Program 51-05-02), the Fail-Over feature of the SV9100 searches for the Primary SV9100 as if this option were set to 0.

The link between SV9100s is dynamically established based on the node list set in Program 51-03-01. The Primary SV9100 is selected in the order in which the SV9100s 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 SV9100 is connected to the Primary SV9100, 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 SV9100. 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 SV9100. 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 SV9100 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 SV9100 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 SV9100 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 SV9100, the communications server using the IP address defined here as the new Primary.

  ➨ *After a Forced Change of Primary SV9100 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 SV9100 by entering the system ID. If this ID is set to 0, the Top Priority SV9100 is selected as the new Primary.

- 51-09-01 : NetLink TCP Port Settings – Primary Port

  Define the port the Primary SV9100 uses to communicate with the Secondary SV9100 (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 SV9100 to communicates to the Primary SV9100 (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 SV9100 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 SV9100. 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 SV9100 can replicate the Primary SV9100 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 SV9100 may synchronize the Secondary SV9100 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 SV9100.) This option is not user-definable and is view-only.

- 51-11-01 - 51-11-06 : NetLink System Information

  View the SV9100 information for communication servers connecting with the Primary SV9100. 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 SV9100 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 SV9100 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 SV9100 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 SV9100 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 SV9100 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).

- 51-17-01 : Registrar Port

  This is the SIP registrar port of each system when NetLink is used. Open range is 0-65535, default is 5080.

- 51-17-02 : Subscribe Session Port

  This is the SIP subscribe session port number for each system when NetLink is used. Open range is 0-65535, default is 5081.

- 51-18-01 : NetLink Fail-Over Limit

  When tear-down of network is repeated more than the specified times, NetLink is operated in stand-alone. Input range is 0, 2~10 (0 = No limit). Default is 0.

- 51-19 : Net-Link IP Trunk(SIP) Calling Party Number Setup for Extension

  This program assigns transmission of Calling Party Number from PRG21-19 for each secondary system. The transmission applies for every extension.

SECTION 6 FAIL-OVER

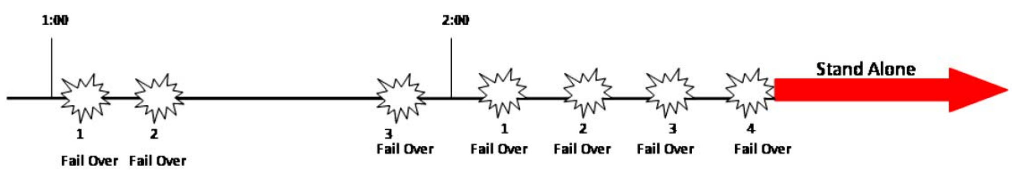
With NetLink, a connection to the Primary System may be lost (disconnected or powered down) from the TCP/IP connection. When this occurs, there could be a loss of control at all Secondary Systems.

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.

Netlink has been enhanced with PRG 51-18-01 (Netlink Fail Over Limit). This setting allows the system to be programmed for the maximum number of specified times that Fail Over occurs before the Secondary Systems remain in Stand Alone mode (Own Primary). Settings are 0: Infinity and 2~10. The following is an example when PRG 51-18-01 has a setting of “4”.

Figure 10-5 PRG 51-18-01 with setting of “4”



Once the specified number of Fail-Over times have been reached, the Secondary Systems will remain in stand alone mode until the Forced Change Primary System Enabling (PRG-51-07-01) and IP Address of New Primary System (PRG-51-08-01) are performed.

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 SV9100 in the NetLink network. Each SV9100 must be a unique number in the network. When this option is set to 0, NetLink is disabled.

➨ *The SV9100 must be reset when changing this option.*

❍ 51-01-02 : NetLink System Settings – Primary Candidate Order

When the Primary SV9100 is turned off or disconnected from the network, this value (1~50) is used to select the new Primary SV9100 with the Fail-Over feature. The smaller the number, the higher the priority for the SV9100. When the value for two SV9100s is the same, lower NetLink System ID (51-01-01) is selected as the Primary SV9100.

❍ 51-03-01 : NetLink Internet Protocol Address List Setting – Internet Protocol Address List

The communications server sets the Primary SV9100 using this list. When no Primary SV9100 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 SV9100). After the communications server connects to the Primary SV9100, this setting is updated by the Primary SV9100 when Program 51-13-01 is enabled. This allows any new or changed SV9100s 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 SV9100s is established based on this setting (0, 1). When enabled, the SV9100 connects with the top priority Primary SV9100 (address set in Program 51-04-01). If the SV9100 is not found during the defined time (Program 51-05-02), the Fail-Over feature of the SV9100 searches for the Primary SV9100 as if this option were set to 0.

The link between SV9100s is dynamically established based on the node list set in Program 51-03-01. The Primary SV9100 is selected in the order in which the SV9100s 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 SV9100. This setting is needed to use the Primary SV9100 Automatic Integration Feature (Program 51-06-01). If the secondary flag is set in Program 51-01-03 secondary SV9100s connect with this IP address.

- 10-12-09 : GCD-CP10 Network Setup – VoIP Daughter Board IP Address

  Set the LAN IP address for the VoIP daughter board. 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 SV9100 uses to communicate with the Secondary SV9100 (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 SV9100 to communicates to the Primary SV9100 (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 SV9100 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.

- 51-18-01 : NetLink Fail-Over Limit

  When tear-down of network is repeated more than the specified times, NetLink is operated in stand-alone. Input range is 0, 2~10 (0 = Infinity). Default is 0.

Conditions

- 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.

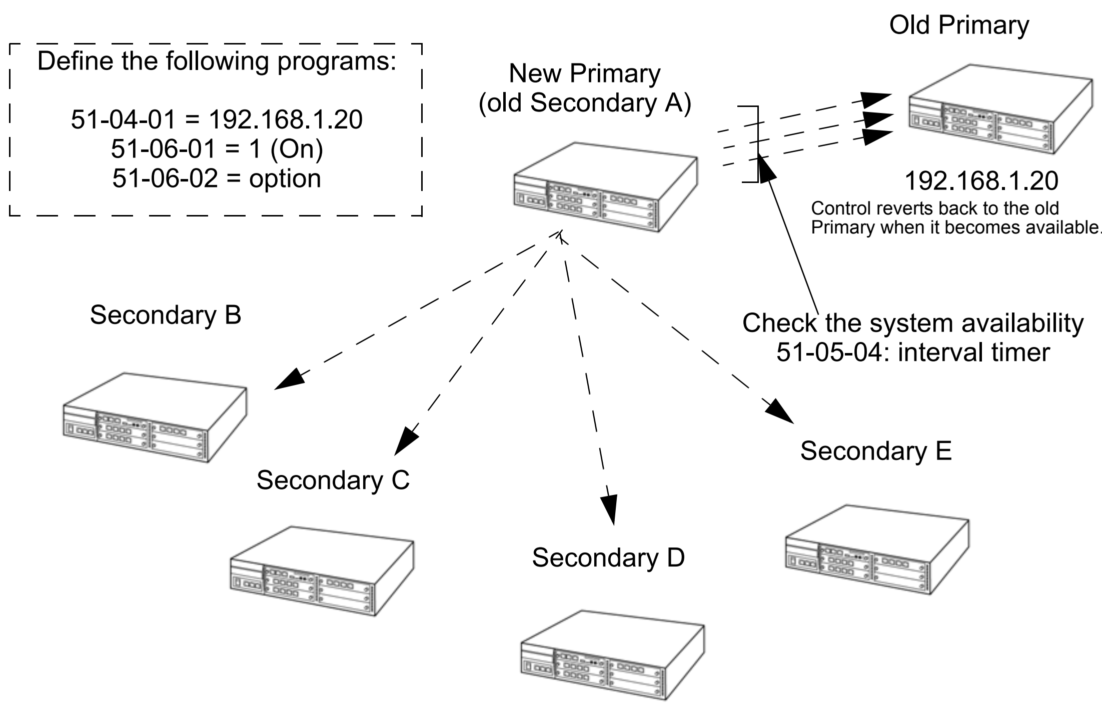
- When Fail Over occurs between the Primary System and two or more Secondary Systems, the attendant telephone displays the System ID of the system that went into Fail-Over last.

- ❍ When the attendant's telephone exists on a secondary system, alarm information cannot be displayed on the attendant's telephone.
- ❍ 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.
- ❍ Only the voice mail connected to the Primary System can be used. During Fail-Over if a InMail Flash is installed in the new Primary it will now function. This is done automatically, and no programming is required. During the Fail-Over the messages do not synchronize to the new InMail Flash. After Recovery the Original InMail Flash will now function, and any messages stored on the other Inmail flash will not synchronize.

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**.



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 SV9100. This setting is needed to use the Primary SV9100 Automatic Integration Feature (Program 51-06-01). If the secondary flag is set in Program 51-01-03 secondary SV9100s connect with this IP address.

- 51-06-01 : NetLink Primary System Automatic Integration Setting – Primary Automatic Integration

  When Fail-Over occurs, multiple Primary SV9100s may be established. When the connection is recovered, with this option enabled (1), the NetLink feature automatically reestablishes around the top priority Primary SV9100.

- 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 SV9100s are reset. This option determines if the secondary SV9100 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 SV9100. 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 SV9100 searches for the new Primary SV9100 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 SV9100s are reset. This option determines if the secondary SV9100 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 SV9100 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 SV9100 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 SV9100, the communications server uses the IP address defined here as the new Primary.

  ➨ *After a Forced Change of Primary SV9100 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 SV9100 by entering the system ID. If this ID is set to 0, the Top Priority SV9100 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.

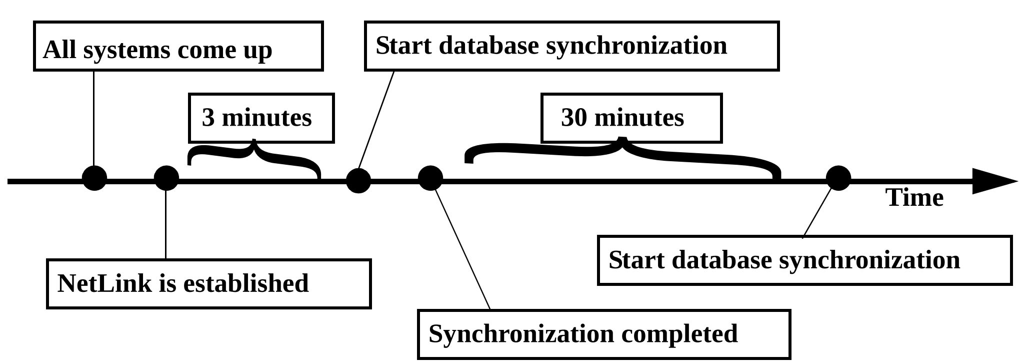
- ○ SMDR
- ○ ACD-MIS
- ○ SIP Terminal
- ○ Soft Phone
- ○ K-CCIS-IP

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 lose 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 SV9100 can replicate the Primary SV9100 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 SV9100 may synchronize the Secondary SV9100 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 SV9100 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.

IMPORTANT — ***Communication server reset is required.***

Related Programming:

- **90-57-01 : Backup Recovery Data**

  Back up the SV9100 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 SV9100 data. After executing this option, the SV9100 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.

The VoIP Channel for each Secondary must be licensed in PRG 10-54 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 28 days. After the specified duration, 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.
- After 28 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 the specified duration based on the software version listed above)
- 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 SV9100 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 SV9100 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 GCD-CP10), 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 (GCD-CP10)
- 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 SV9100.

(See the example)

Primary System

Slot	GCD-CP10	
Slot	GCD-DLCA	Ext port: 1 16
Slot	GCD-4COT	Trunk port: 1 4
Slot	GCD-4PRTA	Trunk port: 5 12

Secondary System 1 Connected First

Slot	GCD-CP10	
Slot	GCD-16DLCA	Ext port: 17 32
Slot	GCD-8LCA	Ext port: 33 40
Slot	GCD-4PRTA	Trunk port: 13 20
Slot 5		

Secondary System 2 Connected Second

Slot	GCD-CP10	
Slot	GCD-16DLCA	Ext port: 41 56
Slot	GCD-2BRIA	Trunk port: 21 24

As described above, port assignment is performed in the order which the communication server connects. With this condition, if a GCD-4COT 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.

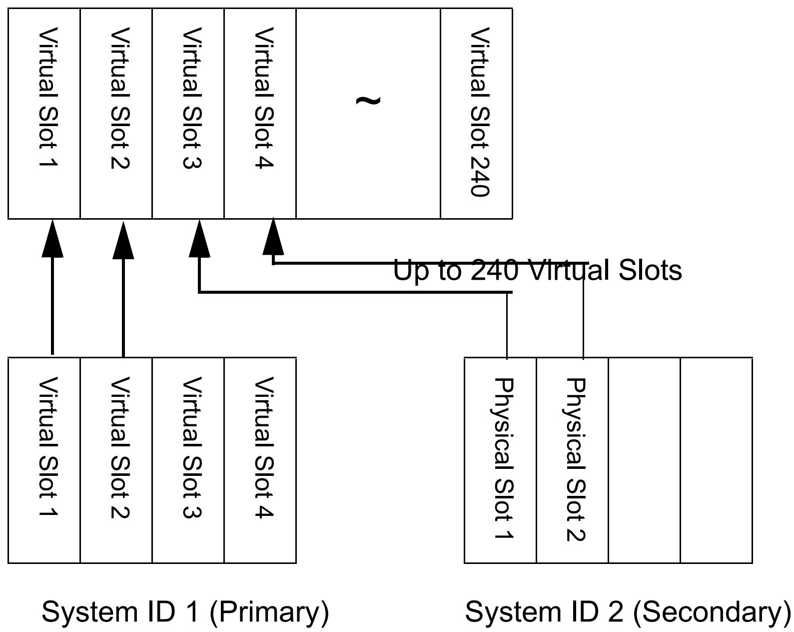
SECTION 14 SLOT CONTROL

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 SV9100.) *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.

IMPORTANT

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 SV9100 Time and Date through SV9100 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 SV9100. 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 SV9100. 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 256 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.

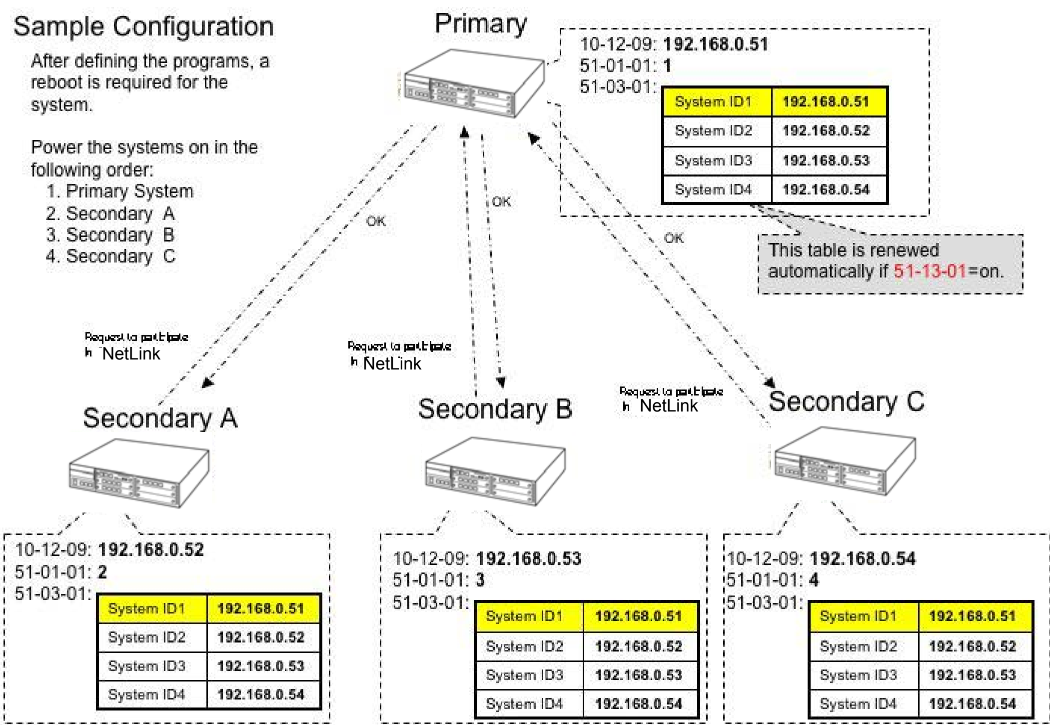
		Primary System				Secondary 1		Secondary 2	
		TDM Terminal	IP Terminal	CO Analog/ Digital	IP Trunk	TDM Terminal	CO Analog/ Digital	TDM Terminal	CO Analog/ Digital
Primary System	TDM Terminal	0	P:1	0	P:1	P :1 S1 :1	P :1 S1 :1	P :1 S2 :1	P :1 S2 :1
	IP terminal	P:1	0	P:1	P:2	S1 :1	S1 :1	S2 :1	S2 :1
Secondary System 1	TDM Terminal	P:1 S1:1	S1:1	P: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: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

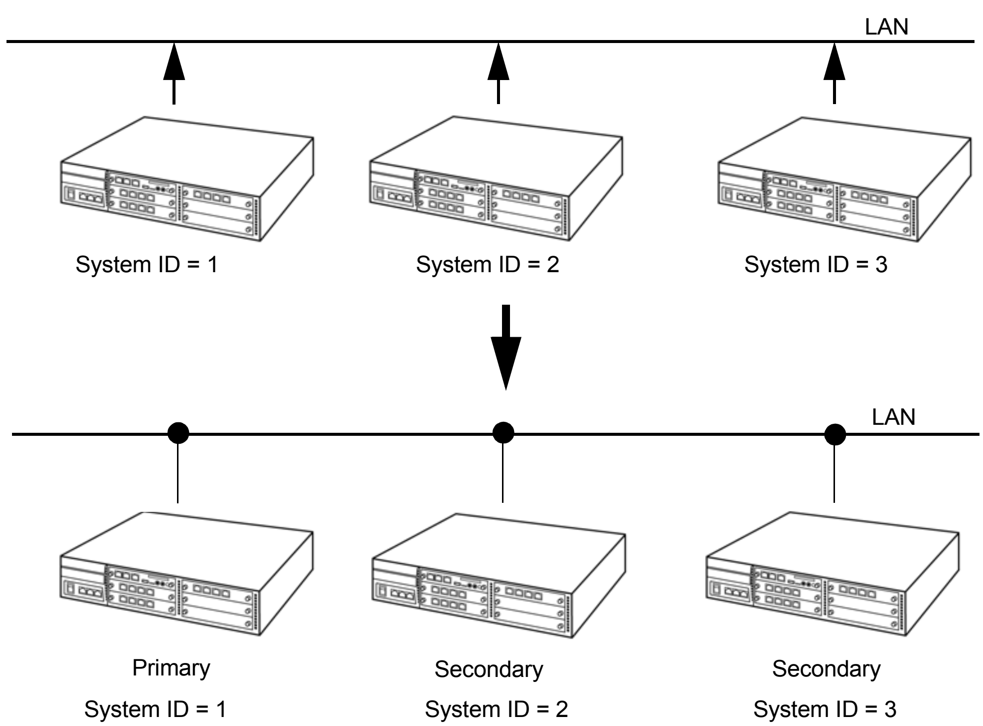
About VOIP DSP Settings

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
Primary
10-12-09: 192.168.0.51
51-01-01: 1
51-03-01:
System ID1 192.168.0.51
System ID2 192.168.0.52
System ID3 192.168.0.53
System ID4 192.168.0.54
This table is renewed automatically if 51-13-01=on.
OK
OK
OK
NetLink
NetLink
NetLink
Secondary A
10-12-09: 192.168.0.52
51-01-01: 2
51-03-01:
System ID1 192.168.0.51
System ID2 192.168.0.52
System ID3 192.168.0.53
System ID4 192.168.0.54
Secondary B
10-12-09: 192.168.0.53
51-01-01: 3
51-03-01:
System ID1 192.168.0.51
System ID2 192.168.0.52
System ID3 192.168.0.53
System ID4 192.168.0.54
Secondary C
10-12-09: 192.168.0.54
51-01-01: 4
51-03-01:
System ID1 192.168.0.51
System ID2 192.168.0.52
System ID3 192.168.0.53
System ID4 192.168.0.54



LAN
System ID = 1
System ID = 2
System ID = 3
LAN
Primary
System ID = 1
Secondary
System ID = 2
Secondary
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 configuration 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 10-37 and Section 9 Create Recovery Database on page 10-39 sections.

   CAUTION

   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***
   - ***Program 10-12-09 : VoIPDB IP Address***
   - ***Program 10-12-10 : Subnet Mask***
   - ***Program 84-26-01 : VoIPDB DSP IP Address***
   - ***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 GatewayProgram 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***
   - ***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 10-36.

   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 10-35 function.

➨ *For centralized voice mail the voice mail must be installed 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 10-37 and Section 9 Create Recovery Database on page 10-39 sections.

2. 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.

3. Program the following NetLink settings:

Primary - Site A	Secondary - Site B
51-01-01 System ID 51-04-01 Top Priority System Enter own IP address 51-06-01 Enable Automatic Integration	51-01-01 System ID 51-04-01 Top Priority System Enter site A (Primary) IP Address 51-01-03 Set Secondary Flag

4. After entering the System ID, the communication server asks to reboot the communication server. Reboot.

5. When the communication servers come up, the Primary System and Secondary Systems are connected and start working.
6. For database synchronization, wait three minutes (or more).

SECTION 17 NETLINK MULTI-SIP CARRIER

- When the Secondary NetLink system calls out using its own SIP Trunk, no DSPs are used from the Primary system.
- The NetLink Nodes which have their own SIP trunks can use Register IDs independently of each other. A Secondary NetLink system is able to utilize its own SIP trunks independently to the Primary system.

Conditions

- When Programming SIP Trunks on a secondary system only SIP Profile One can be utilized.
- It is possible to set Register ID for trunks that belong to that specific system. For example, a Register ID set in the Primary system cannot be assigned to a trunk in the Secondary system. The allocation of the trunk and Register ID of Program14-12 must be in the same system.
- In order to use CPN in a Secondary system, Program 51-19 must be turned on for those extensions. Once enabled, CPN may be sent on a per station basis using Program 21-19.
- Each NetLink system can use either SIP trunks to a provider or SIP trunk TIE line mode, but not both.
- Once NetLink is established, PCPro or WebPro must be used to change the system data related to the SIP trunks.
- Any SIP trunks that are built in a system before establishing NetLink will be deleted after establishing NetLink.
- SIP trunks are assigned in the order of system set up. System ID's are needed to assign Program 10-40 data.
- Registered SIP trunks can be utilized by any system in the NetLink network, as long as trunk route programming allows it.
- When a Secondary system becomes the primary after fail over, the SIP trunks will work for the effective license time.

Default Setting

No SIP Trunks are defined

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 18 How to Enter Programming Mode on page 10-52 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. Dial 150701 from the telephone dial pad. The message 15-07-01 TEL is displayed on the first line of the telephone display.

```
15-07-01    TEL
KY01 = *01
 ←                  →
```

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 18 How to Enter Programming Mode on page 10-52.

SECTION 18 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
-----	**-----**	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 19 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 GCD-CP10 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 20 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.

Table 10-4 Keys for Entering Data (Continued)

Keys for Entering Data	
Use this key...	**When you want to...**
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.
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 21 PROGRAMMING NAMES AND TEXT MESSAGES

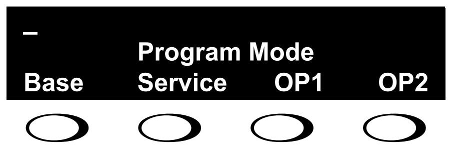
Several programs (e.g., Program 20-16 : Selectable Display Messages) require you to enter text. Use the following chart when entering and editing text. When using the keypad digits, press the key once for the first character, twice for the second character, etc. For example, to enter a C, press **2** 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

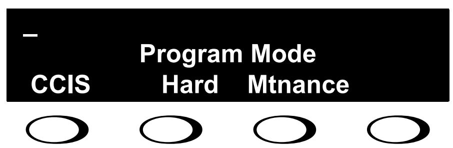
<table>
<tr><th>Use this keypad digit . . .</th><th>When you want to. . .</th></tr>
<tr><td>1</td><td>Enter characters:
1 @ [¥] ^ _ ` { | } ∅ ♦ Á À Â Ã Ç É Ê ì ó</td></tr>
<tr><td>2</td><td>Enter characters: A-C, a-c, 2.</td></tr>
<tr><td>3</td><td>Enter characters: D-F, d-f, 3.</td></tr>
<tr><td>4</td><td>Enter characters: G-I, g-i, 4.</td></tr>
<tr><td>5</td><td>Enter characters: J-L, j-l, 5.</td></tr>
<tr><td>6</td><td>Enter characters: M-O, m-o, 6.</td></tr>
<tr><td>7</td><td>Enter characters: P-S, p-s, 7.</td></tr>
<tr><td>8</td><td>Enter characters: T-V, t-v, 8.</td></tr>
<tr><td>9</td><td>Enter characters: W-Z, w-z, 9.</td></tr>
<tr><td>0</td><td>Enter characters:
0 ! “ # $ % & ’ ( ) ô Õ ú ä ö ü α ε θ</td></tr>
<tr><td>✱</td><td>Enter characters:
✱ + , - . / : ; < = > ? π Σ σ Ω □ ¢ £</td></tr>
<tr><td>#</td><td># = 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.)</td></tr>
<tr><td>Conf</td><td>Clear the character entry one character at a time.</td></tr>
<tr><td>Hold</td><td>Clear all the entries from the point of the flashing cursor and to the right.</td></tr>
</table>

SECTION 22 USING SOFTKEYS FOR PROGRAMMING

Each UNIVERGE SV9100 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.



Press the VOLUME ▲ or VOLUME ▼ to scroll between the menus.



SECTION 23 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.

Softkey Display Prompts	
If you press this Softkey . . .	**The system will. . .**
back	Go back one step in the program display. You can press VOLUME ▲ or VOLUME ▼ to scroll forward or backward through a list of programs.
↑	Scroll down through the available programs.
↓	Scroll up through the available programs.
select	Select the currently displayed program.
←	Move the cursor to the left.
→	Move the cursor to the right.
-1	Move back through the available program options.
+1	Move forward through the available program options.

Program 10 : System Configuration Setup

10-12 : GCD-CP10 Network Setup

Level:
SA

Description

Use **Program 10-12 :** GCD-CP10 **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 GCD-CP10.
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 (GCD-CP10)
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 IPLE.
10	**Subnet Mask**	128.0.0.0 192.0.0.0 224.0.0.0 240.0.0.0 248.0.0.0 252.0.0.0 254.0.0.0 255.0.0.0 255.128.0.0 255.192.0.0 255.224.0.0 255.240.0.0 255.248.0.0 255.252.0.0 255.254.0.0 255.255.0.0 255.255.128.0 255.255.192.0 255.255.224.0 255.255.240.0 255.255.248.0 255.255.252.0 255.255.254.0 255.255.255.0 255.255.255.128 255.255.255.192 255.255.255.224 255.255.255.240 255.255.255.248 255.255.255.252 255.255.255.254 255.255.255.255	255.255.0.0	Set for IPLE.

Input Data (Continued)

Item No.	Item	Input Data	Default	Description
11	**NIC Setup**	0 = Auto Detect 1 = 100Mbps, Full-Duplex 3 = 10Mbps, Full-Duplex 5 = 1 Gbps, Full-Duplex	0	Set for IPLE.

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

10-54 : License Configuration for Each Package

Level:
IN

Description

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

Input Data

Slot Number	1~24

License Index Number	1~32

Item No.	Item	Read Data
01	**License Code**	0000~9999
02	**License Quantity**	0~255

Conditions

❍ If applying more than 255 licenses to a slot the licenses must be applied across multiple indexes. For example assigning 256 VoIP resource licenses (5103) to the CPU slot could be assigned using different methods as long as the total for the CPU slot is 256:

1. Index 1 has 128 of feature code 5103 and index 2 also has 128 of feature code 5103 for a total of 256.
2. Index 1 has 255 of feature code 5103 and index 2 has 1 of feature code 5103 for a total of 256.

❍ When using IP devices IP Resource licenses (5103) must be assigned to the CPU Slot (1) for them to be available for use. If this is not done, IP related features will not work.

Feature Cross Reference

None

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.

NOTE

- *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

Input Data (Continued)

04	**Signal Transmit Method** 0 = Immediate This is the default setting which does not use Nagle Algorithm. When this is enabled data packets are immediately sent across the network with no buffering delay. 1 = Buffering Nagle Algorithm enabled. This means that small data packets will not be transmitted immediately across the network. The smaller data packets will be buffered and then sent across as larger data packets therefore decreasing the number of packets sent across the network. When the number of packets sent across the network decreases, the amount of bandwidth also decreases.	0 = Immediate 1 = Buffering	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.

NOTE

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

NEC

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)	0
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	0
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

Program 51 : NetLink Service

51-09 : NetLink Communication Port Settings

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 (Continued)

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.

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

Level:
IN

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.

Data Input

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

Program 51 : NetLink Service
51-13 : NetLink Options

Level:
IN

Description

Use **Program 51-13: NetLink Options** to enable automatic IP address List Operation updates, time zone information, and MAC address authorization.

Data Input

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.

NOTE

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

Input Data

System ID	1~50

Input Data

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

Program 51 : NetLink Service
51-15 : Demonstration Setting

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..

NOTE

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

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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

Level:
IN

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	0
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

.

IMPORTANT

In a NetLink system with In-Mail, replication should be scheduled during non-peak hours of operation

Conditions

None

Feature Cross Reference

None

Program 51 : NetLink Service

51-17 : NetLink DT800/DT700 Server Individual Information Setup

Level:
IN

Description

Use **Program 51-17: NetLink DT800/DT700 Server Individual Information Setup** to set the NetLink port information.

Input Data

System ID	1~50

Input Data

Item No.	Item	Input Data	Default
01	**Register Port** Use to set the SIP Register Port of each system.	0 ~ 65535	5080
02	**Subscribe Session Port** Use to set the SIP Subscribe Session Port number of each system when NetLink is used.	0 ~ 65535	5081

Conditions

None

Feature Cross Reference

None

Program 51 : NetLink Service
51-18 : NetLink Configuration Options

Level:
IN

Description

Use **Program 51-18: NetLink Configuration Options** to set the NetLink Fail-Over limits.

Input Data

Item No.	Item	Input Data	Default
01	**NetLink Fail-Over Limit** When tear-down of network was repeated more than the specified times, NetLink is operated stand-alone.	0, 2~10 (0 = Infinity)	0

Conditions

None

Feature Cross Reference

None

Program 51 : NetLink Service

51-19 : NetLink IP Trunk (SIP) Calling Party Number Setup for Extension

Level:
IN

Description

Use **Program 51-19: NetLink IP Trunk (SIP) Calling Party Number Setup for Extension** to set CPN transmission for each secondary system.

Input Data

Item No.	Item	Input Data	Default
01	**NetLink CPN Transmission** This program assigns transmission of Calling Party Number (CPN) from PRG 21-19 for each secondary system. The transmission applies for every extension.	0 = Disable 1 = Enable	1

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 GCD-CP10 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 GCD-CP10. 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 GCD-CP10.

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

NAPT

Chapter 11

Section 1 NAPT

1.1 Introduction

NAPT, or Network Address Port Translation, is a method by which a private address, or addresses, and their TCP/UDP ports are translated into a single public address and its TCP/UDP ports. In the case of DT800/DT700 IP phones with the SV9100, it allows their connection to a public (Internet) IP address which is then converted back to the private (non-Internet) IP address on the customers network. The translation is available at the SV9100 end as well as at the remote IP Phone end of the connection if required. The feature is NOT available for IP-CCIS and Netlink connections.

The router that the SV9100 resides behind still requires Port Forwarding statements. However the router that the IP terminal/terminals reside behind may not require any port forwarding.

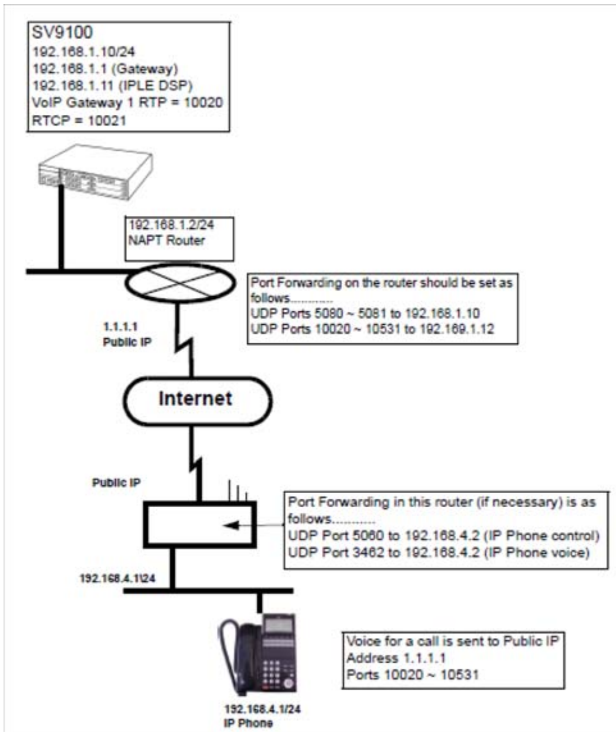
Due to the fact that there are many manufacturers producing routers there may still be times when port forwarding is required.

This feature is only effective when PRG 15-05-45 set to a "1" (ON).

In all software versions, SIP ALG's (Application Level Gateway) or other SIP Applications MUST be disabled in all routers. If a SIP ALG or a similar SIP application is enabled, IP phone service WILL be interrupted.

The example below is with 256 DSP Resources licensed to the GCD-CP10. Your Port forwarding range could change depending on number of DSP Resource Licenses.

Figure 11-1 NAPT Configuration Example



1.2 Requirements

The following information provides requirements for NAPT.

1.2.1 Main Software

NAPT is supported with the V1000/V1.00 or higher release of the SV9100.

1.2.2 Hardware

The SV9100 requires the following hardware:

- GCD-CP10
- GPZ-IPLE

1.2.3 Capacity

896 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 DT800/DT700 ITL Terminals.
- IP Terminal License (5111)
- System Port License (0300)
- Version 1 License (0411)

1.3 Installation

The following settings have been added for NAT traversal in the DT800/DT700 ITL Terminal.

- Terminals must be V3.0.0.0 or higher to support the NAPT feature.

1.3.1 Settings for Terminals

NAT Traversal 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.
2. Network Area Name	A character string	No value	Available	Available	The name of the network area to which terminals belong. Specify up to 32 alphanumeric characters. **(Optional step)**
3. WAN Settings					See table below.

WAN Settings: 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 SV9100 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 SV9100 uses for SIP registration in PRG 10-46-06.
3. WAN Self IP Address	IP Address	0.0.0.0	Available	Available	Only used when Static NAT is enabled. This setting is the WAN address of the router that the NAPT Terminal resides behind.

Self Port Settings: 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: At default this is assigned to port 3462. The first IP phone on this local LAN can use this port. The second IP phone would need to be changed to port 3464.
2. SIP Self Port	1024~65535	5060	Available	Not available	The number of the port receiving SIP data. Note:At default this is assigned to port **5060**. The first IP phone on this local LAN can use this port. The second IP phone would need to be changed to port **5062**.

1.4 Programming Example

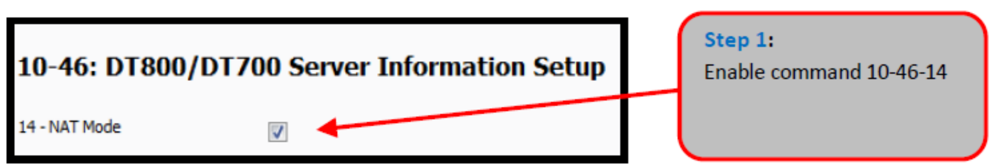
NAPT

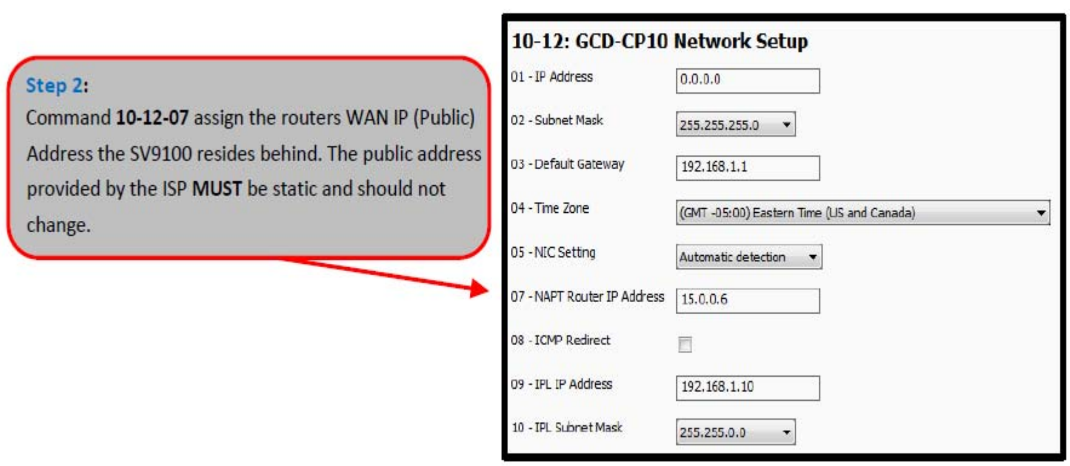
SV9100

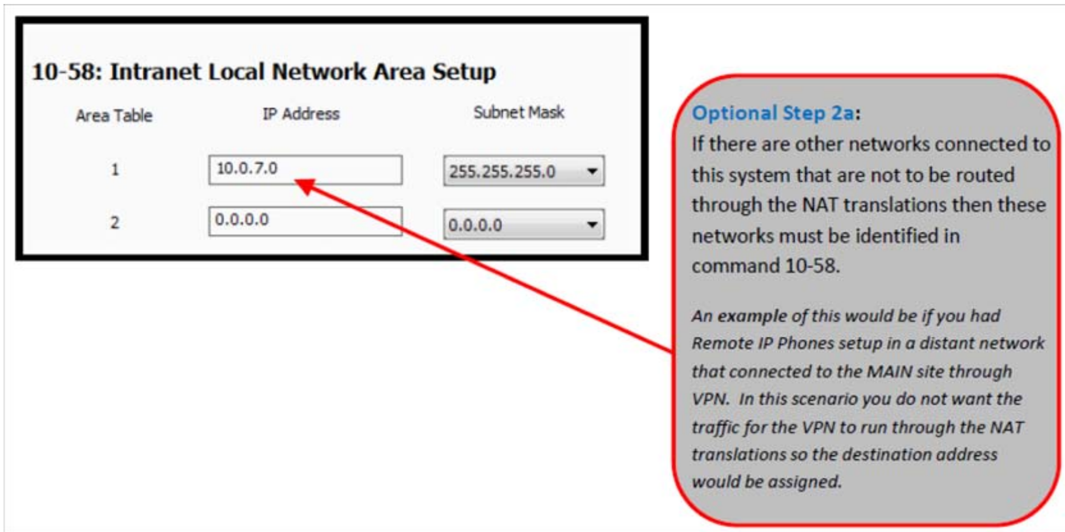
NAPT, or Network Address Port Translation, is a method by which a private address or addresses and their TCP/UDP ports are translated into a single public address and its TCP/UDP ports. In the case of IP phones with the SV9100 it allows their connection to a public (internet) IP address which is then converted back to the private (non-internet) IP address on the customers network. The translation is available at the SV9100 end as well as at the remote IP Phone end of the connection if required. The feature is NOT available for IP-CCIS and Netlink connections.

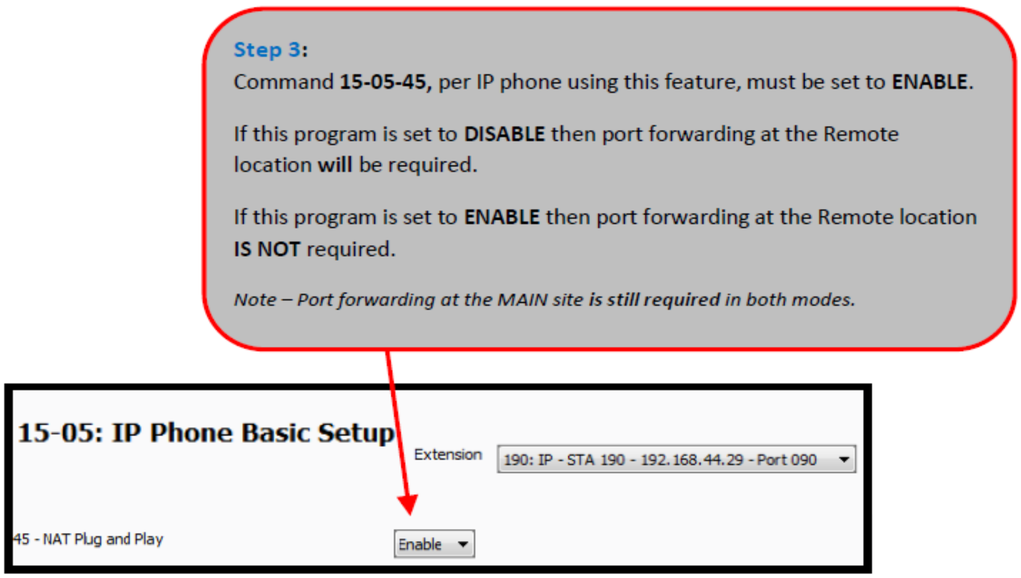
Note 1: The NAPT feature requires the following licenses to be loaded to the CPU: IP Terminal (5111), Version 1000 (0411), VoIP Channel (5103), and System Port (0300).

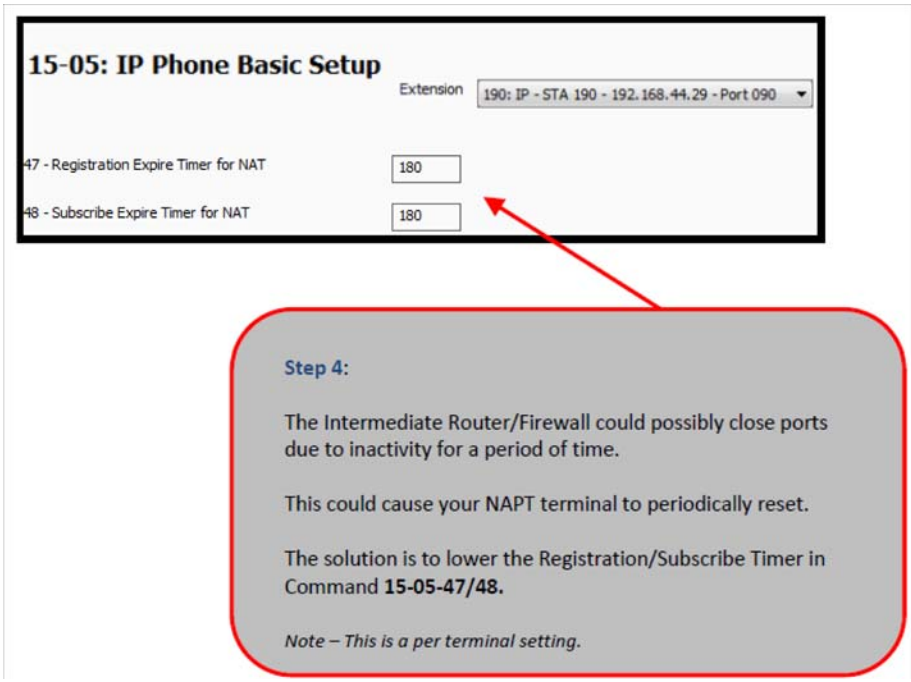
CPU Setup

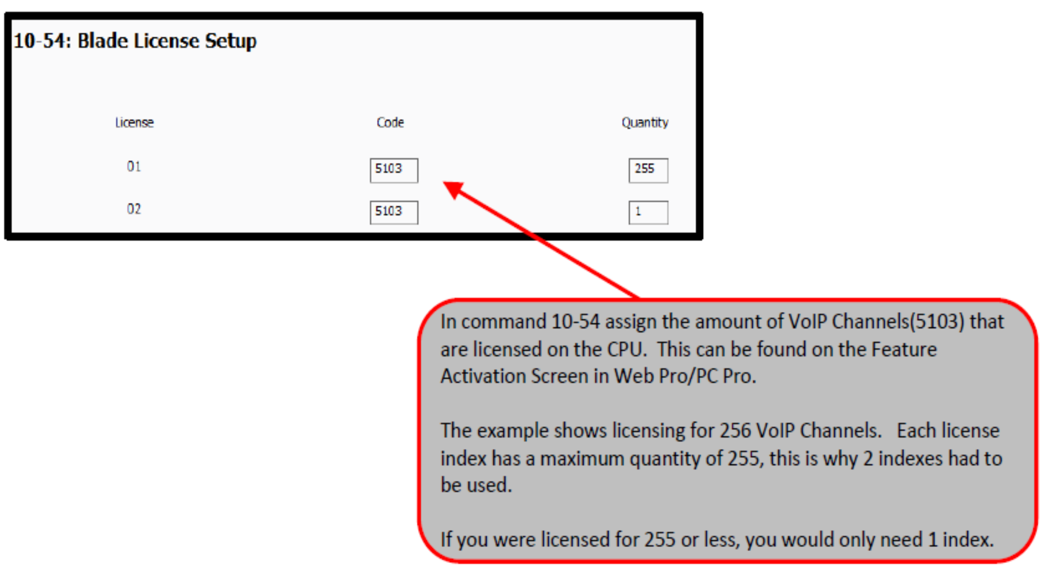


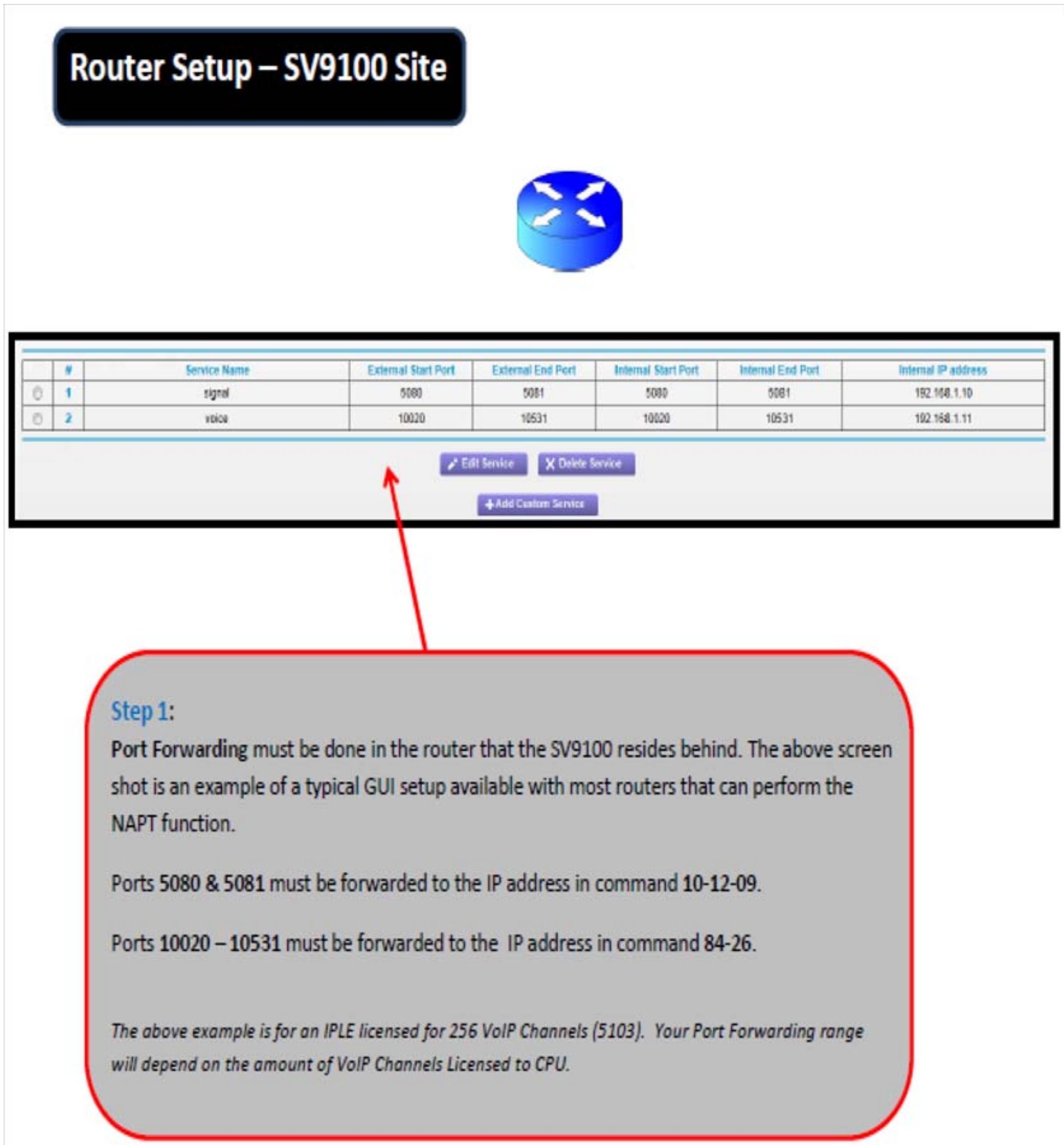



10-58: Intranet Local Network Area Setup
Area Table
IP Address
Subnet Mask
1
10.0.7.0
255.255.255.0
2
0.0.0.0
0.0.0.0
Optional Step 2a:
If there are other networks connected to this system that are not to be routed through the NAT translations then these networks must be identified in command 10-58.
An example of this would be if you had Remote IP Phones setup in a distant network that connected to the MAIN site through VPN. In this scenario you do not want the traffic for the VPN to run through the NAT translations so the destination address would be assigned.


Step 3:
Command 15-05-45, per IP phone using this feature, must be set to ENABLE.
If this program is set to DISABLE then port forwarding at the Remote location will be required.
If this program is set to ENABLE then port forwarding at the Remote location IS NOT required.
Note – Port forwarding at the MAIN site is still required in both modes.
15-05: IP Phone Basic Setup
Extension
190: IP - STA 190 - 192.168.44.29 - Port 090
45 - NAT Plug and Play
Enable


15-05: IP Phone Basic Setup
Extension
190: IP - STA 190 - 192.168.44.29 - Port 090
47 - Registration Expire Timer for NAT
180
48 - Subscribe Expire Timer for NAT
180
Step 4:
The Intermediate Router/Firewall could possibly close ports due to inactivity for a period of time.
This could cause your NAPT terminal to periodically reset.
The solution is to lower the Registration/Subscribe Timer in Command 15-05-47/48.
Note – This is a per terminal setting.


10-54: Blade License Setup
License
Code
Quantity
01
5103
255
02
5103
1
In command 10-54 assign the amount of VoIP Channels(5103) that are licensed on the CPU. This can be found on the Feature Activation Screen in Web Pro/PC Pro.
The example shows licensing for 256 VoIP Channels. Each license index has a maximum quantity of 255, this is why 2 indexes had to be used.
If you were licensed for 255 or less, you would only need 1 index.


Router Setup – SV9100 Site
| Service Name | External Start Port | External End Port | Internal Start Port | Internal End Port | Internal IP address
1 | signal | 5080 | 5081 | 5080 | 5081 | 192.168.1.10
2 | voice | 10020 | 10531 | 10020 | 10531 | 192.168.1.11
Edit Service
Delete Service
Add Custom Service
Step 1:
Port Forwarding must be done in the router that the SV9100 resides behind. The above screen shot is an example of a typical GUI setup available with most routers that can perform the NAPT function.
Ports 5080 & 5081 must be forwarded to the IP address in command 10-12-09.
Ports 10020 – 10531 must be forwarded to the IP address in command 84-26.
The above example is for an IPLE licensed for 256 VoIP Channels (5103). Your Port Forwarding range will depend on the amount of VoIP Channels Licensed to CPU.

IP Terminal Setup

Note: Terminals must be version 3.0.0.0 or higher to support the NAPT feature

The below settings are assigned via the configuration mode of the IP Terminal. They can also be set up via a GUI by browsing to the IP address of the terminal.

To enter this mode hold down the ***MENU*** key. The login is ***ADMIN*** and password ***6633222***

Step 1:

(2) SIP Settings

(8) NAT Traversal

(1) NAT Traversal Mode

(1) Disable: *This disables the NAPT feature in the terminal*

(2) Dynamic:
This setting is used to automatically acquire the WAN IP address of the router that the IP terminal resides behind.

(3) Static:
This setting would be used when the IP terminal could not acquire the WAN IP address of the router that it resides behind. Some routers do not support Dynamic NAT and these routers would require you to statically assign the routers WAN IP Address here.

Note: *It is recommended to use Dynamic NAT and to leave this command unassigned. The reason for using Dynamic NAT is the local router may not have a static IP address assigned and periodically receives a different Public IP address. With Dynamic NAT set the phones will update the change in address when it happens. With static NAT selected any change in the Public IP address would require the user to update the phone each time a change is detected.*

Step 1 continued:
It is recommended to use Dynamic NAT.

Step 2:

(2) SIP Settings

(8) NAT Traversal

(3) WAN Settings

(1) WAN Mate IP Address:

Assign the **WAN IP address** *that is assigned in command* ***10-12-07****.*

Note: *This is the WAN Address of the router the SV8100 resides behind.*

(2) WAN SIP Mate Port:

Change this to **5080**

Note: *This is the port number assigned in command* ***10-46-06***

(3) WAN Self IP Address:

If the phone is set to **Static NAT**, *then assign the* **WAN IP Address** *of the router that the IP Phone resides behind.*

Note: *If the phone is set to Dynamic NAT, leave this set to 0.0.0.0*

Step 3:

Save all the settings and allow the phone to reset and test.

The IP terminal should then come online and have speech path in both directions on a call in progress.

Multiple IP Phones behind the same NAT

Step 1:

(1) Network Settings

(6) Advanced Settings

(5) Self Port Settings

(1) RTP Self Port:

At default this is assigned to port 3462 *The First IP phone on this local LAN can use this port. The second IP phone would need to be changed to port* 3464, *the third IP phone would be changed to* 3466, *the fourth IP phone would be changed to* 3468, *etc, etc.*

(2) SIP Self Port:

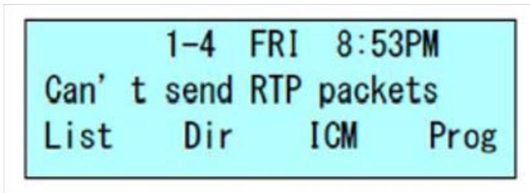
At default this is assigned to port 5060. *The first IP phone on this local LAN can use this port. The second IP phone would need to be changed to port* 5062, *the third IP phone would be changed to* 5064, *the fourth IP phone would be changed to* 5066, *etc, etc.*

Save these settings and reset the IP phone. If the first IP phone came online using Dynamic NAT then the other phones should follow also using Dynamic NAT.

Note – The above settings are only required when multiple NAPT phones are setup on the same Remote location. If there are NAPT phones at multiple remote locations, containing only 1 phone at each site, then the ports do not have to be re-assigned.

Section 2 Conditions

- ❑ NAPT is supported for DT800/DT700, and Standard SIP Terminals.
- ❑ "License Exceeded" will display on the DT800/DT700 terminal when trying to register a NAT phone, if the feature is not licensed.
- ❑ 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 SV9100 side will require port forwarding.
- ❑ K-CCIS IP does not use NAPT translations.Softphone (SP310) is not supported.
- ❑ If NAPT enabled phones become unresponsive after being idle, the timer in PRG 15-05-47 and 15-05-48may need to be changed to a shorter interval.   NAPT terminals will ignore the timers in PRG 84-23-01 and 84-23-02 and use the timers in PRG 15-05-47 and 15-05-48 instead.
- ❑ It is necessary to set Program 10-46-14 to **OFF** when the IPLE 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, however they require the same WAN Address in PRG 10-12-07.
- ❑ When Program 15-05-45 is set to "1" the manual table setting for port forwarding may not be required on the remote side Router B, but the router must support the NAT function setting itself. If Program 15-05-45 is set to "0" port forwarding at the Remote side router is required.
- ❑ The router may close the port being used if packet exchange is not performed during a certain time frame. In this case, change Programs 15-05-47 and 15-05-48 to a shorter interval. These programs are changed on a per station basis. No- NAPT phones will still use Programs 84-23-01 and 84-23-02 while only NAPT phones will use Programs 15-05-47 and 15-05-48.
- ❑ The SIP-ALG or similar SIP application function of all routers/firewalls in the network must be disabled.
- ❑ After starting SIP negotiation for the call, if any RTP packet can't be received from the terminal within 10 seconds, the call is dropped and the IP Terminal displays the following:



- ❑ Each NAPT terminal can have a separate Register and Subscribe expire timer. The load of the CPU will increase with each NAPT terminal using a short timer. The following chart shows the minimum timer settings based on the number of NAPT terminals using PRG 15-05-47 and 15-05-48.

Program Number	Number of DT700 terminals using PRG's 15-05-47 and 15-05-48			
	1 ~ 144 (terminals)	145 ~ 192 (terminals)	193 ~ 464 (terminals)	465 ~ 512 (terminals)
15-05-47 (minimum setting)	60 sec	90 sec	180 sec	180 sec
15-05-48 (minimum setting)	60 sec	90 sec	180 sec	480 sec

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.)
- ❑ Encryption with the SV9100 IPLE is not supported with IP terminals connected via NAPT.
- ❑ 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	2	3
10-12-01	**GCD-CP10 Network Setup – IP Address**	Assign the IP Address for the CPU NIC card. When a IPLE card is installed in the system it is recommended to set this PRG to **0.0.0.0**	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 = 172.16.0.10	✓		
10-12-03	**GCD-CP10 Network Setup – Default Gateway**	Assign the default gateway IP address for the GCD-CP10.	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	**GCD-CP10 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)	✓		

Program Number	Program Name	Description/Comments	Assigned Data	Level		
				1	2	3
10-12-09	**GCD-CP10 Network Setup – IP Address**	Set for IPLE *The IP Address assigned in Program 10-12-01 cannot start with the same leading digits as the IP Address assigned here.*	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254 (default = 172.16.0.10)	✓		
10-12-10	**GCD-CP10 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)	✓		
15-05-50	**IP Phone Basic Setup - Peer-to-Peer Mode**	Enable/Disable the Peer-to-Peer feature for the IP station.	0 = Off 1 = On (default)			✓

Program Number	Program Name	Description/Comments	Assigned Data	Level		
				1	2	3
10-46-01	**DT800/DT700 Server Information Setup - Register Mode**	**Normal mode:** When the phone boots up, it reports the ext. assigned in the phone or chooses the next available extension in the system. No password is required. **Auto mode**: If set to Auto, the SIP user name and password must be entered in the actual IP phone. These settings must match 84-22/ 15-05-27, for the phone to come on-line. **Manual mode:** When the phone boots up it prompts a user to enter a user ID and password before logging in. It checks the user ID/password against 84-22/15-05-27. If there is no match, the phone does not come on-line.	0 = Normal (default) 1 = Auto 2 = Manual		✓	
10-46-06	**Register Port** Assign the port number to which the SIP messages are sent to on the IPLE. 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	**Network Address**	This program sets the local area network of the DT800/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	**Network Address - Subnet Mask**	This program sets the local area network of the DT800/DT700 terminal when the system has a NAT router and a local router.	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		✓	
15-05-45	**IP Telephone Terminal Basic Data Setup - NAT Plug and Play**	This Program is valid when Program 10-46-14 is On (NAT feature activated). Select sending RTP port number to remote Router, use from negotiation result (0) or received RTP packet (1). SV9100 uses this program to decide a destination port of RTP transmitting packets from IPLE to a remote IP terminal. If “0:OFF” is selected, the destination port of RTP transmitting packets will be a SIP/SDP negotiation result.(same behavior as before). If you chose “1:ON”, the destination port of RTP transmitting packet will be the same port of a source port of a receiving RTP packet on IPLE.	0 = Disable (default) 1 = Enable	✓		

Program Number	Program Name	Description/Comments	Assigned Data	Level		
				1	2	3
15-05-47	**Registration Expire Timer for NAPT**	On a per station basis, this setting defines the SIP registration expiry timer. This setting only applies to DT800/DT700 stations connected via NAPT. If this value is set to 0, for a NAPT terminal, the value in PRG 84-23-01 is applied.	0 = Disable 1 = 60 - 65535 (sec) (default - 180)		✓	
15-05-48	**Subscribe Expire Timer for NAPT**	On a per station basis, this setting defines the SIP subscribe expiry timer. This setting only applies to DT800/DT700 stations connected via NAPT. If this value is set to 0, for a NAPT terminal, the value in PRG 84-23-02 is applied.	0 = Disable 1 = 60 - 65535 (sec) (default - 180)		✓	
84-26-01	**IPL Basic Setup – IP Address**	When using a GPZ-IPLE assign only one IP address.	Slot 1 = 172.16.0.20	✓		
84-26-02	**IPL Basic Setup – RTP Port Number**	Assign the RTP port number to be used for each DSP on the IPLE. ✎ *Only even numbered ports are supported.*	VoIP GW1 = 10020	✓		
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		✓	

All DSP Busy Indication

Chapter 12

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 12-1 Alarm Types

Parameters	Description
STA	DSP for IP Station Calls Were All Busy.
TRK	DSP for Trunk Calls Were All Busy, includes SIP 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 IPLE blade.
Clock/Calendar	NetLink System
VoIP DSP All Busy YY-XX	YY = System ID
Soft Key	XX = Slot number of CCPU with IPLE 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 SV9100 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
- ➥ SV9100 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)		✓	

AspireNet

Chapter 13

Section 1 Introduction

1.1 What is AspireNet?

The AspireNet package provides a seamless connection of multiple systems into a single “virtual” communications system using ISDN (PRI/BRI) and VoIP lines with a unified numbering plan.AspireNet will allow many companies to connect their telephone systems so they appear as one. This will give them the ability to have only one operator to manage the system and share one voice mail within the network. An extension user in the network can easily dial another extension or transfer a call within the AspireNet 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.

Centralized Voice Mail
Centralized Voice Mail allows multiple networked systems to share a single voice mail system. This centralized voice mail can receive calls from and transfer calls to any destination in any network node. Unanswered calls recall and route as if they were part of a single, much larger system.

Flexible Network Routing
Use network routes to set up “single channel” networking between many separate systems - or use multiple networking channels per system for greater network performance. Data tables in the system program define the routing for each extension in each network node. These tables are easily customized to meet the requirements of each networking configuration.

Users may place an intercom call or transfer a call to any extension at any location by simply dialing an extension number. The system analyses each extension number received and determines how to route the call to its final destination. The feature which handles this route selection is called Flexible Routing (F-Routing). Once an extension number is dialed, the system checks the routing, accesses the assigned trunk group and places the call. Each extension is assigned a route or routes that decides which trunk group to access and any modified dialed data if required.

Busy Lamp Indication
The status of an extension will be shown at a Hotline key/DSS Console on another networked system. This allows a Centralized Operator to have lamp indication of extensions in the network or an extension user to have a Hotline key for a co-worker on another system.

Centralized Operation
Centralized Operator allows multiple networked systems to share a single operator. The operator can be accessed by a single digit code, if the operator is busy your call will automatically queue until the operator becomes free. The operator can have a DSS console to show the status of users anywhere in the network.

Paging
AspireNet allows a user to place Paging call to a networked system. If you need to get through to a co-worker who is not at their desk then place a paging call to their system.

Call Forwarding
You can forward your calls to an extension at another networked extension. If you visit another site within your network but forgot to set a call forward then use Follow Me to have your calls forwarded to you. Follow Me is also useful if you have an DECT handset. If your handset is subscribed at the site you are visiting then you can use Follow Me to have your calls forwarded to your DECT handset.

Direct Dial In(DDI)
A DDI call can be routed to any extension within the network. This allows an extension to receive a DDI call from any other system in the network. Along with trunk access it is possible for an extension on a system that has no trunk lines to use the trunks of another system in the network.

Trunk Access
An extension can access a trunk line at another system in the network. The user dials the standard trunk access code and the system will automatically route the call to the system that has trunks connected.

Conference
An extension can have a conference call that includes co-workers at another system within the network.

1.2 SV9100 PCBs

There are two methods available for AspireNet connection as shown in the following table.

Interface	Description	Note
ISDN	Using Q.931 based proprietary protocol, Basic rate interface and Primary rate interfaces are available.	A PRIA or BRIA Blade is required for connection.
VoIP	Using H.323 protocol for voice transmit protocol.	An IPLE daughter board is required.

ISDN AspireNet
Using ISDN AspireNet the system provides up to 256 B-channel ports which can be used for Networking.
A PRIA circuit will take 30 ports and each BRI circuit will take 2 ports.
These ISDN AspireNet ports are independent of the trunk and station ports available on the system.

IP AspireNet

Using IP AspireNet the maximum quantity of simultaneous calls is limited by the availability of resources on the IPLE PCB’s installed. A PZ-IPLE PCB’s giving a maximum of 256 speech channels. The maximum quantity of calls may also be reduced by the compression mode (CODEC type) of the IPLE PCB’s, this is selectable by the installer in Program 84-12-28.

SV9100 System Requirements
AspireNet is supported on the GCD-CP10 blade

1.3 Available Features

Feature Name
ARS/F-Route
Barge In
BLF Indication (at Hotline / DSS Console key)
Call Forwarding Override
Caller ID Display
Call Forwarding: Immediately Busy No Answer Both Ring
Call Forwarding, Follow Me
Call Forwarding, Off-Premise
Call Waiting / Camp On
Callback
Central Office Calls, Placing: Seizing a trunk in networked system
Conference
Department Step Calling
Direct Dial In (DDI)
Direct Inward Line (DIL)
Direct Inward System Access (DISA)
Department Calling
Hold
Hotline / DSS Console
Intercom: Change Voice/Signal Ring
Keep Alive Operation
Last Number Redial
Message Waiting
Operator, Centralized
Paging
Park
Ringdown Extension, Internal/External
Selectable Display Messaging
Toll Restriction
Transfer
Voice Mail, Centralized

1.4 Using This Manual

This Chapter is in four sections:

Section 1: Introduction
This section provides an Introduction for the AspireNet.

Section 2: Setting Up AspireNet
This section guides you step by step in setting up a basic AspireNet system.

Section 3: AspireNet Features
This section provides a description of each of the AspireNet Features

Section 4: Programming
This section describes all of the programming commands required to install an AspireNet system.

Section 5: Examples of AspireNet Configurations
This section shows diagrams and simple programming instructions for AspireNet.

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

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

```
15-07-01    TEL301
KY01 = *01
 ←                 →
```

tells you to enter the programming mode, dial 150701 from the telephone dial pad. After you do, you'll see the message "15-07-01 TEL301" 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 extension 301. 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, see 4.3 How to Enter Programming Mode on page 13-55.

1.5 Unique Considerations

Read These Notes

Simplifying Keyset Operation with One-Touch Keys...

A system phone user can access many features through Service Codes (e.g., Service Code *0 answers a Message Waiting from a co-worker). To streamline the operation of their phone, a system phone user can store these codes under One-Touch Keys. This provides one-button operation for almost any feature. To find out more, read the One-Touch Calling feature in your Feature & Specifications Manual.

Programmable Keys...

When reading an instruction using programmable keys, you will see a notation similar to ***(PGM 15-07 or SC 751: 06).*** This means that the key requires function code 06, and you can program this code through Program 15-07 or by dialing Service Code 751. Refer to the Programmable Function Keys feature in your Features and Specifications Manual if you need more information.

Using Handsfree...

The manual assumes each extension has Automatic Handsfree. This lets a user just press a line key or Speaker key to answer or place a call. For extensions without Automatic Handsfree, the user must:

Lift the handset or press SPK for Intercom dial tone

Lift the handset or press SPK, then press a line key for trunk dial tone

SECTION 2 SETTING UP ASPIRENET

2.1 Required System Programming

The selection of an ISDN PRIA, BRIA, or IPLE Blade determines the type of programming you must do on the SV9100.

Refer to either the AspireNet ISDN or AspireNet IP section below.

Also refer to the examples in the Section 5.

2.2 AspireNet ISDN

- **10-03-01 : PCB Setup - ISDN Line Mode**

Determine the line mode of the ISDN. If Basic Rate Interface (BRI) is chosen, the setting must be done for each line. The settings must match at each end of the networked line. The following entries are acceptable for Networking.

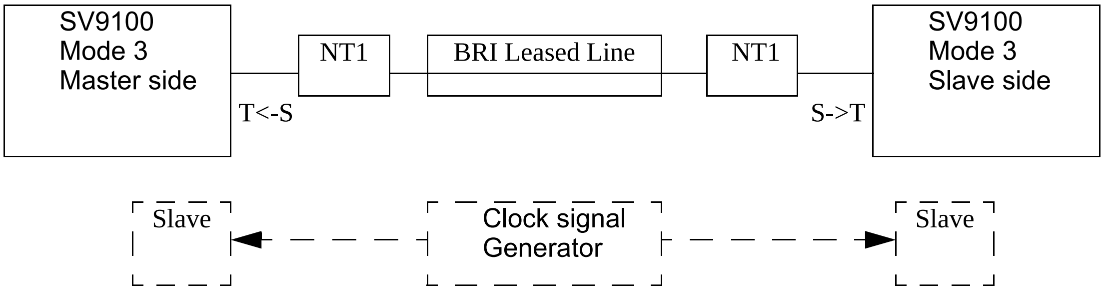
ISDN Line Number	1-4

Item No.	Item	Input Data	Default
01	ISDN Line Mode	1 = T-Point 2 = S-Point 3 = Network Mode (Leased Line) 4 = Network Mode (Interconnected Line) 5 = Interconnection (Interconnected Line, Fixed Layer 1 Forced NT Mode)	1

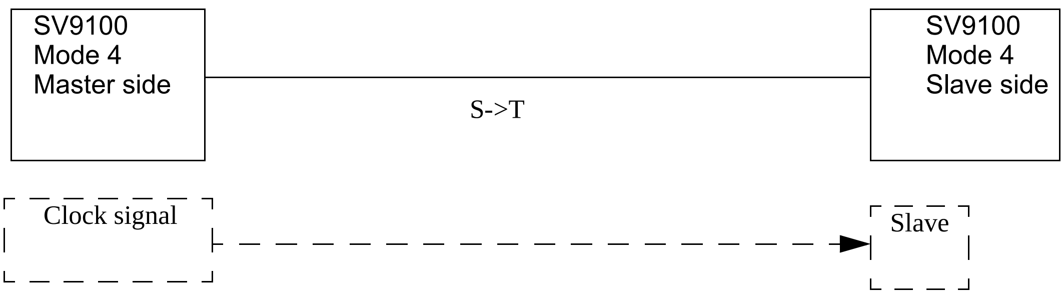
A PRIU interface will provide up to 30 channels, the BRIU interface will provide 2 channels. Program 10-32-01 can limit the quantity of channels available for PRIU interfaces.

The installation of each mode is shown in the following diagrams.

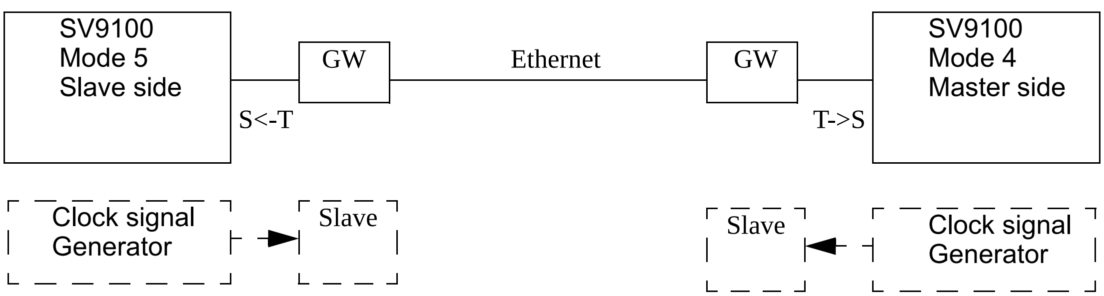
Mode 3 BRI/PRI Network Mode (Leased Line)



Mode 4 BRI/PRI Network Mode (Interconnected Line).



Mode 5 BRI/PRI Interconnection Mode (Interconnected Line, Layer 1 = NT).



- **10-03-03: PCB Setup - Connection Type**

The connection type should be changed if Basic Rate Interface (BRI) is used. **Only Point-to-Point connection (1) is available for system interconnection.**

ISDN Line Number	1-4

Item No.	Item	Input Data	Default

03	Connection Type	0 = Point-to-Multipoint (not available for Networking) 1 = Point-to-Point	0

Example:

System – A	System – B
1: Point-to-Point	1: Point-to-Point

10-03-10 : PCB Setup - Master/Slave System

Determine which system will be the master system and which one(s) will be the slave system(s). If one end of the line is set as the Master, the other end of the line must be set as the Slave. The choice of Master/Slave is determined by the ISDN clock available at the SV9100 System, see the Section 5 for further detail.

ISDN Line Number	1-4

Item No.	Item	Input Data	Default
10	Master/Slave System (Network Mode Only)	0- Slave System 1- Master System	0

Example:

System – A	System – B
1: Master	2: Slave

10-03-11 : PCB Setup - Networking System Number

The Networking ID is used to select the access route for dialed digits. You can choose any number 0 to 50 (0 equals no operation). This ID is used when setting the numbering plan for the networked systems. The same ID number must be set in both 10-03-11 and 11-01. Refer to **2.5 Numbering Plan in on page -12** for more on the numbering plan settings.

ISDN Line Number	1-4

Item No.	Item	Input Data	Default
10	Networking System Number (Network Mode Only)	0-50	0

Example:

System – A	System – B
Networking ID: 1	Networking ID: 1

2.3 AspireNet IP

CAUTION

As with any Voice Over IP (VoIP) implementation, there are several issues that should be considered when setting up AspireNet IP

2.3.1 System IP Address

- **10-12-09 : GCD-CP10 Network Setup - IPL IP Address (IPLE)**

  Select the IP address of the IPLE Blade (default: 172.16.0.10). A static IP address is required. Each SV9100 system within the network must have a unique IP address.

  The system must be reset in order for the change to take effect.

- **10-12-10 : GCD-CP00 Network Setup - Subnet Mask (IPLE)**

  Select the Subnet Mask to be used by the IPLE Blade (default: 255.255.0.0).

Item No.	Item	Input Data			Default	Conditions
09	IP Address (IPLE)	1.0.0.1 - 126.255.255.254 128.1.0.1 -191.254.255.254 192.0.1.1 - 223.255.254.254			172.16.0.10	Set for IPLE.
10	Subnet Mask (IPLE)	128.0.0.0 240.0.0.0 254.0.0.0 255.192.0.0 255.252.0.0 255.255.128.0 255.255.248.0 255.255.255.0 255.255.255.224 255.255.255.252	192.0.0.0 248.0.0.0 255.0.0.0 255.224.0.0 255.254.0.0 255.255.192.0 255.255.252.0 255.255.255.128 255.255.255.240 255.255.255.254	224.0.0.0 252.0.0.0 255.128.0.0 255.248.0.0 255.255.0.0 255.255.224.0 255.255.254.0 255.255.255.192 255.255.255.248 255.255.255.255	255.255.0.0	Set for IPLE.

Example:

System – A	System – B
IP Address: 172.16.0.10	IP Address: 172.16.0.11
Subnet Mask: 255.255.0.0	Subnet Mask: 255.255.0.0

2.3.2 IPLE Card IP Address

- **84-26-01 : IPLE Basic Setup - IP Address (DSP)**

Slot Number	1

Item	Input Data	Default	Description	Related Program
IP Address	1.0.0.1–126.255.255.254 128.1.0.1–191.254255.254 192.0.1.1–223.255.254.254	Slot 1: 172.16.0.20	Assign the IP address for each DSP on the GPZ-IPLE.	

Example:

System – A	System – B
IP Address: 172.16.0.20	IP Address: 172.16.5.20
Subnet Mask: 255.255.0.0	Subnet Mask: 255.255.0.0

2.3.3 IP Address of other systems within the network

- **10-27 : IP System ID - IP Address**

The IP Addresses of all other systems within the network must be defined in this program. Each system is identified by a 'System Number' (1 to 50), this is used within the numbering plan to route calls to another system.

System ID	01-50

Item No.	Item	Input Data	Default	Related Program
01	**IP Address** System ID is related with the System ID in the Numbering Plan (Program 11-01-03). When the digits are analyzed and the system ID is determined from the system data set in the Numbering Plan, the Networking call will be sent to the IP Address set in this program. The IP Address should be the IP Address of the peer GCD-CP10 (Program 10-12-01).	1.0.0.1 - 126.255.255.254 128.1.0.1 - 191.254.255.254 192.0.1.1 - 223.255.254.254	0.0.0.0	11-01-03 10-12-01
02	**Call Procedure Port** The Port Number should be set with the same value as the H.225 setup port in Program 84-02-33.	1-65535	1730	84-02-33

Example:

System – A	System – B
System ID: 1	System ID: 1
IP Address: 172.16.5.10	IP Address: 172.16.0.10
Port: 1730	Port: 1730

2.3.4 Networking TCP Port

- **10-20-01 : LAN Setup for External Equipment - TCP Port**

Define the TCP port number for communicating to external equipment. The port number defined should be the same in each networked system.

Type of external equipment	1- CTI Server 2- ACD MIS 3- - Reserve - 4- Network System 5- - Reserve -

Item No.	Item	Input Data	Default
01	TCP Port	0-65535	External Device 1 = 0 External Device 2 = 0 External Device 3 = 0 External Device 4 = 0 External Device 5 = 0 :

Example:

System – A	System – B
External Equipment: 4	External Equipment: 4
Port: 30000	Port: 30000

2.3.5 Networking TCP Port

- **84-02-35 : H.225, H.245 Information Basic Setup - Fast Start Mode**

  If IPLA is used for networking, the Fast Start option must be enabled.

Item No.	Item	Input Data	Default
35	Fast Start	0: Disable 1: Enable	1

2.4 AspireNet Multi Site

AspireNet Multi Site is a network of three or more systems connected by either AspireNet ISDN or AspireNet IP. The configuration for three or more sites is the same as for two sites, refer to AspireNet ISDN or AspireNet IP for details of setting up the SV9100 systems.

The example below shows three systems, each has a IPLE Blade installed.

The IPLE Blade in each system are connected to an Ethernet hub.

System A has extension numbers in the range 200-299, system B has 300-399 and system C has 400-499.

System A		System B		System C
Ethernet Hub				

System Configuration

System A		System B		System C
IPLA IP Address 172.16.0.10		IPLA IP Address 172.16.0.11		IPLA IP Address 172.16.0.12
Address for Remote System System 01 = 172.16.0.11 System 02 = 172.16.0.12		Address for Remote System System 01 = 172.16.0.10 System 02 = 172.16.0.12		Address for Remote System System 01 = 172.16.0.10 System 02 = 172.16.0.11
System Numbering Plan Dial 2x = Intercom Dial 3x = Networking Dial 4x = Networking		System Numbering Plan Dial 2x = Networking Dial 3x = Intercom Dial 4x = Networking		System Numbering Plan Dial 2x = Intercom Dial 3x = Intercom Dial 4x = Networking
VoIP Gateway IP Address 172.16.0.21		VoIP Gateway IP Address 172.16.0.22		VoIP Gateway IP Address 172.16.0.23

• All systems within the network must have a direct connection to all other systems in the network.This makes AspireNet IP the more practical type for Multi Site networks.

• IPLE Resource will be used at each system when a call is transferred. For example a call that originates from site A and calls site B is held and transferred to site C will use IPLE resources at all three sites.

The same also applies for AspireNet ISDN, an ISDN channel will be used from site A to B and also from site B to C.

2.5 Numbering Plan

- **11-01-01 : System Numbering**

Set the system's internal (Intercom) numbering plan and system ID to route to networked systems. The numbering plan assigns the first and second digits dialed and affects the digits an extension user must dial to access other extensions and features, such as service codes and trunk codes, within a networking node or to reach another node.

Consider using a "Unified Numbering Plan" for extensions. This gives every extension in the network a unique extension number. The extension number can then be used to route a call to the correct node. This also allows the same extension number to be dialed at any node to reach a given extension.

CAUTION

Improperly programming this option can adversely affect system operation. Make sure you thoroughly understand the default numbering plan before proceeding. If you must change the standard numbering, use the chart for ***Table 13-5 System Numbering Default Settings on page 13-83*** ***to keep careful and accurate records of your changes.***

Before changing your numbering plan, use PC Pro to make a backup copy of your system data.

Changing the numbering plan consists of three steps:

1. Enter the digits you want to change.
2. Specify the length of the code you select to change.
3. Assign a function to the code selected.

Step 1: Enter the digit(s) you want to change

You can make either single or two digit entries. In the dialed Number column in the table, the nX rows (e.g., 1X) are for single digit codes. The remaining rows (e.g., 11, 12, etc.) are for two digit codes.

--Entering a single digit affects all the dialed Number entries beginning with that digit. For example, entering 6 affects all number plan entries beginning with 6. The entries you make in step 2 and step 3 below affect the entire range of numbers beginning with 6. (For example, if you enter 3 in step 2 the entries affected would be 600-699. If you enter 4 in step 2 below, the entries affected would be 6000-6999.)

--Entering two digits lets you define codes based on the first two digits a user dials. For example, entering 60 allows you to define the function of all codes beginning with 60. In the default program, only * and # use two-digit codes. All the other codes are single digit. If you enter a two digit code between 0 and 9, be sure to make separate entries for all the other two digit codes within the range as well. This is because in the default program all the two digit codes between 0 and 9 are undefined.

Step 2: Specify the length of the code you want to change

After you specify a single or two digit code, you must tell the system how many digits comprise the code. This is the ***Number of Digits Required*** column in the table. In the default program, all codes from 100-999 are three digits long. Codes beginning with 0 are one digit long. Codes beginning with * are 3 digits long and codes beginning with # are 4 digits long.

Step 3: Assign a function to the code selected

After entering a code and specifying its length, you must assign its function. This is the Dial Type column in the table. The choices are:

Dial Types	Dial Type Description	Related Program
0	- Not Used -	
1	Service Code	11-10: Service Code Setup (for System Administrator) 11-11: Service Code Setup (for Registration) 11-12: Service Code Setup (for Service Access) 11-13: Service Code Setup (for ACD) 11-14: Service Code Setup (for HOTEL) 11-15: Service Code Setup (Special access)
2	Extension Number	11-02: Extension Numbers 11-04: Virtual Extension Numbers 11-06: 2PGDAD (ACI) Extension Numbers 11-07: Department Calling Group Numbers 11-08: 2PGDAD (ACI) Group Pilot Numbers
3	Trunk Access Code	11-09: Trunk Access Code.
4	Special Trunk Access	11-09: Trunk Access Code.
5	Operator Access	20-17: Operator's Extension
6	ARS/F-Route Access	44-xx
8	Networking	10-03: PCB Setup 10-12: GCD-CP10 Network Setup 10-20: LAN Setup for External Equipment 10-27: IP System ID

--Changing the ***Dial Type*** for a range of codes can have a dramatic affect on how your system operates. Assume, for example, the site is a hotel that has room numbers from 100-399. In order to make extension numbers correspond to room numbers, you should:

- In Program 11-02, reassign extension numbers on each floor from 100 to 399.

(Other applications might also require you to change entries in Program 11-10 through 11-16.)

Example:

This example shows two separate extension numbers assigned for the networked systems. System A dials 4xx to reach System B, while system B dials 3xx to reach System A.

System – A	System – B
Dial "3x": Digit "3" Type 2 (Intercom)	Dial "3x": Digit "3" Type 8 (Networking) System ID "1"
Dial "4x": Digit "3" Type 8 (Networking) System ID "1"	Dial "4x": Digit "3" Type 2 (Intercom)

The following example shows a unified extension number assignment. All users dial a 4-digit extension number (2xxx) to reach anyone within the network, regardless of which system they are connected. System A users have extension numbers 20xx, while system B users have extension numbers 23xx.

Programming	System – A	System – B
Program 11-01	Dial "2": 2x = Digit "0", Type "0" 20 = Digit "4", Type 2 (Intercom) 23 = Digit "4", Type 8 (Network), System ID "1"	Dial "2": 2x = Digit "0", Type "0" 20 = Digit "4", Type 8 (Network), System ID "1" 23 = Digit "4", Type 2 (Intercom)
Program 11-02	Port 1 = extension number 2001 Port 2 = extension number 2002 Port 3 = extension number 2003, etc.	Port 1 = extension number 2301 Port 2 = extension number 2302 Port 3 = extension number 2303, etc.

It is also possible to use F-Route to select the correct node for the destination extension number.

The example below shows a numbering scheme where the user must dial an additional digit 7 before the extension number, this is routed by F-Route to the correct node and analyzed again in the F-Route tables at the remote SV9100.

When using F-Route you must translate the dialed number (e.g 72301 translates to 2301) otherwise the call will not 'exit' from the F-Route tables.

Programming	System – A	System – B
Program 11-01	Dial "7": 7x = Digit "5", Type 6 (F-Route)	Dial "7": 7x = Digit "5", Type 6 (F-Route)
Program 11-02	Port 1 = extension number 2001 Port 2 = extension number 2002 Port 3 = extension number 2003, etc.	Port 1 = extension number 2301 Port 2 = extension number 2302 Port 3 = extension number 2303, etc.

Program 44-01-01	0 (Not Used)		0 (Not Used)	
Program 44-02	Table 1	Table 2	Table 1	Table 2
Program 44-02-01	720@@	723@@	720@@	723@@
Program 44-02-02	2	2	2	2
Program 44-02-03	1	2	1	2
Program 44-05	Table 1	Table 2	Table 1	Table 2
Program 44-05-01	255	101	101	255
Program 44-05-02	1	0	0	1

- **Program 11-02-01 : Extension Numbering**

  Assign the extension numbers to the ports. The extension number can be up to eight digits long. The first/second digit(s) of the number should be assigned in Program 11-01. This lets an employee move to a new location (port) and retain the same extension number.

- **44-01-01 : System Options for ARS/F-Route - ARS/F-Route Time Schedule**

  Set this option to ‘0’ so that the F-Route table selected is determined only by the digits dialed without any relation to the day or time of the call.

- **44-02-01 : Dial Analysis Table for ARS/F-Route Access - Dial**

  Set the number of digits to be analyzed by the system for ARS routing. Using the 4-digit extension number example in Program 11-01-01, the entry would be: Analysis Table 1: 720@@; Analysis Table 2: 723@@.

- **44-02-02 : Dial Analysis Table for ARS/F-Route Access - Service Type**

  Select the Service Type (0=no setting, 1=extension call, 2=ARS/F-Route table, 3=Dial extension analyze table). Using the 4-digit extension number example in Program 11-01-01, the entry would be: Analysis Table 1: 2 (ARS/F-Route table); Analysis Table 2: 2 (ARS/F-Route table).

- **44-02-03 : Dial Analysis Table for ARS/F-Route Access - Additional Data**

  Enter the additional data required for the service type selected in Program 44-02-02, either the number of digits to be deleted or the table number to be used. Using the 4-digit extension number example in Program 11-01-01, the entry would be: 1 (delete 1 digit).

- **44-02-04 : Dial Analysis Table for ARS/F-Route Access - Dial Tone Simulation**

  If enabled (1), this option sends dial tone to the calling party once the routing is determined. This may be required if the central office at the destination does not send dial tone. Using the 4-digit extension number example in Program 11-01-01, the entry would be: 0 (disabled).

- **44-05-01 : ARS/F-Route Table - Trunk Group Number**

  Select the trunk group number to be used for the outgoing ARS call (0-100, 101-150, 255 [0 = No setting, 101-150 = Networking, 255 = Extension Call]). Using the 4-digit extension number example in Program 11-01-01, the entry would be: ARS/F-Route Table Number 1: Priority Number 1, Trunk Group = 255; ARS/F-Route Table Number 2: Priority Number 1, Trunk Group = 101.

- **44-05-02 : ARS/F-Route Table - Delete Digits**

  Enter the number of digits to be deleted from the dialed number (0-255 [255 = Delete all]). Using the 4-digit extension number example in Program 11-01-01, the entry would be: ARS/F-Route Table Number 1: Priority Number 1, Delete Digits = 1; ARS/F-Route Table Number 2: Priority Number 1, Delete Digits = 0.

- **44-05-03 : ARS/F-Route Table - Additional Dial Number Table**

  Enter the table number (defined in Program 44-06) for additional digits to be dialed (0-1000). Using the 4-digit extension number example in Program 11-01-01, the entry would be: ARS/F-Route Table Number 1: Priority Number 1, Additional Dial Number Table = 0; ARS/F-Route Table Number 1: Priority Number 1, Additional Dial Number Table = 0.

SECTION 3 ASPIRENET FEATURES

Refer to the SV9100 Feature & Specifications Manual for complete description and programming information for the following features. The information detailed here applies only to the feature when used in a Networked system

3.1 ARS/F-Route

Digits dialed by a user can be sent to the F-Route tables and specified as an AspireNet number by entering the Networked node ID (Trunk Group 101-150 correspond to Network ID's 1-50) as the target trunk group number, calls will be routed to the target system via the node ID specified. The dialed digits will then be analyzed by the F-Route tables in the target system. At the target system the call will be analyzed within F-Route for the following call types:

- Outgoing call to a trunk
- Extension access call (you must translate the dialed digits)
- Access to the other system via AspireNet
- No defined dial

Alternate route selection is not available when the primary AspireNet route is busy. When all channels are busy the call will return busy tone.

Compared With Single System Configuration

In a single system with F-Route used, the dialing is analyzed when the call is initially dialed.

Operation

With the sample programming shown below, dialing the F-Route access number, which is defined in the F-Route table (7300), the system calls extension number 300 within System 1 (System ID 1). The telephone's display initially indicates the F-Route number in progress, then changes to appear as a normal intercom call.

(Sample Programming for calls from System 1 to System 2, to simplify the example the programming for calls from System 2 to System 1 has been omitted)

Kind of setting	System 1	System 2
PRG11-01-01 System Numbering	Dial "7" Digit "4" (4 digit) Type "6" (F-Route)	Dial "7" Digit "4" (4 digit) Type "6" (F-Route)

PRG44-02- Dial analysis table for F-Route access	01	"7300" "73@@" (using wildcard) is available.	"7300" "73@@" (using wildcard) is available.
	02	"2" (F-Routing)	"2" (F-Routing)
	03	Add Code = "1"	Add Code = "1"
PRG44-05- System Numbering	01	F-Route table-1 Trunk Group No. "101" (Route to System ID 1)	F-Route table-1 Trunk Group "255" (Route to Intercom)
	02	Delete Digit: 0	Delete Digit: 1

Notice: *In the example above Dial 7 is set to 4 digit number length, this is important for AspireNet as it is possible to* *set call forward to an F-Route number. The full digit length must be set in Program 11-01 to allow the user to enter the full F-Route destination number.*

For example if 11-01 was set to 1 digit number length in the example above a user who sets call forward can only enter 1 digit as the destination (e.g 741+1+7). This would cause all calls to the extension to fail whilst the call forward is set to an invalid destination number.

Notice : *When a call is routed via AspireNet by using F-Route the dialed digits MUST be translated otherwise the call will not 'exit' from the F-Route tables.*

Notice : *When an AspireNet call is routed via F-Route it is possible to translate the dialed digits at either (or both) the originating or destination system. Programs 44-05-02 (Delete digits) and 44-05-03 (Additional Dial Table) are available for digit translation.*

Related Programs

Program Number	Title
11-01	Numbering Plan
44-01	System Options for F-Route Service
44-02	Dial Analysis Table for F-Route Access
44-04	F-Route Selection for time schedule
44-05	F-Route Table
44-06	Additional Dial Table
44-07	Gain Table for F-Route Access
44-08	Time Schedule for F-Route Service
44-09	Weekly Schedule for F-Route Service
44-10	Holiday Schedule for F-Route Service

3.2 Barge In

Barge In is available in the Networking feature with the following options:

- Barge into a conversation between an extension's own system and a networked system
- Barge into a conversation between callers in a networked system
- Barge into a call between two networked systems

Barge In can be used in either Monitor Mode (Silent Monitor) or Speech Mode (determined in Program 20-13-10).

Barge In ***cannot*** barge into calls across the network in the following instances:

- Conference calls
- Off hook signaling a telephone in the other system
- Barge into an extension's call without first calling the busy extension in the other system

Operation

To Barge In after calling a busy extension:

An analog trunk call must be set up for 10 seconds before you can Barge In.
Listen for busy/ring or busy tone.

1. Call busy extension.
2. Dial Barge In service code 710

OR

3. Press Barge In key (PGM 15-07 or SC 751: 34).

Related Programs

Program Number	Title
11-12-08	Service Code Setup (for Service Access) - Barge In
11-16-02	One Digit Service Code Setup - Barge In
20-13-15	Class of Service Options (Supplementary Service) - Barge In, Initiate
20-13-16	Class of Service Options (Supplementary Service) - Barge In, Receive
20-13-17	Class of Service Options (Supplementary Service) - Barge In Tone/ Display

3.3 Call Forwarding

Call Forwarding Immediate / Busy / No Answer / Busy-No Answer / Both Ring options are available with the Networking feature.

With a networked system, when Call Forwarding enabled, there is a slight difference in the telephone's display. With a single system, the extension name is displayed on the extension which has Call Forwarding. With a networked system, the extension number is displayed.

Operation

To activate or cancel Call Forwarding:

1. Press idle SPK key (or lift handset).
2. Dial Call Forwarding condition :
   745 = Call Forward Both Ring
   742 = Call Forward Busy
   744 = Call Forward Busy or not answered
   743 = Call Forward No answer
   741 = Immediate
3. Dial 1 to set Call Forwarding, Dial 0 to cancel Call Forwarding.
4. Dial destination extension, Voice Mail Master Number.
   When you enable Call Forwarding, your Call Forwarding key lights.
   Your Call Forwarding (Station) Programmable Function Key flashes when Call Forwarding is activated.

OR

1. Press Call Forwarding key.
   PGM 15-07 or SC 751: code 10 for Forward All Calls Immediately
   PGM 15-07 or SC 751: code 11 for Forward when Busy
   PGM 15-07 or SC 751: code 12 for Forward when Unanswered
   PGM 15-07 or SC 751: code 13 for Forward Busy/No Answer
   PGM 15-07 or SC 751: code 14 for Forward with Both Ringing
   PGM 15-07 or SC 751: code 15 for Follow Me
2. Dial 1 plus extension to enable; dial 0 to disable.
   Once you activate Call Forwarding, only your Call Forwarding destination can place an Intercom call to you.
3. Dial destination extension, Voice Mail master number.
   You'll hear stutter dial tone when placing a new call.
   Your Call Forwarding Programmable Function Key flashes when Call Forwarding is activated.

Related Programs

Program Number	Title
11-11-01	Service Code: Call Forward – Immediate
11-11-02	Service Code: Call Forward – Busy
11-11-03	Service Code: Call Forward – No Answer
11-11-04	Service Code: Call Forward – Busy/No Answer
11-11-05	Service Code: Call Forward – Both Ring
11-11-06	Service Code: Call Forwarding (Select Option)
20-11-01	Call Forward – Immediate Class of Service
20-11-02	Call Forward – Busy Class of Service
20-11-03	Call Forward – No Answer Class of Service
20-11-04	Call Forward – Both Ring Class of Service

3.4 Call Forwarding/ Do Not Disturb Override

The extension user may be able to call an extension which has Call Forwarding or Do Not Disturb set.

Operation

To override an extension's Call Forwarding or Do Not Disturb:

1. Call the forwarded or DND extension.
2. Dial Call Forward / DND Override service code 707
   OR
2. Press Override key (PGM 15-07 or SC 751: 37).

Related Programs

Program Number	Title
11-12-01	Service Code Setup (for Service Access) : Call Forwarding / Do Not Disturb Ovveride
11-16-06	Single Digit Service Code Setup : DND/Call Forward Override
15-07-01	Programming Function Keys
20-06-01	Class of Service for Extensions
20-13-04	Class of Service Options (Supplementary Service) - Call Forwarding/ DND Overrride

3.5 Call Forward/ Off-Premise

Off-Premise Call Forwarding allows an extension user to forward their calls to an off-site location. This fea-

ture works the same in a networked system as it does in a single system.
A call to an extension at a remote system will forward to the Abbreviated Dial bin using a trunk at the remote system.

Operation

To activate Call Forwarding Off-Premise

1. At system phone, press SPK key + Dial Call Forward Service Code (741, 742, 744, 743).
   OR
   Press Call Forward key (PGM 15-07 or SC 751: 10, 11, 13 or 12)
   OR
   At an SLT, lift handset Dial 741, 742, 744 or 743.
2. Dial 1.
3. Dial the Abbreviated Dial Bin number (000 to 999) which your calls should be for warded.
4. Press SPK (or hang up at SLT) to hang up.

Your DND and Call Forwarding Programmable Function Key flashes.

To cancel Call Forwarding Off-Premise

1. At system phone, press SPK key + Dial 741, 742, 744 or 743.
   OR
   Press Call Forward key (PGM 15-07 or SC 751: 10, 11, 13 or 12)
   OR
   At an SLT, lift handset and dial 741, 742, 744 or 743.
2. Dial 0.
3. Press SPK (or hang up at SLT) to hang up.

Your DND or Call Forwarding Programmable Function Key stops flashing.

Related Programs

Program Number	Title
13-01	Abbreviated Dial Function Setup
14-05	Trunk Group
14-06	Trunk Group Routing
20-11-12	Enable call forward to Abbreviated Dial bin
21-02	Trunk Group Routing for Extensions
21-16	Trunk Group Routing for Networking

3.6 Call Forwarding with Follow Me

The extension user can program Call Forward Follow-Me to extension in a networked system. When the extension with the Follow Me setting receives an incoming call, both the original extension and the programmed destination extension starts ringing.

With a networked system, when Call Forward Follow-Me is enabled, there is a slight difference in the telephone's display. With a single system, the destination extension displays the extension name for the phone with Follow-Me enabled. With a networked system, the extension number is displayed.

Operation

To activate Call Forward Follow Me:

1. At a system phone other than your own, press idle SPK key and dial 846.
   OR
   Press Call Forward (Station) key (PGM 15-07 or SC 751: 15).
   OR
   At an SLT other than your own, lift handset and dial 746.
2. Dial 1 to set.
3. Dial the Extension to forward.
4. SPK (or hang up at SLT).

Your Call Forwarding (Station) Programmable Function Key flashes when Call Forwarding is activated.

To cancel Call Forward Follow Me:

1. At system phone, press SPK key.
   OR
   Press Call Forward (Station) key (PGM 15-07 or SC 751: 15).
   OR
   At an SLT, lift handset and dial 846.
2. Dial 0 to cancel.
3. Dial 0 (Cancel All Forward Follow Me).

Your Call Forwarding (Station) Programmable Function Key goes out.

Related Programs

Program Number	Title
11-11-07	Service Code of Follow Me
11-11-06	Call Forwarding (Select Option)

3.7 Camp On

With Camp On, an extension user may call a busy extension and wait in line (Camp-On) without hanging up. The call goes through when the busy extension becomes free. Camp On helps extension users to get through as soon as a busy extension becomes free. It also lets callers wait in line for a busy extension without being forgotten.
When you have set Camp-On you can choose to wait off hook or go on hook. If you go on hook your phone will ring when the extension becomes free.

With a networked system, camping on to an idle extension and Trunk Queuing/Camp On for a trunk port are not supported.

With a networked system, when Camp On is enabled, there is a slight difference in the telephone's display. With a single system, the target extension's name is displayed on the phone which has enabled Call Waiting. With a networked system, the extension number is displayed.

Operation

To Camp-On to a busy extension:

1. Call busy extension.
2. Dial Camp-On service code 750
   OR
   Press Camp-On key (PGM 15-07 or SC 751: 35).
3. You can choose to hang up or not.

To cancel a Camp-On request:

1. Hang up or SPK key.
2. At system phone, press SPK key and dial 770.
   OR
   At system phone, press Camp-On key (PGM 15-07 or SC 751: 35).
   OR
   At single line set, lift handset and dial 770.

Related Programs

Program Number	Title
11-12-05	Service Code: Setting Camp On
11-12-06	Service Code: Cancelling Camp On
11-16-05	One-Digit Service Code: Camp On

3.8 Caller ID Display

Caller ID information can be sent to the target extension in a networked system and show the Caller ID on the phone's display. The ABB Name is also shown on LCD by searching ABB table at the target system.

Operation

A DDI call routed directly to a remote extension will send the Caller ID information to the remote system. The DDI name set in Program 22-11-03 is also sent to the remote system.
At the remote system the Caller ID information will be displayed or, if the Caller ID number matches an Abbreviated Dial entry the Abbreviated Dial name will be displayed.

A trunk call that is first answered and then transferred to a remote extension will display the Caller ID number and the DDI name. The Abbreviated Dial name will NOT be displayed.

Related Programs

- **20-09-02 : Class of Service Options (Incoming Call Service) - Caller ID Display**

Define the option whether display Caller ID or not in each station port.

Item No.	Item	Input data	Default	
			COS 01-14	COS 15
02	**Caller ID Display** Enables/disables the Caller ID display at an extension.	0-Off 1-On	1	1

- **20-09-03 : Class of Service Options (Incoming Call Service) - Sub-Address Identification**

Define whether an extension displays the Caller Sub-Address.

Item No.	Item	Input data	Default	
			COS 01-14	COS 15
03	**Sub Address Identification** Define whether an extension displays the Caller Sub-Address.	0-Off 1-On	0	0

- **13-04 : Abbreviated Dialing Number and Name**

Define the abbreviated dialing number and name. The ABB name will be shown on a phone's LCD when the ABB table has a matching number with the incoming Caller ID. The common abbreviated number table is used to search for a match.

Item No.	Item	Input Data	Default
01	Abbreviated Dialing Data	1-9, 0, *, #, **P**ause (Press line key 1), **R**ecall/Flash (Press line key 2), **@** for Additional Digit for ISDN Functionality (Press line key 3) (max. 24 digits)	No Setting
02	Name	Max. 12 Characters	No Setting

3.9 Central Office Calls, Placing: Seizing a trunk in a networked system

The system allows a user to seize a trunk in a networked system using the following methods:

Method of Outgoing	Available	Note
Specified Trunk Access (#0 + the trunk number)	No	
Specified Trunk Group Access (704 + group number)	No	
Trunk Route Access (0)	Yes	
ARS/F-Route	Yes	See ARS/F-Route

Operation

The operation is automatic, the user dials the trunk access code in the normal way.
Abbreviated Dial numbers will follow the trunk routing if set to TRG 0 in Program 13-05.
For IP AspireNet ensure that the VOIPU 'trunk' ports are in their own trunk group in Program 14-05, do not create a trunk group with a mix of VOIP trunk ports and any other trunk port type. VOIP trunk ports should not be seized directly via line keys or trunk access (SC 0, #0 or 704).

Related Programs

The following example indicates the setting required to seize the trunk in a networked system (Extension in System A tries to make an external call using a trunk in System B).

Program	System – A	System – B
Program 14-06-01 Trunk Group Routing	Route 1 101 (Route to System ID 1)	Route 1 1 (Route to Trunk Group 1)
Program 21-02-01 Trunk Group Routing for Extensions	Extension 301 (which make a call via networking) 1 (Route 1) This setting is referenced in Program 14-06-01	
Program 21-16-01 Trunk Group Routing for Networking		System ID 1 Route 1 This setting is referenced in Program 14-06-01

3.10 Conference

The user can create a Conference call to include a user in a networked system.

Operation

To establish a Conference:

<u>Keyset</u>

1. Establish Intercom or trunk call.
2. Press Conf or Conference key (PGM 15-07 or SC 751: 07) or press HOLD and dial #1.
3. Dial extension you want to add.
   OR
   Access outside call
   OR
   Retrieve call from Park orbit.

To get the outside call, you can either press a line key or dial a trunk/trunk group code.
You can optionally go back to step 2 to add more parties to your Conference.

4. When called party answers, press CONF or Conference key twice or HOLD key twice.

If you cannot add additional parties to your Conference, you have exceeded the system's Conference limit.

5. Repeat steps 2 - 4 to add more parties.

<u>Single Line Set / 2-Button Telephone</u>

1. Establish Intercom or trunk call.
2. ***<u>Single Line Set</u>***
   Hookflash and dial #1.
   <u>2-Button Telephone</u>
   Press HOLD and dial #1
3. Dial extension you want to add.
   OR
   Access trunk call.
   OR
   Retrieve call from Park orbit.
4. ***<u>Single Line Set</u>***
   Hookflash and repeat step 3 to add more parties.
   OR
   Hookflash twice to set up the Conference.
   <u>2-Button Telephone</u>
   Press HOLD and repeat step 3 to add more parties.
   OR
   Press HOLD twice to set up the Conference.

If you cannot add additional parties to your Conference, you have exceeded the system's Conference limit.

Related Programs

Program Number	Title
15-02-24	CONF key operation mode (for system phones)
15-07-01	Programming Function Keys - Conference (code 07)
20-06-01	Calls of Service for Extensions
20-13-08	Class of Service Options (Supplementary Service) - Conference

3.11 Department Calling

Department Group access is available via Networking. When the extension at System A tries to make a Department Call to System B, System A should have a numbering plan which defines the Department Access code at System-B (must be defined as dial type 8, Networking in Program 11-01: System Numbering).

The following Department Calling options are supported with the Networking feature.

Program Number	Mode	Same as Single System
16-01-02	Department Calling Cycle	Yes
16-01-03	Department Routing When Busy	Yes
16-01-04	Hunting Mode	No
16-01-05	All Ring Mode	When a call is transferred to a Department Group with All Ring, there is a difference in operation. In a single system, an extension within the same system can transfer a call to a Department Group and the call will ring an extension within the Department Group once the transferring user hangs up. In a networked system, the transfer will not go through and the call will recall the extension performing the transfer.
16-01-06	STG Withdraw Mode	No
16-01-07	Call Recall Restriction for STG	No
16-01-08	Maximum Queuing Number	No
16-01-10	Enhanced Hunting Mode	No

Operation

1. When dialing the Department access code for the networked system, the call is dialed in the same way as normal.
2. A Department Call from the outgoing system will be routed to an available extension in the Department Group.

Related Programs

Program Number	Title
11-07-01	Department Group Pilot Numbers
16-01	Department Group Basic Data Setup
16-02-01	Department Group Assignment for Extensions
22-07-01	DIL Assignment
22-11-02	DDI

3.12 Department Step Call

After calling a busy Department Calling Group member in a networked system, an extension user can have Department Step Calling Quickly call another member in the group.

Operation

To make a Step Call:

You step through Extension Groups set in Program 16-02.

1. Place call to busy Department Group member.
   OR
   Place call to Department Group pilot number.
2. 751Repeat step 2 to call other Department Group members.

Related Programs

Program Number	Title
11-12-07	Service Code Setup (for Service Access) - Step Call
11-16-01	Single Digit Service Code Setup - Step Call
20-06-01	Class of Service for Extensions
20-08-12	Class of Service Options (Outgoing Call Service) - Department Step Calling

3.13 Direct Inward Dialing (DDI)

An incoming DDI call can be routed to an extension in a networked system.

Operation

For incoming DDI calls, the system refers to Program 22-11: DDI Translation Number Conversion to determine how to route the call. If the extension number is determined to be in the networked system, the call will be routed to the appropriate system node.
It is possible to route to a Department Group Pilot number at a remote system, the group can be set to all ring mode.
It is not possible to route a DDI call to an IRG at the remote system.

The timer value is determined by the system data at the incoming trunk side if the incoming DDI call is transferred to a ring group due to the no-answer timer expiring.

Related Programs

Program Number	Title
22-02	Incoming Service Type
22-09	DDI Basic Data Setup
22-10	DDI Conversion Area Setup
22-11	DDI Conversion Table Data Setup
22-12	DDI Transferred Destination Setup

3.14 Direct Inward Line (DIL)

A Direct Inward Line (DIL) is a trunk that rings an extension, virtual extension or Department Group directly. Since DILs only ring one extension or group (i.e., the DIL destination), employees always know which calls are for them. For example, a company operator can have a Direct Inward Line for International Sales Information. When outside callers dial the DIL's phone number, the call rings the operator on the International Sales line key. The DIL does not ring other extensions.

The outside party can call an extension at a networked system, if the DIL trunk is set to route to the other system.

Operation

1. An outside caller places a call to a DIL trunk.
2. The call will be routed to the networked system if the DIL target is defined as an extension in the networked system in Program 22-07: DIL Assignment.

Related Programs

Program Number	Title
22-02	Incoming Service Type Setup
22-07	DIL Assignment
22-08	Second IRG Setup for Unanswered DIL/IRG

3.15 Direct Inward System Access (DISA)

Networking allows DISA callers to place a call to an extension in a networked system. Some system features can also be accessed from the networked system. The Class of Service is determined by the password entered by the DISA caller. The password table is referred to by the system on the incoming trunk side.

The Networking feature allows the following DISA services as allowed in Program 20-14: Class of Service Options for DISA/E&M.

Program Number	Service Name	Available
20-14-02	Trunk Route Access	Yes
20-14-03	Trunk Group Access	No
20-14-04	Common Abbreviated dialing	No
20-14-05	Operator access	Yes
20-14-06	Internal Paging	No
20-14-07	External Paging	Yes
20-14-08	Specified trunk access	No
20-14-09	Forced trunk disconnect	No
20-14-10	Call Forward setting	No
20-14-11	Break In	No

Operation

To place a DISA call into the system:

1. Dial the telephone number that rings the DISA trunk.
2. Wait for the DISA trunk to automatically answer with a unique dial tone.
3. Dial the 6-digit DISA password (access code).
4. Wait for a second unique dial tone.
5. Dial an extension number.
   OR
   Dial 0 for Trunk Group Routing or ARS.
   OR
   Dial Alternate Trunk Route Access Code (if enabled).
   OR
   Dial #0 + a trunk number (1-400) for an outside call.
   OR
   Dial 9 for the operator.
   OR
   Dial 703 + an External Paging Zone number (1-8 or 0 for All Call).

If the received digits are analysed as a networked extension number, the call is routed to the proper network node.

Related Programs

Program Number	Title
20-14	Class of Service Options for DISA/E&M
22-02	Incoming Service Type Setup
25-01	DDI/DISA Line Basic Data Setup
25-02	DDI/DISA VRS Error Message
25-03	DDI/DISA Transfer Ring Group with Incorrect Dialing
25-04	DDI/DISA Transfer Ring Group with No Answer/Busy
25-05	DDI/DISA Error Message Setup
25-06	DDI/DISA One-Digit Code Attendant Setup
25-07	System Timers for DDI/DISA
25-08	System timer for DDI/DISA Service
25-09	Class of Service for DISA User
25-10	Trunk Group Routing for DISA
25-11	Toll Restriction Class for DISA
25-12	Individual Trunk Group Routing for DISA
25-13	System Option DISA Service

3.16 Hold

This feature is available with no changes in programming or operation.

The MOH tone is sourced at the local system where the caller is hearing the hold tone. For example a user at System A places a call to system B and puts the call on hold, the MOH source at system B will be heard by the held user.

Whilst the caller is on hold the AspireNet speech path will be reserved, waiting for the call to be taken off hold.

3.17 Hotline / Direct Station Selection (DSS)

An extension user can have a Hotline key to a networked extension. The Hotline or DSS console keys can display the status lamp indication of an extension in a networked system.
The key will show the status for idle, busy, DND set and Call Forward Immediate set.

Lamp status may not be updated immediately. Status will be updated in the time interval specified in Program 20-01-04.

The status for ACD extensions or Virtual extensions will not be sent via AspireNet . It is still possible to have a Hotline / DSS key for an ACD or Virtual extension but it will not show any BLF information, the key can only be used to call the extension.

The basic status for a Hotel extension is sent via AspireNet e.g idle, busy, DND set, Call forward set. The Hotel room status is not sent e.g. check in/out, room status etc.

Operation

To Set a Function Key to your Hotline partner:

1. Press SPK to go off hook.
2. Dial service code 751.
3. Press Hotline key + 01 +partner's extension number + HOLD.

To place a call to your Hotline partner:

1. Press Hotline key

You can optionally lift handset after this step for privacy.

To transfer your outside call to your Hotline partner:

1. Press Hotline key.
2. Announce call and press the Transfer button.
   OR
   Press Transfer to have the call wait at your Hotline partner unannounced.

If unanswered, the call recalls like a regular transferred call.

To answer a call from your Hotline partner:

1. If you hear two beeps, speak toward phone.
   OR
2. If your telephone rings, lift handset.

Calling an extension from your 60 button DSS Console:

1. Press DSS Console key.

If the call voice-announces, you can make it ring by dialing 1.
If you don't have Handsfree, you must lift handset to speak.

Related Programs

Program Number	Title
20-01-04	System Options - Network BLF Indication
20-02-03	System Options for Multi-Line Telephones - BLF Control
20-13-06	Class of Service Options (Supplementary Service) - Automatic Off Hook Signaling
20-06-01	Class of Service for Extensions
30-02-01	110 button DSS Console Key Assignment
30-05-01	DSS Console Lamp Table

3.18 Intercom

An extension user can make an intercom call to a networked system if the networked extensions are defined with the Network Access Code (Program 11-01, Dial Type = 8)

A user can change the signaling type for the intercom call they place to either a voice announce or ringing call to extension in a networked system.

Operation

To place an Intercom call:

1. At system phone, press SPK key.
   OR
   At single line telephone, lift handset.
2. Dial extension number (or 0 for your operator).

Your call may voice-announce or ring the called extension. Dial 1 to change the way your call alerts the called extension.
If the extension you call is busy or doesn't answer, you can dial another extension without hanging up.

Related Programs

Program Number	Title
11-12-06	Service code of Voice/Signal Change
11-16-03	One digit service code of Voice/Signal Change
20-06-01	Class of Service for Extensions
20-08-01	Class of Service Options (Outgoing Call Service) - Intercom Calls

3.19 Keep Alive Operation

The Keep Alive operation will check that the distant end is available. A dummy message is sent that the distant end must respond to, if no response is received the line will be taken out of service.
Keep alive operation is effective on ISDN BRI/PRI and IP AspireNet connections.

Operation

The response to the keep alive message is automatic.
The generation of the keep alive message is set by Program 10-31-01, when the timer is set. If the timer is set to 0 the keep alive generation is turned off.
The retry count for a keep alive message that is not responded to is also set at the originating system.
The line will be placed back in service when the line is active and a keep alive message is responded to a keep alive message.
The keep alive operation will only take place if the message is sent and not responded to by the distant end, if the message is not sent (for example if ISDN layer 1 is not active) then the keep alive operation will not take place.

When the keep alive operation occurs the link will be taken out of service:

- Any calls that are in progress will be released.
- Park Hold orbits will be released.
- No further Park Hold information will be sent until the link is active.

Related Programs

Program Number	Title
10-31-01	The interval of the keep alive time. The interval can be set from 0 to 65535 Seconds. When set to 0 the keep alive is disabled.
10-31-02	Retry Count. The system will retry the keep alive message, after the distant end does not respond after this count the line will be taken out of service.

3.20 Last Number Redial

Last Number Redial allows an extension user to quickly redial the last number dialed. When used with a networked system, the system can use the same trunk route on which the call was originally placed, even if the trunk is a trunk in another system.

Operation

To redial your last call:

1. Without lifting the handset, press Redial key.

The last dialed number is displayed.

2. To redial the last number, press #.
   OR
   Search for the desired number from the Redial List by pressing the Redial or *VOL* ▼ *or VOL* ▲ keys.
3. Lift the handset or press SPK to place the call.

The system automatically selects a trunk from the same group as your original call and dials the last number dialed.

OR

1. At system phone, press idle line key (optional).

The system automatically selects a trunk from the same group as your original call.

2. Press Redial.
   OR

1. At system phone, press SPK key.
   OR
   At single line telephone, lift handset.
2. Dial #5.

The system automatically selects a trunk from the same group as your original call and dials the last number dialed.

Related Programs

Program Number	Title
11-12-12	Service Code Setup (for Service Access) - Last Number Dial
15-02-13	Multi-Line Telephone Basic Data Setup - Redial List Mode
20-06-01	Class of Service for Extensions

3.21 Message Waiting

This feature can be used when placing an intercom call to a networked extension and receives either no answer or hears a busy tone.

With a networked system, when a Message Waiting has been left, there is a slight difference in the telephone's display. With a single system, the extension's name which left the message is displayed. With a networked system, the extension number is displayed.

Operation

To leave a Message Waiting:

1. Call busy or unanswered extension.
2. Dial the Message Waiting service code *0
   OR
3. ress Message Waiting key (PGM 15-07 or SC 751: 38)
4. Hang up.

With system phones, the MW LED lights.

To answer a Message Waiting:

When you have a message, your MW LED flashes fast for system phones.

1. At a system phone, press SPK key and dial *0.
   OR
   Press Message Waiting key (PGM 15-07 or SC 751: 38).
   OR
   At single line telephones, lift the handset and dial *0.

If the called extension doesn't answer, dial 0 or press your Message Waiting key to automatically leave them a message.
Normally, your MW LED goes out. If it continues to flash, you have new messages in your "Voice Mail" mailbox or a new "General Message". Go to "To check your messages" below.

Related Programs

Program Number	Title
11-11-09	Service Code Setup (for Setup/Entry Operation) - Answer Message Waiting
11-11-10	Service Code Setup (for Setup/Entry Operation) - Cancel All Messages Waiting
11-11-11	Service Code Setup (for Setup/Entry Operation) - Cancel Message Waiting
11-16-07	Single Digit Service Code Setup: Message Waiting
20-13-07	Class of Service Options (Supplementary Service) - Message Waiting

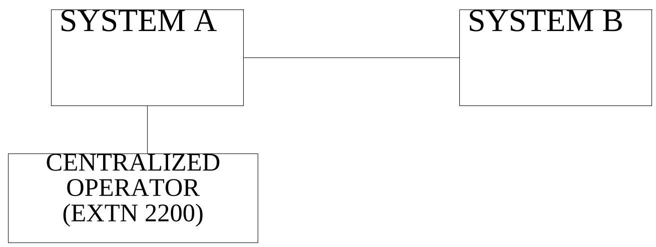
3.22 Operator, Centralized

It is possible to have a centralized network operator extension that can be dialed with the operator access code (0).
Calls to the operator will be queued and answered in order. Up to 32 calls can be queued at the operator extension. The quantity of network calls that can queue at the operator may be limited by the quantity of AspireNet channels available.

Operation

The network numbering plan must be set up to route the operator access code (0) to the system that has the operator extension.
The operator extension must be set in Program 20-17-01.

Centralized Operator



- At System-A extension 2200 must be set as the operator extension in Program 20-17-01.
- At system-B dial 0 must be set as networking in Program 11-01 and routed to the node ID of System-A.
- Users at System A and B can access the operator by dialing 9.

3.23 Paging

An extension user can make internal or external pages to a networked system. Paging to a networked system can only be activated by dialing a service code and the target network's system ID.

Operation

To Make an Internal Page

1. Dial 701.
2. Dial # and the system ID.

The system ID must be dialed as 2 digits (ex: #02).

3. Dial the Paging Zone number (00-64).

Dialing 00 calls All Call Internal Paging.

4. Make announcement to the networked system.
5. Press SPK to hang up.

To Make an External Page

1. Dial 703.
2. Dial # and the system ID.

The system ID must be dialed as 2 digits (ex: #02).

3. Dial the Paging Zone number (0-9).

Dialing 0 calls All Call External Paging.

4. Make announcement to the networked system.
5. Press SPK to hang up.

To Make a Combined Page

1. Dial *1.
2. Dial # and the system ID.

The system ID must be dialed as 2 digits (ex: #02).

3. Dial the Paging Zone number (0-9).

Dialing 0 calls All Call Combined Paging.

4. Make announcement to the networked system.
5. Press SPK to hang up.

Related Programs

Program Number	Title
11-12-19	Service Code Setup (for Service Access) - Internal Group Paging
11-12-20	Service Code Setup (for Service Access) - External Paging
11-12-24	Service Code Setup (for Service Access) - Combined Paging
31-01	System Options for Internal/External Paging
31-02	Internal Paging Group Assignment
31-03	Internal Paging Group Settings
31-04	External Paging Zone Group
31-07	Combined Paging Assignment

3.24 Park

Park places a call in a waiting state (called a Park Orbit) so that an extension user may pick it up. Any extension user who is in the same Park Group as the extension which placed the call in Park can answer the call. This includes extension users in a networked system. For example, when an extension user in Park Group 3 within System A places a call in Park, the extension users in Park Group 3 at any connected system can retrieve the call by pressing the flashing park key or dialing a service code.
If you do not want the park hold orbits to be available to other users within the AspireNet network, then place the extension at each site in a different park hold group in Program 24-03.

With a single system, when two users within the same Park Group try to place a call in the same park orbit at the same time, one user will get the orbit while the other user's call will either ring back or it will remain an active call, depending on how the park orbit was accessed. With Networking, if both users try to access the same orbit, one user will get the orbit, while the other will hear ringback, at which time they can park the call in a different orbit.

Operation

To Park a call in a system orbit:

You can Park Intercom or trunk calls.

1. Press Park key (PGM 15-07 or SC 752: *04 + orbit).

The Park key LED lights.
If you hear busy tone, the orbit is busy. Try another orbit.

2. Use Paging to announce call.
3. Press SPK to hang up.

If not picked up, the call will recall to you.
OR

1. At system phone or 2-Button telephone, press HOLD.
   OR
   At a SLT single line telephone, hookflash.
2. Dial #6 and the Park orbit (01-64).

If you hear busy tone, the orbit is busy. Try another orbit.

3. Use Paging to announce call.
4. Press SPK to hang up.

If not picked up, the call will recall to you.
Note: The parked call recalls after the Park Hold Time (Program 24-01-06). The call rings the extension to which it recalled for the Hold Recall Callback Time (Program 24-01-02). The call then goes on Hold for the Park Hold Time - then recalls again for the Hold Recall Callback Time. The call continues to cycle between Hold and recall until the extension user answers the call or the outside party hangs up.

To pick up a parked call:

1. Lift handset.
2. Press Park key (PGM 15-07 or SC 752: *04 + orbit).
   OR

   At single line telephone, lift handset.
1. Dial *6 and the Park orbit (1-64).

Related Programs

Program Number	Title
11-12-31	Service Code Setup (for Service Access) - Placing a Call in Park
11-12-32	Service Code Setup (for Service Access) - Retrieving Call from Park
20-11-19	Class of Service Options (Hold/Transfer Service) – Normal/Extended Park If enabled, the recall timer set in Program 24-01-07 is used. If this option is disabled, the timer in Program 24-01-06 is used.
24-01-06	System Options for Hold - Park Hold Recall Timer – Normal
24-01-07	System Options for Hold - Park Hold Recall Timer – Extended
24-03-01	Extension Park Group assignment

3.25 Ringdown Extension, Internal/External

A networked system can have a phone defined as a Ringdown Extension to dial either an internal or external number.

Operation

To place a call if your extension has ringdown programmed:

1. Lift handset.

If you want to place a trunk call, press a line key before lifting the handset.
Depending on the setting of your ringdown timer, you may be able to dial an Intercom call before your ringdown goes through.
If the destination has Handsfree Answerback enabled, your call will voice announce. If the destination has Forced Intercom Ringing enabled, your call will ring.

Related Programs

Program Number	Title
20-06-01	Class of Service for Extensions
20-08-09	Class of Service Options (Outgoing Call Service) - Hotline/Extension Ringdown
21-01-09	System Options for Outgoing Calls - Ringdown Extension Timer
21-11-01	Extension Ringdown (Hotline) Assignments

3.26 Selectable Display Messaging

An extension user can select a preprogrammed Selectable Display Message for their extension. This message will be displayed on an incoming intercom caller's LCD when they call the extension in a networked system.

Operation

To select a message:

1. Press SPK key + dial by PRG11-11-14.
   OR
   Press SPK key + press Text Message key (PGM 15-07 or SC 751: 18 + Message Number).
2. Enter Message number (01-20) if needed.
   Use VOL ▼ or VOL ▲ to scroll through the messages.
3. (Optional for messages 1-8 and 10)
   Dial the digits you want to append to the message.
   You can append messages 1-8 and 10 with digits (e.g., the time when you will be back). You enter the time in 24-hour format, but it displays in 12-hour format.
4. Press SPK to hang up.

To cancel a message:

1. Press SPK key + dial 0.
   OR
   Press SPK key + press Text Message key (PGM 15-07 or SC 751: 18)
2. Press SPK to hang up.

Related Programs

Program Number	Title
20-13-19	Class of Service Options (Supplementary Service) - Selectable Display Messaging
20-16	Selectable Display Messages

3.27 Toll Restriction

Toll Restriction limits the numbers an extension user may dial. When accessing a trunk at a remote system the Toll Restriction Class of Service is defined by the calling extension's system, but the Toll Restriction tables will be used from the system which has the outgoing trunk. The Toll restriction class number is sent to the remote system, the remote system will use the class number to define the Toll Restriction tables to use.

Since the restriction table is used for the system which has the outgoing trunk, the definition of the Class of Service may be different, unless all Toll Restriction Classes of Service and Toll Restriction Tables are defined the same between systems.

Operation

Example:
The extension user in System A, which has a Toll Restriction Class 2, dials an outside party after seizing a trunk from a networked system (System B). The received digits are compared to the Class 2 Restriction Table in System B. The call is then either allowed or rejected based on this table.

Related Programs

Program Number	Title
21-04	Toll Restriction Class Assignment for Extensions
21-05	Toll Restriction Class Setup
21-06	Toll Restriction Table Setup

3.28 Transfer

The following types of Transfer are available with Networking:

- Screened Transfer
- Unscreened Transfer
- Transfer to busy extension

Operation

Transferring Trunk Calls

To Transfer a trunk call to a co-worker's extension:

1. At system phone, press HOLD.
   OR
   At a single line telephone, hookflash.
   You hear Transfer dial tone.
2. Dial co-worker's extension number.
   If the extension is busy or doesn't answer, you can dial another extension number or press the line key to return to the call.
   *SLT users can retrieve the call by pressing hookflash. If a call has been transferred and the SLT user has hung up the handset, the call be can retrieved by dialing ** and the extension number to which it had been transferred.*
3. Announce call and hang up.
   If you don't have Automatic On Hook Transfer, you must press Conf or your Transfer Programmable Function Key to Transfer the call.
   If your co-worker doesn't want the call, press the flashing line key to return to the call.
   *SLT users can retrieve the call by pressing hookflash. If a call has been transferred and the SLT user has hung up the handset, the call be can retrieved by dialing ** and the extension number to which it had been transferred.*
   If you don't want to screen the call, hang up without making an announcement.

Transferring to a busy extension

Will require setting in Program number 24-01-01, must be enabled at the originating system and the target system.

1. Dial the co-worker's extension number.
   Busy tone is heard.
2. Press the Transfer button.
   The call will wait for the busy co-worker to become free and then will ring.

Transferring Intercom Calls

To Transfer your Intercom call:

1. At system phone, press HOLD.
   OR
   At single line telephone, hookflash.
2. Dial extension to receive your call.
   If the extension is busy, doesn't answer or does not want the call, you can dial another extension number. In addition, you may be able to transfer the call to the busy extension.
   *SLT users can retrieve the call by pressing hookflash. If a call has been transferred and the SLT user has hung up the handset, the call be can retrieved by dialing ** and the extension number to which it had been transferred.*
3. Announce your call and press Transfer (Program 15-07 or SC 751: 06) or hang up.
   With Automatic On Hook Transfer
   When you hang up, the call is automatically transferred.
   Without Automatic On Hook Transfer
   You must press your Transfer Programmable Function Key to Transfer the call.
   To Transfer the call unscreened, press your Transfer Programmable Function Key and hang up without making an announcement.

Related Programs

Program Number	Title
20-11-06	Class of Service Options (Hold/Transfer Service) - Unscreened Transfer
20-11-07	Class of Service Options (Hold/Transfer Service) - Transfer Without Holding
20-11-08	Class of Service Options (Hold/Transfer Service) - Transfer Display
20-11-18	Class of Service Options (Hold/Transfer Service) - No Recall
24-02-01	System Options for Transfer - Busy Transfer

3.29 Voice Mail, Centralized

Networking will support the use of a single voice mail for the entire network. A user may call into the voice mail from anywhere in the network and perform most functions as if the voice mail were located on their premises.

With a networked system, when voice mail is busy, there is a slight difference in the telephone's display. With a single system, the extension calling a busy voice mail will see WAITING VOICE MAIL on their display. With a networked system, the extension will display CALLING XXX (XXX = extension number).

Operation

Pilot Call
When the extension in System B dials the centralized voice mail access number (Program 45-01-07), then the voice mail in System A is accessed as the centralized voice mail.

Service Codes
When the voice mail service code (Program 11-12-51, SC *8) is dialed, the system calls either the voice mail at the same site as the user, or if the centralized voice mail access number is defined in Program 45-01-07, then the centralized voice mail is called.

One-Digit Service Code
If an extension user hears either a busy signal or a ring back tone when placing an intercom call and dials the one-digit service code (or SC*8) for voice mail, the call is connected to the voice mail at the same site, or if the centralized voice mail access number is defined in Program 45-01-07, then the centralized voice mail is called.

Voice Mail Message Key
If an extension user presses their Voice Mail Message key (Program 15-07 or SC 751: 77), the voice mail at the same site as the user is called, or if the centralized voice mail access number is defined in Program 45-01-07, then the centralized voice mail is called.

Conversation Recording
If an extension user presses their Conversation Recording key (Program 15-07 or SC 751: 78), the voice mail at the same site as the user is called, or if the centralized voice mail access number is defined in Program 45-01-07, then the centralized voice mail is called and the conversation is recorded to that voice mail.

Automatic Attendant
If an extension user presses their Automatic Attendant Message key (Program 15-07 or SC 751: 79), the voice mail at the same site as the user is called, or if the centralized voice mail access number is defined in Program 45-01-07, then the centralized voice mail is called.

Incoming Calls - Normal Trunks
If centralized voice mail is set in Program 22-05:103 as the destination of an incoming call, this call is automatically transferred to the centralized voice mail.
If 102 (in-skin/external voice mail) is selected for the ring group instead of 103, the voice mail within the extension's own system will be called.

No Answer at Incoming Ring
If centralized voice mail is set in Program 22-08:103 as the destination, normal incoming calls or DIL calls which receive no answer are transferred to the centralized voice mail
If 102 (in-skin/external voice mail) is selected for the ring group instead of 103, the voice mail within the extension's own system will be called.

Transferred Destination for Each DDI Translation Table
If centralized voice mail is set in Program 22-11-05: and 22-11-06:103 for the transferred destination for each DDI translation table, then the call will be transferred to the system which has the centralized voice mail.

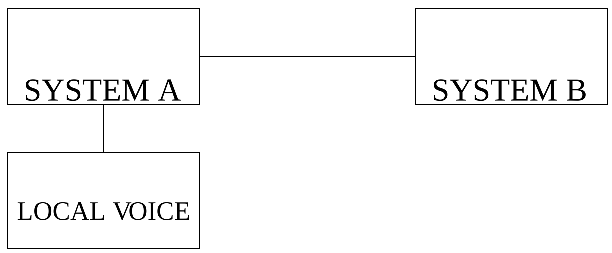
If 102 (in-skin/external voice mail) is selected for the ring group instead of 103, the voice mail within the extension's own system will be called.

DDI, DISA: Mis-Dial Calls
If the centralized voice mail is set in Program 25-03:103 as the transferred destination for DDI/DISA mis-dial calls, then the call will be transferred to the centralized voice mail.
If 102 (in-skin/external voice mail) is selected for the ring group instead of 103, the voice mail within the extension's own system will be called.

DDI, DISA: No Answer and Busy Calls
If the centralized voice mail is set in Program 25-04:103 as the transferred destination for DDI/DISA no answer and mis-dial calls, then the call will be transferred to the centralized voice mail.
If 102 (in-skin/external voice mail) is selected for the ring group instead of 103, the voice mail within the extension's own system will be called.

System Configuration Examples

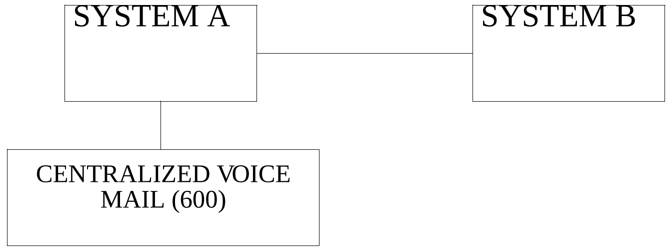
Only Local Voice Mail



System A		System B	
16-02-01	VM ports = Group 64	16-02-01	None
45-01-01	64	45-01-01	None
11-07-01	Group 64 = 600	11-07-01	None
45-01-07	None	45-01-07	None
11-01-01	Dial 6x = Type 2 (Inter-com)	11-01-01	Dial 6x = Type 8 (Net-working)

- The inbound and outbound calls in System-A can access the local voice mail (600), but the inbound and outbound calls in System-B can not reach the local voice mail (600). Access from System-B to the voice mail is available only when a called telephone (at system-A) has Call Forward set to the local voice mail (600).
- Users at System-B can not use any voice mail services (SC *8, voice mail function key 77, conversation record key 78 or auto attendant key 79).
- The voice mail system should not have any mail boxes set for extension numbers at System-B.

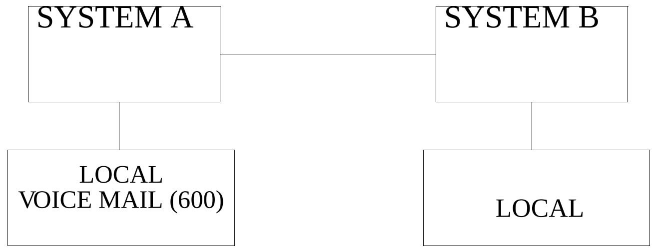
Only Local Voice Mail



System A		System B	
16-02-01	VM ports = Group 64	16-02-01	None
45-01-01	64	45-01-01	None
11-07-01	Group 64 = None	11-07-01	None
45-01-07	600	45-01-07	600
45-01-08	64	45-01-08	None
11-01-01	Dial 6x = Type 2 (Intercom)	11-01-01	Dial 6x = Type 8 (Networking)

- The inbound and outbound calls in Systems A and B can access the centralized voice mail (600).
- Program 11-07-01 must be set to none (for the VM group) in System-A otherwise the operation of centralized voice mail will be disabled.
- Trunk calls must be routed to IRG 103 (Centralized Voice Mail) in System A or B. A trunk routed to IRG 102 (Local Voice Mail) will fail.
- The DDI voice mail tag (@xxx) can be used in Program 22-11-02 at System A and B.
- The voice mail function key (SC 751+77), conversation record key (SC 751+78) and auto attendant key (SC 751+79) can be used at System A and B.

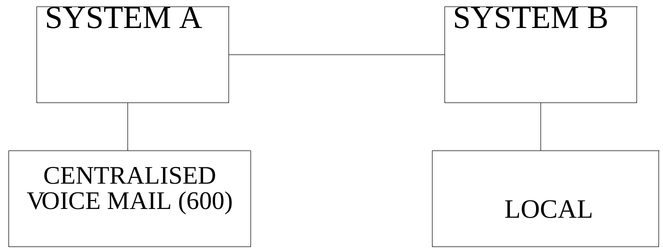
Local Voice Mail at each SV9100 System



System A		System B	
16-02-01	VM ports = Group 64	16-02-01	VM ports = Group 64
45-01-01	64	45-01-01	64
11-07-01	Group 64 = 600	11-07-01	Group 64 = 500
45-01-07	None	45-01-07	None
11-01-01	Dial 5x = Type 8 (Networking) Dial 6x = Type 2 (Intercom)	11-01-01	Dial 5x = Type 2 (Intercom) Dial 6x = Type 8 (Networking)

- The inbound and outbound calls in System-A can access the local voice mail (600),
- The inbound and outbound calls in System-B can access the local voice mail (500).
- Access to the voice mail of the remote system is available only when a called telephone has Call Forward set to their local voice mail.
- Conversation record key (SC 751+78) and auto attendant key (SC751+79) can be used.
- The voice mail function key (SC 751+77) can be used for access to the mail box and message waiting.
- Users at System A or B can not use the voice mail function key (SC 751+77) or SC *8 to access to called party's mail box when placing calls to a remote extension. The voice mail function key should only be used when calling a local extension.
- The voice mail systems should not have any mail boxes set for extension numbers at the remote system.

Local and Centralized Voice Mail



This configuration is not supported, the local voice mail should be removed and only the centralized voice mail used.

Related Programs

Program Number	Title
11-01	System numbering plan
11-07	ICM Pilot Calling Number
11-12-51	Service Code (Call Own Mailbox)
16-02	ICM Group
22-05	Destination for normal incoming ringing
22-08	Destination for no answering
22-11-05	Transferred destination-1 for each DDI translation table
22-11-06	Transferred destination-2 for each DDI translation table
22-12	Transferred destination for each DDI translation table area
25-03	Destination for DDI/DISA: Mis-Dial
25-04	Destination for DDI/DISA: No Answer/Busy Status
45-01	Voice Mail Integration Options

SECTION 4 PROGRAMMING

4.1 Before Reading This Section

This section provides you with detailed information about the system programs. By changing a program, you change the way the feature associated with that program works. In this section, you find out about each program, the features that the program affects and how to enter the program data into system memory.

CAUTION

Do not start customizing your system without first reading Section 1, Setting Up the Networking Feature.

When you want to customize a feature, find it in Section 1 and learn about it. (If you have trouble finding the feature, try cross-referencing it in the Index at the back of this book.) Section 1 will tell you what programs you have to change to get the operation you want. Then, look the program up in this section if you have any questions about how to enter the data.

4.2 How to Use This Section

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 only use the program if your access level meets or exceeds the level the program requires. Refer to 4.3 How to Enter Programming Mode on page 13-55 for a list of the system access levels and passwords.

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

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

1. Enter the programming mode.

2. 15-07-01

15-07-01 TEL
KY01 = *01
← →

tells you to enter the programming mode, dial 150701 from the telephone dial pad. After you do, you will see the message “15-07-01 TEL” on the first line of the telephone display. This indicates the program number (15-07), item number (01), and that the options are being set for the extension . The second row of the display “KY01 = *01” indicates that Key 01 is being programmed with the entry of *01. The third row allows you to move the cursor to the left or right, depending on which arrow is pressed. To learn how to enter the programming mode, refer to 4.3) How To Enter Programming Mode below.

4.3 How to Enter Programming Mode

To enter programming mode:

1. Go to any working display telephone.

✎ *In a newly installed system, use extension 101 (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
------	**------**	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

4.4 How to Exit Programming Mode

To exit the programming mode:

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

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

```
    Program Mode
Base  Service  OP1   OP2
```

2. Press **Speaker**. If changes were to the system programming, "Saving System Data" is displayed.

3. The display shows "Complete Data Save" when completed and exits the telephone to an idle mode.

✎ *To save a customer's database, a blank USB Drive is required. Insert the USB Drive into the GCD-CP10 and, using Program 90-03, save the software to the USB Drive. (Program 90-04 is used to reload the customer data if necessary.) Note that a USB Drive can only hold one customer database. Each database to be saved requires a separate drive.*

4.5 Using Keys to Move Around in the Programs

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

Table 13-2 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, 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.

Table 13-2 Keys for Entering Data (Continued)

Keys for Entering Data	
Use this key...	**When you want to...**
MIC	Switch between the different input data fields by pressing **MIC**. The cursor moves up to the top row of the display. Pressing **MIC** 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 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.
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.

4.6 Programming Names and Text Messages

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

Table 13-3 Keys for Entering Names

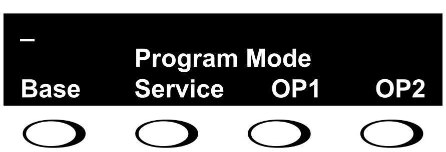
Use this keypad digit . . .	**When you want to. . .**
1	Enter characters: **1 @ [¥] ^ _ ` { | } Æ ¨ Á À Â Ã Ç É Ê ì ó**
2	Enter characters: **A-C, a-c, 2**.

Table 13-3 Keys for Entering Names

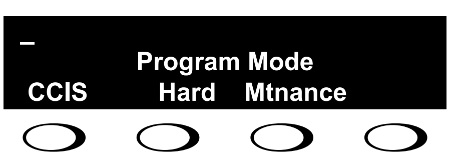
Use this keypad digit . . .	**When you want to. . .**
3	Enter characters: **D-F, d-f, 3**.
4	Enter characters: **G-I, g-i, 4**.
5	Enter characters: **J-L, j-l, 5**.
6	Enter characters: **M-O, m-o, 6**.
7	Enter characters: **P-S, p-s, 7**.
8	Enter characters: **T-V, t-v, 8**.
9	Enter characters: **W-Z, w-z, 9**.
0	Enter characters: 0 ! “ **#** $ % & ’ ( ) ô Õ ú ä ö ü α ε θ
*	Enter characters: * + , - . / : ; < = > ? **π** Σ σ Ω □ ¢ £
#	**#** = 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.)
Recall	Delete the entry to the left.
Hold	Clear all the entries from the point of the flashing cursor and to the right.

4.7 Using Softkeys For Programming

Each UNIVERGE SV9100 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.



Pressing the VOLUME ▲ or VOLUME ▼ will scroll between the menus.



4.8 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 13-4 Softkey Display Prompts

Softkey Display Prompts	
If you press this Softkey . . .	**The system will. . .**
back	Go back one step in the program display. You can press VOLUME ▲ or VOLUME ▼ to scroll forward or backward through a list of programs.
↑	Scroll down through the available programs.
↓	Scroll up through the available programs.
select	Select the currently displayed program.
←	Move the cursor to the left.
→	Move the cursor to the right.
-1	Move back through the available program options.
+1	Move forward through the available program options.

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 reset 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**	0~960	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)-U13 (Paging) 7 =PGD(2)-U13 (Tone Ringer) 8 =PGD(2)-U13 (Door Box) 9 =PGD(2)-U13 (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 (1~960) 2 = SLT Adapter (1~960) 3 = Bluetooth Cordless Handset (BCH) (1~960) 6 = PGD(2)-U13 (Paging) (1~8) 7 = PGD(2)-U13 (for Tone Ringer) (1~8) 8 = PGD(2)-U13 (for Door Box) (1~8) 9 = PGD(2)-U13 (for ACI) (1~96) 10 = DSS (1~32) 11 = --- Not Used ---	0
03	**--- Not Used ---**		
04	**--- Not Used ---**		
05	**--- Not Used ---**		

B-Channel 2			
Item No.	**Item**	**Input Data**	**Default**
06	**Terminal Type (B2)**	0 =Not set 6 =PGD(2)-U13 (Paging) 7 =PGD(2)-U13 (Tone Ringer) 8 =PGD(2)-U13 (Door Box) 9 =PGD(2)-U13 (ACI) 12 =APR (B2 Mode)	0
07	**Logical Port Number (B2)**	0 =Not set 6 =PGD(2)-U13 (Ext. Speaker) 7 =PGD(2)-U13 (Paging/Tone Ringer) = (1~8) 8 =PGD(2)-U13 (for Door Box) = (1~8) 9 =PGD(2)-U13 (ACI) = (1~96) 12 =APR (for B2 mode) (193~512)	0
08	**Multiline Telephone Type**	0 =DT3** 1 =DT4**	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 4 =No Used 5 =BCA	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**	0~960	0
03	**Transmit Gain Level (S-Level)**	1~57 (-15.5 +12.5dB)	32 (0dB)
04	**Receive Gain Level (R-Level)**	1~57 (-15.5 +12.5dB)	32 (0dB)

For COTC Unit Setup

Physical Port Number	1~8

Item No.	Item	Input Data	Default
01	**Logical Port Number**	0~400	0

For ODTB PKG Setup

Physical Port Number	01~04

Item No.	Item	Input Data	Default
01	**Logical Port Number**	0~400	0
02	**2/4 Wire**	0 = 2 Wire 1 = 4 Wire	1
03	**E&M Line Control Method**	0 = TYPE I 1 = TYPE V	1

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 =1~400 (LD Trunk) 1 =1~960 (OPX)	0

For BRIA PKG Setup

ISDN Line Number	01~04

Item No	Item	Input Data	Default
01	**ISDN Line Mode**	0 =No setting 1 =T-Point 2 =S-Point 3 =NW Mode (Leased Line) 4 =NW Mode (Interconnected Line) 5 =NW Mode (Interconnected Line, Fixed Layer1=NT) 6 =S-Point (Leased Line)	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:No setting] = 0 [1:T-Point] = 1-400 [2:S-Point] = 1-960 [3:NW Mode (Leased Line)] = 1-256 [4:NW Mode (Interconnected Line)] = 1-256 [5:NW Mode (Interconnected Line, Fixed Layer1=NT)] = 1-256 [6:S-Point(Leased Line)] = 1-960	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	**Connection Bus Mode**	0 =Extended passive bus 1 =Short passive bus	0
07	**S-point DDI digits**	0 - 4	0
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

Item No	Item	Input Data	Default
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	**Networking System No.**	0 - 50	0
14	**Service Protocol for S-point**	0 =Keypad facility 1 =Specified Protocol for Aspire system	0
15	**Call Busy Mode for S-point**	0 =Alerting 1 =Disconnect	0
17	**ISDN Line Ringback Tone** If Telco does not provide ringback tone, SV9100 can if set to 1:Enable.	0 =Disable 1 =Enable	0
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	0
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	0
22	**QSIG Operation Mode**	0 =Disable 1 =Enable	0
23	**Straight/Cross Wiring**	0 =Auto 1 =Manual (Cross) 2 =Manual (Straight)	0
24	**Power feeding for S-point**	0 =Disable 1 =Enable	0

For PRTA PKG Setup

ISDN Line Number	01~24

Item No.	Item	Input Data	Default
01	**ISDN Line Mode**	0 =No setting 1 =T-Point 2 =S-Point 3 =NW Mode (Leased Line) 4 =NW Mode (Interconnected Line) 5 =NW Mode (Interconnected Line, Fixed Layer1=NT) 6 =S-Point (Leased Line)	1
02	**Logical Port Number** ✎ *The start port number of a PRI line is displayed.*	1 =for T-Bus 1~400	1
03	**CRC Multi-frame(CRC4)** **(Only for E1[30B+D] Mode)**	0 =off 1 =on	1
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	**S-point DDI digits**	0 - 4	0
08	**Dial Sending Mode** ISDN Protocol definition	0 =Enbloc Sending 1 =Overlap Sending	0
09	**Dial Information Element** ISDN Protocol definition Only when Dial Sending Mode (10-03-08) is set for 1 (Overlap Sending).	0 =Keypad Facility 1 =Called Party Number	0
10	**Master/Slave System (Network Mode only)**	0 =Slave System 1 =Master System	0

Item No.	Item	Input Data	Default
11	**Networking System Number (Network Mode only)**	0 - 50	0
12	**--- Not Used ---**		
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	**Service Protocol for S-point**	0 =Keypad facility 1 =Specified Protocol for Aspire system	0
15	**Call Busy Mode for S-point**	0 =Alerting 1 =Disconnect	0
16	**Two B-Channel Transfer for PRI Service**	0 =off 1 =on	0
17	**ISDN Ringback Tone** If Telco does not provide ringback tone, SV9100 can if 10-03-17 is set to 1:Enable.	0 =Disable 1 =Enable	0
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	0
19	**Numbering Plan Identification** ISDN Protocol definition. Select the Numbering Plan used for the ISDN circuit.	0 =Unknown 1 =ISDN numbering plan 2 =Data numbering plan 3 =Telex numbering plan 4 =National standard numbering plan 5 =Private numbering plan	0
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

Item No.	Item	Input Data	Default
21	**Number of Ports**	0 =Auto 1 =4 Ports 2 =8 Ports 3 =12 Ports 4 =16 Ports 5 =20 Ports 6 =24 Ports 7 =28 Ports	0
22	**QSIG Operation Mode**	0 =Disable 1 =Enable	0
23	**Straight/Cross Wiring**	0 = Auto 1 = Manual (Cross) 2 = Manual (Straight)	0

For DTI (T1) PKG Setup

Physical Port Number	01~30

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.	0~400	0
02	**T1 Signal Format Selection**	0 =D4 (12 Multi Frame) 1 =ESF (24 Multi Frame)	0
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 IPLE PKG Setup

Item No.	Item	Input Data
01	**VoIP Type**	IPLE
02	**Number of Channel**	256
03	**Number of Voice Channels**	256

For VM00 PKG Setup

Physical Port Number	01~16

Item No.	Item	Input Data	Default
01	**Logical Port Number**	0~480	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.	0~400	0
02	**T1 Signal Format Selection**	0 =D4 (12 Multi Frame) 1 =ESF (24 Multi Frame)	0
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

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

10-12 : GCD-CP10 Network Setup

Level:
SA

Description

Use **Program 10-12 : GCD-CP10 Network Setup** to setup the IP Address, Subnet-Mask, and Default Gateway addresses.

CAUTION

If any IP Address or NIC setting is 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 GCD-CP10.
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)	12	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 (GCD-CP10)
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	Sets 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	

Input Data (Continued)

Item No.	Item	Input Data	Default	Description
10	**Subnet Mask**	128.0.0.0 192.0.0.0 224.0.0.0 240.0.0.0 248.0.0.0 252.0.0.0 254.0.0.0 255.0.0.0 255.128.0.0 255.192.0.0 255.224.0.0 255.240.0.0 255.248.0.0 255.252.0.0 255.254.0.0 255.255.0.0 255.255.128.0 255.255.192.0 255.255.224.0 255.255.240.0 255.255.248.0 255.255.252.0 255.255.254.0 255.255.255.0 255.255.255.128 255.255.255.192 255.255.255.224 255.255.255.240 255.255.255.248 255.255.255.252 255.255.255.254 255.255.255.255	255.255.0.0	
11	**NIC Setup**	0 = Auto Detect 1 = 100Mbps, Full Duplex 3 = 10Mbps, Full Duplex 5 = 1 Gbps, Full Duplex	0	Set for GPZ-IPLE.
13	**DNS Primary Address**	0.0.0.0 ~ 126.255.255.254 128.0.0.1 ~ 191.255.255.254 192.0.0.1 ~ 223.255.255.254	0.0.0.0	Set for adding a function for DNS.
14	**DNS Secondary Address**			
15	**DNS Port**	0~65535	53	
17	**IPL NIC Port Setting**	0 = MDI 1 = MDI-X	0	

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

10-19 : 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	01~256

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 = Networking(CCIS) 4 = Use for NetLink 5 = Blocked 6 = Common without Unicast Paging 7 = Multicast Paging 8 = Unicast Paging	Resource 1~256 = 0

Conditions

None

Feature Cross Reference

None

Program 10 : System Configuration Setup

10-20 : LAN Setup for External Equipment

Level:
IN

Description

Use **Program 10-20 : LAN Setup for External Equipment** to define the TCP port/address/etc. for communicating to external equipment.

Input Data

Type of External Equipment	1 = CTI Server 2 = ACD MIS 3 = Not Used 4 = Networking System 5 = SMDR Output 6 = DIM Output 7 = Reserved 8 = Reserved 9 = 1st Party CTI 10 = ACD Agent Control 11 = O&M Server 12 = Traffic Report Output 13 = Room Data Output for Hotel Service 14 = IP-DECT Directory Access 15 = Presence

Item No.	Item	Input Data	Default
01	**TCP Port**	0~65535	External Device 1 (CTI Server) = 0 External Device 2 (ACD MIS) = 0 External Device 4 (Networking System) = 0 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
03	**Keep Alive Time**	1~255 (sec)	30

Conditions

None

Feature Cross Reference

None

Program 10 : System Configuration Setup

10-27 : H.323 System Interconnection with Application Setup

Level:
SA

Description

Use Program **10-27 : H.323 Interconnection with Application Setup** to set the IP address of the networked IP systems.

Input Data

System ID	01-50

Input Data

Item No.	Item	Input Data	Default	Default
01	**IP Address** System ID is related with the System ID in the Numbering Plan (Program 11-01-03). When the digits are analyzed and the system ID is determined from the SV9100 data set in the Numbering Plan, the networking call will be sent to the IP Address set in this program. The IP Address should be the IP Address of the peer CPU (Program 10-12-01)	1.0.0.1_126.255.255.254 128.1.0.1 _191.254.255.254 192.0.1.1 _223.255.254.254	0.0.0.0	11-01-01 10-12-01
02	**Call Procedure Port** The Port Number should be set with the same value as the H.225 setup port in Program 84-02-33.	1-65535	1730	84-02-33

Conditions

None

Feature Cross Reference

None

Program 10 : System Configuration Setup

10-31 : Networking Keep Alive Setup

Level:
IN

Description

Use **Program 10-31 : Networking Keep Alive Setup** to set the interval and retry count of the AspireNet networking keep alive message. The keep alive is used for ISDN and IP networking.

The keep alive message is automatically responded to by the destination SV9100, if the response is not received the retry count will start. If a response is not received within the number of retries the networking link will be taken out of service. When the link is taken out of service:

- ❑ Any calls that are in progress will be released.
- ❑ Park Hold orbits will be released.
- ❑ No further Park Hold information will be sent until the link is active.
  The link will automatically become active when the next keep alive response is received.

Input Data

Item No.	Item	Input Data	Default
01	**Keep Alive Interval** This program is used to set the interval of Keep Alive. The SV9100 does not send Keep alive when this item is set to "0". If this entry is greater than "0", networked PRI spans which are using Kentrox DSUs will not re-sync when removed from service then returned to service.	0-65535	0
02	**Keep Alive Retry Timer** Set how many times the SV9100 resends Keep Alive.	1-255	5

Conditions

The keep alive message must be sent and a response not received for the retry count, for the link to be taken out of service and the calls in progress and Park Hold orbits to be released.
For example: If an ISDN Net Link connection is disconnected at Layer 1 then the keep alive message can not be sent, therefore the keep alive operation will not occur.

Feature Cross Reference

- ❑ Networking - AspireNet

Program 10 : System Configuration Setup

10-32 : Networking Maximum PRI Channel Setup

Level:
IN

Description

Use **Program 10-32 : PRI Networking Maximum PRI Channel Setup** to assign the number of B-channels to be used for each ISDN blade. This allows for fractional PRIs when used with multiple site networking.

If this program is limited to less than "30" on one side of the network, then it also limits both inbound and outbound network calls. For example, when you select 10 channels then only channels 1 to 10 will be available. If a call is attempted on channels 11 to 30 the caller will receive busy tone. This also applies on the other side of the network as well.

The setting is for each slot within the SV9100; ensure that you select the correct slot before making any changes.

This program will not affect a PRI card set as Trunk or Station mode.

Input Data

Slot Number	1-24

Item No.	Item	Input Data	Default
01	**Maximum Channels** Set the maximum number of channels which can be used with PRI NetLink.	1 - 30	30

Conditions

None

Feature Cross Reference

❑ Networking - AspireNet

Program 11 : System Numbering

11-01 : System Numbering

Level:
IN

Description

Use **Program 11-01 : System Numbering** to set the system numbering plan. The numbering plan assigns the first and second digits dialed and affects the digits an extension user must dial to access other extensions and features, such as service codes and trunk codes. If the default numbering plan does not meet the site requirements, use this program to tailor the system numbering to the site.

CAUTION

Improperly programming this option can adversely affect system operation. Make sure you thoroughly understand the default numbering plan before proceeding. If you must change the standard numbering, use the chart for Table 13-5 System Numbering Default Settings on page 13-83 to keep careful and accurate records of your changes.

Before changing your numbering plan, use PC Pro to make a backup copy of your system data.

Changing the numbering plan consists of three steps:

Step 1: Enter the digit(s) you want to change

You can make either single or two digit entries. In the Dialed Number column in the Table 1-4 System Numbering Default Settings on page 1-91 table, the nX rows (e.g., 1X) are for single digit codes. The remaining rows (e.g., 11, 12, etc.) are for two digit codes.

- ❑ Entering a single digit affects all the Dialed Number entries beginning with that digit. For example, entering 6 affects all number plan entries beginning with 6. The entries you make in step 2 and step 3 below affect the entire range of numbers beginning with 6. (For example, if you enter 3 in step 2 the entries affected are 600~699. If you enter 4 in step 2 below, the entries affected are 6000~6999.)

❑ Entering two digits lets you define codes based on the first two digits a user dials. For example, entering 60 allows you to define the function of all codes beginning with 60. In the default program, only ✱ and # use 2-digit codes. All the other codes are single digit. If you enter a two digit code between 0 and 9, be sure to make separate entries for all the other two digit codes within the range as well. This is because in the default program all the two digit codes between 0 and 9 are undefined.

✎ *Defining codes based on more than 2 digits require a secondary program (PRG 11-20) to define the codes.*

Step 2: Specify the length of the code you want to change

After you specify a single or two digit code, you must tell the system how many digits comprise the code. This is the ***Number of Digits Required*** column in the Table 1-4 System Numbering Default Settings on page 1-91 table.

Step 3: Assign a function to the code selected

After entering a code and specifying its length, you must assign its function. This is the Dial Type column in the Table 1-4 System Numbering Default Settings on page 1-91 table. The choices are:

Dial Types	Dial Type Description	Related Program
0	**--- Not Used ---**	
1	**Service Code**	11-10 : Service Code Setup (for System Administrator) 11-11 : Service Code Setup (for Setup/Entry Operation) 11-12 : Service Code Setup (for Service Access) 11-13 : Service Code Setup (for ACD) 11-14 : Service Code Setup (for Hotel) 11-15 : Service Code Setup, Administrative (for Special Access) 11-16 : Single Digit Service Code Setup
2	**Extension Number**	11-02 : Extension Numbering 11-04 : Virtual Extension Numbering 11-06 : ACI Extension Numbering 11-07 : Department Group Pilot Numbers 11-08 : ACI Group Pilot Number 11-17 : ACD Group Pilot Number
3	**Trunk Access Code**	11-09-01 : Trunk Access Code
4	**Special Trunk Access**	11-09-02 : Trunk Access Code
5	**Operator Access**	20-17 : Operator Extension
6	**ARS/F-Route Access**	44-xx
8	**Networking System Access**	

Dial Types	Dial Type Description	Related Program
9	**Dial Extension Analyze**	11-20 : Dial Extension Analyze Table

✎ *Changing the Dial Type for a range of codes can have a dramatic affect on how your system operates. Assume, for example, the site is a hotel that has room numbers from 100-399. To make extension numbers correspond to room numbers, you should use Program 11-02 to reassign extension numbers on each floor from 100 to 399. (Other applications might also require you to change entries in Program 11-10 ~ 11-16.)*

Default

See the following tables for default settings.

Table 13-5 System Numbering Default Settings

Dial Types: 1=Service Code, 2=Extension Number, 3=Trunk Access, 4=Special Trunk Access, 5=Operator Access, 6=Flexible Routing, 8 = Networking 9 = Dial Extension Analyze, 0=None					
Dialed	**Number of Digits Required**		**Dial Type**		**Network System ID [if type 8] 0~50**
	Default	**New**	**Default**	**New**	
1X	3		2		
11	0		0		
12	0		0		
13	0		0		
14	0		0		
15	0		0		
16	0		0		
17	0		0		
18	0		0		
19	0		0		
10	0		0		
1✱	0		0		
1#	0		0		
2X	3		2		
21	0		0		
22	0		0		
23	0		0		
24	0		0		
25	0		0		
26	0		0		

Table 13-5 System Numbering Default Settings (Continued)

Dial Types: 1=Service Code, 2=Extension Number, 3=Trunk Access, 4=Special Trunk Access, 5=Operator Access, 6=Flexible Routing, 8 = Networking 9 = Dial Extension Analyze, 0=None					
Dialed	**Number of Digits Required**		**Dial Type**		**Network System ID [if type 8] 0~50**
	Default	**New**	**Default**	**New**	
27	0		0		
28	0		0		
29	0		0		
20	0		0		
2✱	0		0		
2#	0		0		
3X	4		2		
31	0		0		
32	0		0		
33	0		0		
34	0		0		
35	0		0		
36	0		0		
37	0		0		
38	0		0		
39	0		0		
30	0		0		
3✱	0		0		
3#	0		0		
4X	3		1		
41	0		0		
42	0		0		
43	0		0		
44	0		0		
45	0		0		
46	0		0		
47	0		0		
48	0		0		
49	0		0		

Table 13-5 System Numbering Default Settings (Continued)

Dial Types: 1=Service Code, 2=Extension Number, 3=Trunk Access, 4=Special Trunk Access, 5=Operator Access, 6=Flexible Routing, 8 = Networking 9 = Dial Extension Analyze, 0=None					
Dialed	Number of Digits Required		Dial Type		Network System ID [if type 8] 0~50
	Default	New	Default	New	
40	0		0		
4∗	0		0		
4#	0		0		
5X	3		1		
51	0		0		
52	0		0		
53	0		0		
54	0		0		
55	0		0		
56	0		0		
57	0		0		
58	0		0		
59	0		0		
50	0		0		
5∗	0		0		
5#	0		0		
6X	3		1		
61	0		0		
62	0		0		
63	0		0		
64	0		0		
65	0		0		
66	0		0		
67	0		0		
68	0		0		
69	0		0		
60	0		0		
6∗	0		0		
6#	0		0		

Table 13-5 System Numbering Default Settings (Continued)

Dial Types: 1=Service Code, 2=Extension Number, 3=Trunk Access, 4=Special Trunk Access, 5=Operator Access, 6=Flexible Routing, 8 = Networking 9 = Dial Extension Analyze, 0=None

Dialed	Number of Digits Required		Dial Type		Network System ID [if type 8] 0~50
	Default	New	Default	New	
7X	3		1		
71	0		0		
72	0		0		
73	0		0		
74	0		0		
75	0		0		
76	0		0		
77	0		0		
78	0		0		
79	0		0		
70	0		0		
7✱	0		0		
7#	0		0		
8X	1		1		
81	0		0		
82	0		0		
83	0		0		
84	0		0		
85	0		0		
86	0		0		
87	0		0		
88	0		0		
89	0		0		
80	0		0		
8✱	0		0		
8#	0		0		
9X	1		5		
91	0		0		
92	0		0		

Table 13-5 System Numbering Default Settings (Continued)

Dial Types: 1=Service Code, 2=Extension Number, 3=Trunk Access, 4=Special Trunk Access, 5=Operator Access, 6=Flexible Routing, 8 = Networking 9 = Dial Extension Analyze, 0=None					
Dialed	Number of Digits Required		Dial Type		Network System ID [if type 8] 0~50
	Default	New	Default	New	
93	0		0		
94	0		0		
95	0		0		
96	0		0		
97	0		0		
98	0		0		
99	0		0		
90	0		0		
9∗	0		0		
9#	0		0		
0X	1		3		
01	0		0		
02	0		0		
03	0		0		
04	0		0		
05	0		0		
06	0		0		
07	0		0		
08	0		0		
09	0		0		
00	0		0		
0∗	0		0		
0#	0		0		
∗X	42		1		
∗1	0		0		
∗2	0		0		
∗3	0		0		
∗4	0		0		
∗5	0		0		

Table 13-5 System Numbering Default Settings (Continued)

Dial Types: 1=Service Code, 2=Extension Number, 3=Trunk Access, 4=Special Trunk Access, 5=Operator Access, 6=Flexible Routing, 8 = Networking 9 = Dial Extension Analyze, 0=None					
Dialed	Number of Digits Required		Dial Type		Network System ID [if type 8] 0~50
	Default	New	Default	New	
∗6	0		0		
∗7	0		0		
∗8	0		0		
∗9	0		0		
∗0	0		0		
∗∗	0		0		
∗#	0		0		
#X	0		0		
#1	2		1		
#2	2		1		
#3	2		1		
#4	2		1		
#5	2		1		
#6	2		1		
#7	2		1		
#8	2		1		
#9	2		1		
#0	2		1		
#∗	4		1		
##	2		1		

Conditions
None

Feature Cross Reference

❏ Flexible System Numbering

Program 11 : System Numbering

11-02 : Extension Numbering

Level: IN

Description

Use **Program 11-02 : Extension Numbering** to set the extension number. The extension number can have up to eight digits. The first/second digit(s) of the number should be assigned in Program 11-01 or Program 11-20. This allows an employee to move to a new location (port) and retain the same extension number.

Input Data

Extension Port Number	001 ~ 960

Item No.	Extension Number	Description
01	**Dial (Up to 8 digits)**	Set up extension numbers for multiline telephones, single line telephones (including APR), and IP telephones. Extension number assignments cannot be duplicated in Programs 11-02, 11-06, 11-07, 11-08, and 11-17.

Default

Extension Port Number	Extension Number
1	101
2	102
3	103
≀	≀
99	199
100	3101
≀	≀
960	3961

Conditions

None

Feature Cross Reference

- Department Calling
- Flexible System Numbering
- Intercom

Program 11 : System Numbering

11-07 : Department Group Pilot Numbers

Level:
IN

Description

Use **Program 11-07 : Department Group Pilot Numbers** to assign pilot numbers to each Department Group set up in Program 16-02. The pilot number is the number users dial for Department Calling and Department Step Calling. The pilot number can have up to eight digits. The first and second digits of the number should be assigned in Program 11-01 or Program 11-20 as type 2.

Input Data

Department (Extension) Group Number	01~64

Item No.	Extension Group Pilot Number	Description	Related Program
01	**Dial (Up to 8 digits)**	Use this program to assign department group pilot numbers. The number set up by Program 11-02 (Extension Numbering) cannot be used. The extension number cannot be duplicated in Programs 11-02, 11-04, 11-06, 11-08, and 11-17.	❍ 16-01 : Department (Extension) Group Basic Data Setup ❍ 16-02 : Department Group Assignment for Extensions ❍ 16-03 : Secondary Department Group

Default

No Setting

Conditions

None

Feature Cross Reference

- ❑ Department Calling
- ❑ Department Step Calling

Program 11 : System Numbering

11-10 : Service Code Setup (for System Administrator)

Level:
IN

Description

Use **Program 11-10 : Service Code Setup (for System Administrator)** to customize the Service Codes for the System Administrator. You can customize additional Service Codes in Programs 11-11~11-16. The following chart shows:

- The number of each code (01~42).
- The function of the Service Code.
- The type of telephones that can use the Service Code.
- The default entry. For example, dialing (item 26) allows users to force a trunk line to disconnect.

Input Data

Item No.	Item	Terminals	Default	Related Program
01	**Night Mode Switching**	MLT, SLT	718	12-xx 20-07-01
02	**Change of music on hold tone**	MLT	No set	
03	**Setting the System Time**	MLT	728	
04	**Storing Common Speed Dialing Numbers**	MLT	753	
05	**Storing Group Speed Dialing Numbers**	MLT	754	
06	**Setting the Automatic Transfer for Each Trunk Line**	MLT	733	24-04-01
07	**Canceling the Automatic Transfer for Each Trunk Line**	MLT	734	24-04-01
08	**Setting the Destination for Automatic Trunk Transfer**	MLT	735	24-04-01
09	**Charging Cost Display by the Supervisor**	MLT	Not Set	
10	**--- Not Used ---**			
11	**Entry Credit for Toll Restriction**	MLT	Not Set	
12	**Night Mode Switching for Other Group**	MLT	618	12-xx 20-07-01

Input Data (Continued)

Item No.	Item	Terminals	Default	Related Program
13	--- Not Used ---			
14	--- Not Used ---			
15	--- Not Used ---			
16	**Leaving Message Waiting (Requires CPU to be licensed for Hotel/Motel)**	MLT	626	11-11-09
17	**Dial Block by Supervisor**	MLT	601	90-19
18	**Off-Premise Call Forward by Door Box**	MLT	722	13-05
19	--- Not Used ---			
20	**VRS - Record/Erase Message**	MLT, SLT	616	20-07-13
21	**VRS - General Message Playback**	MLT, SLT	611	20-07-14
22	**VRS - Record or Erase General Message**	MLT, SLT	612	20-07-15
23	**SMDR - Extension Accumulated Printout Code**	MLT	621	20-07-18
24	**SMDR - Group Accumulated Printout Code**	MLT	622	20-07-19
25	**Account Code Accumulated Printout Code**	MLT	623	20-07-20
26	**Forced Trunk Disconnect**	MLT, SLT	Not Set	20-07-11
27	**Trunk Port Disable for Outgoing Calls**	MLT, SLT	645	20-07-12
28	--- Not Used ---			
29	--- Not Used ---			
30	**Register DECTPP**	MLT	Not Set	
31	**Delete DECTPP**	MLT	Not Set	
32	**Set Private Call Refuse**	MLT, SLT	Not Set	
33	**Entry Caller ID Refuse**	MLT	Not Set	
34	**Set Caller ID Refuse**	MLT, SLT	Not Set	
35	**Dial-In Mode Switching**	MLT, SLT	Not Set	
36	**Change the Guidance Message Number on Voice Mail Auto Attendant**	MLT, SLT	Not Set	
37	--- Not Used ---			
38	--- Not Used ---			

Input Data (Continued)

Item No.	Item	Terminals	Default	Related Program
39	**--- Not Used ---**			
40	**--- Not Used ---**			
41	**Date Setting**	MLT	Not Set	20-07-30
42	**Maintenance Service**	MLT	Not Set	
43	**--- Not Used ---**			
44	**--- Not Used ---**			
45	**--- Not Used ---**			
46	**Watch Message Setting**	MLT, SLT	614	
47	**Warning Message Setting**	MLT	615	
48	**Auto Dial Setting for Sensor**	MLT	617	
49	**Auto Dial Setting for Remote Watch**	MLT	619	
51	**Power Saving for Power Save Group**	MLT, SLT	731	

➨ *MLT = Multiline Terminal*

➨ *SLT = Single Line Telephone*

❑ Conditions

None

Feature Cross Reference

❑ Refer to Input Data chart on the previous pages.

Program 11 : System Numbering

11-11 : Service Code Setup (for Setup/Entry Operation)

Level:
IN

Description

Use **Program 11-11 : Service Code Setup (for Setup/Entry Operation)** to customize the Service Codes which are used for registration and setup. You can customize additional Service Codes in Programs 11-10, and 11-12 ~ 11-16.

The following chart shows:

- The number of each code (01~65).
- The function of the Service Code.
- What type of telephones can use the Service Code.
- The default entry. For example, dialing 724(Item 19) allows users to turn on or turn off Key Touch Tone.

Input Data

Item No.	Item	Terminals	Default	Related Program
01	**Call Forward – All**	MLT, SLT	741	
02	**Call Forward – Busy**	MLT, SLT	742	
03	**Call Forward – No Answer**	MLT, SLT	743	
04	**Call Forward – Busy/No Answer**	MLT, SLT	744	
05	**Call Forward – Both Ring**	MLT, SLT	745	
06	**--- Not Used ---**			
07	**Call Forwarding – Follow-Me**	MLT, SLT	746	
08	**Do Not Disturb**	MLT, SLT	747	
09	**Answer Message Waiting**	MLT, SLT	∗0	11-10-16
10	**Cancel All Messages Waiting**	MLT, SLT	773	
11	**Cancel Message Waiting**	MLT, SLT	771	
12	**Alarm Clock**	MLT, SLT	727	20-01-06
13	**Display Language Selection for Multiline Terminal**	MLT	678	15-02

Input Data (Continued)

Item No.	Item	Terminals	Default	Related Program
14	**Text Message Setting**	MLT	No Setting	
15	**Enable Handsfree Incoming Intercom Calls**	MLT	721	20-09-05 20-02-12
16	**Force Ringing of Incoming Intercom Calls**	MLT	723	20-09-05 20-02-12
17	**Programmable Function Key Programming (2-Digit Service Codes)**	MLT	751	15-07 11-11-38
18	**BGM On/Off**	MLT	No Setting	
19	**Key Touch Tone On/Off**	MLT	724	
20	**Change Incoming CO and ICM Ring Tones**	MLT	720	15-02
21	**Check Incoming Ring Tones**	MLT	711	
22	**Extension Name Programming**	MLT	700	15-01
23	**Second Call for DID/DISA/DIL**	MLT	679	
24	**Change Station Class of Service** Allow an extension user to change the COS of another extension. Must be allowed in Program 20-13-28.	MLT	677	20-13-28
25	**Automatic Transfer Setup for Each Extension Group**	MLT, SLT	602	20-11-17 24-05
26	**Automatic Transfer Cancellation for Each Extension Group**	MLT, SLT	603	
27	**Destination of Automatic Transfer Each Extension Group**	MLT	604	20-11-17 24-05
28	**Delayed Transfer for Every Extension Group**	MLT, SLT	605	20-11-17 24-05 24-02-08
29	**Delayed Transfer Cancellation for Each Extension Group**	MLT, SLT	606	20-11-17
30	**DND Setup for Each Extension Group**	MLT, SLT	607	
31	**DND Cancellation for Each Extension Group**	MLT, SLT	608	
32	**--- Not Used ---**			
33	**Dial Block**	MLT, SLT	600	
34	**Temporary Toll Restriction Override**	MLT, SLT	775	21-07
35	**Pilot Group Withdrawing**	MLT, SLT	650	
36	**Toll Restriction Override**	MLT, SLT	663	21-14

Input Data (Continued)

Item No.	Item	Terminals	Default	Related Program
37	**Ring Volume Set**	MLT	729	
38	**Programmable Function Key Programming (3-Digit Service Codes)**	MLT	752	15-07 11-11-17
39	**Station Speed Dial Number Entry**	MLT, SLT	755	
40	**--- Not Used ---**			
41	**Tandem Ringing**	MLT, SLT	No Setting	15-07 30-03
42	**--- Not Used ---**			
43	**Headset Mode Switching**	MLT, SLT	688	
44	**Auto Attendant**	MLT, SLT	No Setting	
45	**Set/Cancel Call Forward All (Split)**	MLT, SLT	No Setting	
46	**Set/Cancel Call Forward Busy (Split)**	MLT, SLT	No Setting	
47	**Set/Cancel Call Forward No Answer (Split)**	MLT, SLT	No Setting	
48	**Set/Cancel Call Forward Busy No Answer (Split)**	MLT, SLT	No Setting	
49	**Set/Cancel Call Forward Both Ring (Split)**	MLT, SLT	No Setting	
50	**Set Message Waiting Indication**	SLT	No Setting	15-03-03 45-01-01
51	**Cancel Message Waiting Indication**	SLT	No Setting	15-03-03 45-01-01
52	**Set/Cancel Call Forward All Destination (No Split)**	MLT, SLT	790	
53	**Set/Cancel Call Forward Busy Destination (No Split)**	MLT, SLT	791	
54	**Set/Cancel Call Forward No Answer Destination (No Split)**	MLT, SLT	792	
55	**Call Forward Busy No Answer Destination (No Split)**	MLT, SLT	793	
56	**Telephone Book Lock Service**	MLT	No Setting	
57	**Set Do Not Call Table**	MLT, SLT	No Setting	

Input Data (Continued)

Item No.	Item	Terminal s	Default	Related Program
58	**Call Forward with Personal Greeting**	MLT, SLT	713	
59	**Call Forward to Attendant except Busy**	MLT, SLT	No Setting	15-01-08
60	**Call Forward to Attendant/No Answer**	MLT, SLT	No Setting	15-01-09
62	**Headset Ring Volume Adjustment**	MLT	662	11-11-37 15-02-12 15-02-41 15-02-42
63	**Double Height Character Indication**	MLT	No Setting	15-02-45
64	**Reverse Display Indication**	MLT	No Setting	15-02-44
65	**Headset Mode Switching**	MLT	No Setting	
68	**IntraMail Language Selection for own Extension**	MLT, SLT	No Setting	47-02-16
69	**IntraMail Language Selection for Specific Extension**	MLT, SLT	No Setting	20-13-53 47-02-16

➨ *MLT = Multiline Terminal*

➨ *SLT = Single Line Telephone*

❑ Conditions

None

Feature Cross Reference

❑ Refer to the Input Data chart above.

Program 11 : System Numbering

11-12 : Service Code Setup (for Service Access)

Level:
IN

Description

Use **Program 11-12 : Service Code Setup (for Service Access)** to customize the Service Codes which are used for service access. You can customize additional Service Codes in Programs 11-10, 11-11, and 11-13 through 11-16.

The following chart shows:

- The number of each code (01~59).
- The function of the Service Code.
- The type of telephones that can use the Service Code.
- The default entry. For example, dialing 770 (Item 05) cancels a previously set Camp-On.
- Programs that may be affected with the changing the code.

Input Data

Item No.	Item	Terminals	Default	Related Program
01	**Bypass Call** Activate Call Forwarding/Do Not Disturb Override. This code is available only if you disable the voice mail Single Digit dialing code in Program 11-16-09.	MLT, SLT	707	11-16-09
02	**Conference**	MLT, SLT	#1	
03	**Override (Off-Hook Signaling)**	MLT, SLT	709	
04	**Set Camp-On**	MLT, SLT	750	
05	**Cancel Camp-On**	MLT, SLT	770	
06	**Switching of Voice Call and Signal Call**	MLT, SLT	712	
07	**Step Call**	MLT, SLT	708	
08	**Barge-In**	MLT, SLT	710	
09	**Change to STG (Department Group) All Ring**	MLT, SLT	No Setting	16-02
10	**Station Speed Dialing**	MLT, SLT	#2	
11	**Group Speed Dialing**	MLT, SLT	#4	

Input Data (Continued)

Item No.	Item	Terminals	Default	Related Program
12	**Last Number Dial**	MLT, SLT	#5	
13	**Saved Number Dial**	MLT, SLT	715	
14	**Trunk Group Access**	MLT, SLT	704	
15	**Specified Trunk Access**	MLT, SLT	#9	
16	**Trunk Access Via Networking**	MLT, SLT	No Setting	
17	**Clear Last Number Dialing Data**	MLT, SLT	776	
18	**Clear Saved Number Dialing Data**	MLT, SLT	785	
19	**Internal Group Paging**	MLT, SLT	701	31-01-01
20	**External Paging**	MLT, SLT	703	
21	**Meet-Me Answer to Specified Internal Paging Group**	MLT, SLT	764	31-02-01
22	**Meet-Me Answer to External Paging**	MLT, SLT	765	
23	**Meet-Me Answer in Same Paging Group**	MLT, SLT	763	31-02-01
24	**Combined Paging**	MLT, SLT	∗1	31-02-01 31-07
25	**Direct Call Pickup - Own Group**	MLT, SLT	756	
26	**Call Pickup for Specified Group**	MLT, SLT	768	23-02
27	**Call Pickup**	MLT, SLT	∗ #	23-02
28	**Call Pickup for Another Group**	MLT, SLT	769	23-02
29	**Direct Extension Call Pickup**	MLT, SLT	∗ ∗	
30	**Specified Trunk Answer**	MLT, SLT	672	
31	**Park Hold**	MLT, SLT	#6	24-03
32	**Answer for Park Hold**	MLT, SLT	∗6	24-03
33	**Group Hold**	MLT, SLT	732	
34	**Answer for Group Hold**	MLT, SLT	762	
35	**Station Park Hold**	MLT, SLT	757	
36	**Door Box Access**	MLT, SLT	702	
37	**Common Canceling Service Code**	MLT, SLT	620	
38	**General Purpose Indication**	MLT	783	15-07-56 15-07-57
39	**--- Not Used ---**			

Input Data (Continued)

Item No.	Item	Terminals	Default	Related Program
40	**Station Speed Dialing**	MLT, SLT	#7	
41	**Voice Over**	MLT, SLT	690	11-16-08
42	**Flash on Trunk lines**	SLT	#3	
43	**Answer No-Ring Line (Universal Answer)**	MLT, SLT	#0	14-05 14-06
44	**Callback Test for SLT**	SLT	799	
45	**Enabled On Hook When Holding (SLT)**	SLT	749	15-03-07
46	**Answer On Hook When Holding (SLT)**	SLT	759	15-03-08
47	**Call Waiting Answer/Split Answer** Splitting (switching) between calls	SLT	794	11-12-03
48	**Account Code**	SLT	# #	
49	**--- Not Used ---**			
50	**General Purpose Relay**	MLT, SLT	780	
51	**VM Access (InMail and VMS)**	MLT, SLT	✱8	
52	**Live Monitoring (InMail)**	MLT	No Setting	
53	**Live Recording at SLT**	MLT, SLT	654	
54	**VRS Routing for ANI/DNIS** Use when setting up ANI/DNIS Routing to the VRS Automated Attendant. Using the Transfer feature, this also allows a call to be transferred to the VRS.	MLT, SLT	782	
55	**--- Not Used ---**			
56	**E911 Alarm Shut Off** Enter the Service Code that an extension user can dial to shut off the E911 Alarm Ring.	MLT	786	21-01-13 21-01-14
57	**Tandem Trunking**	MLT, SLT	#8	
58	**Transfer Into Conference** Assign the Service Code a user dials to Transfer a call to a Conference call.	MLT, SLT	624	20-13-10 20-13-15 20-13-16
59	**Trunk Drop Operation for SLT**	SLT	No Setting	
60	**--- Not Used ---**			
61	**--- Not Used ---**			
62	**Security Sensor Rest**	MLT, SLT	716	

Input Data (Continued)

Item No.	Item	Terminals	Default	Related Program
63	**Watch Mode Start**	MLT, SLT	717	
64	**Security Sensor Mode Start**	MLT, SLT	719	

➨ *MLT = Multiline Terminal*

➨ *SLT = Single Line Telephone*

❏ Conditions

None

Feature Cross Reference

Refer to the Input Data chart on the previous pages.

NEC

Program 11 : System Numbering

11-16 : Single Digit Service Code Setup

Level:
IN

Description

Use **Program 11-16 : Single Digit Service Code Setup** to customize the one-digit Service Codes used when a busy or ring back signal is heard. You can customize additional Service Codes in Programs 11-10 ~ 11-15.

The following chart shows:

- The number of each code (01~11).
- The function of the Service Code.
- The default entry. For example, dialing 1 (code 03) when calling an extension switches the call from either a voice or signal call (depending on how it is currently defined).
- Programs that may be affected by changing these codes.

Input Data

Item No.	Item	Default	Related Program
01	**Step Call**	2	11-12-07
02	**Barge-In**	No Setting	11-12-08
03	**Switching of Voice/Signal Call**	1	11-12-06
04	**Intercom Off-Hook Signaling**	✱	11-12-03
05	**Camp-On**	#	11-12-04
06	**DND/Call Forward Override Bypass**	No Setting	11-12-01
07	**Message Waiting**	0	11-12-09
08	**Voice Over**	6	11-12-41
09	**Access to Voice Mail**	8	11-12-51
10	**(Department) STG All Ring Mode**	No Setting	11-12-09 16-01-05
11	**Station Park Hold**	No Setting	11-12-35

❑

Conditions

None

Feature Cross Reference

Refer to the Input Data chart on previous pages.

Program 14 : Trunk, Basic Setup

14-01 : Basic Trunk Data Setup

Level:
IN

Description

Use **Program 14-01 : Basic Trunk Data Setup** to set the basic options for each trunk port. Refer to the chart below for a description of each option, its range and default setting.

Input Data

Trunk Port Number	001~400

Item No.	Item	Input Data	Default	Related Program
01	**Trunk Name** Set the names for trunks. The trunk name displays on a multiline terminal for incoming and outgoing calls.	Up to 12 Characters	Line 001 Line 002 Line 003 : Line 400	
02	**Transmit Level** Use this option to select the CODEC gain for the trunk. The option sets the gain (signal amplification) for the trunk you are programming.	1~63 (-15.5dB~ +15.5dB in 0.5dB intervals)	32 (0dB)	
03	**Receive Level** Use this option to select the CODEC gain for the trunk. The option sets the gain (signal amplification) for the trunk you are programming.	1~63 (-15.5dB ~ +15.5dB in 0.5dB intervals)	32 (0dB)	

Item No.	Item	Input Data	Default	Related Program
04	**Transmit Gain Level for Conference and Transfer Calls** Use this option to select the CODEC gain type used by the trunk when it is part of an Unsupervised Conference.	1~63 (-15.5dB ~ +15.5dB in 0.5dB intervals)	32 (0dB)	
05	**Receive Gain Level for Conference and Transfer Calls** Use this option to select the CODEC gain type used by the trunk when it is part of an Unsupervised Conference.	1~63 (-15.5dB ~ +15.5dB in 0.5dB intervals)	32(0dB)	
06	**SMDR Printout** Use this option to have the system include/exclude the trunk you are programming from the SMDR printout. Refer to Program 35-01 and 35-02 for SMDR printout options.	0 =No Print Out 1 =Prints Out	1	35-01 35-02
07	**Outgoing Calls** Use this option to allow/prevent outgoing calls on the trunk you are programming.	0 =Deny (No) 1 =Allow (Yes)	1	
08	**Toll Restriction** Use this option to enable/disabled Toll Restriction for the trunk. If enabled, the trunk follows Toll Restriction programming (example: Programs 21-05, 21-06). If disabled, the trunk is a toll free line.	0 =Restriction Disabled (No) 1 =Restriction Enabled (Yes)	1	21-04 21-05 21-06
09	**Private Line**	0 =Disable Private Line (Normal) 1 =Enable Private Line (Private Line)	0	

Item No.	Item	Input Data	Default	Related Program
10	**DTMF Tones for Outgoing Calls** Use this option to enable (1) or disable (0) DTMF tones for outgoing trunk calls.	0 =Disable (No) 1 =Enable (Yes)	0	
11	**Account Code Required**	0 =Disable (No) 1 =Enable (Yes)	1	
12	**--- Not Used ---**			
13	**Trunk-to-Trunk Transfer** Use this option to enable (1) or disable (0) loop supervision for the trunk. This option is required for Call Forwarding Off-Premise and Tandem Trunking only.	0 =Disable (No) 1 =Enable (Yes)	1	
14	**Long Conversation Cutoff** Use this option to enable or disable the Long Conversation Cutoff feature for each trunk.	0 =Disable (No) 1 =Enable (Yes)	0	20-21-03 20-21-04
15	**Long Conversation Alarm Before Cutoff** Use this option to enable or disable the Long Conversation Alarm for each trunk.	0 =Disable (No) 1 =Enable (Yes)	0	20-21-01 20-21-02
16	**Forced Release of Held Call** Use this option to enable/disable forced release for calls on Hold. If enabled, the system disconnects a call if it is on Hold longer than a programmed interval (Program 24-01-05). If disabled, forced disconnection does not occur. Program 24-01-01 also affects this option.	0 =Disable (No) 1 =Enable (Yes)	0	24-01-01 24-01-05

Item No.	Item	Input Data	Default	Related Program
17	**Trunk to Trunk Warning Tone for Long Conversation Alarm** Use this option to enable or disable the Warning Tone for Long Conversation feature for DISA callers.	0 =Disable (No) 1 =Enable (Yes)	0	
18	**Warning Beep Tone Signaling**	0 =Disable (No) 1 =Enable (Yes)	0	
19	**Privacy Mode Toggle Option** Use this option to enable or disable a trunk ability to be switched from private to non-private mode by pressing the line key or Privacy Release function key.	0 =Disable (No) 1 =Enable (Yes)	0	
20	**Block Outgoing Caller ID** Allow (1) or prevent (0) the system from automatically blocking outgoing Caller ID information when a user places a call. If allowed (i.e. block, enabled), the system automatically inserts the Caller ID block code (defined in 14-01-21) before the user dialed digits.	0 =Disable (No) 1 =Enable (Yes)	0	14-01-21
21	**Caller ID Block Code** Enter the code, up to 8 digits, that should be used as the Caller ID Block Code. This code is automatically inserted before dialed digits if Program 14-01-20 is set to **1**.	Dial (up to eight digits)	1831	14-01-20

Item No.	Item	Input Data	Default	Related Program
22	**Caller ID to Voice Mail** Enable or disable the system ability to send the Caller ID digits (Remote Log-On Protocol) to voice mail.	0 =Disable (No) 1 =Enable (Yes)	0	
23	**Least Cost Routing**	0 =LCR Off 1 =LCR On 2 =LCR On (Cost Center Code only)	0	
24	**Trunk-to-Trunk Outgoing Caller ID through Mode** Enable (1) or Disable (0) the ability to send the original Caller ID through when the call is Forward Off-Premise.	0 =Disable (No) 1 =Enable (Yes)	0	
25	**Continued/ Discontinued Trunk-to-Trunk Conversation** Enable (1) or Disable (0) the ability to dial a service code code to continue or disconnect the Trunk-to-Trunk conversation after the alert tone is heard.	0 =Disable (No) 1 =Enable (Yes)	0	20-28-01 20-28-02 20-28-03 24-02-07 24-02-10 25-07-07 25-07-08
26	**Automatic Trunk-to-Trunk Transfer Mode**	0 = Normal Transfer (Normal) 1 = Step Transfer (Step)	0	24-02-11 24-02-12
27	**Caller ID Refuse Setup**	0 =Disable (No) 1 =Enable (Yes)	0	
28	**Effectivity of "Conversation Recording Destination for Extension"**	0 = No Effect (No) 1 = Available (Yes)	1	15-12
30	**Flexible Ringing by Caller ID**	0 =Disable (No) 1 =Enable (Yes)	1	13-04
32	**Anti-trombone Function**	0 = No Effect (No) 1 = Available (Yes)	0	

Item No.	Item	Input Data	Default	Related Program
33	**APSU Trunk Receive Gain** Additional PAD when a trunk call connects to APSU Voice Mail.	1~57 (-15.5dB ~ +12.5dB in 0.5dB intervals)	32 (0dB)	
35	**DT800/DT700 Large LED Illumination Setup** Sets LED color for incoming trunk call. In DT800/DT700 local terminal setting menu, illumination setting must be ‘Automatic’, otherwise the terminal will ignore PRG 14-01-35, PRG 15-05-37 and PRG 15-23 settings.	2 = Red 3 = Green 4 = Blue 5 = Yellow 6 = Purple 7 = Light Blue 8 = White 9 = Rotation	2	
36	**Calling Party Name Indication (ISDN Trunk)** Shows sending caller name on outgoing ISDN calls.	0 = Disable 1 = Enable	0	

Item No.	Item	Input Data	Default	Related Program
38	**Outgoing CLI Selection** Select CLI (Calling Party Number) sending way to trunk. When set to 0, extension CLI number set in PRG21-13-01, PRG21-18-01, or PRG21-19-01is sent according to seized trunk type (ISDN/ H.323/SIP) automatically. When set to 1, calling extension number is sent as CLI. When set to 2, extension table number set in PRG21-25-01 is sent as CLI. When set to 3, 4, or 5, extension CLI number set in PRG21-13-01, PRG21-18-01, or PRG21-19-01is sent to seized trunk regardless of trunk type.	0 = Contract Number 1 = Extension Number 2 = Extended Table 3 = PRG 21-13 4 = PRG 21-18 5 = PRG 21-19 6 = No digits	0	21-13-01 21-18-01 21-19-01 21-25-01
39	**CLI Composition** If select default value 0:”prefer extension”, the extension's CLI is sent out, if it is not empty. If it is empty, the trunk's CLI is sent instead. If select value 1:”combine trunk + extension”, the trunk's CLI is stored in the sending buffer, padded with the extension's CLI.	0 = Prefer Extension 1 = Combine Trunk + Extension	0	
40	**ISDN Queue Announcement Connect Mode**	0 = Send CONNECT 1 = Send PROGRESS #8	0	22-14 22-15 41-11 41-19

Default

Trunk Port Number	Name
1	Line 001
2	Line 002
:	:
400	Line 400

Conditions

None

Feature Cross Reference

Refer to features in the Input Data table.

Program 14 : Trunk, Basic Setup

14-06 : Trunk Group Routing

Level:
IN

Description

Use **Program 14-06 : Trunk Group Routing** to set up an outbound routing table for the trunk groups you assigned in Program 14-05. When users dial 9, the system routes their calls in the order (priority) specified. For example, if a user dials 9 and all calls in the first group are busy, the system may route the call to another group. Trunk Access Map programming (Programs 14-07) may limit this option. The system contains 100 routing tables for trunk access. Each table has four priority orders for trunk access. There are 100 available Trunk Group Numbers.

Example for setting:

With less than four trunk groups,

Route Number 1	: Order 1 – Trunk Group 1
	: Order 2 – Trunk Group 2

For the above setting, if all the lines in trunk group 1 are busy, the system searches for an idle line in trunk group 2.

With more than four trunk groups,

Route Number 1	: Order 1 – Trunk Group 1
	: Order 2 – Trunk Group 2
	: Order 3 – Trunk Group 3
	: Order 4 – 1002 (Jump To Route Number 2)
Route Number 2	: Order 1 – Trunk Group 4
	: Order 2 – Trunk Group 5

For the above setting, if all the lines in the trunk groups 1, 2 and 3 are busy, the system searches for an idle line in trunk groups 4 and 5.

Input Data

Route Table Number	001~100

Item No.	Priority Order Number	Input Data	Related Program
01	1~4	0 = Not set 001-100 = Trunk group No. 101-150 = 100+Networking System No. 1001-1100 = 1000+Route Table No.	14-01-07 14-05 15-01-02 21-02

Default

- Route 1, Order Number 1 = 1 (Trunk Group 1).
- Order Numbers 2, 3, 4 = 0 (Not Specified).
- All Other Routes (2~100) and Order Numbers (1~4) = 0 (Not Specified).

Conditions

None

Feature Cross Reference

None

Program 16 : Department Group Setup

16-01 : Department Group Basic Data Setup

Level:
IN

Description

Use **Program 16-01 : Department Group Basic Data Setup** to set the function mode for each department group. There are 64 available Department Groups.

Input Data

Department Group Number	1~64

Item No.	Item	Input Data	Default	Related Program
01	**Department Name**	Maximum 12 characters	No setting	11-07
02	**Department Calling Cycle** Use this option to set the call routing for Department Calling. Routing can be either circular (cycles to all phones in group) or priority (cycles to highest priority extensions first).	0 =Normal Routing (Priority) 1 =Easy – UCD Routing (Circular)	0	16-02
03	**Department Routing when Busy (Auto Step Call)** Use this option to set how the system routes an Intercom call to a busy Department Group member. Intercom callers to the extension can either hear busy or route to the first available department number. This only occurs for calls to the extension directly, not the department number assigned in Program 11-07.	0 =Normal (Intercom caller to busy department member hears busy) 1 =Circular (Intercom callers to busy department member routes to idle member)	0	16-02
04	**Hunting Mode** Use this option to set the action taken when a call reaches the last extension in the Department Group (0=hunting stopped, 1 =hunting repeats with circular routing through the Department Group).	0 =Last extension is called and hunting is stopped 1 =Circular	0	

Item No.	Item	Input Data	Default	Related Program
05	**Extension Group All Ring Mode Operation** Determine whether calls ringing a Department Group should ring all extensions in the group simultaneously automatically or manually when using the service code defined in Program 11-12-09. ✎ *When set to (1) Automatic, only ICM Calls and DID Calls will ring all the stations in the Department Group.*	0 =Manual 1 =Automatic	0	11-16-10
06	**STG Withdraw Mode**	0 = Disable (Camp On) 1 = Enable (Overflow Mode)	0	
07	**Call Recall Restriction for STG** Determine whether or not an unanswered call transferred to a Department Group should recall the extension from which it was transferred.	0 =Disable (Recall) 1 =Enable (No Recall)	0	
08	**Maximum Queuing number for Department Group Call** To have Department Group calls queue when busy, set this entry to maximum queuing number.	0 =No Queuing 1 =Queuing	0	
09	**Department Hunting No Answer Time** Set how long a call rings a Department group extension before hunting occurs.	0~64800 seconds	15	
10	**Enhanced Hunt Type** Set the type of hunting for each Extension (Department) Group.	0 =No queuing 1 =Hunting When Busy 2 =Hunting When Not Answered 3 =Hunting When Busy or No Answer	0	

Conditions

None

Feature Cross Reference

- ❑ Department Calling

Program 16 : Department Group Setup

16-02 : Department Group Assignment for Extensions

Level:
IN

Description

Use **Program 16-02 : Department Group Assignment for Extensions** to set the Department Groups. The system uses these groups (64 Department Groups) for Department Calling. Assign pilot numbers to Department Groups you set up in Program 11-07. This lets system users place calls to the departments. Use Program 16-01 to set the priority of each extension in each Department Group. When a call comes to the group, the extensions ring in order of their priority.

Input Data

Extension Number	Maximum eight digits

Item No.	Group Number	Priority	Default	Description	Related Program
01	1~64	1~999	1 – xxx (See Note)	Set up the Department Group called by the pilot number and the extension priority when a group is called. Call Pickup Groups are set up in 23-02.	11-07 16-01

✎ *The initial value of a priority becomes the ports numerical order assigned in Program 11-02 and 11-04. (Extension ports are 1~ 960. Virtual extension ports are 961~1472.)*

Conditions

None

Feature Cross Reference

❑ Department Calling

Program 20 : System Option Setup

20-01 : System Options

Level:
IN

Description

Use **Program 20-01 : System Options** to set various system options.

Input Data

Item No.	Item	Input Data	Default	Description	Related Program
01	**Operator Access Mode**	0 =Step Call 1 =Circular	0	Use this program to set up priority of a call when calling an operator telephone.	20-17
02	**Text Message Mode**	0 =Call mode 1 =No Answer/ Busy mode	1	Use this program to select the mode when calling the telephone which set up the text message.	11-11-14 15-07-08
04	**Network BLF Indication**	0-64800 in 100ms increments	0	Used to determine how often the SV9100 updates the DSS key BLF indications. For NetLink, the entry should be “30” in all SV9100s.	30-05
05	**DTMF Receive Active Time**	0~64800 seconds	10	For OPXs, analog telephones and certain analog trunks (like DISA), the system attaches a DTMF receiver to the port for this interval. The system releases the receiver after the interval expires.	25-07-01
06	**Alarm Duration**	0~64800 seconds	30	This time sets the duration of the alarm signal.	11-12-05
07	**Callback Ring Duration Time**	0~64800 seconds	15	Callback rings an extension for this time.	11-12-05 15-07-35
08	**Trunk Queuing Callback Time**	0~64800 seconds	15	Trunk Queuing callback rings an extension for this time.	11-12-05 15-07-35
09	**Callback/Trunk Queuing Cancel Time**	0~64800 seconds	64800	The system cancels an extension Callback or Trunk Queuing request after this time.	11-12-05 15-07-35
10	**Trunk Guard Timer**	0~64800 seconds	1	The amount of time the system waits to seize the next outside line after the system releases an outside line.	
12	**Telephone/Web Pro Logout Time**	1~84600 seconds (84600sec = 1 day)	900	The system automatically logs out of a Telephone/Web Pro session after inactivity lasting this time.	

Conditions

None

Feature Cross Reference

Refer to the Input Data table at the beginning of this section.

Program 21 : Outgoing Call Setup

21-16 : Trunk Group Routing for Networks

Level:
IN

Description

Use **Program 21-16 : Trunk Group Routing for Networks** to assign Program 14-06 routes for a networked system. This is required to seize the trunk in a networked system (Extension in System A tries to make an external call using a trunk in System B).
The route number is specified for each system ID (01-50).

Input Data

System ID	01-50

Item No.	Day/Night Mode	Route Table Number	Default	Related Program
01	1~8	0-100 (0=No setting)	1	14-06

Conditions

None

Feature Cross Reference

- ❑ Central Office Calls, Placing
- ❑ Networking - NetLink
- ❑ Networking - AspireNet

Program 22 : Incoming Call Setup

22-05 : Incoming Trunk Ring Group Assignment

Level:
IN

Description

Use **Program 22-05 : Incoming Trunk Ring Group Assignment** to assign trunks to incoming Ring Groups.There are 100 available Ring Groups.

Input Data

Trunk Port Number	001~400

Item No.	Day/Night Mode	Incoming Group Number	Default	Description	Related Program
01	1~8	0 (No Setting) 001~100 (Incoming Group) 102 (In-Skin/ External Voice Mail or InMail) 103 Centralized VM	1	Use this program to assign Normal Ring Trunks (22-02) to Incoming Ring Groups (22-04).	22-04 22-06

Conditions

None

Feature Cross Reference

- ❑ Ring Groups

Program 22 : Incoming Call Setup

22-08 : DIL/IRG No Answer Destination

Level:
IN

Description

Uor DIL Delayed Ringing, use **Program 22-08 : DIL/IRG No Answer Destination** to assign the DIL No Answer Ring Group. An unanswered DIL rings this group after the DIL No Answer Time expires (Program 22-01-04). DIL Delayed Ringing can also reroute outside calls ringing a Ring Group.

Make eight assignments, one for each Night Service mode.

Input Data

Trunk Port Number	001~400

Item No.	Day/Night Mode	Incoming Group Number	Default
01	1~8	0 (No Setting) 001~100 (Incoming Ring Group) 102 (In-Skin/External Voice Mail or InMail)	0

Conditions

None

Feature Cross Reference

- ❑ Direct Inward Line (DIL)
- ❑ Ring Group

Program 22 : Incoming Call Setup

22-10 : DID Translation Table Setup

Level:
IN

Description

Use **Program 22-10 : DID Translation Table Setup** to specify the size of the DID Translation Tables. There are 4000 Translation Table entries that you can allocate among 20 Translation Tables.

Input Data

Conversion Table Area Number	01~20

Item No.	Item	Input Data
01	1st Area Setup (Start Address)	0~2000 (0 = No Setting)
	1st Area Setup (End Address)	Default Table
	2nd Area Setup (Start Address)	Default Table
	2nd Area Setup (End Address)	Default Table

Default Table

Conversion Table Area	1st		2nd	
	Start Table	End Table	Start Table	End Table
1	1	200	0	0
2	201	400	0	0
3	401	600	0	0
4	601	800	0	0
5	801	1000	0	0
6	1001	1200	0	0
7	1201	1400	0	0
8	1401	1600	0	0
9	1601	1800	0	0
10	1801	2000	0	0
:	:	:	:	:
20	0	0	0	0

Conditions

None

Feature Cross Reference

❏ Direct Inward Dialing (DID)

Program 22 : Incoming Call Setup

22-11 : DID Translation Number Conversion

Level:
SA

Description

Use **Program 22-11 : DID Translation Table Number Conversion** to specify for each Translation Table entry (2000).

- ❑ The digits received by the system (eight maximum)
- ❑ The extension the system dials after translation (24 digits maximum)
- ❑ The name that should show on the dialed extension display when it rings (12 characters maximum)
- ❑ The Transfer Target – 1 and 2

  ✎ *If the Transfer Targets are busy or receive no answer, those calls are transferred to the final transfer destination (Program 22-10).*

- ❑ Operation Mode

Use the following chart when entering and editing text for names. Press the key once for the first character, twice for the second character, etc. For example, to enter a C, press 2 three times.

Key for Entering Names	
When entering names in the procedures below, refer to this chart. Names can have up to 12 digits.	
Use this keypad digit . . .	**When you want to. . .**
1	Enter characters: 1 @ [¥] ^ _ ` { | } → ← 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

Key for Entering Names	
When entering names in the procedures below, refer to this chart. Names can have up to 12 digits.	
Use this keypad digit . . .	**When you want to. . .**
*	Enter characters: ✱ + , - . / : ; < = > ? **B** **E** σ **S** □ ¢ £
#	**#** = 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 soft key instead to accept and/or add a space.)
Recall	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.

Input Data

Conversion Table Number	1~4000

Item No.	Item	Input Data	Default
01	**Received Number** This is the received DID digits.	Maximum eight digits	See Default Value
02	**Target Number** Enter the destination number to which the DID number is sent.	Maximum 24 digits	See Default Value
03	**DID Name** This is the name that is assigned to the DID digits when it rings the extension.	Maximum 12 characters	No Setting
04	**Transfer Operation Mode**	0 =No Transfer 1 =Busy 2 =No Answer 3 =Busy/No Answer	0

Item No.	Item	Input Data	Default
05	**Transfer Destination Number 1**	0 = No Setting 1~100 = Incoming Group 102 = In-Skin/External Voice Mail or InMail 103 = Centralized VM 201~264 = Extension Group 400 = Valid Extension Number 401 = DISA 501~599 = DISA/VRS Message 1000~1999 = Speed Dial Number (000~999)	0
06	**Transfer Destination Number 2** 400 = Allows the outside party to dial a different extension number in the translation table (for example, ring no answer to a dialed number, the caller then hears a dial tone, allowing them to enter another Valid Extension Number). 401 = Provides the caller with DISA dialing options (requires using the DISA password). ✎ *This applies to 22-11-05 and 22-11-06.*		0
07	**Call Waiting** PRG 20-09-07 overrides this setting.	0 = Disable (No) 1 = Enable (Yes)	0
08	**Maximum Number of DID Calls**	0~200 (0 = No Limit)	0
09	**Music on Hold Source**	0 = IC/MOH Port 1 = BGM Port 2 = ACI Port	0
10	**ACI Music Source Port**	When a sound source type is 2 in above : (0~96)	0
11	**Ring Group Transfer** Enable (1) or disable (0) each conversation tables ability to follow the Ring Group programming defined in Program 22-12-01 : DID Intercept Ring Group. If Program 22-11-05 : DID Translation Number Conversion, Transfer Destination Number 1 and Program 22-11-06 : DID Translation Number Conversion, Transfer Destination Number 2 are set, the priority of transferring is in this order: Program 22-11-05 then Program 22-11-06 then if Program 22-11-11 is enabled, Program 22-12-01.	0 = Disable (Caller will hear Ringback) 1 = Enabled (Go to normal ring)	1

Default

The default value of PRG22-11-01/PRG22-11-02 is shown as below.

Conversion Table	Received Number	Target Number
1	01	101
:	:	:
99	99	199
100	00	100
101	No Setting	No Setting
:	:	:
200	No Setting	No Setting
201	01	101
:	:	:
299	99	199
300	00	100
301	No Setting	No Setting
:	:	:
400	No Setting	No Setting
401	01	101
:	:	:
499	99	199
500	00	100
501	No Setting	No Setting
:		
600	No Setting	No Setting
601	01	101
:		
699	99	199
700	00	100
701	No Setting	No Setting
:	:	:
4000	No Setting	No Setting

Conditions

- When the trunk type is set to 3 (DID) in 22-02-01, the DID Transfer Destination for each DID feature is not supported. This feature is supported only for DID trunks when assigned as VRS.

Feature Cross Reference

❏ Direct Inward Dialing (DID)

Program 22 : Incoming Call Setup

22-12 : DID Intercept Ring Group

Level:
IN

Description

For each DID Translation Table, use **Program 22-12 : DID Intercept Ring Group** to define the first destination group for DID calls.

Depending on the entry in Program 22-09-02 and 22-11-04, the incoming calls route to the first destination group by the following:

❑ Vacant number intercept (vacant number means that no phone is connected, no station blade is installed, or the extension number is not defined in Program 11-02)

❑ Busy intercept

❑ Ring-no-answer intercept

If the destination is 0, the calls are forwarded to the trunk ring group defined in Program 22-11 based on the table assigned to the DID trunk.

✎ ***If Programs 22-11-05 and 22-11-06 are set, the priority of transferring is in this order: Program 22-11-05 + Program 22-11-06 + Program 22-12.***

For busy and no-answer calls, if the first and third destinations are programmed, but the second destination is not, the incoming call goes to the third destination after the first destination. If the first and second destinations are not defined, but the third destination is, the call goes directly to the third destination.

Input Data

Conversion Table Area Number	01~20

Item No.	Day/Night Mode	Incoming Group Number	Default
01	1~8	0 (No Setting) 1~100 (Incoming Ring Group) 102 (In-Skin/External Voice Mail or InMail) 103 = Centralized VM	0

Conditions

None

Feature Cross Reference

- Direct Inward Dialing (DID)

Program 25 : VRS/DISA Setup

25-03 : VRS/DISA Transfer Ring Group With Incorrect Dialing

Level:
IN

Description

Use **Program 25-03 : VRS/DISA Transfer Ring Group With Incorrect Dialing** to set what happens to a call when the DISA or Automated Attendant caller dials incorrectly or waits too long to dial. The call can either disconnect (0) or Transfer to an alternate destination (a ring group or voice mail). When setting the DISA and DID Operating Mode, make an entry for each Night Service mode.

Input Data

Trunk Port Number	001~400

Item No.	Day/Night Mode	Incoming Group Number	Default	Related Program
01	1~8	0 (Disconnect) 1~100 (Incoming Ring Group) 102 (In-Skin/External Voice Mail or InMail) 103 = Centralized VM	0	22-04

Conditions

None

Feature Cross Reference

❑ Direct Inward System Access (DISA)

Program 25 : VRS/DISA Setup

25-04 : VRS/DISA Transfer Ring Group With No Answer/Busy

Level:
IN

Description

Use **Program 25-04 : VRS/DISA Transfer Ring Group With No Answer/Busy** to set the operating mode of each DISA trunk. This sets what happens to the call when the DISA or Automated Attendant caller calls a busy or unanswered extension. The call can either disconnect (0) or Transfer to an alternate destination (a ring group or voice mail). When setting the DISA and DID Operating Mode, make an entry for each Night Service mode.

Input Data

Trunk Port Number	001~400

Item No.	Day/ Night Mode	Incoming Group Number	Default	Related Program
01	1~8	0 (Disconnect) 1~100 (Incoming Ring Group) 102 (In-Skin/External Voice Mail or InMail)	0	22-04

Conditions

None

Feature Cross Reference

- ❑ Direct Inward System Access (DISA)

Program 25 : VRS/DISA Setup

25-08 : DISA User ID Setup

Level:
SA

Description

Use **Program 25-08 : DISA User ID Setup** to set the 6-digit DISA password for each user. There are 15 users each with one 6-digit password.

Input Data

DISA User Number	1~15

Item No.	Password	Default	Related PRG
01	Dial (Fixed – six digits) 0~9, ✱, #	DISA User No.1-15: DISA User ID 000001-000015	49-10-11

Conditions

None

Feature Cross Reference

❑ Direct Inward System Access (DISA)

Program 44 : ARS/F-Route Setup

44-01 : System Options for ARS/F-Route

Level:
IN

Description

Use **Program 44-01 : System Options for ARS/F-Route** to define the system options for the ARS/F-Route feature.

Input Data

Item No.	Item	Input Data	Default
01	**ARS/F-Route Time Schedule** If this option is set to **0**, the F-Route table selected is determined only by the digits dialed without any relation to the day or time of the call. If this option is set to **1**, the system first refers to Program 44-10. If there is a match, the pattern defined in that program is used. If not, the F-Route pattern in Program 44-09 and time setting in 44-08 are used.	0 = Not Used 1 = Used	0

Conditions

None

Feature Cross Reference

- ❑ Automatic Route Selection (ARS)
- ❑ Uniform Numbering Network

Program 44 : ARS/F-Route Setup

44-02 : Dial Analysis Table for ARS/F-Route Access

Level:
IN

Description

Use **Program 44-02 : Dial Analysis Table for ARS/F-Route Access** to set the Pre-Transaction Table for selecting ARS/F-Route.

Input Data

Dial Analysis Table Number	1~120

Item No.	Item	Input Data	Default
01	**Dial** Set the number of digits to be analyzed by the system for ARS routing.	Up to eight digits (Use line key 1 for a Don't Care digit, @)	No Setting
02	**Service Type** **- Service Type 1 (Extension Number)** The number goes to an extension after deleting the front digit(s). *Additional Data* Assign the digit(s) to be deleted on top of the number for extension number usage. At least one digit must be deleted. **- Service Type 2 (ARS/F-Route)** The number is controlled by ARS/F-Route table. *Additional Data:* If the ARS/F-Route Time Schedule is not used, assign the ARS/F-Route table number for Program 44-05. If the ARS/F-Route Time Schedule is used, assign the ARS/F-Route selection number for Program 44-04. **- Service Type 3 (Dial Extension Analyze Table)** The total length of the number exceeds more than 8 digits. *Additional Data:* Assign the Dial Extension Analysis Table number to be used in Program 44-03.	0 =No setting (None) 1 =Extension Call (Own) 2 =ARS/F-Route Table (F-Route) 3 =Dial Extension Analyze Table (Option)	0

Item No.	Item	Input Data	Default
03	**Additional Data** For the Service Type selected in 44-02-02, enter the additional data required. 1: Delete Digit = 0~255 (255 = Delete All Digits) 2: [Program 44-01 : 0] ARS/F-Route Table Number = 0~500 (0 = No Setting) Refer to Program 44-05. [Program 44-01 : 1] ARS/F-Route Select Table Number = 0~500 (0 = No Setting) Refer to Program 44-04. 3: Dial Extension Analyze Table Number = 0~4 (0 = No Setting) Refer to Program 44-03.	1 =Delete Digit = 0~255 (255 : Delete All Digits) 2 =0~500 (0 = No Setting) 3 =Dial Extension Analyze Table Number = 0~4 (0 = No Setting)	0
04	**Dial Tone Simulation** If enabled, this option sends dial tone to the calling party after the routing is determined. This may be required if the central office at the destination does not send dial tone.	0 =Off 1 =On	0

Conditions

None

Feature Cross Reference

❑ Automatic Route Selection (ARS)

Program 44 : ARS/F-Route Setup

44-03 : Dial Analysis Extension Table

Level: IN

Description

When Program 44-02-02 is set to type 3, use **Program 44-03 : Dial Analysis Extension Table** to set the dial extension analysis table. These tables are used when the analyzed digits must be more than eight digits. If the received digits do not match the digits set in tables 1~250, table number 252 is used to refer to the next Extension Table Area (1~4) to be searched. If the received digits are not identified in tables 1~250, the F-Route selection table number defined in table 251 is used.

Input Data

Extension Table Area Number	1~4

Dial Analysis Table Number	1~252

Dial Analysis Table Number : 1~250

Item No.	Item	Input Data	Default
01	**Dial**	Up to 24 digits Digits = 1~9, 0, *, **#**, @ (Press Line Key 1 for wild character @)	No Setting
02	**ARS/F-Route Select Table Number**	0~500 (ARS/F-Route Table Number) With Program 44-01 set to 0, Program 44-05 is checked. With Program 44-01 set to 1, Program 44-04 is checked.	0

Dial Analysis Table Number : 251

Item No.	Item	Input Data	Default
03	**ARS/F-Route Select Table Number**	0~500 (ARS/F-Route Table Number) With Program 44-01 set to 0, Program 44-05 is checked. With Program 44-01 set to 1, Program 44-04 is checked.	0

Dial Analysis Table Number : 252

Item No.	Item	Input Data	Default
04	**Next Table Area Number**	0~4	0

Conditions

None

Feature Cross Reference

- ❑ Automatic Route Selection (ARS)

Program 44 : ARS/F-Route Setup

44-04 : ARS/F-Route Selection for Time Schedule

Level:
IN

Description

Use **Program 44-04 : ARS/F-Route Selection for Time Schedule** to assign each ARS/F-Route Selection number to an ARS/F-Route table number for each ARS/F-Route time mode. There are eight time modes for ARS/F-Route Access.

Input Data

ARS/F-Route Selection Number	1~500

Item No.	ARS/F-Route Time Mode	ARS/F-Route Table Number	Default
01	1~8	0~500	0

Conditions

None

Feature Cross Reference

❑ Automatic Route Selection (ARS)

Program 44 : ARS/F-Route Setup

44-05 : ARS/F-Route Table

Level:
IN

Description

Use **Program 44-05 : ARS/F-Route Table** to set the ARS/F-Route table. There are four kinds of order. If the higher priority trunk groups are busy, the next order group is used. If a lower priority route is selected, the caller may be notified with a beep tone.

Input Data

ARS/F-Route Table Number	1~500

Priority Number	1~4

Item No.	Item	Input Data	Default
01	**Trunk Group Number** Select the trunk group number to be used for the outgoing ARS call.	0~100, 255 0 = No Setting 255 = Extension Call	0
02	**Delete Digits** Enter the number of digits to be deleted from the dialed number.	0~255 (255 = Delete All)	0
03	**Additional Dial Number Table** Enter the table number (defined in Program 44-06) for additional digits to be dialed.	0~1000	0
04	**Beep Tone** Select whether or not a beep is heard if a lower priority trunk group is used to dial out.	0 = Off 1 = On	0
05	**Gain Table Number for Internal Calls** Select the gain table number to be used for the internal call (defined in Program 44-07).	0~500 0 = No Setting	0
06	**Gain Table Number for Tandem Connections** Select the gain table number to be used for the tandem call (defined in Program 44-07).	0~500 0 = No Setting	0
07	**ARS Class of Service** Select the ARS Class of Service to be used for the table. An extension ARS COS is determined in Program 26-04-01.	0~16	0

Item No.	Item	Input Data	Default
08	**Dial Treatment** Select the Dial Treatment to be used for the table. 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.	0~15	0
09	**Maximum Digit** Input the maximum number of digits to send when using the F-Route.	0~24	0
10	**CCIS over IP Destination Point Code** Input the Destination Point Code to send when using this F-Route.	0~16367	0
11	**Network Specified Parameter Table** Enter a table number from Program 26-12.	0~16	0

Conditions

None

Feature Cross Reference

❑ Automatic Route Selection (ARS)

Program 44 : ARS/F-Route Setup

44-06 : Additional Dial Table

Level:
IN

Description

Use **Program 44-06 : Additional Dial Table** to set the additional dial table to add prior to the dialed ARS/F-Route number. The Additional Dial Table used is determined in Program 44-05-03.

Input Data

Additional Dial Table Number	1~1000

Item No.	Additional Dial	Default
01	Up to 24 digits Enter: 1~9, 0, *, **#**, Pause (press LK 1 to enter a pause)	No Setting

Conditions

None

Feature Cross Reference

❑ Automatic Route Selection (ARS)

Program 44 : ARS/F-Route Setup

44-07 : Gain Table for ARS/F-Route Access

Level:
IN

Description

Use **Program 44-07 : Gain Table for ARS/F-Route Access** to set the gain/PAD table. If an extension dials ARS/F-Route number:

❑ The Extension Dial Gain Table, assigned in Program 44-05, is activated.

❑ The Extension Dial Gain Table follows Outgoing transmit and Outgoing receive settings.

If the incoming call is transferred to another line using ARS/F-Route:

❑ The Tandem Gain Table, assigned in Program 44-05, is activated.

❑ The Tandem Gain Table follows the Incoming transmit and Incoming receive settings for incoming line, and Outgoing transmit and Outgoing receive settings for the outgoing line.

✎ ***For ARS/F-Route calls, the CODEC gains defined in Program 14-01-02 and 14-01-03 are not activated.***

Input Data

Gain Table Number	1~500

Item No.	Item	Input Data	Default
01	**Incoming Transmit**	1~57 (-15.5 ~ +12.5dB)	32 (0dB)
02	**Incoming Receive**	1~57 (-15.5 ~ +12.5dB)	32 (0dB)
03	**Outgoing Transmit**	1~57 (-15.5 ~ +12.5dB)	32 (0dB)
04	**Outgoing Receive**	1~57 (-15.5 ~ +12.5dB)	32 (0dB)

Conditions

None

Feature Cross Reference

- Automatic Route Selection (ARS)

Program 44 : ARS/F-Route Setup

44-08 : Time Schedule for ARS/F-Route

Level:
IN

Description

Use **Program 44-08 : Time Schedule for ARS/F-Route** to define the daily pattern of the ARS/F-Route feature. ARS/F-Route has 10 time patterns. These patterns are used in Program 44-09 and 44-10. The daily pattern consists of 20 time settings.

Input Data

Schedule Pattern Number	01~10

Item No.	Time Number	Start Time	End Time	Mode
01	01~20	0000~2359	0000~2359	1~8

Default

All Schedule Patterns : 0:00 – 0:00, Mode 1

Example:

Pattern 1

0:00	8:00	18:00	22:00	0:00
Mode 3	Mode 1	Mode 2	Mode 3	

Time Number 01 : 00:00 – 08:00   Mode 3
Time Number 02 : 08:00 – 18:00   Mode 1
Time Number 03 : 18:00 – 22:00   Mode 2
Time Number 04 : 22:00 – 00:00   Mode 3

Pattern 2

0:00	0:00
Mode 2	

Time Number 01 : 0:00 – 0:00   Mode 2

Conditions

None

Feature Cross Reference

- Automatic Route Selection (ARS)

Program 44 : ARS/F-Route Setup

44-09 : Weekly Schedule for ARS/F-Route

Level:
IN

Description

Use **Program 44-09 : Weekly Schedule for ARS/F-Route** to define a weekly schedule for using ARS/F-Route. The pattern number is defined in Program 44-08-01.

Input Data

Item No.	Day Number	Schedule Pattern Number	Default
01	1 =Sunday	1~10	Pattern 1
	2 =Monday	1~10	Pattern 1
	3 =Tuesday	1~10	Pattern 1
	4 =Wednesday	1~10	Pattern 1
	5 =Thursday	1~10	Pattern 1
	6 =Friday	1~10	Pattern 1
	7 =Saturday	1~10	Pattern 1

Conditions

None

Feature Cross Reference

- ❑ Automatic Route Selection (ARS)

Program 44 : ARS/F-Route Setup

44-10 : Holiday Schedule for ARS/F-Route

Level:
IN

Description

Use **Program 44-10 : Holiday Schedule for ARS/F-Route** to define a yearly schedule for ARS/F-Route. This schedule is used for setting special days such as national holidays. The pattern number is defined in Program 44-08-01.

Input Data

Item No.	Date	Schedule Pattern Number	Default
01	0101~1231	0~10 (0 = No Setting)	0

Conditions

None

Feature Cross Reference

- ❑ Automatic Route Selection (ARS)

Program 45 : Voice Mail Integration

45-01 : Voice Mail Integration Options

Level:
IN

Description

Use **Program 45-01 : Voice Mail Integration Options** to customize certain voice mail integration options.

Input Data

Item No.	Item	Input Data	Default
01	**Voice Mail Department Group Number** Assign which Extension (Department) Group number is to be assigned as the voice mail group.	0~64 0 = No Voice Mail	0
02	**Voice Mail Master Name** Enter the Voice Mail Master Name.	Up to 12 Characters	VOICE MAIL
04	**Park and Page** Enable/disable the system ability to process the Voice Mail Park and Page (✱) commands. You should normally **enable** this option.	0 = Off 1 = On	1
05	**Message Wait** Enable/disable the system ability to process the Voice Mail Message Wait (**#**) commands. You should normally **enable** this option. If enabled, be sure that the programmed Message Notification strings don't contain the code for trunk access.	0 = Off 1 = On	1
06	**Record Alert Tone Interval Time** This time sets the interval between Voice Mail Conversation Record alerts	0~64800 (sec)	0
07	**Centralized Voice Mail Pilot Number** This number is the same as the extension number or pilot number.	Dial (up to eight digits)	No Setting
08	**Centralized Voice Mail Department Group Number** Assigns which Extension (Department) Group Number is to be assigned as the Centralized Voice Mail group.	0~64	0

Input Data (Continued)

Item No.	Item	Input Data	Default
09	**Centralized Voice Mail Master Name** Assigns the Centralized Voice Mail Master Name.	Up to 12 characters	"C.V.M."
10	**New NSL Protocol support**	0 = Off 1 = On	0
11	**Prefix for Call Screening**	Dial (One digit)	1
12	**Prefix for Park and Page**	Dial (One digit)	✱
13	**Prefix for Message Wait**	Dial (One digit)	#
14	**CCIS Centralized Voice Mail Number** Assign the pilot number to Centralized Voice Mail over CCIS Link. This is assigned only in the remote switches.	Dial (up to eight digits)	No Setting
15	**Analog Voice Mail Protocol Selection** Assigns whether fixed codes are used or the codes used in PRG 45-04 are used for analog voice mail protocol.	0: Fixed 1: Program	0
16	**Voice Mail Fax Digit Add Assignment** Assign up to four digits in front of the station number sent to the SLT port when a call is forwarded.	Up to four digits	None
17	**Reply Mailbox Number** Whether or not to include the mailbox number in the analog voice mail protocol.	0: No 1: Yes	1
18	**Trunk Number Mapping** Assign the digits of trunk number mapping.	2~3	2
19	**Centralized Voice Mail Type** Assign which Centralized Voice Mail types to use, Retro (Aspire) or Enhanced (Cygnus).	0 = Retro 1 = Enhanced	1

Conditions

None

Feature Cross Reference

- ❑ Voice Mail Integration (Analog)

Program 84 : Hardware Setup for VoIP

84-01 : H.323 Trunk Basic Information Setup

Level:
IN

Description

Use **Program 84-01 : H.323 Trunk Basic Information Setup** to set the basic information of the H.323 Trunk.

Input Data

Item No.	Item	Input Data	Default
02	**Number of G.711 audio frames**	1~4	3
03	**G.711 VAD mode**	0 = Disable 1 = Enable	0
04	**G.711 Type**	0 = A-law 1 = μ-law	0
05	**Number of G.729 audio frames**	1~6 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms 5 = 50ms 6 = 60ms	3
06	**G.729 VAD mode**	0 = Disable 1 = Enable	0
07	**G.729 Jitter Buffer(min)**	0~300ms	30
08	**G.729 Jitter Buffer (average)**	0~300ms	60
09	**G.729 Jitter Buffer (max)**	0~300ms	120
15	**Jitter Buffer Mode** Setting 2 (Self adjusting-silent period-is not valid by IPLB. If this value is set, the system will operate as Setting 3.	1 = Static 3 = Self adjusting	3
16	**G.711 Jitter Buffer(min)**	0~300ms	30
17	**G.711 Jitter Buffer (average)**	0~300ms	60
18	**G.711 Jitter Buffer (max)**	0~300ms	120

Input Data (Continued)

Item No.	Item	Input Data	Default
22	**VAD Threshold**	0~30 (-19dB~ +10dB and self adjustment) 0 = Self adjustment 1 = -19dB (-49dBm) : 20 = 0dB (-30dBm) : 29 = 9dB (-21dBm) 30 = 10dB (-20dBm)	20
33	**Priority CODEC setting** Priority of voice encoding method.	0~3 0 = G.711 2 = G.729 3 = G.722	0
63	**Number of G.722 audio frames**	1~4 1 = 10ms 2 = 20ms 3 = 30ms 4 = 40ms	3
65	**G.722 Jitter Buffer (min)**	0~300ms	30
66	**G.722 Jitter Buffer (average)**	0~300ms	60
67	**G.722 Jitter Buffer (max)**	0~300ms	120
68	**RTP Filter** To avoid incorrect voice pass connection, this Program checks the sending side address from received RTP packet at VoIPDB.	0 = Disable 1 = Enable	1

Conditions

None

Feature Cross Reference

❑ Voice Over Internet Protocol (VoIP)

Program 84 : Hardware Setup for VoIP

84-02 : H.225 and H.245 Information Basic Setup

Level:
IN

Description

Use **Program 84-02 : H.225 and H.245 Information Basic Setup** to define the basic setup information of H.225 and H.245.

Input Data

Item No.	Item	Input Data	Default
01	**H.225**	0~255sec	180
02	**H.225 Setup Acknowledge Timer**	0~255sec	9
03	**H.225 Setup Timer**	0~255sec	4
04	**H.225 Info Ack Timer**	0~255sec	9
05	**H.225 Call Proceeding Timer**	0~255sec	10
07	**H.245 Master Slave Determination Timer**	0~255sec	5
08	**H.245 Master Slave Determination Retry Count**	0~255sec	3
09	**H.245 Capability Exchange Timer**	0~255sec	5
10	**H.245 Logical Channel Establishment Timer**	0~255sec	50
11	**H.245 Mode Request Procedures Timer**	0~255sec	50
12	**H.245 Close Logical Channel Timer**	0~255sec	50
13	**H.245 Round Trip Delay Timer**	0~255sec	50
14	**H.245 Maintenance Loop**	0~255sec	50
15	**RAS GRQ Timer**	0~255sec	5
16	**GRQ Retry Count**	0~255	2
17	**RAS RRQ Timer**	0~255sec	5
18	**RRQ Retry Count**	0~255	3
19	**RAS URQ Timer**	0~255sec	3
20	**URQ Retry Count**	0~255	1
21	**RAS ARQ Timer**	0~255sec	5

Input Data

Item No.	Item	Input Data	Default
22	**ARQ Retry Count**	0~255	2
23	**RAS BRQ Timer**	0~255sec	5
24	**BRQ Retry Count**	0~255	2
25	**RAS IRR Timer**	0~255sec	5
26	**IRR Retry Count**	0~255	2
27	**RAS DRQ Timer**	0~255sec	8
28	**DRQ Retry Count**	0~255	2
29	**RAS LRQ Timer**	0~255sec	5
30	**LRQ Retry Count**	0~255	2
31	**RAS RAI Timer**	0~255sec	3
32	**RAI Retry Count**	0~255	2
33	**Call Signaling Port Number**	0~65535: 0~1719, 1721~65535	1730
35	**Fast Start Mode**	0 = Disable 1 = Enable0	1
36	**RAS Unicast Port Number**	0~65535	20001
37	**Terminal Type setting**	0~255	60

Conditions

None

Feature Cross Reference

- ❑ Voice Over Internet Protocol (VoIP)

Program 84 : Hardware Setup for VoIP

84-12 : Networking CODEC Information Basic Setup

Level:
IN

Description

Use **Program 84-12: Networking CODEC Information Basic Setup** to define the CODEC Information for Networking.

Input Data

Item No.	Item	Input Data	Default
01	**Number of G.711 Audio Frame** Maximum number of G711 Audio Frames. When the voice is encoded using the PCM (Pulse Code Modulation) method, a unit is a frame of 10ms.	1 =10 ms 2 =20 ms 3 =30 ms 4 =40 ms	3
02	**G.711 Silence Detection (VAD) Mode** Select whether to compress silence with G.711. When there is silence, the RTP packet is not sent.	0 =Disable 1 =Enable	0
03	**G.711 Type** Set the type of G.711.	0 =A-law 1 =μ-law	0
04	**G.711 Jitter Buffer - Minimum** Set the minimum value of the G.711 Jitter Buffer.	0~160 ms	30
05	**G.711 Jitter Buffer - Standard** Set the average value of the G.711 Jitter Buffer.	0~160 ms	60
06	**G.711 Jitter Buffer - Maximum** Set the maximum value of the G.711 Jitter Buffer.	0~160 ms	120
07	**G.729 Audio Frame** Maximum number of G729 Audio Frames. G.729 assumes the audio signal made by a specimen by 8kHz and the frame of 10ms is assumed to be a unit to 8kbps by the encoding compressed method.	1-6 (1 = 10ms, 2 = 20ms, etc.)	3

Input Data

Item No.	Item	Input Data	Default
08	**G.729 Silence Compression (VAD) Mode** Select whether to compress silence with G.729. When there is silence, the RTP packet is not sent.	0 =Disable 1 =Enable	0
09	**G.729 Jitter Buffer - Minimum** Set the minimum value of the Jitter Buffer of G.729 is set. Jitter is the variation in the time between packets arriving and the buffer allows this variation to be absorbed.	0-270 ms	30
10	**G.729 Jitter Buffer - Standard** Set the average G.729 Jitter Buffer.	0-270 ms	60
11	**G.729 Jitter Buffer - Maximum** Set the maximum G.729 Jitter Buffer.	0-270 ms	120
12	**Number of G.723 Audio Frame** Maximum number of the G.723 Audio Frame.	1 =30 msec 2 =60 msec	1
13	**G.723 Silence Compression (VAD) Mode** If enabled, RTP packets are not sent for the compressed silence.	0 =Disable 1 =Enable	0
14	**G.723 Jitter Buffer - Minimum** Set the minimum value of the G.723 Jitter Buffer.	0~270 ms	30
15	**G.723 Jitter Buffer - Standard** Set the average value of the G.723 Jitter Buffer.	0~270 ms	60
16	**G.723 Jitter Buffer - Maximum** Set the maximum value of the G.723 Jitter Buffer.	0~270 ms	120
17	**Jitter Buffer Mode** Set the mode of the Jitter Buffer. 1 = Size set to the fixed amount for the codec. 2 = The minimum/maximum range for the codec is used. 3 = The minimum/maximum range for the codec is used and adjusts at any time, regardless of silence.	1 =static 2 =adaptive during silence 3 =adaptive immediately	3

Input Data

Item No.	Item	Input Data	Default
18	**Silence Compression (VAD) Threshold** Set the voice level judged to be silence. Change value based .30 This entry is ignored if silence compression is disabled in 84-01-03 with G.711, or 84-01-06 with G.729.	0-30 (self-adjustment and -19db ~ +10db) 0 = self-adjustment 1:-19db (-49dbm) : 20 = 0db (-30dbm) : 29 = 9dbm (-21dbm) 30:10dbm (-20dbm)	20
19	**Idle Noise Level** Set the noise level which is generated when silent.	5000-7000 (-5000 ~ -7000dbm) 5000 = -5000dbm : 7000 = -7000dbm	7000
20	**Echo Canceller Mode** Determine whether or not to use Echo canceller.	0 =Disable 1 =Enable	1
21	**Signal limiter** Set the Signal Limiter Mode.	1 =Mode0 2 =Mode1 3 =Mode2 4 =Mode3 5 =Mode4 6 =Mode5	6
22	**Echo Canceller NLP Mode** Non-linear processing mode. Enable this option to decrease the low level echo. When NLP is enabled, the voice with low level is replacecd with NLP noise. As a result, a low echo of the level is usually removed compared with the conversation level.	0 =2 wire & 4 wire 1 =2 wire only	1
23	**--- Not Used ---**		
24	**Echo Canceller NLP Noise Setting** Becomes invalid item if 84-12-22 is set to Disabled. Set the noise level adjusting method added with NLP. When “0” is set, the level is self-adjusted - when “1” is set, Program 84-13-23 is used.	0 =adaptive 1 =fixed	0
25	**- Not Used -**	-	-

Input Data

Item No.	Item	Input Data	Default
26	**TX (Transmit) Gain** Define the setting to amplify and to attenuate the size of the transmission voice. The gain given when the voice packet is sent from the VOIPDB is set.	0-40 (-20 ~ +20) 0 = -20 dbm 1 = -19 dbm : 20 = 0 dbm : 39 = 19 dbm 40 = 20 dbm	20 (0 dbm)
27	**RX (Receive) Gain** Define the setting to amplify and to attenuate the size of the received voice. The gain given when the voice packet is received from the VOIPDB is set.	0-40 (-20~+20) 0 = -20 dbm 1 = -19 dbm : 20 = 0 dbm : 39 = 19 dbm 40 = 20 dbm	20 (0 dbm)
28	**Priority Codec Setting** The option selected here determines what other codec options are applied by priority.	0 =G711 PT 1 =G723 PT 2 =G729 PT 3 =G.722 PT	0
30	**EchoAuto Gain Control** Define the Auto Gain Control.	0 - 5	0
31	**DTMF Relay Mode** The initial setup information for the VOIPU is set in Program 84-27-02. If this option is set to either 0 or 1, priority is given.	0 =Disable 1 =RFC2833 2 =VoIPU	2
32	**FAX Relay Mode** Select "2" for FAX Relay to SLT (Program 15-03-03:special), Trunk and NetLink. Refer to Program 84-01-36 through 84-01-58 for FAX Relay options.	0 =Disable 1 =Enable 2 =Each Port Mode	0
33	**G.722 Audio Frame** Maximum number of G.722 Audio Frames. G.722 assumes the audio signal made by a specimen by 16kHz and the frame of 10ms is assumed to be a unit to 64kbps by the encoding compressed method.	1 =10 ms 2 =20 ms 3 =30 ms 4 =40 ms	3

Input Data

Item No.	Item	Input Data	Default
34	**G.722 Silence Compression Mode** Select whether to compress silence with G.722. When there is silence, the RTP packet is not sent.	0 =Disable 1 =Enable	0
35	**G.722 Jitter Buffer - Minimum** Set the minimum value of the Jitter Buffer of G.722 is set. Jitter is the variation in the time between packets arriving and the buffer allows this variation to be absorbed.	0-160 ms	30
36	**G.722 Jitter Buffer - Standard** Set the average G.722 Jitter Buffer.	0-160 ms	60
37	**G.722 Jitter Buffer - Maximum** Set the maximum G.722 Jitter Buffer.	0-160 ms	120

Conditions

None

Feature Cross Reference

❏ Voice Over Internet Protocol (VoIP)

Program 84 : Hardware Setup for VoIP

84-26 : IPL Basic Setup

Level:
IN

Description

Use **Program 84-26 : IPL Basic Setup** to set the IP address of IPL and the port.

Input Data

Slot Number	1

Item No.	Item	Input Data	Default
01	**IP Address**	xxx.xxx.xxx.xxx	Slot 1 = 172.16.0.20
02	**RTP Port Number**	0~65534	VoIP GW1 = 10020
03	**RTCP Port Number (RTP Port Number +1)**	0~65534	VoIP GW1 = 10021
12	**Video RTP Port** Sets the starting RTP port used by standard SIP terminal video.	0 ~ 65534	20020
13	**Video RTCP Port** Sets the starting RTCP port used by standard SIP terminal video.	0 ~ 65534	20021

Conditions

None

Feature Cross Reference

❑ Voice Over Internet Protocol (VoIP)

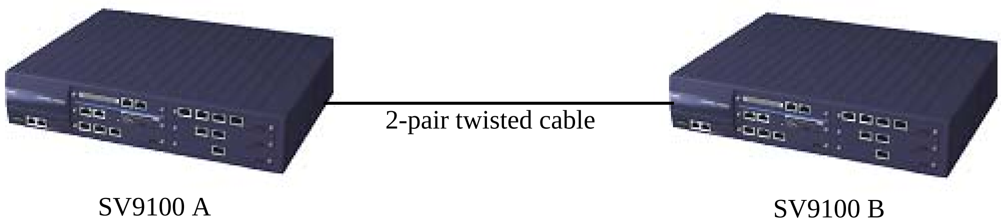
SECTION 5 ASPIRENET EXAMPLE

The examples in this section are only a guide to the configuration required on the SV9100 to install AspireNet Operation.

5.1 AspireNet - ISDN BRI

AspireNet Basic Rate ISDN is supported on all BRI PCB's (2, 4 or 8 BRIU). Each circuit of the BRIU PCB can be set independently to Trunk / S0 / AspireNet mode.
Each circuit set as AspireNet mode will require 2 of the AspireNet ports on the system, there are a total of 256 AspireNet ports available.

The example below shows two SV9100 systems, each has a BRIU card installed with one circuit set to Mode 4 for AspireNet . The two BRIU circuits are cross connected via a 2-pair twisted cable.
SV9100 A has extension numbers in the range 200-299, SV9100 B has 300-399.



SV9100 A	SV9100 B
Program 10-03 BRI PCB Setup 10-03-01 = Mode 4 10-03-03 = P-P 10-03-10 = Master 10-03-11 = 1	Program 10-03 BRI PCB Setup 10-03-01 = Mode 4 10-03-03 = P-P 10-03-10 = Slave 10-03-11 = 1
Program 11-01-01 System Numbering Dial 2x = 3 digit, Type 2 (Intercom) Dial 3x = 3 digit, Type 8 (Networking), ID=1	Program 11-01-01 System Numbering Dial 2x = 3 digit, Type 8 (Networking), ID=1 Dial 3x = 3 digit, Type 2 (Intercom)

More than one circuit required.
If more BRI circuits were added then Program 10-03-xx would copied for each new circuit.

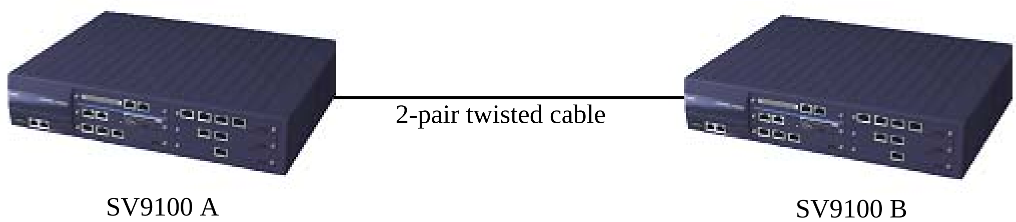
Master/Slave setting
The choice of Master/Slave of each circuit is not important in the example above as there is no ISDN Network connection to either of the systems.
If System A had an ISDN Network connection (BRI trunk or PRI trunk) then System A would be the Master for AspireNet circuits.
If both system A and System B have an ISDN Network connection then the setting of Master/Slave is not important.

When the AspireNet BRI circuit is connected to external equipment then ensure that the equipment is set to Master (Clock generator) or Slave (Clock receiver) accordingly.

5.2 AspireNet - ISDN PRI

AspireNet Primary Rate ISDN is supported on the standard PRTA Blade's. Each PRTA Blade can be set independently to Trunk / S0 / AspireNet mode.
When set to AspireNet a PRTA Blade will require 30 of the AspireNet ports on the system, there are a total of 256 AspireNet ports available.

The example below shows two SV9100 systems, each has a PRTA Blade installed which are set to Mode 4 for AspireNet . The two PRTA Blade's are cross connected via a 2-pair twisted cable.
SV9100 A has extension numbers in the range 200-299, SV9100 B has 300-399.



SV9100 A	SV9100 B
Program 10-03 PRI PCB Setup 10-03-01 = Mode 4 10-03-03 = CRC-4 ON 10-03-10 = Master 10-03-11 = 1	Program 10-03 PRI PCB Setup 10-03-01 = Mode 4 10-03-03 = CRC-4 ON 10-03-10 = Slave 10-03-11 = 1
Program 11-01-01 System Numbering Dial 2x = 3 digit, Type 2 (Intercom) Dial 3x = 3 digit, Type 8 (Networking), ID=1	Program 11-01-01 System Numbering Dial 2x = 3 digit, Type 8 (Networking), ID=1 Dial 3x = 3 digit, Type 2 (Intercom)

More than one circuit required.
If more PRI PCBs were added then Program 10-03-xx would copied for each new PCB.

CRC-4 Setting
The choice of CRC-4 on/off is not important in the example. When connected to external PRI equipment ensure that this option is set the same as the external equipment.

Master/Slave setting
The choice of Master/Slave of each circuit is not important in the example above as there is no ISDN Network connection to either of the systems.
If System A had an ISDN Network connection (BRI trunk or PRI trunk) then System A would be the Master for AspireNet circuits.
If both system A and System B have an ISDN Network connection then the setting of Master/Slave is not important.
When the AspireNet PRI circuit is connected to external equipment then ensure that the equipment is set to Master (Clock generator) or Slave (Clock receiver) accordingly.

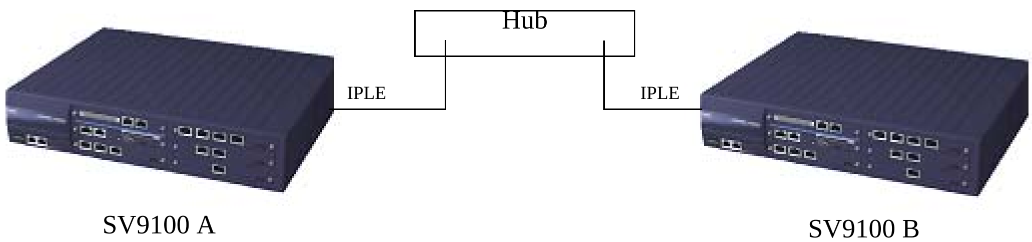
Limit the Quantity of B-Channels available
Use Program 10-32-01 to limit the quantity of channels available.

5.3 AspireNet - IP

5.3.1 VoIP Networking Example

AspireNet IP requires the VOIPDB Blade and SV9100 Networking License. The VOIPDB Blade provides the speech resources.

The example below shows two SV9100 systems, each has a VOIPDB Blade installed.
The VOIPDB Blades are connected to a hub via a standard LAN patch cable.
SV9100 A has extension numbers in the range 200-299, SV9100 B has 300-399.



SV9100 A	SV9100 B
Program 10-12-09 IP Address of the IPLE = 172.16.0.10 Program 10-12-10 Subnet Mask = 255.255.0.0	Program 10-12-09 IP Address of the IPLE= 172.16.5.10 Program 10-12-10 Subnet Mask = 255.255.0.0
Program 10-19-01 DSP Resource is assigned for the Network. = 3	Program 10-19-01 DSP Resource is assigned for the Network. = 3
Program 10-20-01 TCP Port Setup Device 4 = 30000	Program 10-20-01 TCP Port Setup Device 4 = 30000
Program 10-27-01 Destination IP Address and port number for each ID ID 1 = IP Address 172.16.5.10 Procedure port must be set to 1730	Program 10-27-01 Destination IP Address and port number for each ID ID 1 = IP Address 172.16.0.10 Procedure port must be set to 1730
Program 11-01-01 System Numbering Dial 2x = 3 digit, Type 2 (Intercom) Dial 3x = 3 digit, Type 8 (Networking), ID=1	Program 11-01-01 System Numbering Dial 2x = 3 digit, Type 8 (Networking), ID=1 Dial 3x = 3 digit, Type 2 (Intercom)
Program 84-02-35 Enable Fast Start	Program 84-02-35 Enable Fast Start
Program 84-12-31 DTMF Relay Mode 1 = RFC2833 (default: VoIPDB)	Program 84-12-31 DTMF Relay Mode 1 = RFC2833 (default: VoIPDB)
Program 84-26-01 IP address for each DSP 172.16.0.20	Program 84-26-01 IP address for each DSP 172.16.5.20

Explanation of routing for IP AspireNet .

When a user places a call to a remote extension number the dialed digits are checked against the Numbering Scheme in Program 11-01. This will provide the Node ID number of the route to the remote system. The SV9100 will then find the destination IP Address by searching Program 10-27 for the destination IP Address for the given Node ID number.

Quantity of Voice Paths Available.

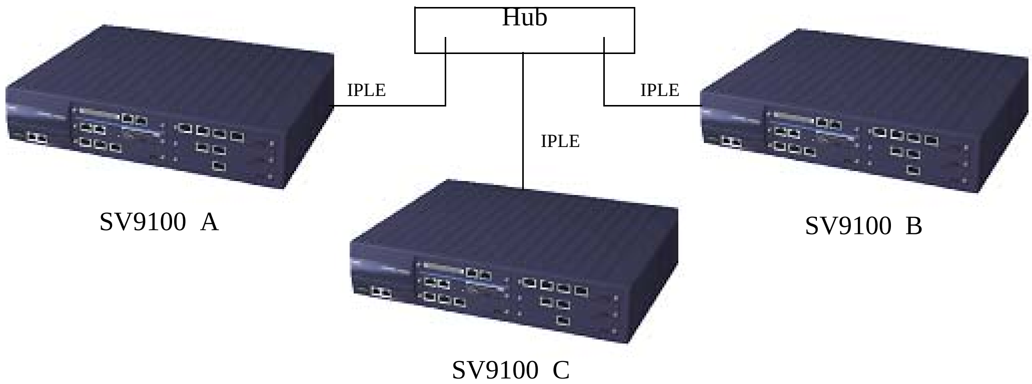
The quantity of voice calls that can be made at any one time are dependent on various factors.
IPLA Resource - A free IPLA resource must be available at each SV9100 system.
Program 10-19 - The mode of each VOIPDB resource can be configured, modes 0 (ICM/Trunk) or 3 (NTW) can be used by AspireNet calls.
IP Network Bandwidth Restrictions - These limitations are beyond the scope of the SV9100 . If there is not enough bandwidth available the call can not be maintained.

5.3.2 Multi-Site Networking Example

Multi-Site Networking is a network of three or more SV9100 systems. The network can consist of either ISDN or IP network connections.

The example below shows three SV9100 systems, each has a VOIPDB Blade installed. The VOIPDB Blades are connected to a hub via a standard LAN patch cable.

SV9100 A has extension numbers in the range 200-299, SV9100 B has 300-399, and SV9100 C has 400-499.



SV9100 A	SV9100 B	SV9100 C
Program 10-12-09 IP Address of the IPLE = 172.16.0.10 Program 10-12-10 Subnet Mask = 255.255.0.0	Program 10-12-09 IP Address of the IPLE = 172.16.5.10 Program 10-12-10 Subnet Mask = 255.255.0.0	Program 10-12-09 IP Address of the IPLE =172.16.10.10 Program 10-12-10 Subnet Mask = 255.255.0.0
Program 10-19-01 DSP Resource is assigned for the Network. = 3	Program 10-19-01 DSP Resource is assigned for the Network. = 3	Program 10-19-01 DSP Resource is assigned for the Network. = 3
Program 10-20-01 TCP Port Setup Device 4 = 30000	Program 10-20-01 TCP Port Setup Device 4 = 30000	Program 10-20-01 TCP Port Setup Device 4 = 30000
Program 10-27-01 Destination IP Address and port number for each ID ID 1 = IP Address 172.16.5.10 Procedure port must be set to 1730 ID 2 = IP Address 172.16.10.10 Procedure port must be set to 1730	Program 10-27-01 Destination IP Address and port number for each ID ID 1 = IP Address 172.16.0.10 Procedure port must be set to 1730 ID 2 = IP Address 172.16.10.10 Procedure port must be set to 1730	Program 10-27-01 Destination IP Address and port number for each ID ID 1 = IP Address 172.16.0.10 Procedure port must be set to 1730 ID 2 = IP Address 172.16.5.10 Procedure port must be set to 1730
Program 11-01-01 System Numbering Dial 2x = 3 digit, Type 2 (Intercom) Dial 3x = 3 digit, Type 8 (Networking), ID=1 Dial 4x = 3 digit, Type 8 (Networking), ID=2	Program 11-01-01 System Numbering Dial 2x = 3 digit, Type 8 (Networking), ID=1 Dial 3x = 3 digit, Type 2 (Intercom) Dial 4x = 3 digit, Type 8 (Networking), ID=2	Program 11-01-01 System Numbering Dial 2x = 3 digit, Type 8 (Networking), ID=1 Dial 3x = 3 digit, Type 8 (Networking), ID=1 Dial 4x = 3 digit, Type 2 (Intercom)
Program 84-02-35 Enable Fast Start	Program 84-02-35 Enable Fast Start	Program 84-02-35 Enable Fast Start

Program 84-12-31 DTMF Relay Mode 1 = RFC2833 (default: VoIPDB)	Program 84-12-31 DTMF Relay Mode 1 = RFC2833 (default: VoIPDB)	Program 84-12-31 DTMF Relay Mode 1 = RFC2833 (default: VoIPDB)
Program 84-26-01 IP address for each DSP 172.16.0.20	Program 84-26-01 IP address for each DSP 172.16.5.20	Program 84-26-01 IP address for each DSP 172.16.10.20

✎ *After programming, each system must be reset in order for the IP address changes to take affect.*

UNIVERGE® SV9100

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NEC Corporation
Issue 2.0