UNIVERGE NEC

UNIVERGE® SV9100

SIP Trunking Service Configuration Guide for Access Line

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Issue 1.0

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Communications Technology Group

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Configuring NEC SV9100 with Access Line SIP Trunking Service

Access Line

SECTION 1 NEC SV9100 AND ACCESS LINE SETUP GUIDE

1.1 This Guide and Related Documents

This guide was created to assist knowledgeable vendors with configuring the NEC SV9100 Communication Server with Access Line SIP Trunking Service. It provides sample entries for the required fields. The actual data is provided by Access Line when service is activated. Questions about software and hardware installation or other PBX configuration issues should be directed to NEC's National Technical Assistance Center (NTAC).

For complete details on using SIP trunks with the SV9100, refer to the SV9100 Networking Manual.

For complete details on using DID features, refer to the DID feature in the SV9100 Features and Specifications Manual.

For details about related hardware, refer to the SV9100 System Hardware Manual.

These manuals can be downloaded from NEC's National Technical Assistance Center (NTAC) web site. You must have a valid dealer ID to access the documents.

1.2 Access Line Account

Contact your Access Line representative.

1.3 SV9100 System Software

The SV9100 requires system software Version 1.70 or higher to use Access Line service.

1.4 Requirements

With the SV9100, a VoIP gateway daughter board is required in addition to licensing for IP (SIP) trunks.

A minimum of four IP (SIP) trunks are required due to the NEC Communications Server infrastructure setup.

The system software for the NEC Communications Server should be Version 1.70 or higher.

NEC recommends that the requirements and programming are completed with as much information as possible before scheduling an activation appointment with Access Line.

1.5 General Information

- Emergency 911/E911 Services Limitations and Restrictions - Although SIP Trunk carrier may provide 911/E911 calling capabilities, the SIP Trunk carrier does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer configuration guide will properly operate with Carrier to complete 911/E911 calls; therefore, it is the customer's responsibility to ensure proper operation with its equipment/software vendor.
- A SIP Trunk carrier services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when that E911/911 service may not be available. Such circumstances include, but are not limited to, relocation of the end-user's CPE, use of non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the customer's location in the automatic location information database.

1.6 Limitations

The following limitations apply:

- Some private IP network ranges conflict with SIP trunking service providers ranges. This can cause issues when connecting to the SIP trunking service provider. Private ranges reserved for the customer's LAN are:

  10.x.x.x
  192.168.0.x through 192.168.10.x
- SIP Diversion header not supported.
- SIP REFER not supported.

- Secondary SIP server for failover not supported.
- Network-based call forward, call transfer, sequential ringing, and simultaneous ringing not supported.
- The interop tested was completed with Non-Registration SIP Trunks, and SIP Profile 1.

SECTION 2 NEC PBX CONFIGURATION

This section provides information to NEC's solution providers and NEC Associates for configuring an NEC UNIVERGE SV9100 to connect to a Access Line SIP Trunk service provider, utilizing a **DYNAMIC** configuration.

2.1 Prerequisites

Before you configure the UNIVERGE SV9100, you must have the following information available.

2.1.1 SIP Trunking Information from Access Line

- ❐ Primary SIP Proxy Server IP Address.
- ❐ Number Plan, if applicable for the Point-to-Point Connection.
- ❐ Trunking DID(s)
  The DID(s) are forwarded to the Public WAN IP address(s), DNS or DNS SRV records of the PBX.

2.1.2 NEC UNIVERGE SV9100

- ❐ SV9100 CPU firmware Version 1.70 or higher
- ❐ GPZ-IPLE
- ❐ Digital, IP and TDM Telephones
- ❐ R1 Version License (0411)
- ❐ System Port License (0300)
- ❐ VoIP Resource License (5301)
- ❐ IP Trunk License (5001)

2.1.3 Installation Worksheet

Use the worksheet to record the information needed for setting up the SIP Trunking service.

Table 1 Installation Worksheet

WAN Side:	
Internet Access Type and Speed:	
WAN IP Address:	
WAN Subnet Mask:	
WAN Gateway IP Address:	

LAN Side:	
LAN IP Address for SIParator or EdgeMarc:	
LAN Subnet Mask:	
LAN IP Address for SV9100:	
VLAN ID:	

PBX Information:	
Model:	
Firmware Version:	
Number of SIP Trunk Licenses:	
Add-on Software Applications:	
Number of Users:	
Number of Concurrent Calls:	

Notes:

SECTION 3 SV9100 PROGRAMMING

When using Access Line as your SIP trunking service provider, the following programs must be changed for SIP trunking service.

When using PCPro or WebPro for programming, enabling an option may be a checkbox option rather than entering a '1' as in terminal programming.

3.1 Trunk Type / Slot Configuration

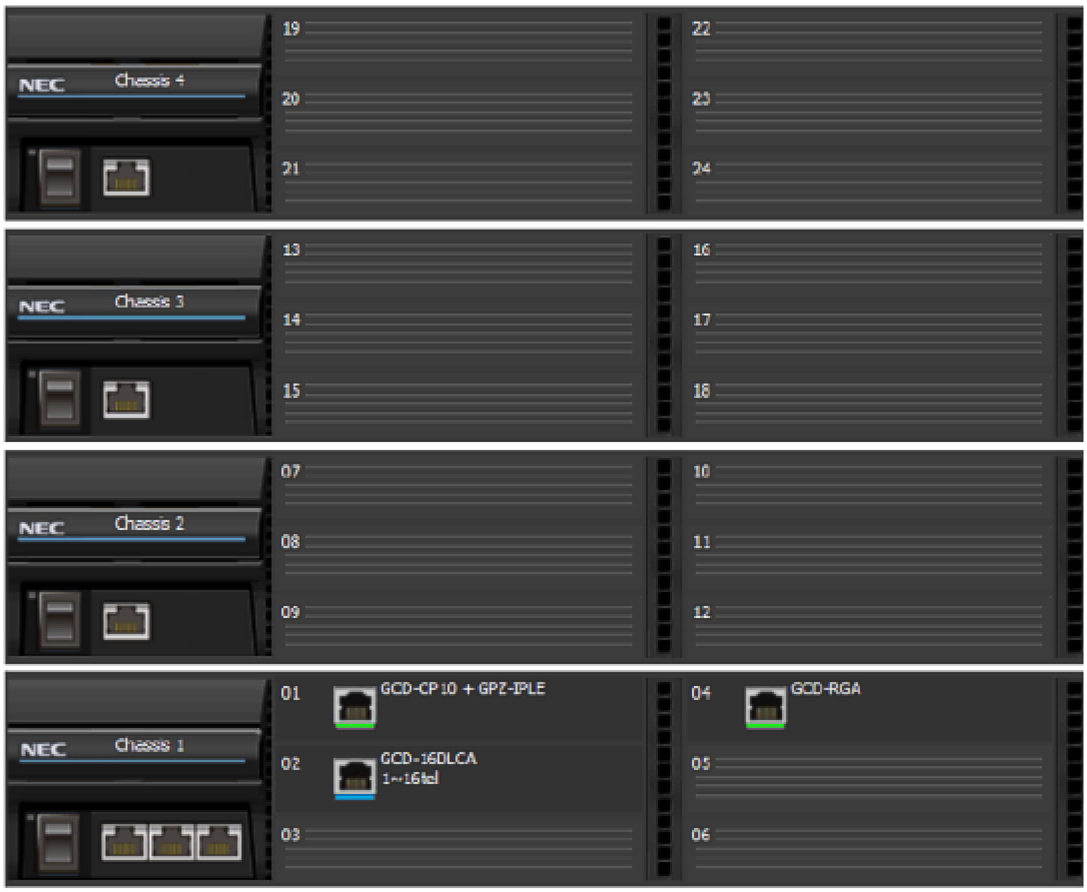


Figure 1 Blade Configuration

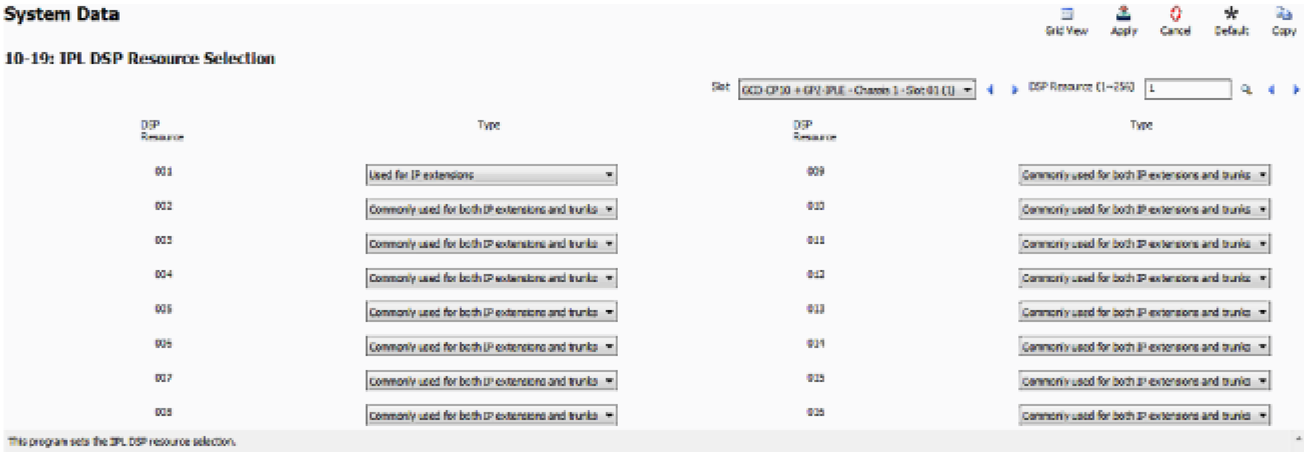


Figure 2 IPL DSP Resource Selection

10-19-01 : VOIP DSP Resource Selection
Specify the operating mode for the DSP resources (0=common use (extensions and trunks), 1=IP extensions only, 2=SIP trunks only, 3=Networking, 4=NetLink, 5=Blocked, 6=Common without Unicast Paging, 7=Multicast, 8=Unicast Paging).

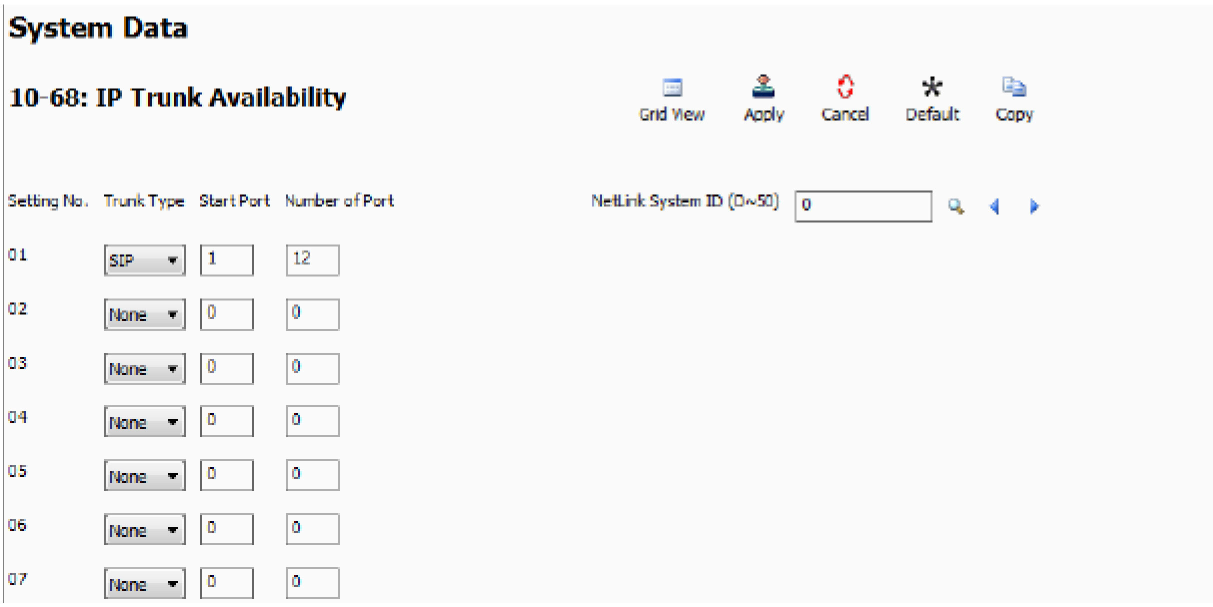


Figure 3 IP Trunk Availability

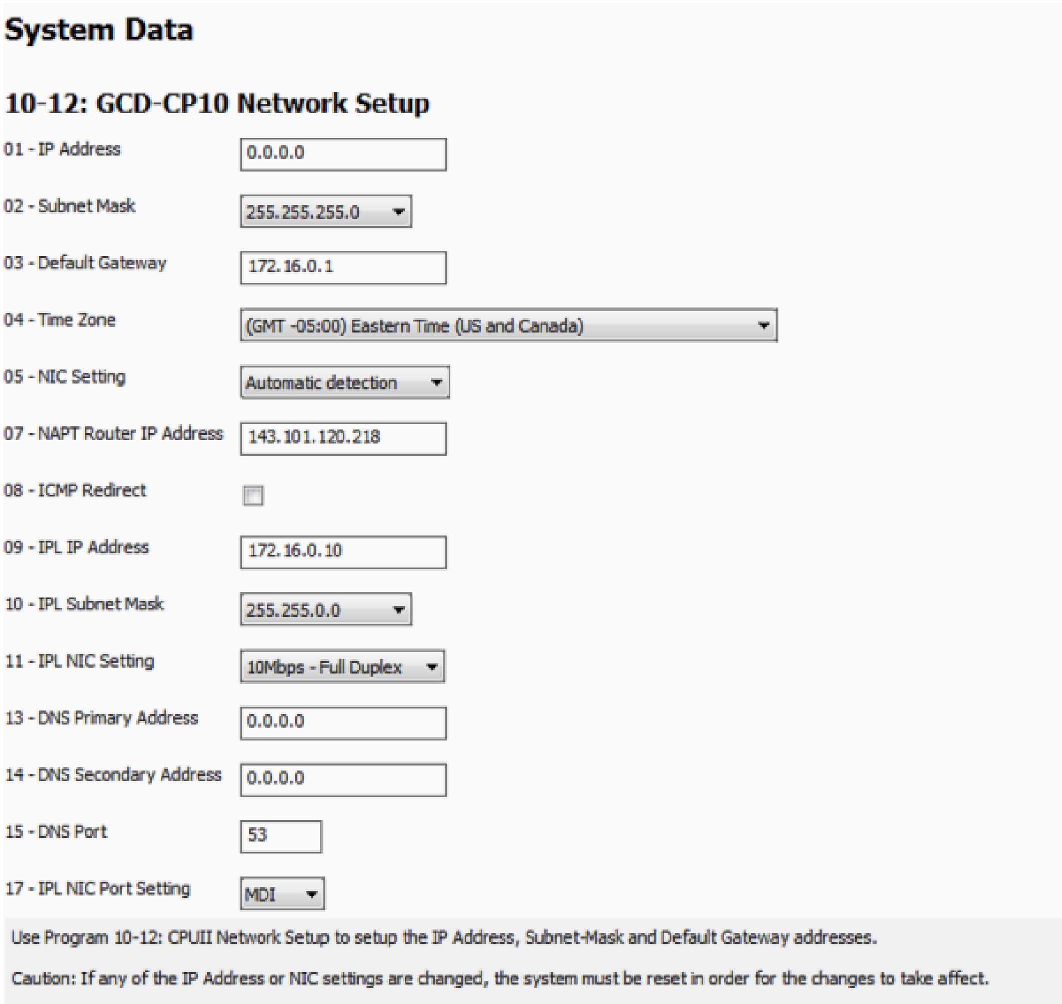
10-68-01 : IP Trunk Availability – IP Trunk Availability
Assign the trunk type as SIP.

10-68-02 : IP Trunk Availability – Start Port
Assign the Starting Port for the SIP Trunks.

10-68-03 : IP Trunk Availability – Number Port
Assign the number to SIP Trunk Ports.

3.2 GCD-CP10 Network Setup

Values shown are for example purposes only. Your actual IP values will be determined by your local LAN administrator.


System Data

10-12: GCD-CP10 Network Setup

01 - IP Address	0.0.0.0
02 - Subnet Mask	255.255.255.0
03 - Default Gateway	172.16.0.1
04 - Time Zone	(GMT -05:00) Eastern Time (US and Canada)
05 - NIC Setting	Automatic detection
07 - NAPT Router IP Address	143.101.120.218
08 - ICMP Redirect	
09 - IPL IP Address	172.16.0.10
10 - IPL Subnet Mask	255.255.0.0
11 - IPL NIC Setting	10Mbps - Full Duplex
13 - DNS Primary Address	0.0.0.0
14 - DNS Secondary Address	0.0.0.0
15 - DNS Port	53
17 - IPL NIC Port Setting	MDI

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

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

Figure 4 GCD-CP10 Network Setup

10-12-01 : GCD-CP10 Network Setup – IP Address
Set the LAN IP address for the system Ethernet port to 0.0.0.0

10-12-02 : GCD-CP10 Network Setup – Subnet Mask
Set the subnet mask for the system Ethernet port to be different than the subnet for the IPLE blade.

10-12-03 : CCD-CP10 Network Setup – Default Gateway
Set the default gateway for the IPLE blade.

<u>If a router or firewall is placed between the SIP Trunk Provider and SV9100</u>, you must also set the following programs:

✎ *All routing and forwarding is done by the Starbox Lite router, so NAPT should not be needed in the SV9100.*

10-12-07 : CD-CP00 Network Setup – NAPT Router IP Address
Set the WAN IP address of the NAT router behind the SV9100. NAT Router must also be enabled in PRG **10-29-21**.

10-12-09 : CD-CP00 Network Setup – IP Address
Select the IP address for the VoIP connection (default: 172.16.0.10). A static IP address is required.

The SV9100 must be reset in order for the change to take effect.

10-12-10 : CD-CP00 Network Setup – Subnet Mask
Select the Subnet Mask to be used by the VoIP server (default: 255.255.0.0).

3.3 VoIP DSP License Assignment

Values shown are for example purposes only. Your actual License quantity will be determined by the License File loaded to GCD-CP10.

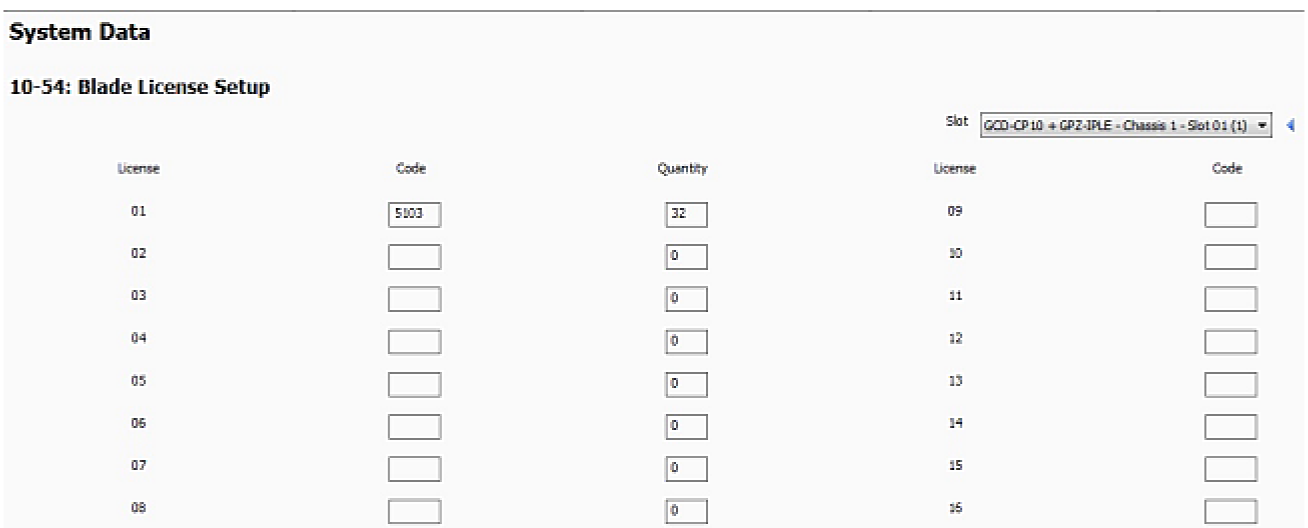


Figure 5 Blade License Setup

10-54-01 : Blade License Setup – Code
Assign License Code 5103 (VoIP DSP Channel)

10-54-02 : Blade License Setup – Quantity
Assign the quantity of VoIP DSP Channel Licenses (5103)

✎ *The License quantity can be found on Feature Activation Page.*

3.4 IPL DSP Basic Setup

Values shown are for example purposes only. Your actual IP values will be determined by your local LAN administrator.

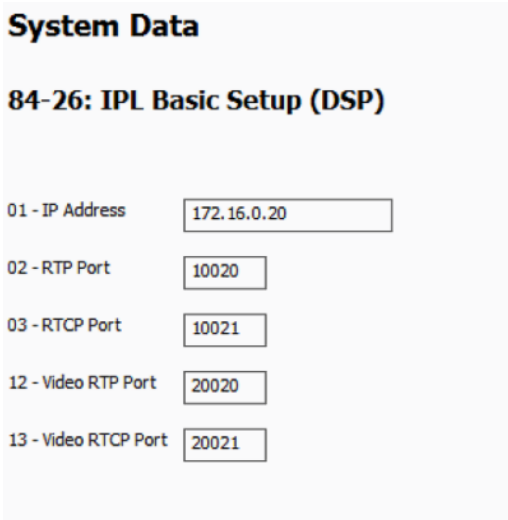


Figure 6 IPL DSP Basic Setup

Port Forwarding:
The Router will require port forwarding rules to be configured.

Port 5060 must be forwarded to the address entered in Program 10-12-09.
Port 5060 is not used for remote terminals - ports 5070 and 5080 are used instead. Port 5060 is only used for trunking so there are no issues with the possible fraudulent usage of unauthorized remote attempts to register remote terminals.

The ports used in Programs 84-26-02 and 84-26-03 must be forwarded to the IP address entered in Program 84-26-01.
The RTP/RTCP ports are forwarded to avoid possible one-way conversation which might occur on inbound calls. The Port Forwarding Range is determined by how many VoIP DSP Resources are licensed to the GCD-CP10. This information can found on the Feature Activation screen in WebPro, and is the same quantity that was entered in PRG 10-54 for feature code 5103.

Table 2 Port Table (UDP)

IPLE Licensed Channels	Begin Port	End Port
12	10020	10043
24	10020	10067
48	10020	10115
64	10020	10147
128	10020	10275
256	10020	10531

Table 3 Router Forwarding (Gateway Table)

IPLE	IP Address	RTP Port	RTCP Port	UDP
IPLE				

Example: Router configuration shown from the NEC InRouter/4300T Router

udp;143.101.120.218/255.255.255.0-10020>172.16.0.20-10020

udp;143.101.120.218/255.255.255.0-10021>172.16.0.20-10021

udp;143.101.120.218/255.255.255.0-10052>172.16.0.20-10052

udp;143.101.120.218/255.255.255.0-10053>172.16.0.20-10053

udp;143.101.120.218/255.255.255.0-10084>172.16.0.20-10084

udp;143.101.120.218/255.255.255.0-10085>172.16.0.20-10085

udp;143.101.120.218/255.255.255.0-10116>172.16.0.20-10116

udp;143.101.120.218/255.255.255.0-10117>172.16.0.20-10117

udp;143.101.120.218/255.255.255.0-5060>172.16.0.10-5060

3.5 SIP System Information Setup

Values shown are for example purposes only. Your actual values will be determined by your implementation team.

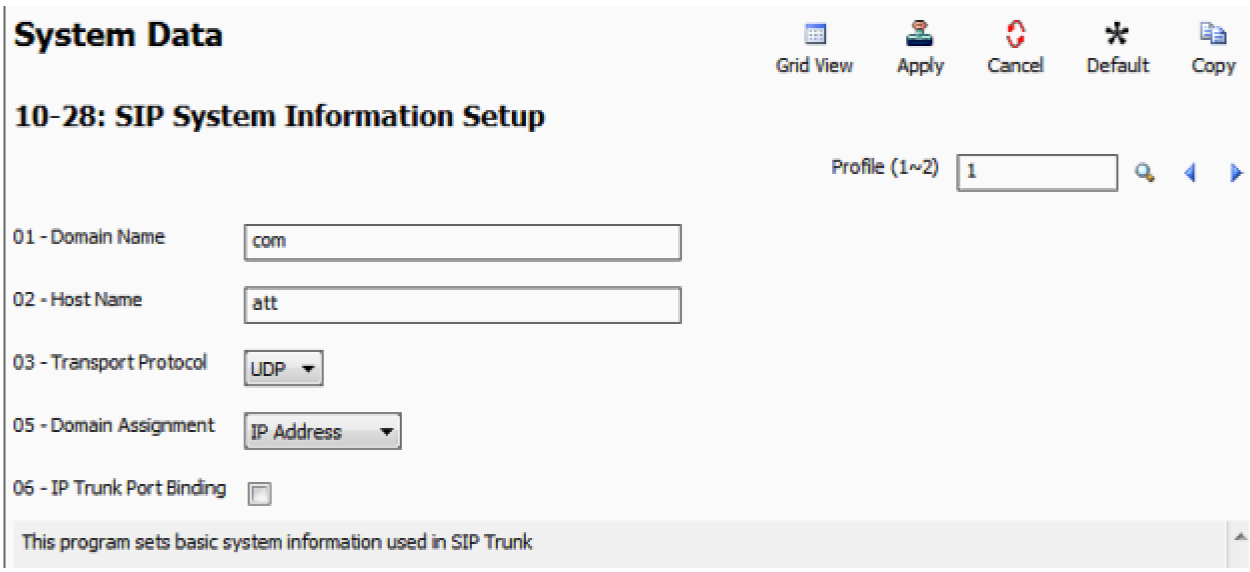


Figure 7 SIP System Information Setup

10-28-01 : SIP System Information Setup – Domain Name
Define the Domain name up to 64 characters. This information is specific to your market and is provided by your SIP Trunking Service Provider.

✎ *When configuring Domain name, the SIP service provider will supply the Proxy/Domain in the following manner - "Host Name" . "Domain Name" . The characters are normally separated by "." The characters **after** "." will be in the Domain Name.*

10-28-02 : SIP System Information Setup – Host Name
Define the Host name, up to 48 characters.

✎ *When configuring Host name, the SIP service provider will supply the Proxy/Domain in the following manner - "Host Name" . "Domain Name" . The characters are normally separated by "." The characters **before** "." will be in the Domain Name.*

10-28-03 : SIP System Information Setup – Transport Protocol
Define the Transport type. This option is always set to 0 (UDP).

10-28-05 : SIP System Information Setup – Domain Assignment
Determine the type of Domain Assignment. Set this entry to 0 (IP Address).

10-28-06 : SIP System Information Setup – IP Trunk Port Binding
Set this entry to 0 (Disable) to allow an incoming call to use the lowest port.

3.6 SIP Server Information Setup

Values shown are for example purposes only. Your actual values will be determined by your implementation team.

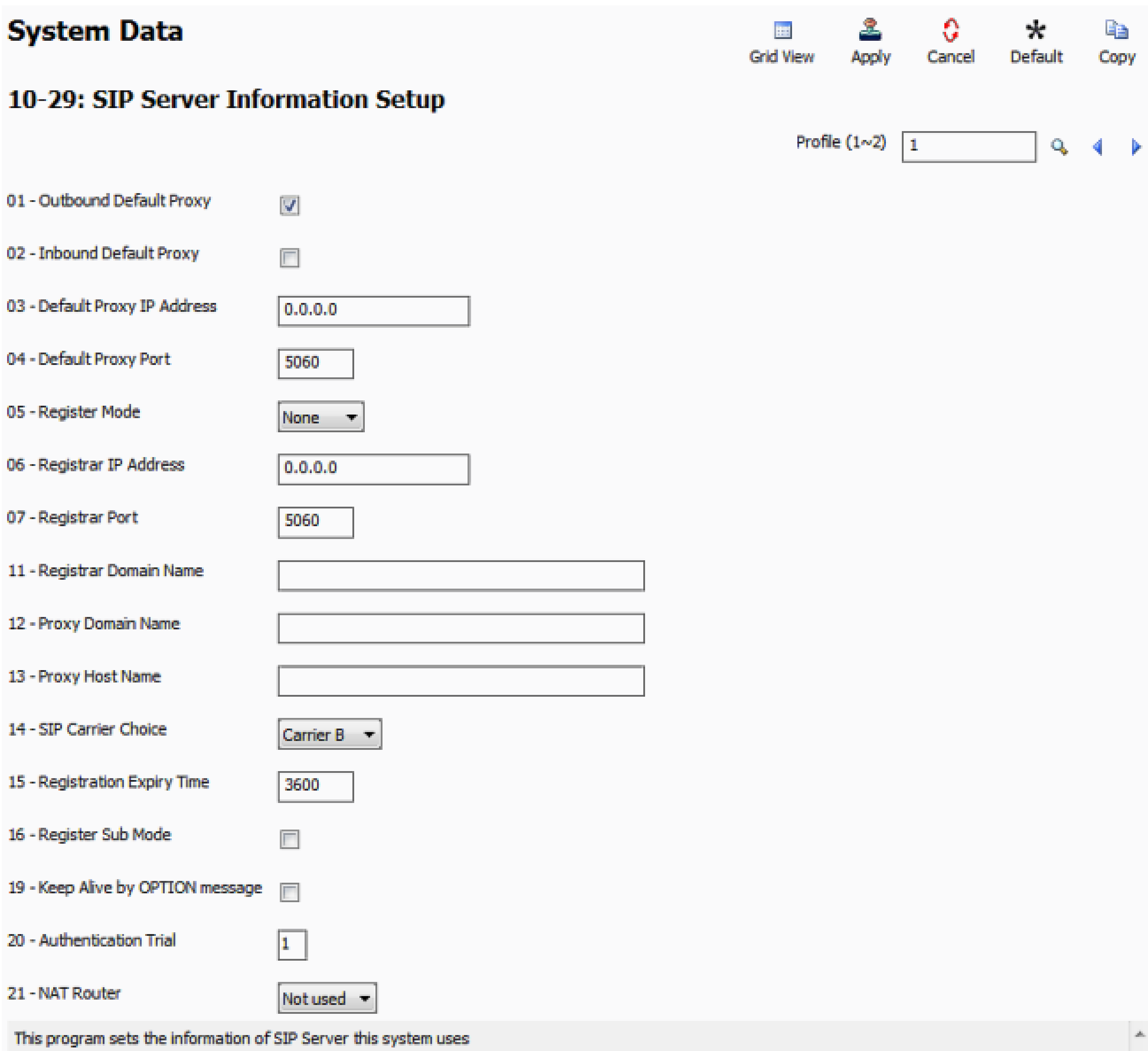


Figure 8 SIP Server Information Setup

10-29-01 : SIP Server Information Setup – Outbound Default Proxy
Enable (1) the SIP Outbound Proxy.

✎ *If entries are made in Program 10-29-xx for a SIP Server and the SIP Server is then removed or not used, the entries in Program 10-29-xx must be set back to their default settings. Even if 10-29-01 is set to .0. (off), the SV9100 will check the settings in the remaining 10-29 programs.*

10-29-03 : SIP Server Information Setup – Default Proxy IP Address
Define the SIP Trunk Service Provider Proxy IP Address. You may resolve the IP address of the Outbound Proxy by pinging the URL.

10-29-05 : SIP Server Information Setup – Registrar Mode
Set the Registrar Mode to 0 (None) with SIP trunking.

10-29-06 : SIP Server Information Setup – Registrar IP Address
Input the IP address of the SIP registrar (if given).

10-29-11 : SIP Server Information Setup – SIP Proxy Setup – Registrar Domain Name
Define the Registrar Domain Name. This information should be provided by your SIP service provider (128 characters maximum).

10-29-12 : SIP Server Information Setup – Proxy Domain Name
Enter the Domain name.

✎ *When configuring the Domain name, the SIP service provider will supply the Proxy/ Domain in the following manner - "Host Name" . "Domain Name" . The characters are normally separated by "." The characters **after** "." will be in the Domain Name.*

10-29-13 : SIP Server Information Setup – Proxy Host Name
Enter the Host name.

✎ *When configuring Domain name the SIP service provider will supply the Proxy/Domain in the following manner - "Host Name" . "Domain Name" . The characters are normally separated by "." The characters **before** "." will be in the Host Name.*

10-29-14 : SIP Server Information Setup – SIP Carrier Choice
Set the SIP Carrier Choice to 2 (Carrier B).

10-29-15 : SIP Server Information Setup – Registration Expiry Time
It is **important** to leave this automatic re-registration time to be 3600 seconds so that the Access Line network does not get flooded.

10-29-16 : SIP Server Information Setup – Register Sub Mode
Unchecking the Register Sub Mode (setting it to "off") will allow all trunk calls to be routed based on routing policies.

10-29-21 : SIP Server Information Setup – NAT Router
Enable this Program if the SV9100 resides behind a NAT router.

3.7 SIP Trunk Registration Information

Values shown are for example purposes only. Your actual values will be determined by your implementation team.

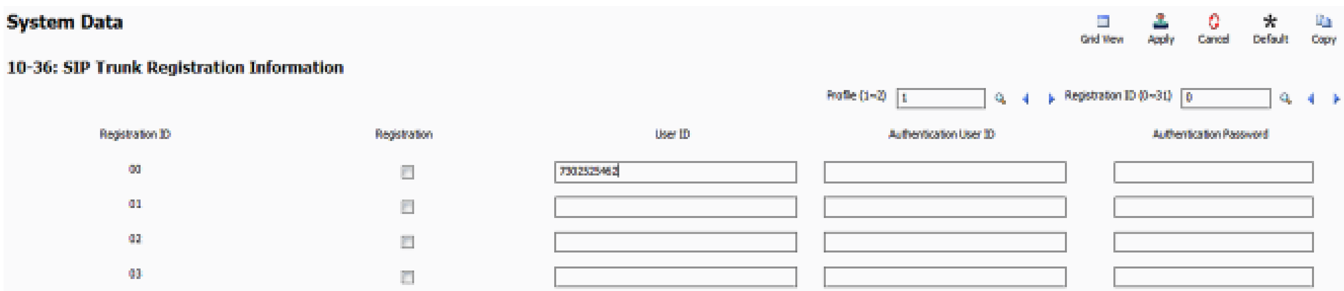


Figure 9 SIP Trunk Registration Information

10-36-1: SIP Trunk Registration Information – Registration
Select whether Registration is enabled/disabled.

10-36-2: SIP Trunk Registration Information – User ID
Enter the NexVortex User ID provided by your SIP Service Provider. This is typical your 10 digit billing number.

10-36-3: SIP Trunk Registration Information – Authentication ID
If required enter the NexVortex Authentication ID. This Information is provided by your SIP Service Provider.

10-36-4: SIP Trunk Registration Information – Authentication Password
If required enter the NexVortex authentication password. This Information is provided by your SIP Service Provider.

3.8 IP System Interconnection Setup

Values shown are for example purposes only. Your actual values will be determined by your implementation team.

Figure 10 IP System Interconnection Setup

10-23-01 : System Interconnection
Enable interconnection to the SIP Server.

10-23-02 : IP Address
Enter the IP Address of the SIP Server.

10-23-04 : Dial Number
Enter the digits to be sent to the SIP Server on an outbound call.

10-23-06 : SIP Profile
Select Profile 1 or Profile 2.

3.9 Calling Party Information (Trunk)

Caller ID - In the Invite message there are two fields that can have caller ID. One field is the “SIP From Address” and the other field is “SIP Display Info”. If both of these fields are left blank the call will not complete.

Below is an example of a SIP Invite Message with outbound CID.

From “2142622000”<sip:test@172.16.0.100>

14-12-01 : SIP Register ID Setup for IP Trunks
On a per trunk basis, you can choose a SIP register ID of 0~31. If the ID is left to 0, the “SIP from Address” would not be assigned on a per trunk basis. If set to 1~31, it then looks at command 10-36-02 to populate the “SIP from Address” field.

14-12-02 : SIP Register ID Setup for IP Trunks
This is for SIP trunks to the provider for inbound purposes. If 10-28-06 (Trunk port Binding) is enabled, inbound calls map to the trunk. If you want to create a hunt group when trunk port binding is enabled, set multiple trunks to the same pilot and then define that number in 10-36.

10-36-02 : SIP Trunk Registration Information
Per registration ID 1~31 you can assign what will be populated in the “SIP from Address” field.

15-16-01 : SIP Register ID Setup for Extensions
Per station you can choose a SIP register ID of 1~31. If left blank the “SIP from Address” would not be assigned on a per station basis. If assigned, it will look at Program 10-36-02 to populate the “SIP from Address” field. This takes priority over command 14-12-01.

10-36-2 : SIP Trunk Registration Information – User ID
This is the default “Display Info” and “From Address” if either of these fields is blank what is assigned in this command will be inserted. This setting has the lowest priority and if any of the next commands are set they will be sent out instead of this command.

3.10 Class of Service Options (Outgoing Call Service)

Values shown are for example purposes only. Your actual values will be determined by your implementation team.

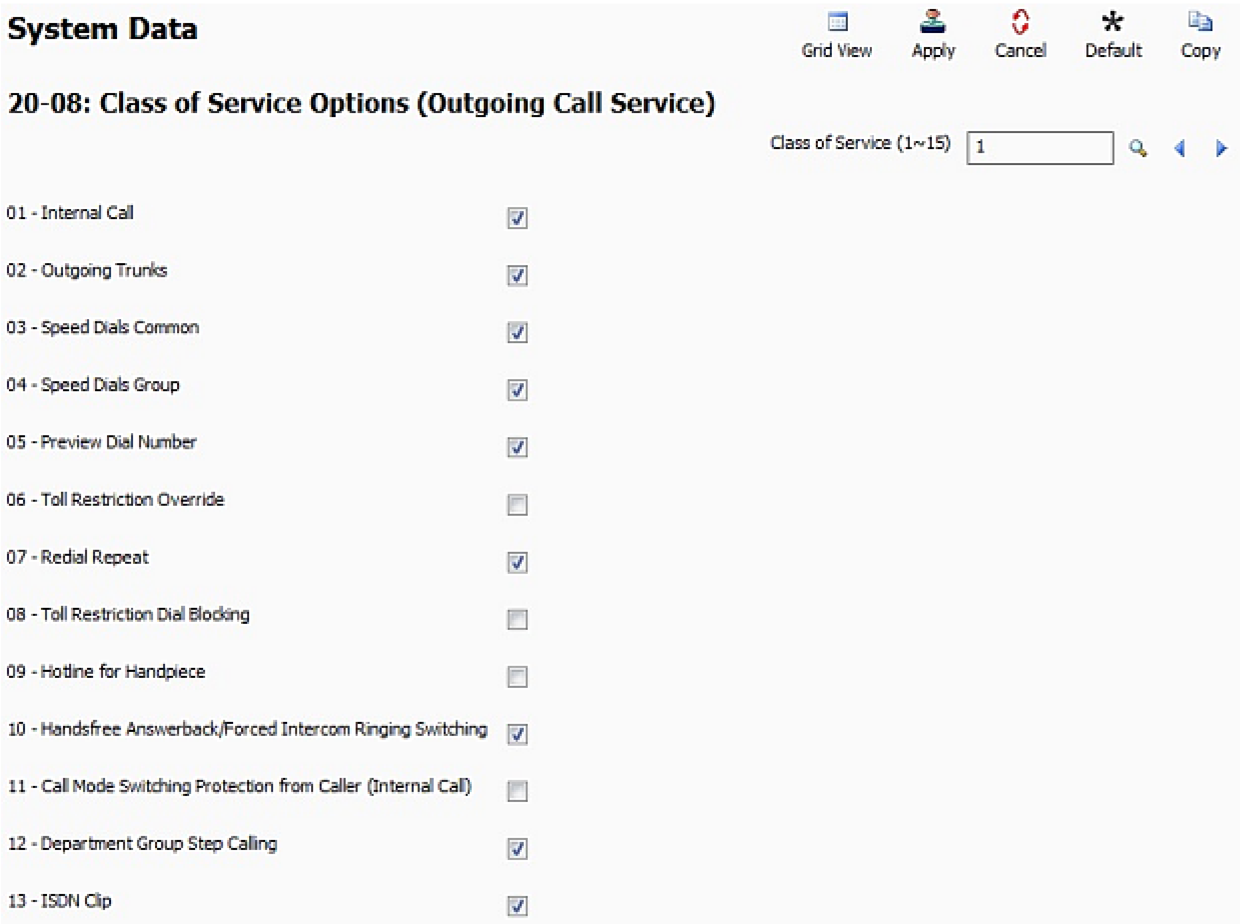


Figure 11 Class of Service Options (Outgoing Call Service)

20-08-13 : Class of Service Options (Outgoing Call Service) – ISDN Clip
This needs to be turned ON per COS, if you are trying to send any information on a per station basis. If turned OFF, it will still send the trunk information if set.

20-09-02: Class of Service Options (Incoming Call Service) Caller ID Display
This needs to be turned ON per COS, if you want to receive caller ID.

3.11 IP Trunk Calling Party Number Setup

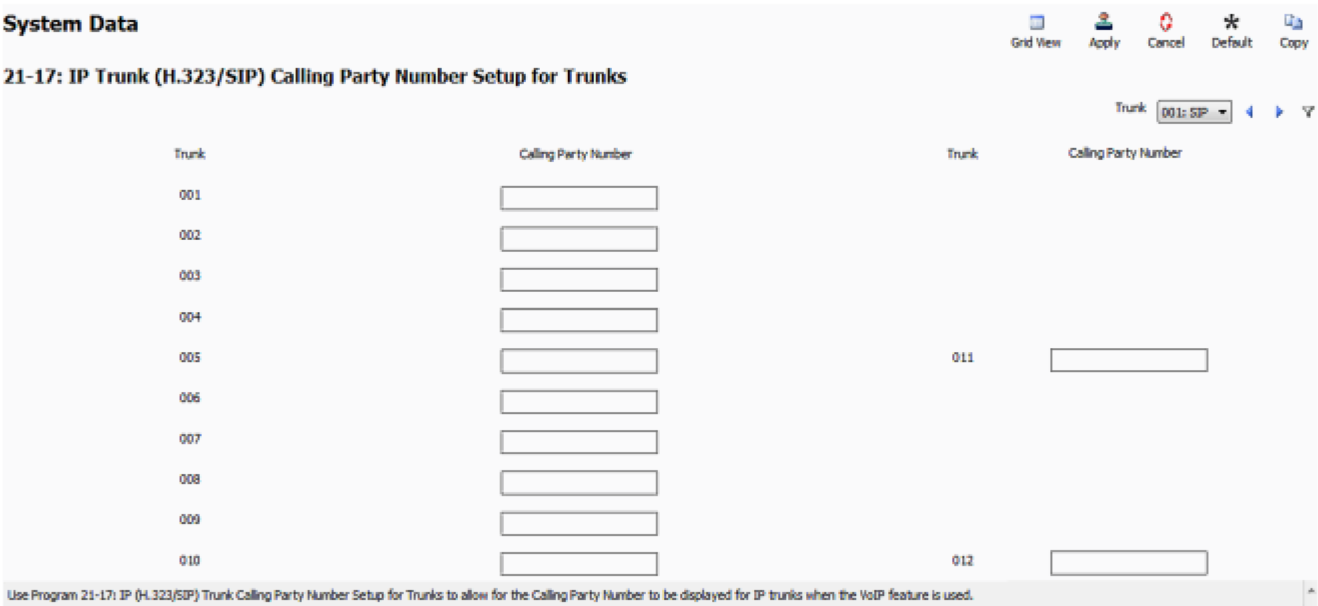


Figure 12 IP Trunk (H.323/SIP) Calling Party Number Setup for Trunks

21-17-01: Calling Party Number Setup for Trunks
On a per trunk basis this populates the “**SIP Display Info**” field. If a station has a setting in 21-19-01, it will override this field.

3.12 IP Trunk (SIP) Calling Party Number Setup for Extensions

Values shown are for example purposes only. Your actual values will be determined by your implementation team.

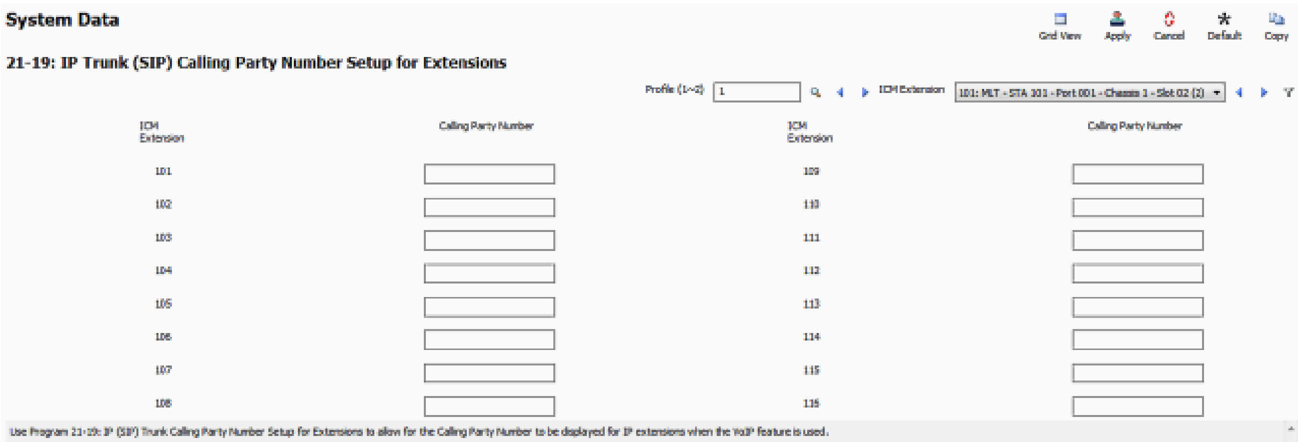


Figure 13 Trunk (SIP) Calling Party Number Setup for Extensions

21-19-01 : IP Trunk (SIP) Calling Party Number Setup for Extensions
On a per station basis this populates the "**SIP Display Info**" field.This setting has the highest priority.

This program is used to assign the Calling Party Number for each extension (Entries: 1~0, *, #). The assigned number is sent to the SIP Trunking Service Provider when the caller places an outgoing call. If the Calling Party Number is assigned by both Program 21-17 and 21-18/21-19, then the system uses the data in Program 21-18/21-19. Do not use Program 21-13 for SIP. This entry must be a 10-digit DID associated with the SIP Trunking Service Provider Account. DID numbers are provided by your SIP Trunking Service Provider Coordinator.

3.13 DID (TN to ext map)

Values shown are for example purposes only. Your actual values will be determined by your implementation team.

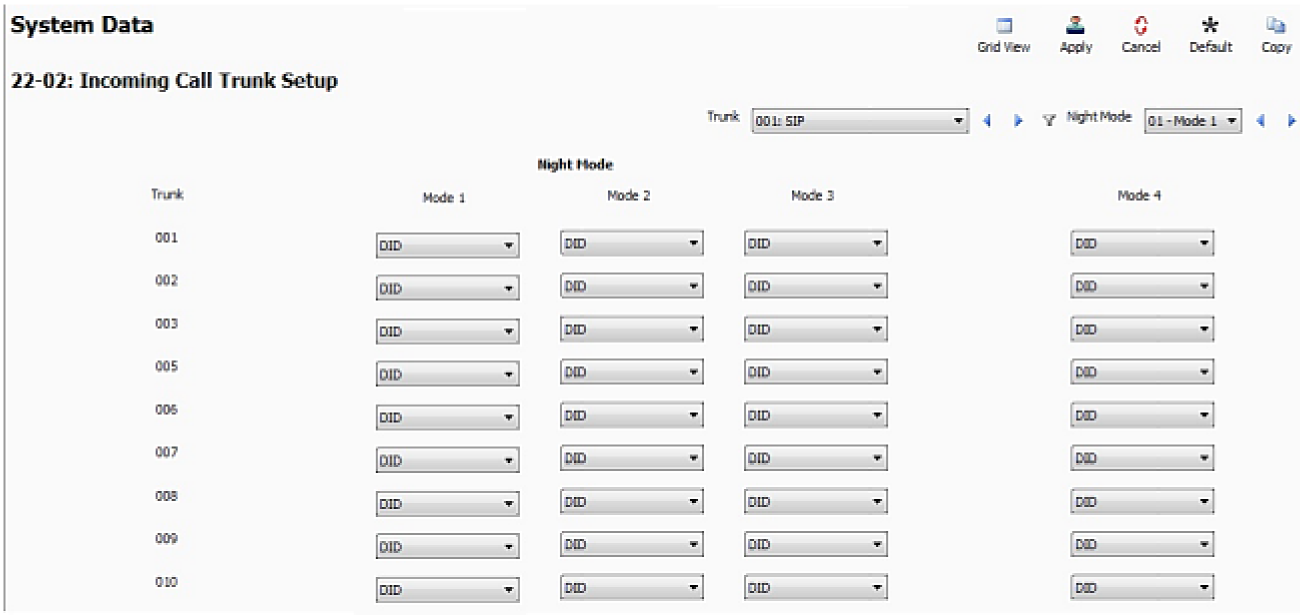


Figure 14 Incoming Call Trunk Setup

22-02-01 : Incoming Call Trunk Setup
Define the SIP trunks as type 3 (DID). In addition to the SIP trunk programming, refer to the DID feature in the SV9100 Features and Specifications Manual for additional DID programming (e.g., 14-05, 22-04, 22-09, 22-10, 22-11, 22-12, 22-13, 22-17, 34-01).

3.14 SIP Trunk CODEC Setup

Values shown are for example purposes only. Your actual values will be determined by your implementation team.

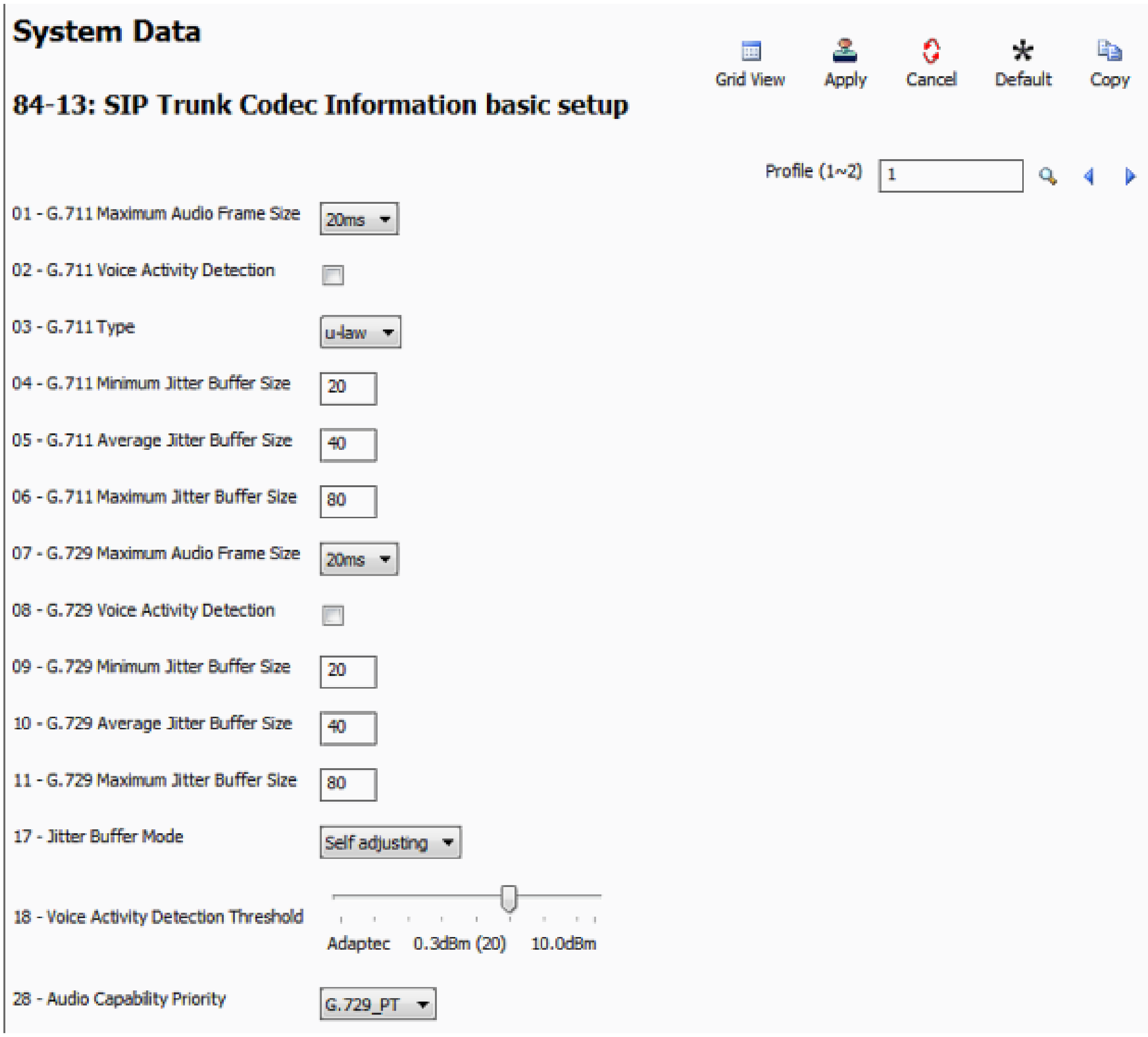


Figure 15 SIP Trunk Codec Information Basic Setup

84-13-28 : SIP Trunk CODEC Setup – Audio Capability Priority
Set the priority to G.729_PT

3.15 ToS Setup

Values shown are for example purposes only. Your actual values will be determined by your implementation team.

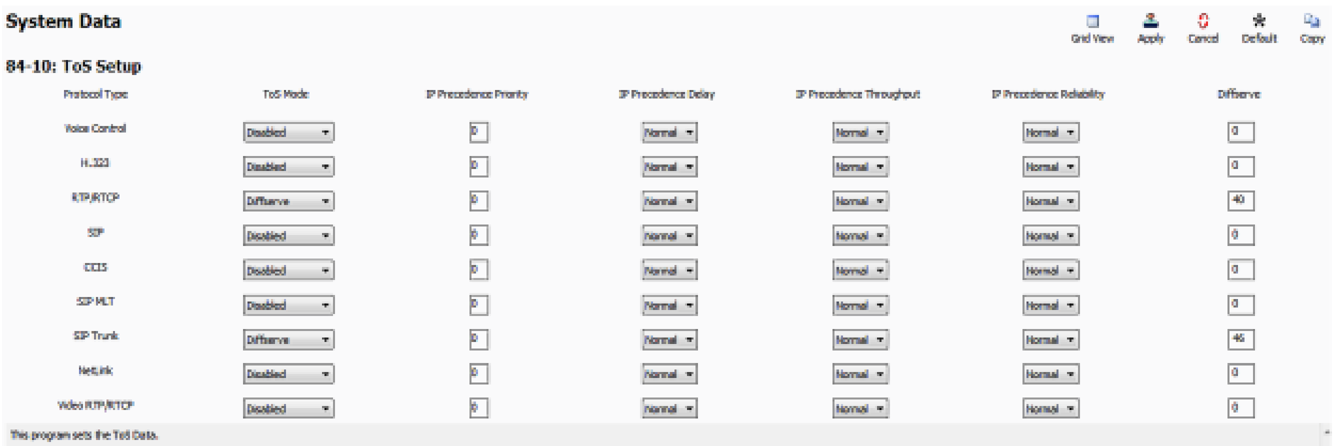


Figure 16 ToS Setup

84-10-01 : ToS Setup – ToS Mode
For the RTP/RTCP (Protocol type 5) and SIP Trunk (Protocol type 9), set the ToS Mode to “2” (Diffserv).

The SV9100 must be reset in order for the change to take effect.

84-10-07 : ToS Setup – Priority (Diffserv)
For each of the following protocol types, set the following priorities:
RTP/RTCP (Protocol type 5): **Priority 40**.
SIP Trunk (Protocol type 9): **Priority 46**.

The SV9100 must be reset in order for the change to take effect.

3.16 SIP Trunk Basic Setup

Values shown are for example purposes only. Your actual values will be determined by your implementation team.

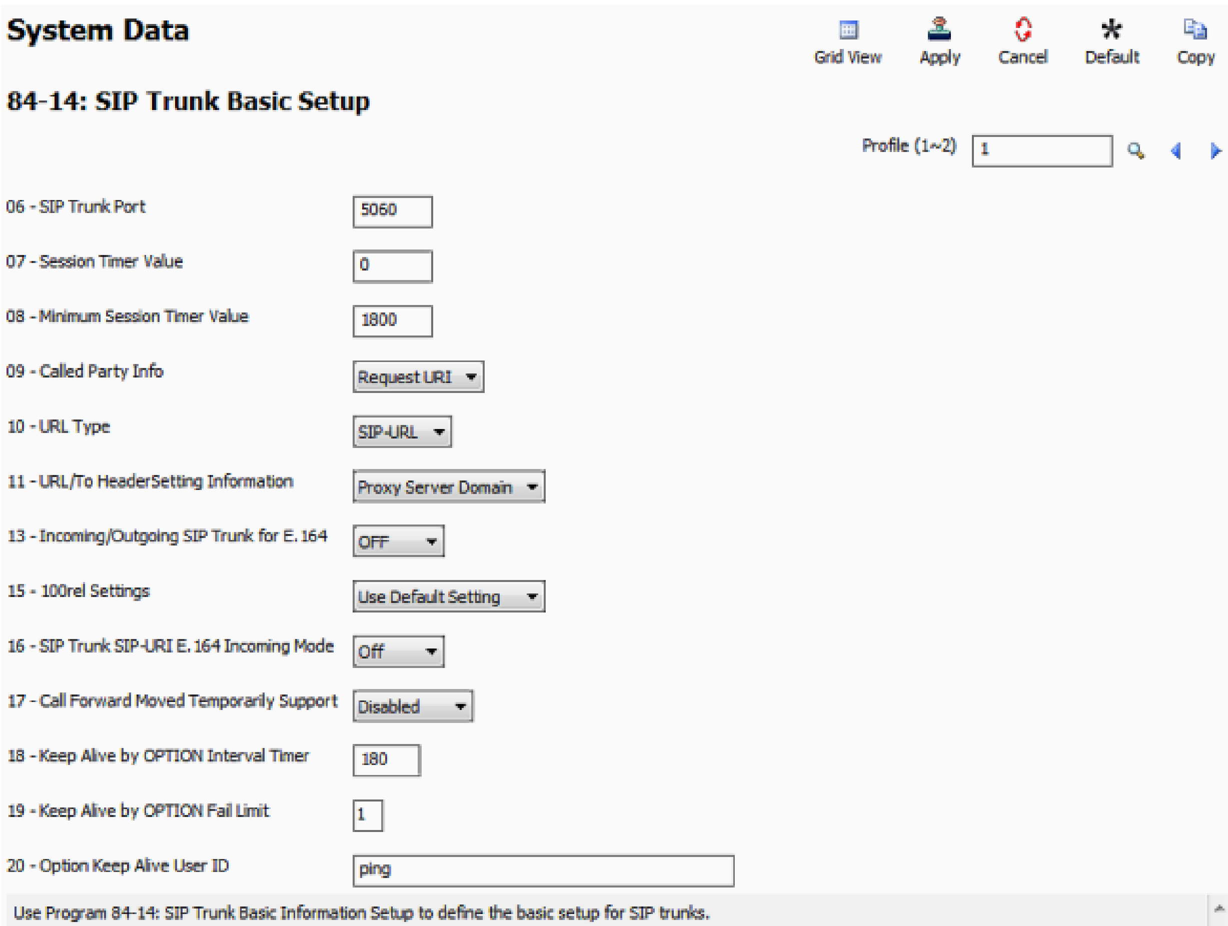


Figure 17 SIP Trunk Basic Setup

84-14-11 : SIP Trunk Basic Setup – URL/TO Header Setting Information
Set this program to Proxy Server Domain.

Changes within this program require the SV9100 be reset in order for the change to take effect.

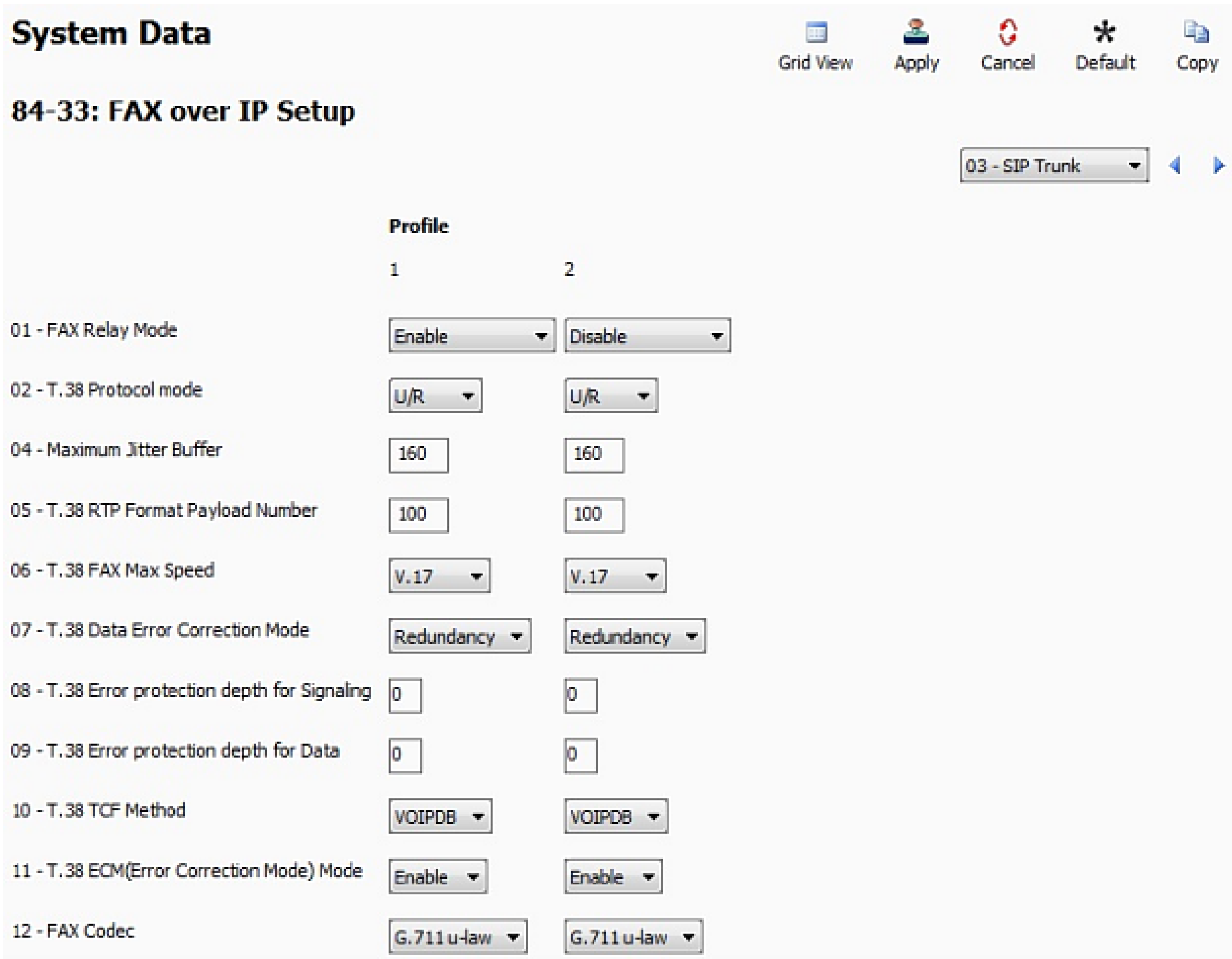


Figure 18 FAX over IP Setup

84-33-01: FAX over IP Setup – Fax Relay Mode
Set this Program to Enable for SIP Trunk.

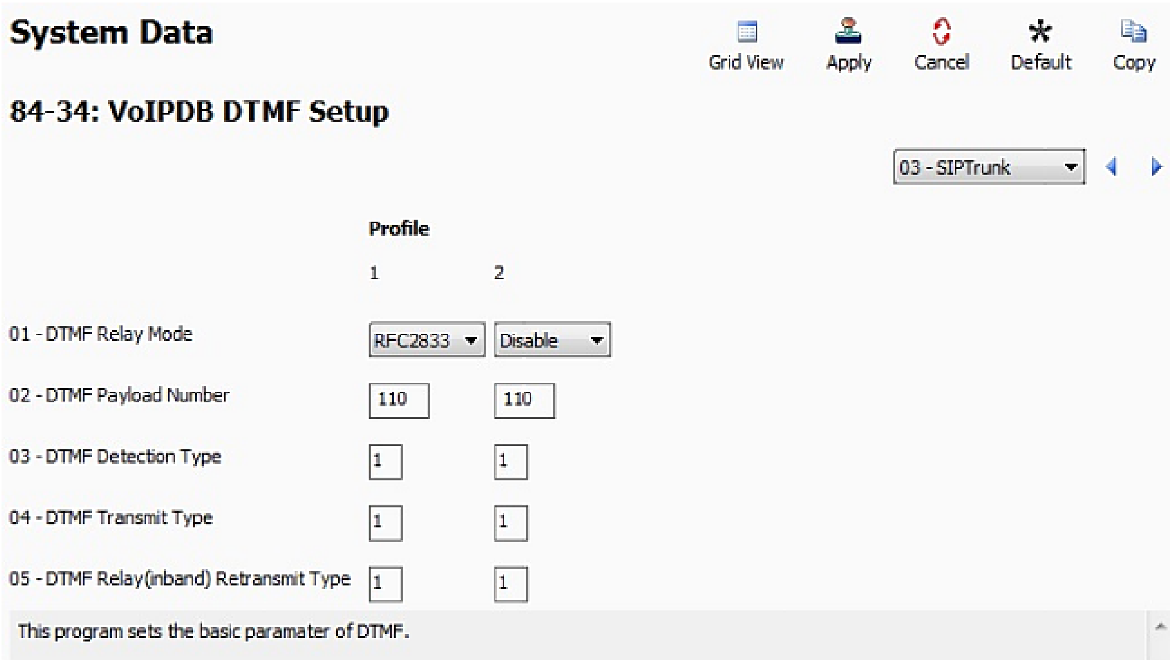


Figure 19 VoIPDB DTMF Setup - DTMF Relay Mode

84-34-01: VoIPDB DTMF Setup – DTMF Relay Mode
Set this to RFC2833 for SIP Trunk.

SECTION 4 INITIAL TESTING AND TROUBLESHOOTING

To confirm that the system is correctly set, perform the following tests:

✎ *If you run into an issue with any of these tests, refer to Table 4 Troubleshooting Guide. Test an outgoing call to a local number. Check for ringback, 2-way audio and quality.*

1. Test an outgoing call to a long distance number. Check for ringback, 2-way audio and quality.
2. Test an outgoing call to an international number. Check for ringback, 2-way audio and quality.
3. Test a outgoing call lasting more than 15 minutes.
4. Test multiple call concurrences on outgoing calls. Setup multiple calls to PSTN.
5. Test an outgoing call to an Operator ‘0’.
6. Test an outgoing call to directory assistance ‘411’.
7. Test a 911 call.

   Identify to the operator that this is a TEST!

8. Test an incoming call to an internal DID. Check for ringback, 2-way audio and quality.
9. Test an incoming call to an auto-attendant. Check DTMF and audio quality.
10. Test transferring calls off-site.
11. Test an outgoing call to an auto-attendant and verify DTMF.

Table 4 Troubleshooting Guide

Issue	Cause	Remedy
No Calls IN/Out	❍ Router Configuration	❍ Check Router Configuration
	❍ NEC Configuration	❍ Check NEC Configuration
	❍ Unqualified IP Address	❍ Note WAN IP Address and Contact Provider
No Calls Out	❍ NEC Configuration	❍ Check NEC Configuration
	❍ Unqualified IP Address	❍ Note WAN IP Address and Contact Provider
No Calls In	❍ NEC Configuration	❍ Check NEC Configuration
	❍ Unqualified IP Address	❍ Note WAN IP Address and Contact Provider
One-Way Audio	❍ NEC Configuration	❍ Check NEC Configuration
Echo	❍ Excessive Delay	❍ Check LAN and WAN for high latency
	❍ Echo Cancellation Issue	❍ Check Echo settings and/or consult Access Line
Call Dropping	❍ Internet Access Issues	❍ Call Internet Access Provider
	❍ Extreme Latency on LAN	❍ Check Latency on LAN
	❍ SIP issue	❍ Contact Provider
Static or HUM on Phones	❍ Power issue	❍ Check power if using AC, should not be issue in PoE
Missing Parts of Words	❍ Packet Loss or Latency on LAN	❍ Check LAN
	❍ Packet Loss or Latency on WAN	❍ Check with Internet Access Provider
	❍ Jitter Buffer Configuration	❍ Check with NEC