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UNIVERGE

NEC

UNIVERGE® SV8100

SIP Trunking Service Configuration Guide for Cable Vision Optimum Business SIP Trunk Utilizing the Edgemarc 4552 SIP Trunk Adapter

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Configuring NEC SV8100 with Optimum Business SIP Trunking Service

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Optimum Business

SECTION 1 NEC SV8100 AND OPTIMUM BUSINESS SETUP GUIDE

1.1 This Guide and Related Documents

This guide was created to assist knowledgeable vendors with configuring the NEC SV8100 Communication Server with Optimum Business' SIP Trunking service. It provides sample entries for the required fields. The actual data is provided by Optimum Business 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 SV8100, refer to the SV8100 Networking Manual.

For complete details on using DID features, refer to the DID feature in the SV8100 Features and Specifications Manual.

For details about related hardware, refer to the SV8100 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 Optimum Business SIP Trunk Account

Please follow the instructions in the Optimum Business SIP Trunk Setup Guide. The SetuP Guide was left by the Optimum Business technician at installation. If you do not have the SetuP Guide, please go to http://www.optimumbusiness.com/SIP to download a copy.

1.3 SV8100 System Software

The SV8100 requires system software 4.00 or higher to use Optimum Business' service.

1.4 Requirements

With the SV8100, 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 4.00 or higher.

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

✎ *The Optimum Business SIP is available with a minimum of four sessions and a maximum of 24 sessions.*

SECTION 2 NEC PBX CONFIGURATION

This section provides information to NEC's solution providers and NEC Associates for configuring an NEC UNIVERGE SV8100 to connect to a Optimum Business SIP Trunk service provider, utilizing a **DYNAMIC** (Registration mode) and **STATIC** (Non-registration mode) configuration.

2.1 Prerequisites

Before you configure the UNIVERGE SV8100, you must have the following information available.

2.1.1 Optimum Business SIP Trunk Adapter

❒ Primary SIP Proxy Server IP Address

✎ *Consult with your network administrator to designate a private IP address assigned to the Optimum Business SIP Trunk Adapter. Refer to step 2 of the Optimum Business SIP Trunk Setup guide for more information on how this address is assigned.*

2.1.2 NEC UNIVERGE SV8100

❒ SV8100 CPU firmware version 4.00 or higher

❒ IPLA-R UNIT (PZ-32IPLA, PZ-64IPLA or PZ-128IPLA)

- ❒ SIP Trunking License (minimum of four licenses)
- ❒ Digital, IP and TDM Telephones

2.1.3 Installation Worksheet

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

Table 1-1 Installation Worksheet

LAN Side:	
LAN IP Address for Optimum Business SIP Trunk Adapter (Edgemarc 4552):	
LAN Subnet Mask:	
LAN IP Address for SV8100:	
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 SV8100 PROGRAMMING

When using Optimum Business 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

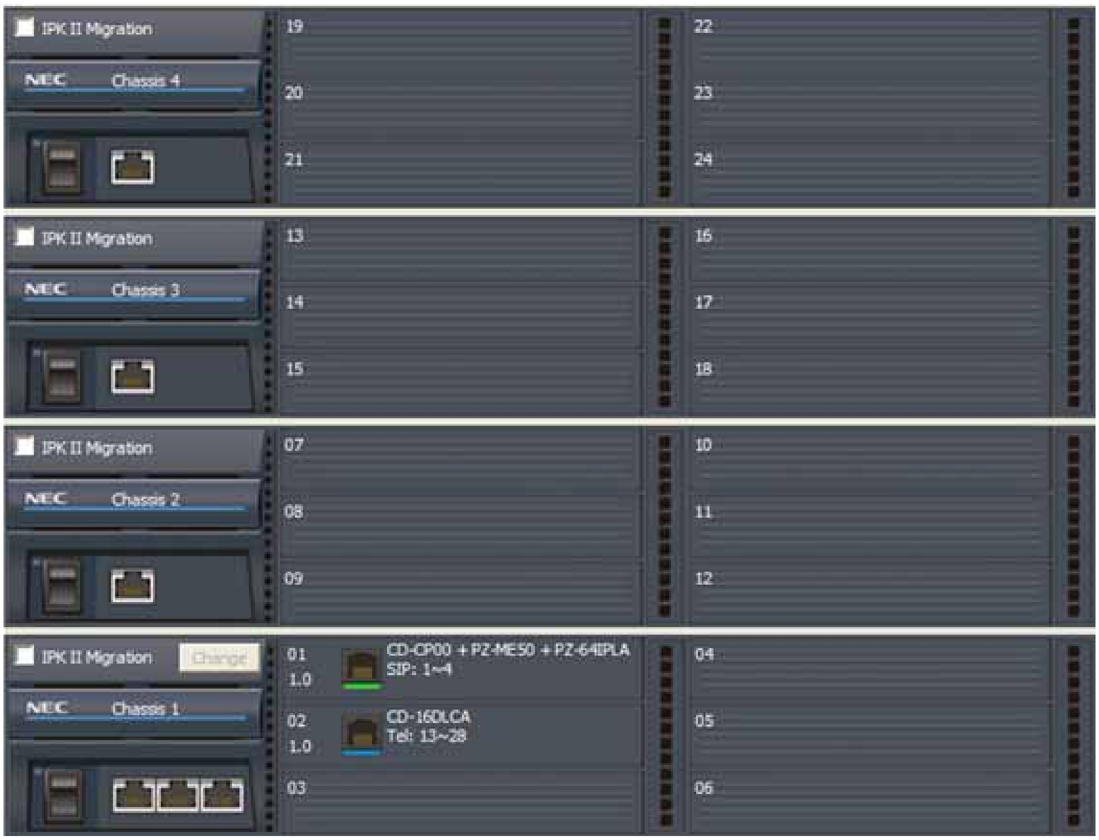

Figure 1-1 Blade Configuration

Figure 1-2 IPLA Configuration

10-03-02: Blade Setup, for IPLA (VoIPDB)

Define the trunks to be used for SIP trunks as 1 (SIP).

Figure 1-3 IPLA 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=CCIS, 4=NetLink, 5=Blocked, 6=Unicast, 7=Multicast, 8=Paging).

Figure 1-4 IP Trunk Availability

10-40-01 : IP Trunk Availability – IP Trunk Availability
Turn this option "on".

10-40-02 : IP Trunk Availability – IP Trunk Port Count
Select the number of trunks being used.

3.2 CD-CP00 Network Setup

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

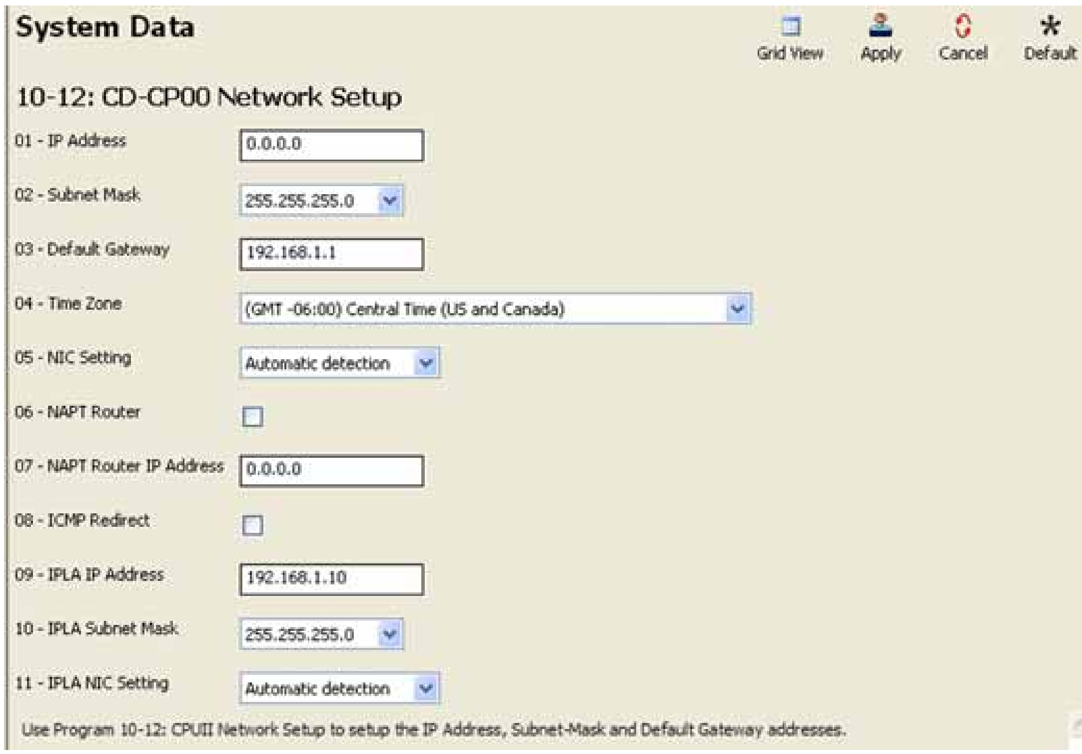

Figure 1-5 CD-CP00 Network Setup

10-12-01 : CD-CP00 Network Setup – IP Address
Set the LAN IP address for the system ethernet port to 0.0.0.0

10-12-02 : CD-CP00 Network Setup – Subnet Mask
Set the subnet mask for the system ethernet port to be different than the subnet for the IPLA blade.

10-12-03 : CD-CP00 Network Setup – Default Gateway
Set the default gateway for the IPLA blade.

If a router or firewall is placed between the SIP Trunk Provider and SV8100, you must also set the following programs:

✎ *The IP address assigned to the Optimum Business SIP Trunk Adapter.*

10-12-06 : CD-CP00 Network Setup – NAPT Router
Turn this program on if the SV8100 resides behind a NAT router.

10-12-07 : CD-CP00 Network Setup – NAPT Router IP Address
Set the WAN IP address of the NAT router behind the SV8100.

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.

✎ *IP address is required by the CD-CP00. Some private IP network ranges (ex: 192.168.0.0/16, 172.16.0.0/12) conflict with SIP Service Provider's Network ranges which may cause issues when connecting SIP connect service. Private ranges reserved for the customer's LAN are 10.x.x.x and 192.168.0.x through 192.168.10.x.*

The SV8100 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 IPLA DSP Basic Setup

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

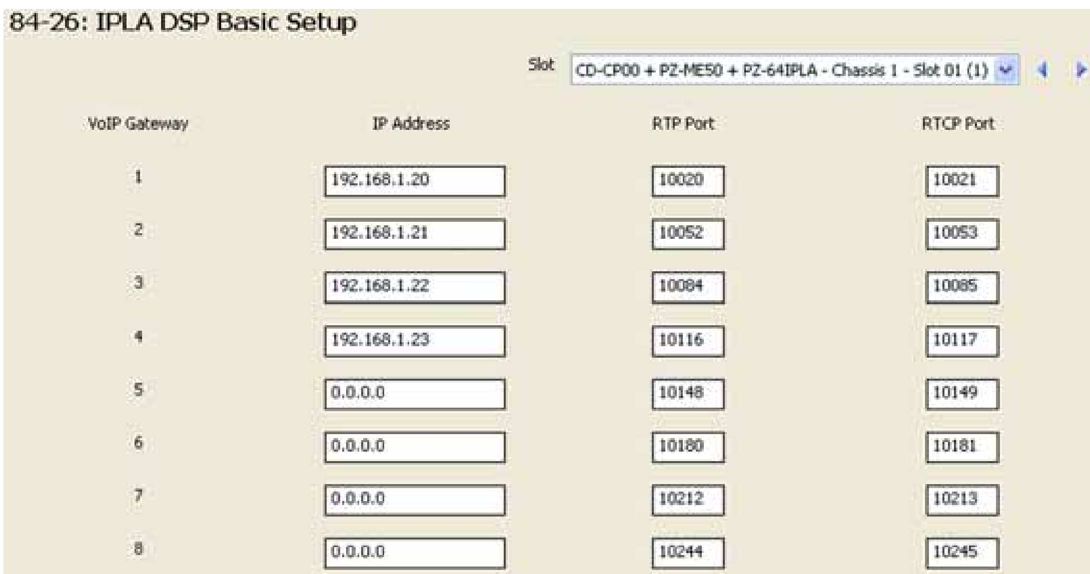

Figure 1-6 IPLA DSP Basic Setup

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. When forwarding the ports, the range for each gateway must be set. The number of gateways to forward will depend on the size of the IPLA.

- Gateway 1 will require ports 10020-10051 forwarded.
- Gateway 2 will require ports 10052-10083 forwarded.
- Gateway 3 will require ports 10084-10115 forwarded.
- Gateway 4 will require ports 10116-10147 forwarded.
- Gateway 5 will require ports 10148-10179 forwarded.
- Gateway 6 will require ports 10180-10211 forwarded.
- Gateway 7 will require ports 10212-10243 forwarded.
- Gateway 8 will require ports 10244-10275 forwarded.

Table 1-2 Port Table

Ports	UDP	TCP
5060	Yes	No
10020	Yes	No
10021	Yes	No
10052	Yes	No
10053	Yes	No
10084	Yes	No
10085	Yes	No
10116	Yes	No
10117	Yes	No

3.4 SIP System Information Setup

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

Figure 1-7 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.

✎ *The Domain name should match IP address that was assigned to the Optimum SIP Trunk Adapter. Refer to step 2 in the Optimum Business SIP trunk Set-up Guide.*

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

✎ *The Host name should match IP address that was assigned to the Optimum SIP Trunk Adapter. Refer to step 2 in the Optimum Business SIP trunk Set-up Guide.*

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

10-28-04 : SIP System Information Setup – User ID
This information is provided by your SIP Trunking Service Provider.
Entries: 32 characters maximum (Default=No Entry).

✎ *Typically the ten digit billing telephone number is used. This entry must be numeric as Program 10-23-04 does not allow text entry - only numeric.*

✎ *This number will also be used as the default caller ID for outbound calls.*

10-28-05 : SIP System Information Setup – Domain Assignment
Determine the type of Domain Assignment. Set this entry to 1 (Domain name).

✎ *Set to Domain when using Dynamic (Registration mode)* ***or*** *IP Address when using Static (Non-registration) mode.*

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.

3.5 SIP Server Information Setup (Dynamic or Static Configuration)

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

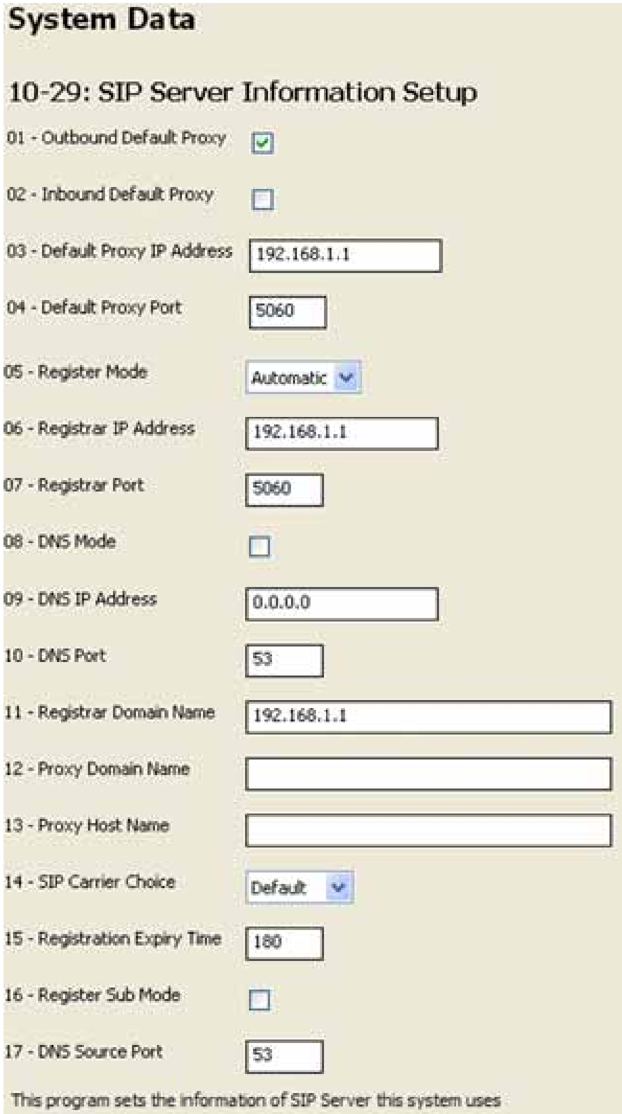

Figure 1-8 SIP Server Information Setup (Dynamic Configuration)

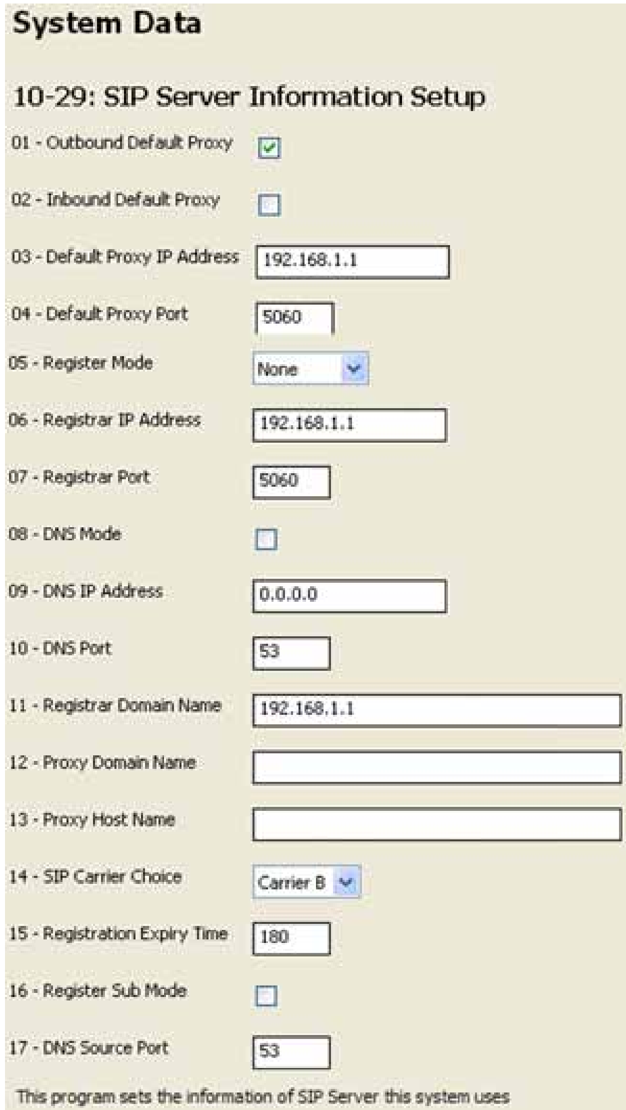

Figure 1-9 SIP Server Information Setup (Static Configuration)

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 SV8100 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 (e.g., 47.234.106.137). You may resolve the IP address of the Outbound Proxy by pinging the URL.

✎ *The IP address assigned should be the same address assigned to the Optimum SIP Trunk Adapter. Refer to step 2 in the Optimum Business SIP Trunk Set-Up Guide.*

10-29-05 : SIP Server Information Setup – Registrar Mode
Set the Registrar Mode to 1(manual) with SIP trunking.

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

10-29-07 : SIP Server Information Setup – Registrar Port
Input the Registrar Port address (5060) at default.

10-29-08 : SIP Server Information Setup – SIP Proxy Setup – DNS Mode
Set the DNS Mode to 1, when the SIP carrier provides a domain name.

10-29-09 : SIP Server Information Setup – SIP Proxy Setup – DNS IP Address
This information should be provided by your SIP service provider.

✎ *The DNS IP Address should be any valid Domain Name Server either SIP provided or within your network.*

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).

✎ *The IP address assigned should be the same address assigned to the Optimum SIP Trunk Adapter. Refer to step 2 in the Optimum Business SIP Trunk Set-Up Guide.*

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 provide 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 4 (Carrier D).

✎ *Changing this program automatically changes program 10-29-16 to "on". **10-29-16 must be turned off in order for incoming calls to route correctly.***

10-29-15 : SIP Server Information Setup – Registration Expiry Time
It is **important** to leave this automatic re-registration time to be 180 seconds so that the Optimum Business 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.

3.6 IP System Interconnection Setup (Static Configuration)

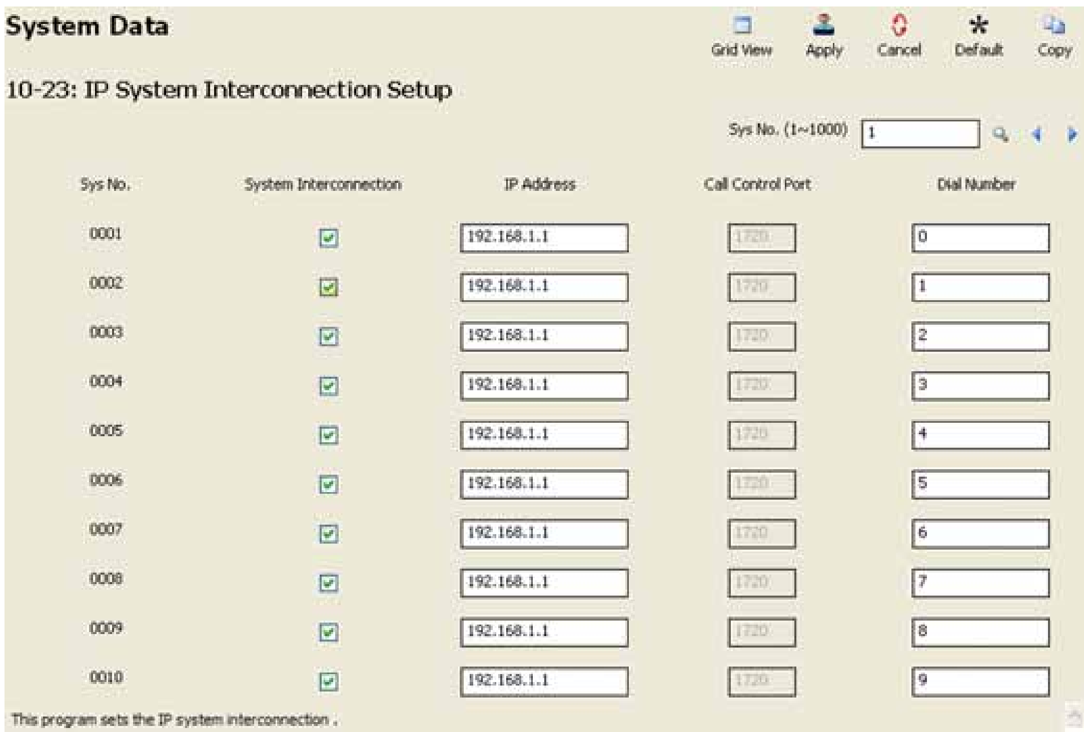

Figure 1-10 IP System Interconnection Setup

10-23 : IP System Interconnection Setup
Sys No. 0001– 0012 Check System Interconnection and assign the SIP Trunking Service Provider Proxy IP Address and Dial Number 0-9 and also (*) and (#).

This field applies for those SIP service providers supporting Static Configuration.

✎ *The IP address assigned should be the same address assigned to the Optimum SIP Trunk Adapter. Refer to step 2 in the Optimum Business SIP Trunk Set-Up Guide.*

3.7 SIP Authentication Information Setup (Utilized with Static and Dynamic Configuration)

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

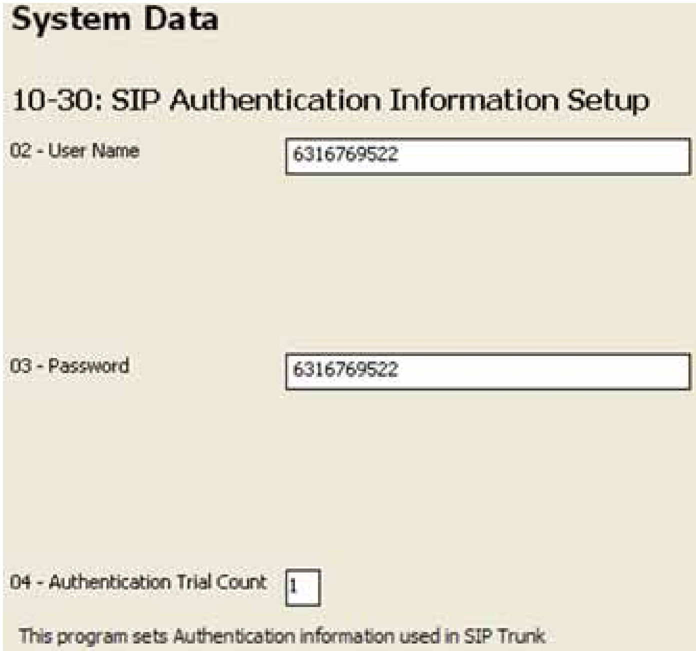

Figure 1-11 SIP Authentication Information Setup

10-30-02 : SIP Authentication Information Setup – User Name
Define the authentication User Name provided by Optimum Business as defined in Program 10-28-04. This information is provided by your SIP Service Provider. Entries: 48 characters maximum.

✎ *This field should be populated with the same user id and password of the Optimum Business SIP Trunk Adapter. This was step 3 of the Optimum Business SIP Trunk Set-Up Guide.*

10-30-03 : SIP Authentication Information Setup – Password

✎ *This field should be populated with the same user id and password of the Optimum Business SIP Trunk Adapter. This was step 3 of the Optimum Business SIP Trunk Set-Up Guide.*

3.8 SIP Trunk Registration

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

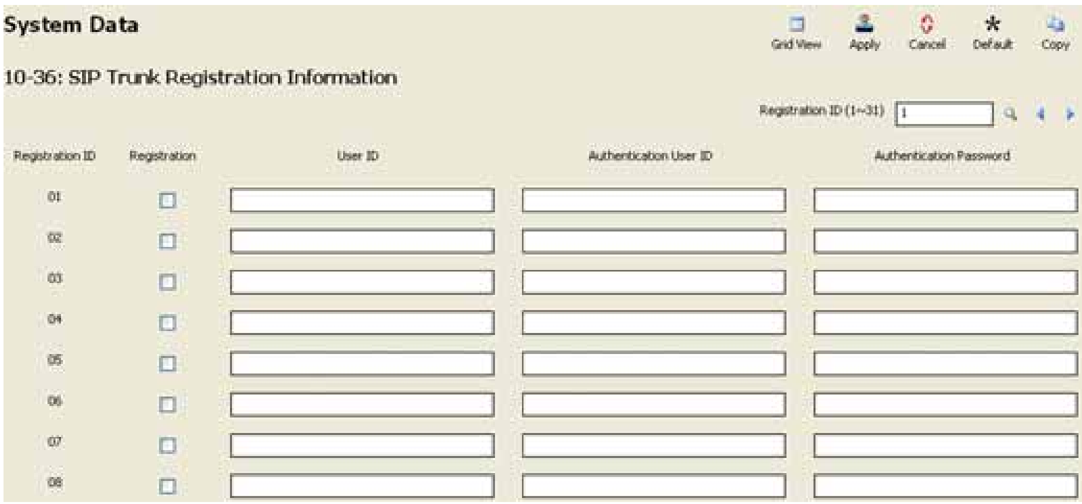

Figure 1-12 SIP Trunk Registration Information

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.

10-28-04 : SIP System Information Setup – 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.

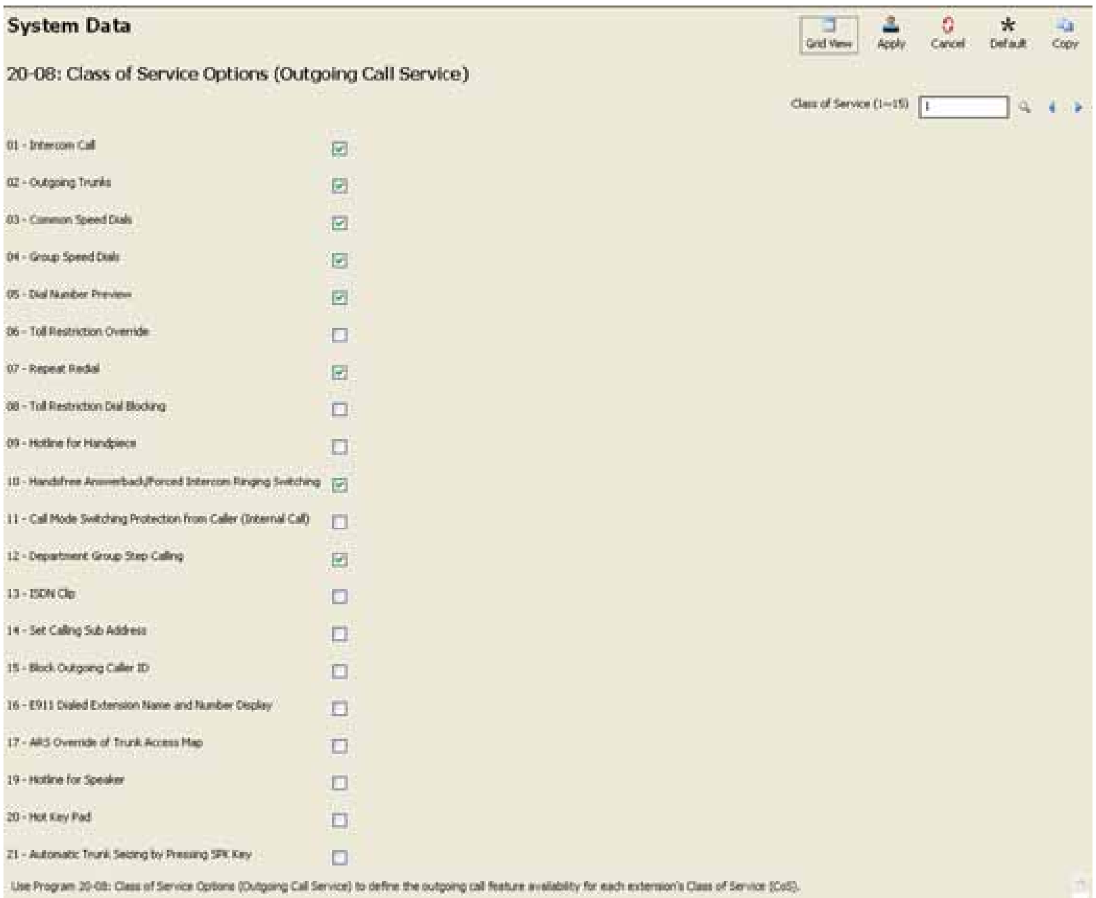

Figure 1-13 Class of Service Options

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

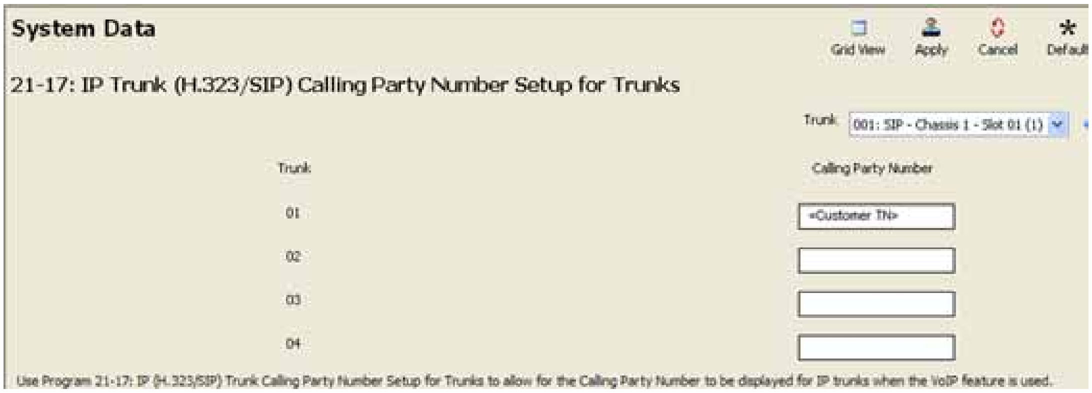

Figure 1-14 IP Trunk (H.323/SIP) Calling Party Number Setup for Trunks

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.

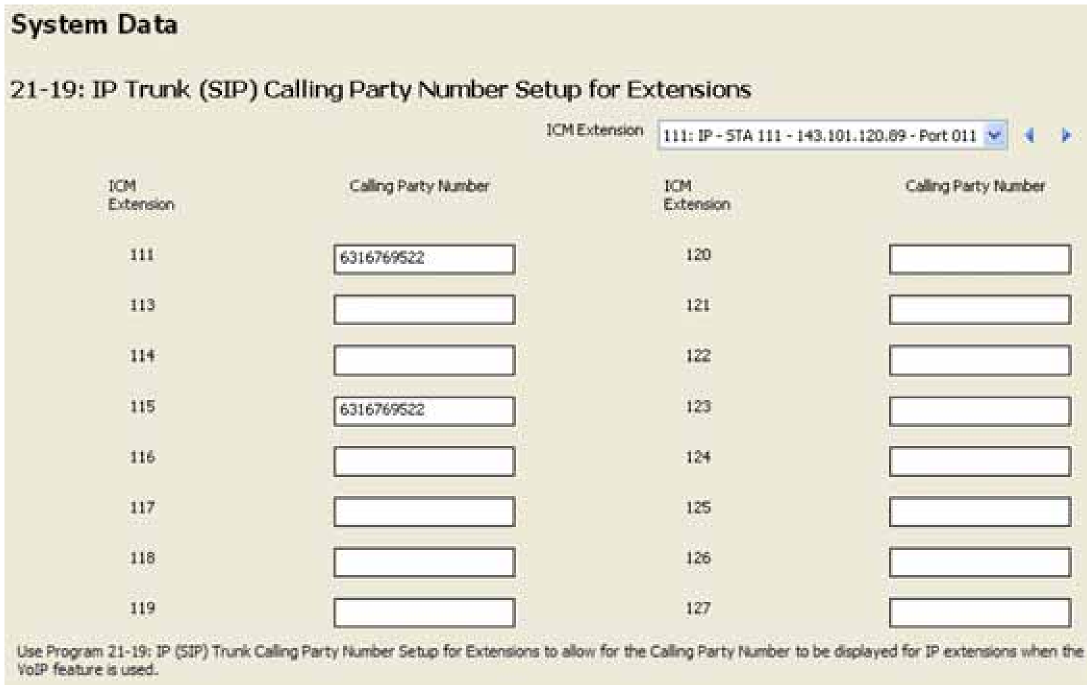

Figure 1-15 IP Trunk (SIP) Calling Party Number Setup for Extensions

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.

✎ *Dynamic Mode all extensions' Calling Party Numbers must be set to the Pilot DID.*

✎ *Outbound calls will not work for extensions where the calling party number does not match the Pilot DID.*

✎ *Static Mode: the calling party number can be set to any DID on the SIP Trunk.*

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 1-16 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 SV8100 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 DTMF Configuration

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

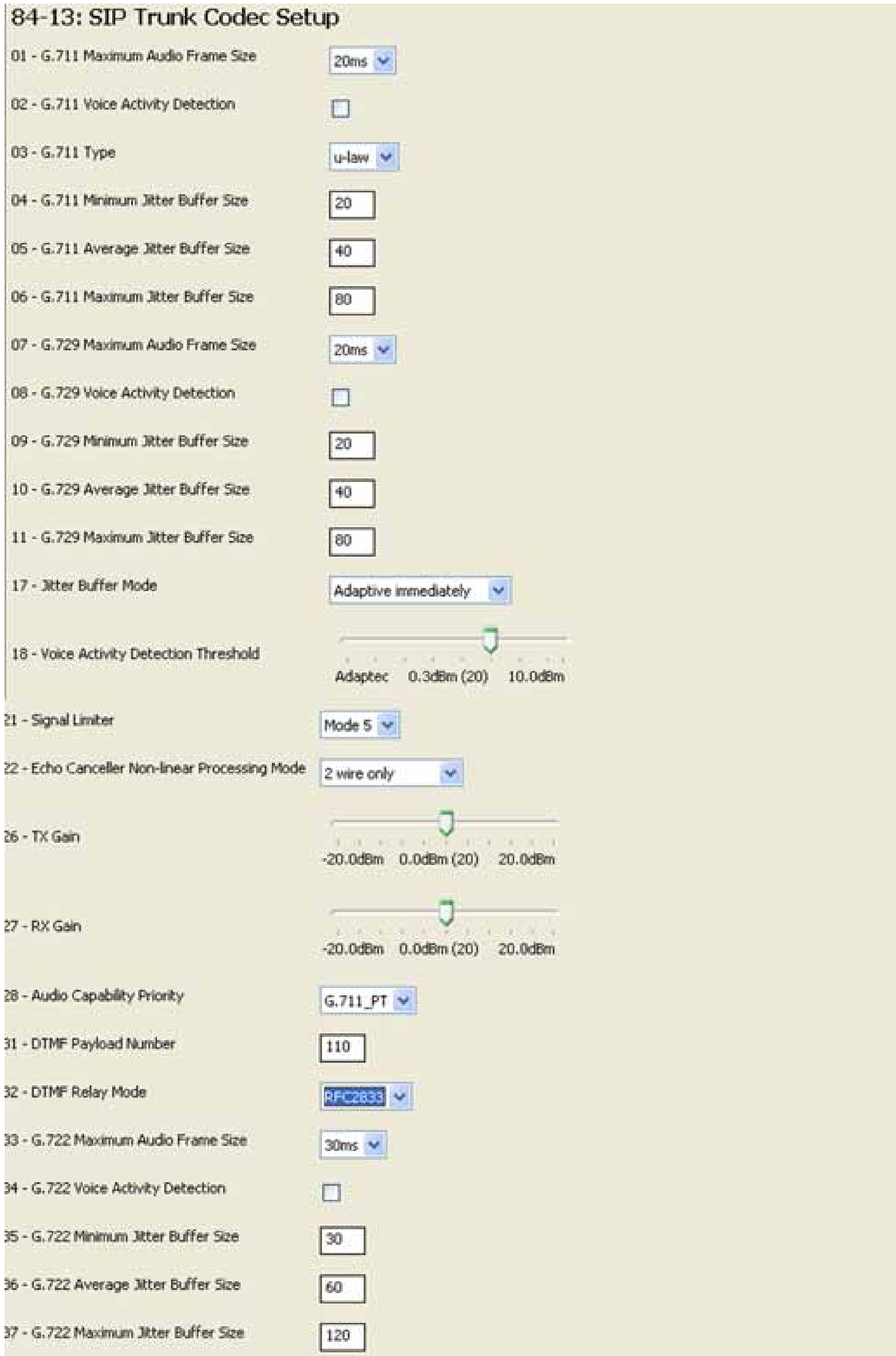

Figure 1-17 SIP Trunk Codec Setup

84-13-32 : SIP Trunk CODEC Information Basic Setup – DTMF Relay Mode
Set the DTMF setup to 1 (RFC2833).

3.15 ToS Setup

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

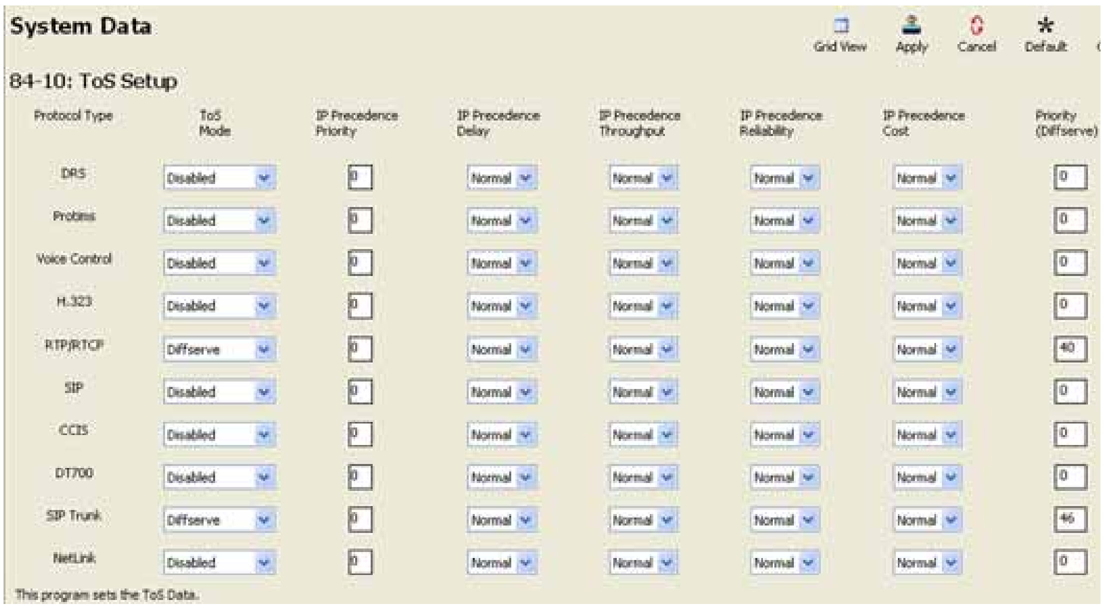

Figure 1-18 ToS Setup

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 SV8100 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 SV8100 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 1-19 SIP Trunk Basic Setup

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

The SV8100 must be reset in order for the change to take effect.

3.17 Enabling DIDs for Outbound Caller-ID Appearance

To enable the use of DIDs for outbound Caller-ID appearance you must perform two steps:

- STEP 1 - change from “Carrier D” to “Carrier B”.
- STEP 2 - change the “Codec Maximum Audio Frame Size” to 20ms.

1. Go to the “10-29:SIP Server Information Setup” screen.

   a Select “Carrier B” for the “14 - SIP Carrier Choice” field. Note that this will allow the PBX to use the DIDs configured for each extension’s caller ID from 21-19. See “21-19:IP Trunk (SIP) Calling Party Number Setup for extensions”.

   b Uncheck the “16 – Register Sub Mode” field. **THIS MUST BE DONE EVERY TIME YOU CHANGE THE SIP CARRIER CHOICE.**

   c Hit the **Apply** icon.

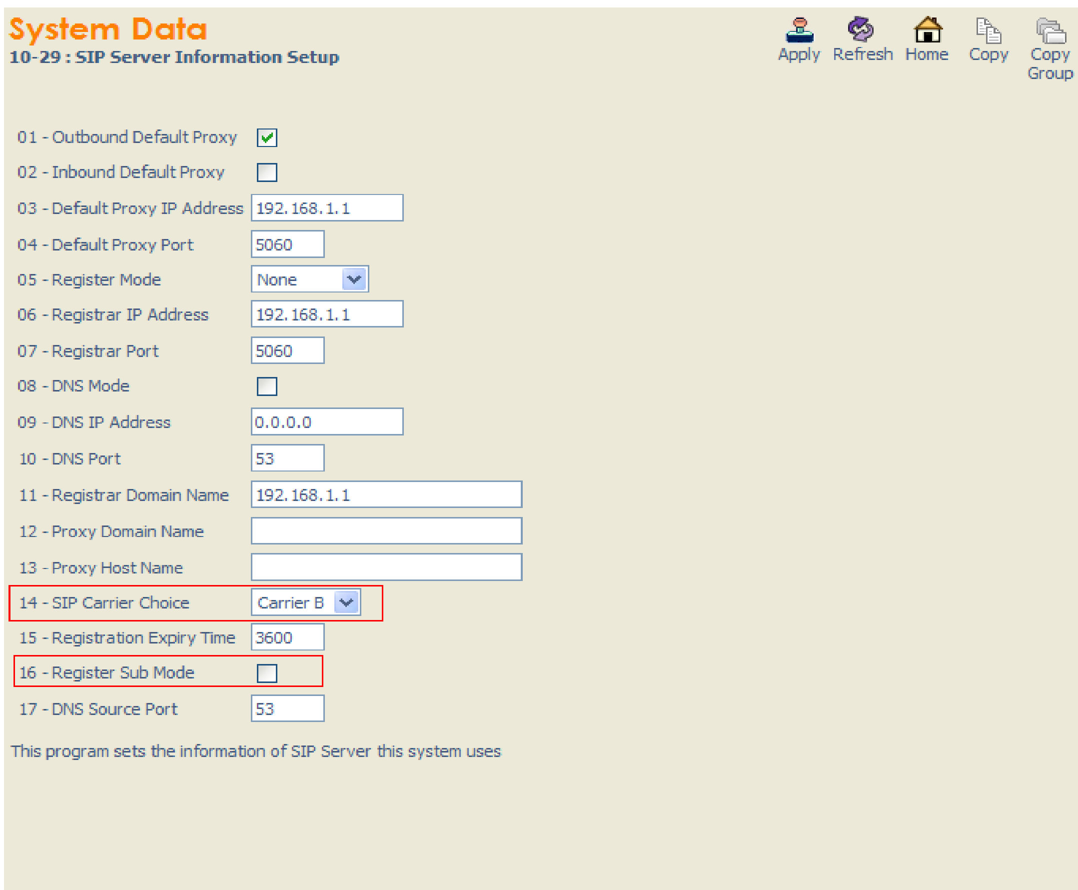

Figure 1-20 SIP Server Information Setup

2. Go to the “84-13 SIP Trunk Codec Setup” screen.

   a Set the “84-13-33 G.722 Maximum Audio Frame Size” field and “84-13-38 G.726 Maximum Audio Frame Size” field to 20ms.

   b Hit **Apply**.

   c Go to the “84-19 SIP Extension Codec Setup” screen.

   d Set the “84-19-33 G.722 Maximum Audio Frame Size” field and “84-19-38 G.726 Maximum Audio Frame Size” field to 20ms.

   e Hit **Apply**.

YOU MUST REPEAT THIS STEP EVERY TIME YOU CHANGE THE SIP CARRIER CHOICE

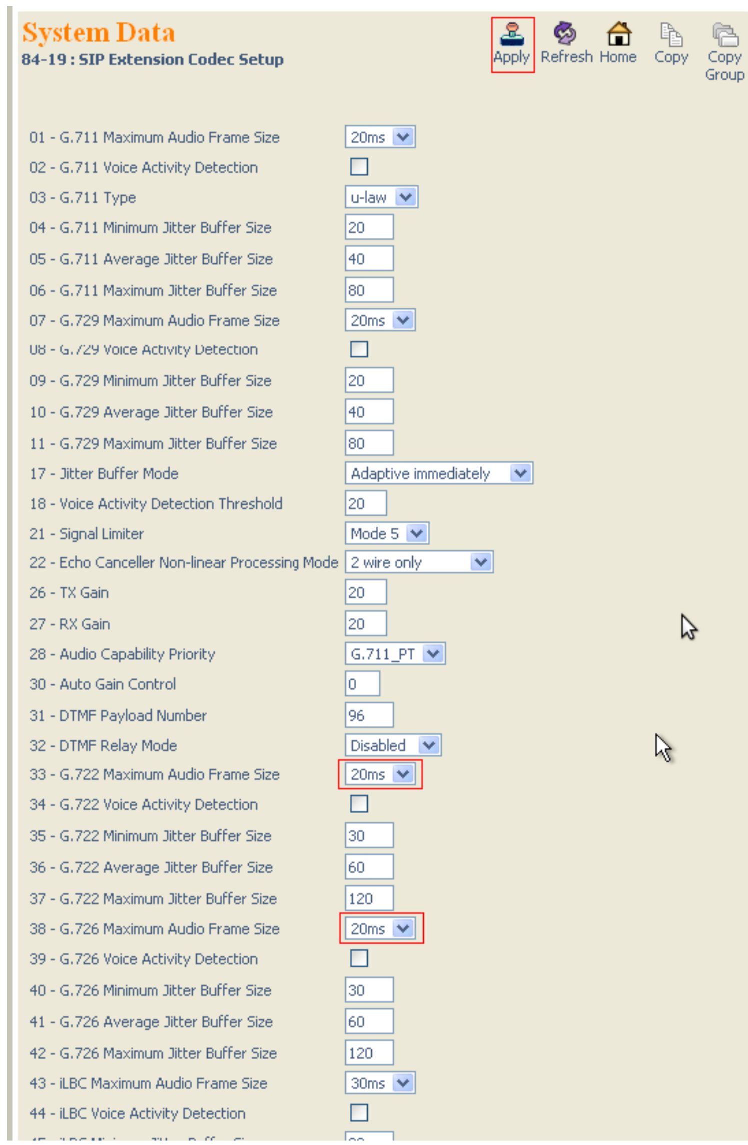

Figure 1-21 SIP Extension Codec Setup

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 1-3 Troubleshooting Guide on page 1-27*. 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 1-3 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 Optimum Business
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

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Appendix A Optimum Business SIP Trunk Network Configuration

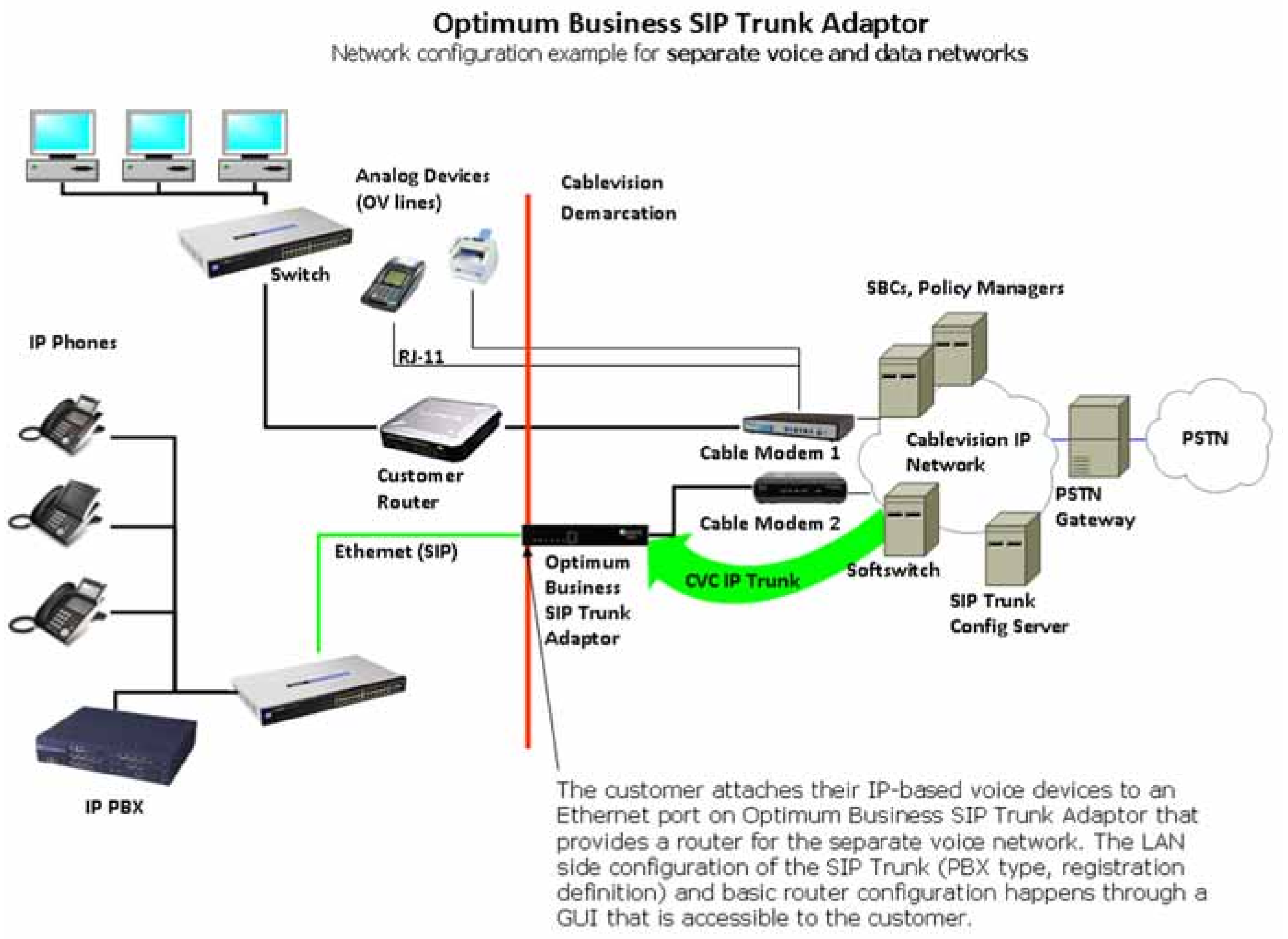

Figure A-1 Network Configuration – Voice and Data Networks

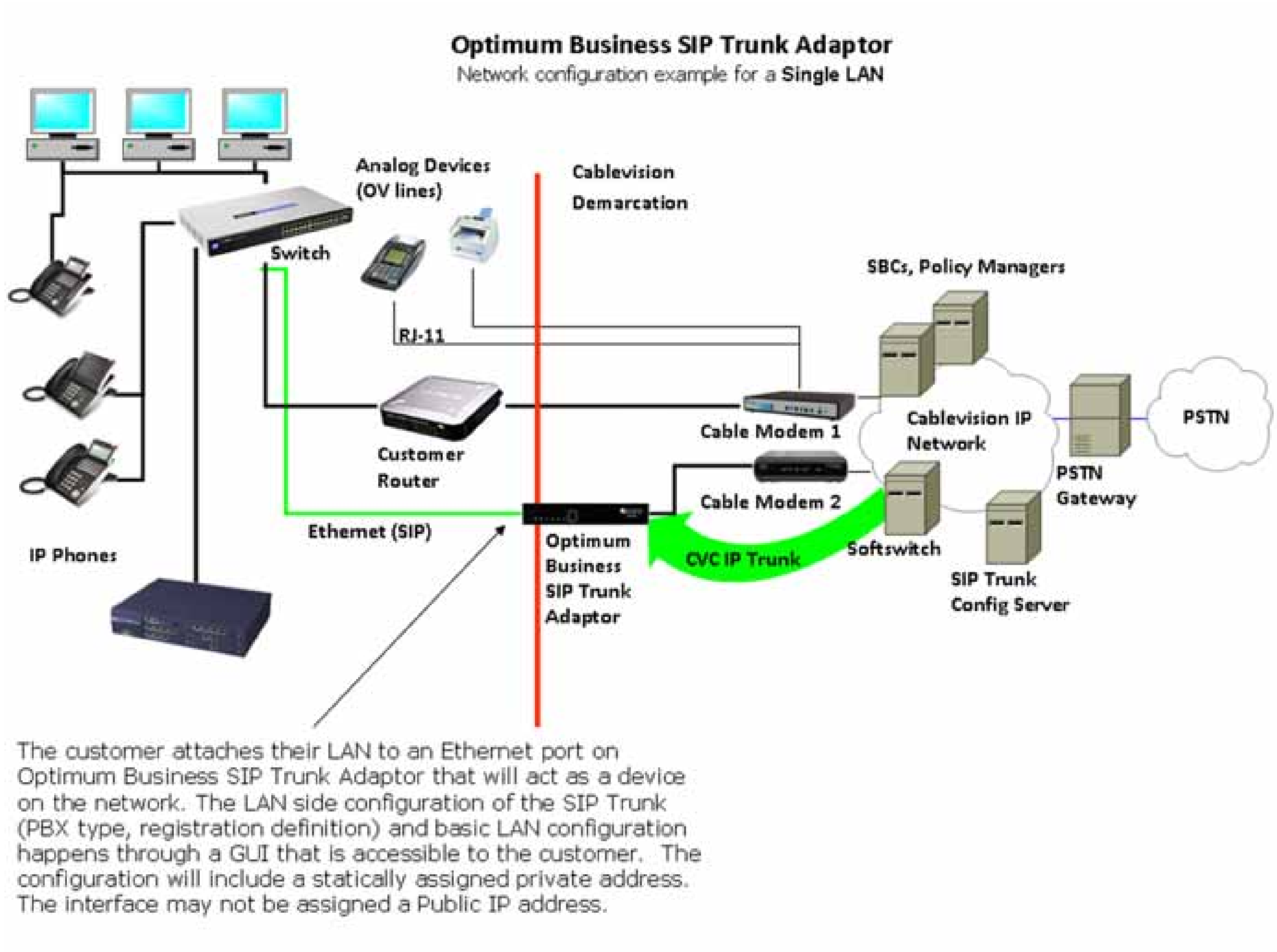

Figure A-2 Network Configuration – Single LAN