

UNIVERGE[®] SV8100

SIP Trunking Service Configuration Guide for American Telephone Company

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**NEC Corporation of America
6535 N. State Highway 161
Irving, TX 75039-2402**

Communications Technology Group

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Configuring NEC SV8100 with American Telephone Company SIP Trunking Service

SECTION 1 NEC SV8100 AND AMERICAN TELEPHONE COMPANY 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 American Telephone Company SIP Trunking service. It provides sample entries for the required fields. The actual data is provided by American Telephone Company 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 American Telephone Company Account

Contact your American Telephone Company representative.

1.3 SV8100 System Software

The SV8100 requires system software Version 8.00 or higher to use American Telephone Company 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 8.00 or higher.

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

1.5 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
- T.38 Faxing is not supported by American Telephone Company.
- G.711 is the only CODEC supported with American Telephone Company.
- The default CODEC Frame Size of 20ms must remain in command 84-13-01.

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 American Telephone Company SIP Trunk service provider, utilizing a **STATIC** configuration.

2.1 **Prerequisites**

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

2.1.1 SIP Trunking Information from American Telephone Company

- 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 SV8100

- SV8100 CPU firmware Version 8.00 or higher
- IPLA/B (PZ-XX)
- 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 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 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 American Telephone Company 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

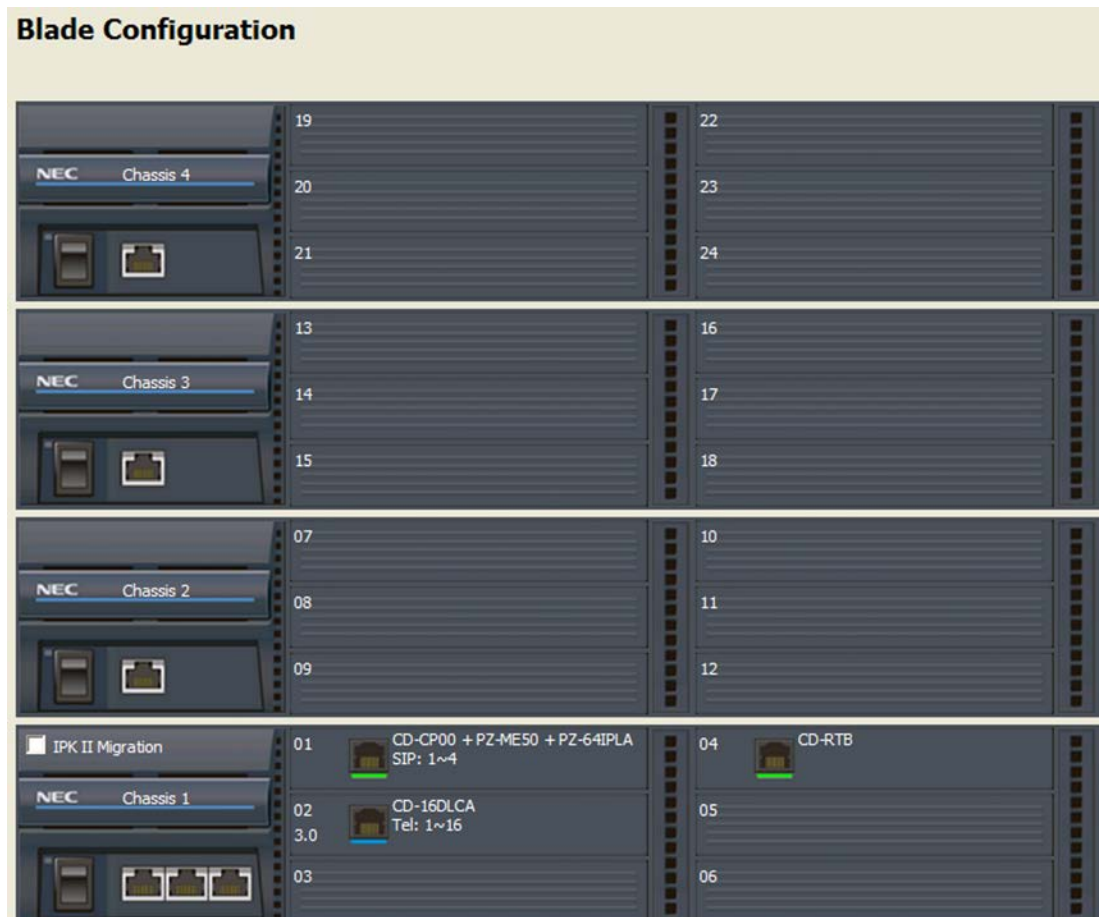


Figure 1 Blade Configuration

System Data Grid View Apply Cancel Default

10-03: IPL Configuration

Slot: CD-CP00 + PZ-3Z1PL - Chassis 1 - Slot 01 (1) Physical Port (1~200): 1

Physical Port	Trunk Logical Port	Trunk Type	CCIS Trunk	Physical Port	Trunk Logical Port	Trunk Type	CCIS Trunk
001	5	SIP	Not CCIS	009	13	SIP	Not CCIS
002	6	SIP	Not CCIS	010	14	SIP	Not CCIS
003	7	SIP	Not CCIS	011	15	SIP	Not CCIS
004	8	SIP	Not CCIS	012	16	SIP	Not CCIS
005	9	SIP	Not CCIS	013	17	SIP	Not CCIS
006	10	SIP	Not CCIS	014	18	SIP	Not CCIS
007	11	SIP	Not CCIS	015	19	SIP	Not CCIS
008	12	SIP	Not CCIS	016	20	SIP	Not CCIS

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

Figure 2 IPL Configuration

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

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

10-19: IPLA DSP Resource Selection

Slot: DSP Resource (1~128):

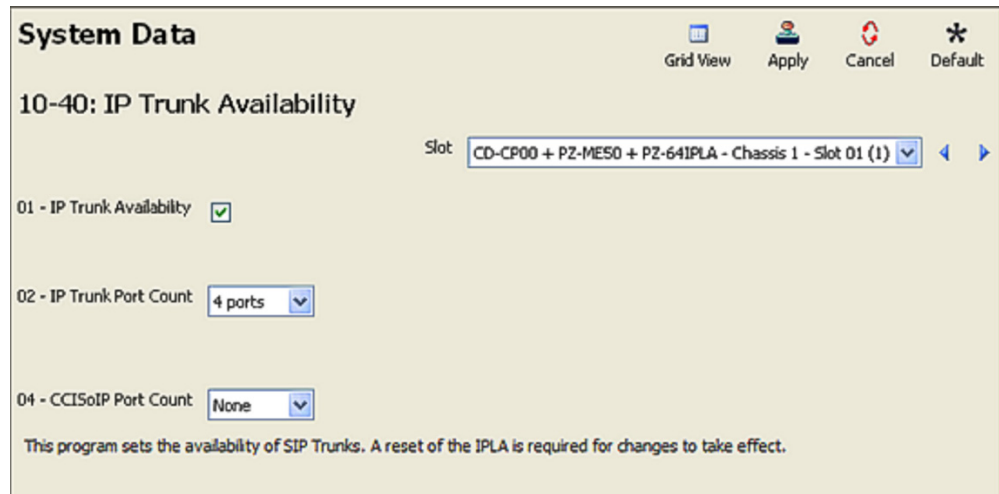
DSP Resource		DSP Resource	
001	<input type="text" value="Used for IP trunks"/>	009	<input type="text" value="Commonly used for both IP extensions and trunks"/>
002	<input type="text" value="Used for IP trunks"/>	010	<input type="text" value="Commonly used for both IP extensions and trunks"/>
003	<input type="text" value="Used for IP trunks"/>	011	<input type="text" value="Commonly used for both IP extensions and trunks"/>
004	<input type="text" value="Used for IP trunks"/>	012	<input type="text" value="Commonly used for both IP extensions and trunks"/>
005	<input type="text" value="Commonly used for both IP extensions and trunks"/>	013	<input type="text" value="Commonly used for both IP extensions and trunks"/>
006	<input type="text" value="Commonly used for both IP extensions and trunks"/>	014	<input type="text" value="Commonly used for both IP extensions and trunks"/>
007	<input type="text" value="Commonly used for both IP extensions and trunks"/>	015	<input type="text" value="Commonly used for both IP extensions and trunks"/>
008	<input type="text" value="Commonly used for both IP extensions and trunks"/>	016	<input type="text" value="Commonly used for both IP extensions and trunks"/>

This program sets the IPLA DSP resource selection.

Figure 3 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).



The screenshot shows a configuration window titled "System Data" with a subtitle "10-40: IP Trunk Availability". In the top right corner, there are four icons: "Grid View", "Apply", "Cancel", and "Default". Below the subtitle, there is a "Slot" dropdown menu showing "CD-CP00 + PZ-ME50 + PZ-64IPLA - Chassis 1 - Slot 01 (1)". The main configuration area contains three items:

- 01 - IP Trunk Availability: A checkbox that is checked.
- 02 - IP Trunk Port Count: A dropdown menu set to "4 ports".
- 04 - CCISoIP Port Count: A dropdown menu set to "None".

At the bottom of the configuration area, there is a note: "This program sets the availability of SIP Trunks. A reset of the IPLA is required for changes to take effect."

Figure 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.

The screenshot shows a configuration window titled "System Data" with a sub-section "10-12: CD-CP00 Network Setup". The fields and their values are as follows:

- 01 - IP Address: 0.0.0.0
- 02 - Subnet Mask: 255.255.0.0
- 03 - Default Gateway: 10.10.3.1
- 04 - Time Zone: (GMT -05:00) Eastern Time (US and Canada)
- 05 - NIC Setting: Automatic detection
- 06 - NAPT Router:
- 07 - NAPT Router IP Address: 143.101.120.218
- 08 - ICMP Redirect:
- 09 - IPL IP Address: 10.10.3.10
- 10 - IPL Subnet Mask: 255.255.255.0
- 11 - IPL NIC Setting: Automatic detection
- 13 - DNS Primary Address: 0.0.0.0
- 14 - DNS Secondary Address: 0.0.0.0
- 15 - DNS Port: 53

At the bottom of the window, there is a note: "Use Program 10-12: CPUUI Network Setup to setup the IP Address, Subnet-Mask and Default Gateway addresses." and a caution: "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 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/IPLB blade.

10-12-03 : CD-CP00 Network Setup – Default Gateway

Set the default gateway for the IPLA/IPLB blade.

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

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 IPL DSP Basic Setup

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

System Data

84-26: IPLA DSP Basic Setup

Slot: CD-CP00 + P2-ME50 + P2-64IPLA - Chassis 1 - Slot 01 (1)

VoIP Gateway	IP Address	RTP Port	RTCP Port
1	10.10.3.20	10020	10021
2	10.10.3.21	10052	10053
3	10.10.3.22	10084	10085
4	10.10.3.23	10116	10117
5	0.0.0.0	10148	10149
6	0.0.0.0	10180	10181
7	0.0.0.0	10212	10213
8	0.0.0.0	10244	10245

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. 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/B.

- 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 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

Table 3 Router Forwarding (Gateway Table)

IPLA/IPLB Size	Gateway	IP Address	RTP Port	RTCP Port	UDP
IPLB32/64/128	1				
IPLA32	2				
	3				
IPLA64	4				
	5				
	6				
	7				
IPLA128	8				

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

```
udp;143.101.120.218/255.255.255.0-10020>10.10.3.20-10020
```

```
udp;143.101.120.218/255.255.255.0-10021>10.10.3.20-10021
```

```
udp;143.101.120.218/255.255.255.0-10052>10.10.3.21-10052
```

```
udp;143.101.120.218/255.255.255.0-10053>10.10.3.21-10053
```

```
udp;143.101.120.218/255.255.255.0-10084>10.10.3.22-10084
```

```
udp;143.101.120.218/255.255.255.0-10085>10.10.3.22-10085
```

```
udp;143.101.120.218/255.255.255.0-10116>10.10.3.23-10116
```

```
udp;143.101.120.218/255.255.255.0-10117>10.10.3.23-10117
```

```
udp;143.101.120.218/255.255.255.0-5060>10.10.3.10-5060
```

3.4 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).

10-28-04 : 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.

10-28-05 : SIP System Information Setup – Domain Assignment

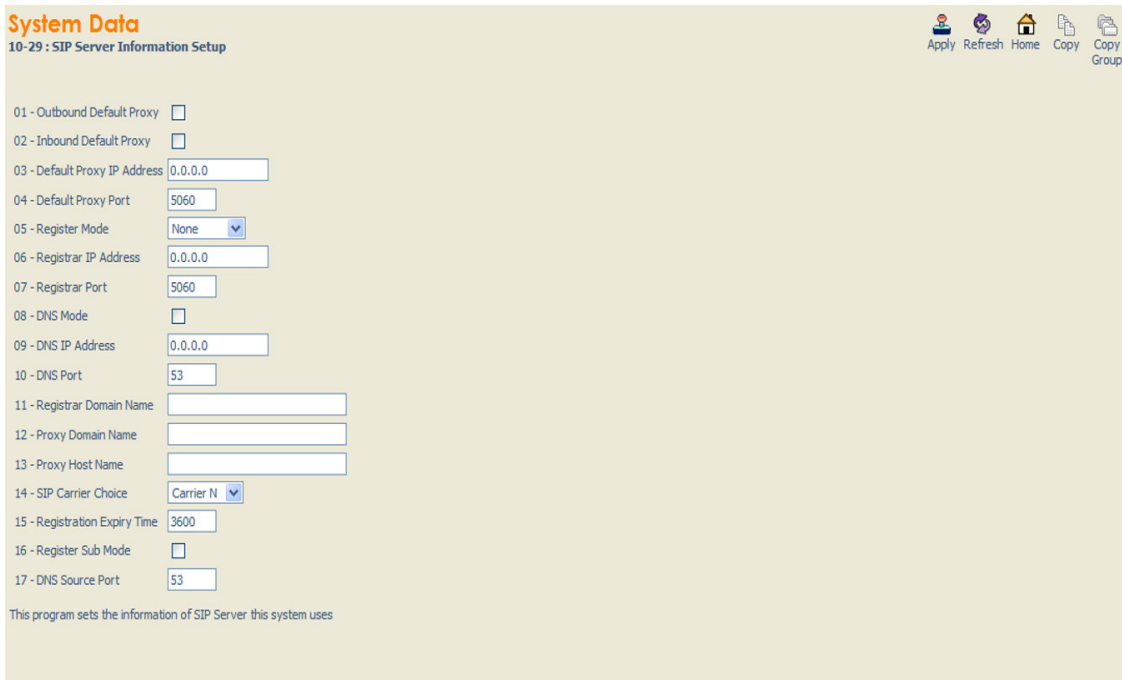
Determine the type of Domain Assignment. Set this entry to 0 (IP Address).

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.

3.5 SIP Server Information Setup

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



System Data
10-29 : SIP Server Information Setup

01 - Outbound Default Proxy

02 - Inbound Default Proxy

03 - Default Proxy IP Address

04 - Default Proxy Port

05 - Register Mode

06 - Registrar IP Address

07 - Registrar Port

08 - DNS Mode

09 - DNS IP Address

10 - DNS Port

11 - Registrar Domain Name

12 - Proxy Domain Name

13 - Proxy Host Name

14 - SIP Carrier Choice

15 - Registration Expiry Time

16 - Register Sub Mode


17 - DNS Source Port

This program sets the information of SIP Server this system uses

Figure 8 SIP Server Information Setup

10-29-01 : SIP Server Information Setup – Outbound Default Proxy

Disable (0) 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. 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).

10-29-08 : 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).


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 Carrier N.

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 American Telephone Company network does not get flooded.

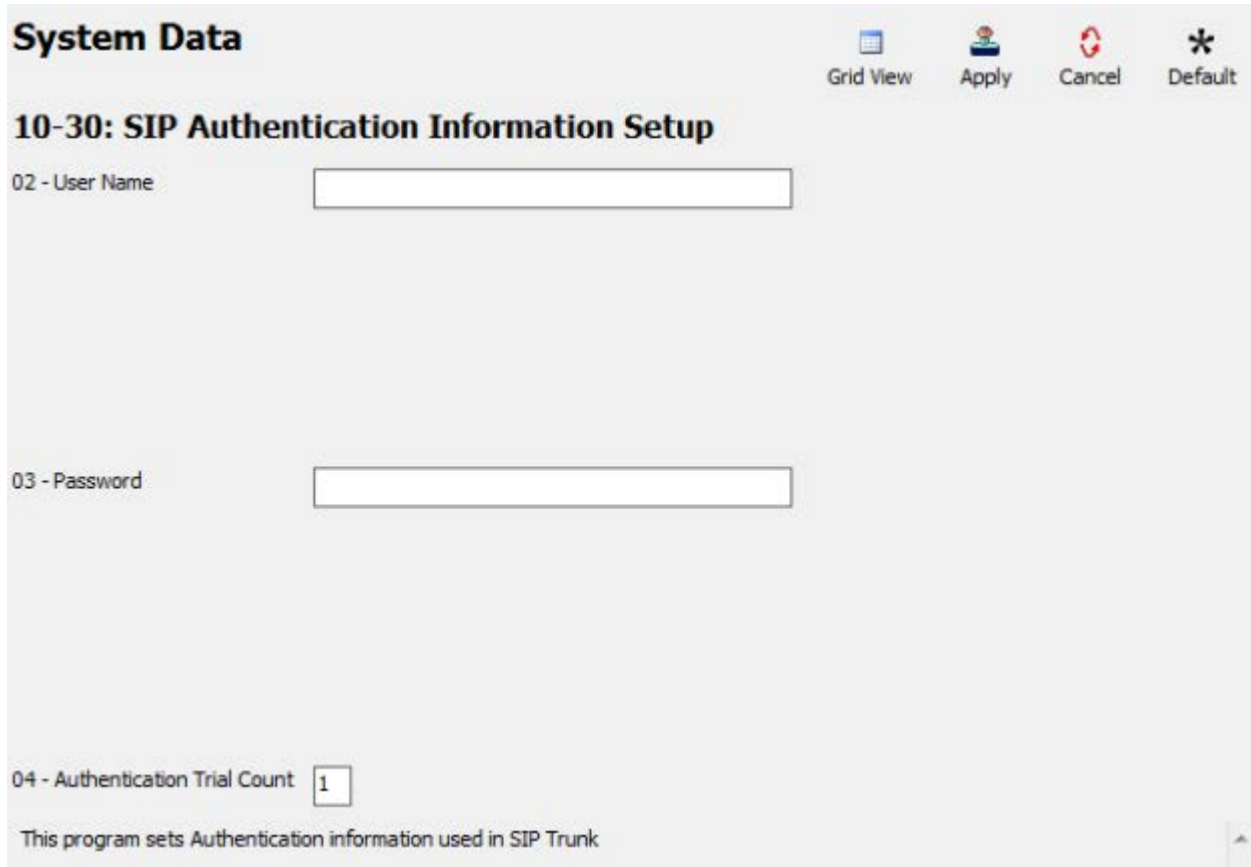
10-29-16 : SIP Server Information Setup – Register Sub Mode

Unchecking the Register Sub Mode (setting it to "off") will allow all trunk calls to be routed based on routing policies.

3.6 SIP Authentication Information Setup (Not Required)

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

 *Not used with IP Authenticated SIP Accounts.*



The screenshot shows a web-based configuration interface titled "System Data". At the top right, there are four buttons: "Grid View" (with a calendar icon), "Apply" (with a person icon), "Cancel" (with a refresh icon), and "Default" (with an asterisk icon). Below the title, the section is labeled "10-30: SIP Authentication Information Setup". There are three main fields: "02 - User Name" with a text input box, "03 - Password" with a text input box, and "04 - Authentication Trial Count" with a numeric input box containing the value "1". At the bottom, there is a note: "This program sets Authentication information used in SIP Trunk".

Figure 9 SIP Authentication Information Setup

10-30-02 : SIP Authentication Information Setup – User Name

Define the authentication User Name provided by American Telephone Company as defined in Program 10-28-04. This information is provided by your SIP Service Provider.

Entries: 48 characters maximum.

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

Enter the American Telephone Company authentication password. This information is provided by your SIP Service Provider.

Entries: 48 characters maximum.

3.7 IP System Interconnection Setup

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

System Data
10-23 : IP System Interconnection Setup

Apply Refresh Home Copy Copy Group

Sys No. (1-1000)

Sys No.	System Interconnection	IP Address	Call Control Port	Dial Number
0001	<input checked="" type="checkbox"/>	8.11.254.251	1720	1
0002	<input checked="" type="checkbox"/>	8.11.254.251	1720	2
0003	<input checked="" type="checkbox"/>	8.11.254.251	1720	3
0004	<input checked="" type="checkbox"/>	8.11.254.251	1720	4
0005	<input checked="" type="checkbox"/>	8.11.254.251	1720	5
0006	<input checked="" type="checkbox"/>	8.11.254.251	1720	6
0007	<input checked="" type="checkbox"/>	8.11.254.251	1720	7
0008	<input checked="" type="checkbox"/>	8.11.254.251	1720	8
0009	<input checked="" type="checkbox"/>	8.11.254.251	1720	9
0010	<input checked="" type="checkbox"/>	8.11.254.251	1720	0

Valid characters are 0-9, *, #

This program sets the IP system interconnection .

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.

3.8 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.

3.9 Class of Service Options (Outgoing Call Service)

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

Option ID	Option Name	Checked
01	Intercom Call	<input checked="" type="checkbox"/>
02	Outgoing Trunks	<input checked="" type="checkbox"/>
03	Common Speed Dials	<input checked="" type="checkbox"/>
04	Group Speed Dials	<input checked="" type="checkbox"/>
05	Dial Number Preview	<input checked="" type="checkbox"/>
06	Toll Restriction Override	<input type="checkbox"/>
07	Repeat Redial	<input checked="" type="checkbox"/>
08	Toll Restriction Dial Blocking	<input type="checkbox"/>
09	Hotline for Handpiece	<input type="checkbox"/>
10	Handfree Answerback/Forced Intercom Ringing Switching	<input checked="" type="checkbox"/>
11	Call Mode Switching Protection from Caller (Internal Call)	<input type="checkbox"/>
12	Department Group Step Calling	<input checked="" type="checkbox"/>
13	ISDN Clip	<input checked="" type="checkbox"/>
14	Set Calling Sub Address	<input type="checkbox"/>
15	Block Outgoing Caller ID	<input type="checkbox"/>

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.

3.10 IP Trunk Calling Party Number Setup

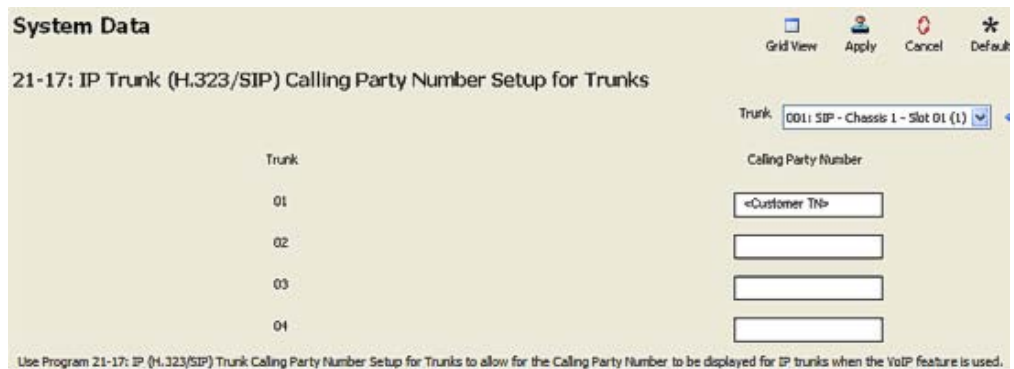


Figure 12 IP Trunk (H.323/SIP) Calling Party Number Setup for Trunks

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.

3.11 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.

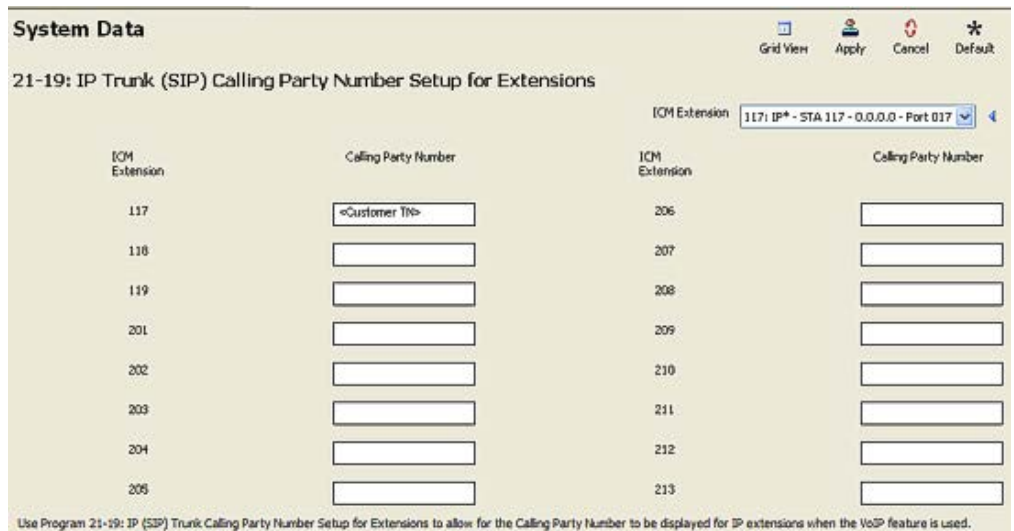


Figure 13 IP Trunk (SIP) Calling Party Number Setup for Extensions

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.

3.12 DID (TN to ext map)

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

System Data

22-02: Incoming Call Trunk Setup

Trunk: 001: SIP - Chassis 1 - Slot 01 (1) | Night Mode: 01 - Mode 1

Trunk	Mode 1	Mode 2	Mode 3	Mode 4
01	DID	DID	DID	DID
02	DID	DID	DID	DID
03	DID	DID	DID	DID
04	DID	DID	DID	DID

Use Program 22-02: Incoming Call Trunk Setup to assign the incoming trunk type for each trunk. There is one item for each Night Service Mode.

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 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).

3.13 SIP Trunk CODEC Setup

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

The screenshot displays a configuration interface for SIP Trunk CODEC Setup. The settings are as follows:

Setting ID	Setting Name	Value
21	Signal Limiter	Mode S
22	Echo Canceller Non-linear Processing Mode	2 wire only
24	Echo Canceller Comfort Noise Generator Configuration	Adaptive
26	TX Gain	0.0dBm (20)
27	RX Gain	0.0dBm (20)
28	Audio Capability Priority	G.711_PT
31	DTMF Payload Number	110
32	DTMF Relay Mode	RFC2833
33	G.722 Maximum Audio Frame Size	30ms
34	G.722 Voice Activity Detection	<input type="checkbox"/>

Figure 15 SIP Trunk CODEC Setup

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

3.14 ToS Setup

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

System Data Grid View Apply Cancel Default

84-10: ToS Setup

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

This program sets the ToS Data.

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 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.

3.15 SIP Trunk Basic Setup

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

The screenshot shows a configuration interface for SIP Trunk Basic Setup. The title is 'System Data' and the specific program is '84-14: SIP Trunk Basic Setup'. The interface includes several configuration items:

- 01 - Invite ReTx Count: 7
- 02 - Request ReTx Count: 11
- 03 - Response ReTx Count: 7
- 04 - Request ReTx Start Time: 5
- 05 - Request Max ReTx Interval: 40
- 06 - SIP Trunk Port: 5060
- 07 - Session Timer Value: 0
- 08 - Minimum Session Timer Value: 1800
- 09 - Called Party Info: Request URI
- 10 - URL Type: SIP-URL
- 11 - URL/TO Header Information: Proxy Server Domain** (highlighted with a red box)
- 13 - Incoming/Outgoing SIP Trunk for E.164: OFF
- 15 - 100rel Settings: Use Default Setting
- 16 - SIP Trunk SIP-URI E.164 Incoming Mode: OFF
- 17 - Call Forward Moved Temporarily Support: Disabled

At the bottom, there is a note: 'Use Program 84-14: SIP Trunk Basic Information Setup to define the basic setup for SIP trunks.'


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

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 American Telephone Company
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