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UNIVERGE® SV8100

SIP Trunking Service Configuration Guide for Bandwidth

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Communications Technology Group

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Configuring NEC SV8100 with Bandwidth SIP Trunking Service

SECTION 1 NEC SV8100 AND BANDWIDTH 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 Bandwidth's SIP Trunking service. It provides sample entries for the required fields. The actual data is provided by Bandwidth 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 Bandwidth Account

Contact your Bandwidth representative.

1.3 SV8100 System Software

The SV8100 requires system software 3.00 or higher to use Bandwidth's 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 3.00 or higher.

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

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

NEC PBX Configuration SECTION 2

This section provides information to NEC's solution providers and NEC Associates for configuring an NEC UNIVERGE SV8100 to connect to a Bandwidth SIP Trunk service provider, utilizing a **DYNAMIC** and **STATIC** configuration.

2.1 **Prerequisites**

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

2.1.1	SIP.	P Trunking Information from Bandwidth	
		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 3.00 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 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 Bandwidth 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

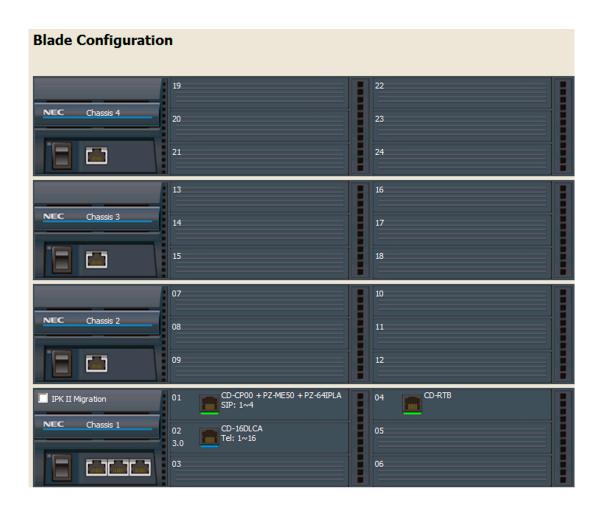


Figure 1 Blade Configuration



Figure 2 IPLA Configuration

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

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

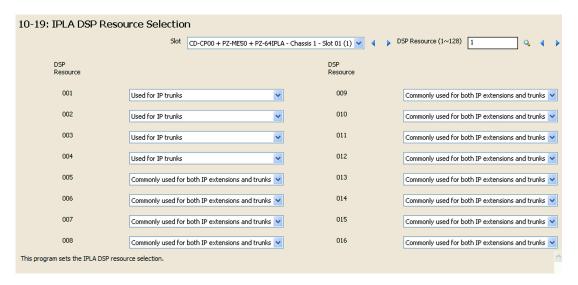


Figure 3 IPLA DSP Resource Selection

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 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.

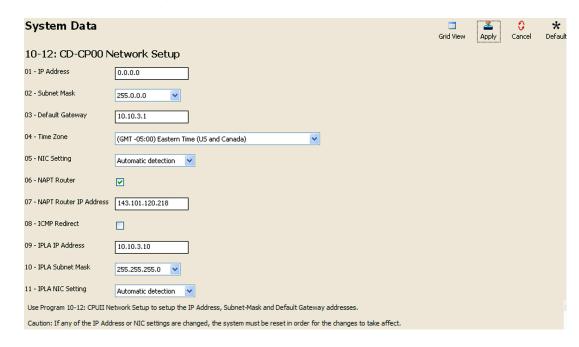


Figure 5 CD-CP00 Network Setup

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:

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.

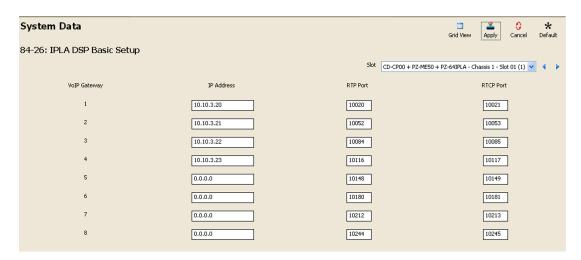


Figure 6 IPLA DSP Basic Setup

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.

- O Gateway 1 will require ports 10020-10051 forwarded.
- O Gateway 2 will require ports 10052-10083 forwarded.
- O Gateway 3 will require ports 10084-100115 forwarded.
- O Gateway 4 will require ports 10116-10147 forwarded.
- O Gateway 5 will require ports 10148-10179 forwarded.
- O Gateway 6 will require ports 10180-10211 forwarded.
- O Gateway 7 will require ports 10212-10243 forwarded.
- O Gateway 8 will require ports 10244-10275 forwarded.

Table 2 Port Table

Ports	UDP	ТСР
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 Size	Gateway	IP Address	RTP Port	RTCP Port	UDP
	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 (Static and Dynamic Configuration)

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

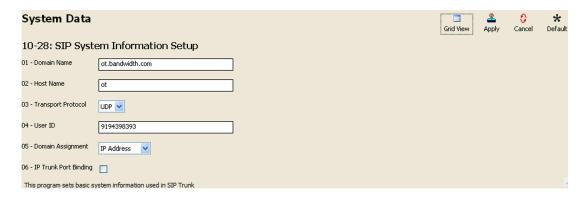


Figure 7 SIP System Information Setup

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 provide 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 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 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).

■ If Program 10-23-04 is set for system interconnection, this entry must be numeric as 10-23-04 does not allow text entry - only numeric.

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

Determine the type of Domain Assignment. Set this entry to 1 (Domain name).

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 Configuration)

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

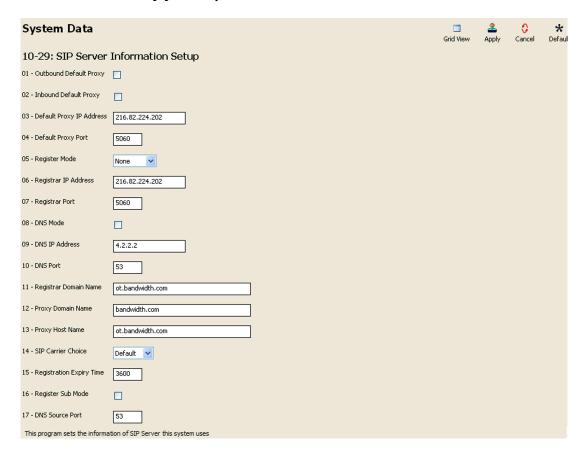


Figure 8 SIP Server Information Setup

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 SV8100 will check the settings in the remaining 10-29 programs.

10-29-03 : SIP Server Information Setup – Default Proxy IP AddressDefine 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.

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).

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 Default.

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 3600 seconds so that the Bandwidth 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)

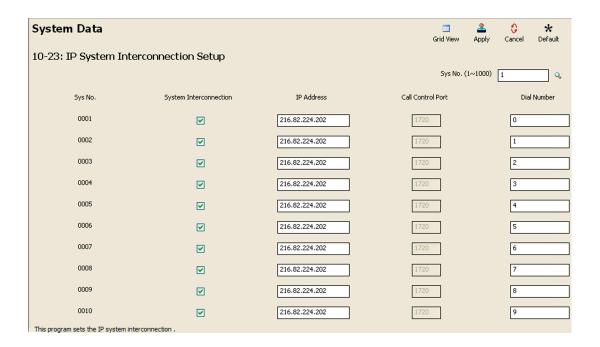


Figure 9 IP System Interconnection Setup

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.

3.7 SIP Authentication Information Setup

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

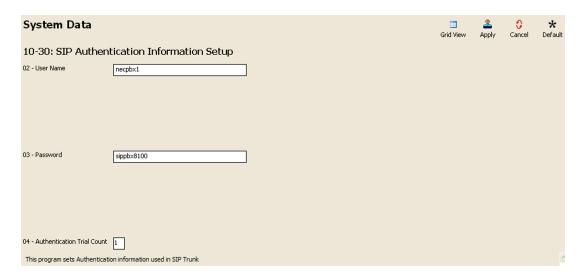


Figure 10 SIP Authentication Information Setup

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

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

Entries: 48 characters maximum.

NEC recommends using "nec8100" if this information is not supplied by your service provider.

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

Enter the Bandwidth authentication password. This information is provided by your SIP Service Provider.

Entries: 48 characters maximum.

3.8 SIP Trunk Registration

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

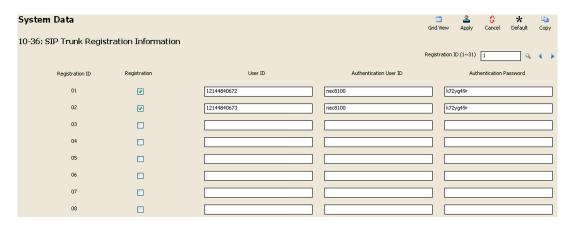


Figure 11 SIP Trunk Registration Information

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.

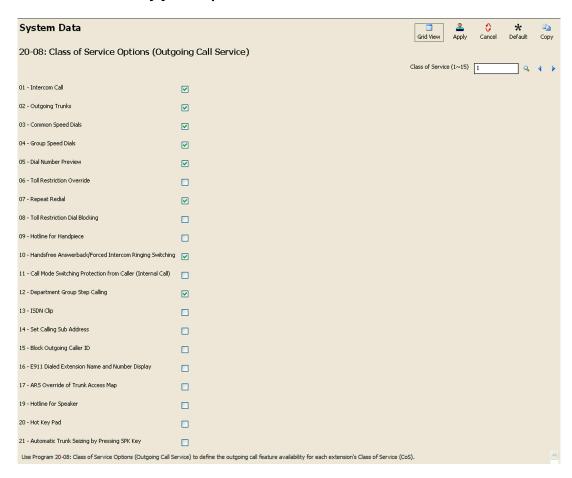


Figure 12 Class of Service Options

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

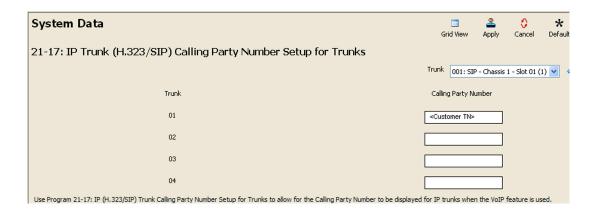


Figure 13 IP Trunk (H.323/SIP) Calling Party Number Setup for Trunks

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.

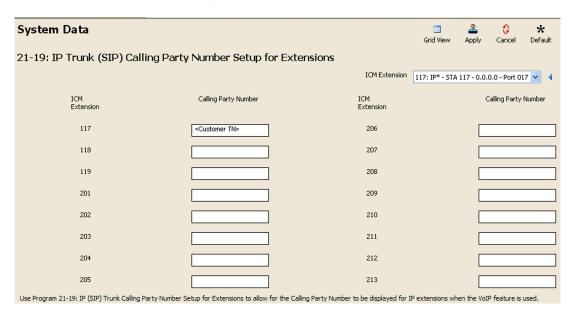


Figure 14 IP Trunk (SIP) Calling Party Number Setup for Extensions

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.

3.13 DID (TN to ext map)

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

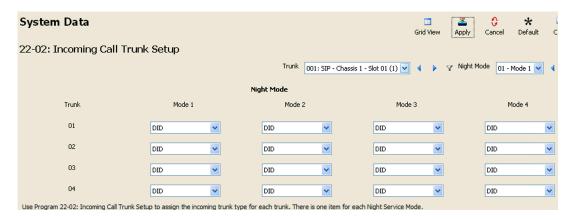


Figure 15 Incoming Call Trunk Setup

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.

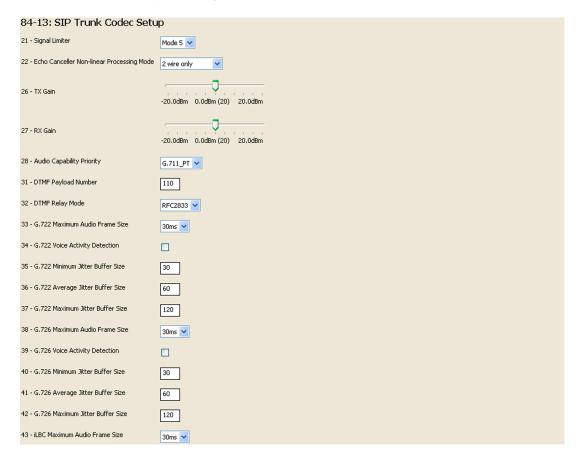


Figure 16 SIP Trunk Codec Setup

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.

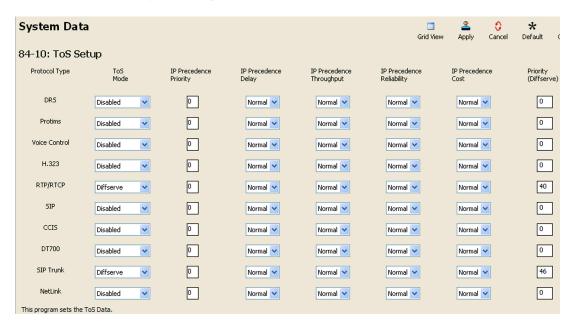


Figure 17 ToS Setup

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 18 SIP Trunk Basic Setup

84-14-11 : SIP Trunk Basic Setup – URL/To Header Setting Information Set this program to SIP UA 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
	Router Configuration	Check Router Configuration
No Calls IN/Out	NEC Configuration	Check NEC Configuration
	 Unqualified IP Address 	Note WAN IP Address and Contact Provider
No Calls Out	NEC Configuration	Check NEC Configuration
No cans out	 Unqualified IP Address 	Note WAN IP Address and Contact Provider
No Calls In	NEC Configuration	Check NEC Configuration
NO Calls III	 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 Bandwidth
	Internet Access Issues	Call Internet Access Provider
Call Dropping	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
	Packet Loss or Latency on LAN	○ Check LAN
Missing Parts of Words	Packet Loss or Latency on WAN	Check with Internet Access Provider
	Jitter Buffer Configuration	O Check with NEC

SECTION 5 NEC INROUTER CONFIGURATION SOFTWARE VERSION 9.1.0

The following are screen shots taken from an active NEC InRouter while interfaced with BandTel SIP service. They are provided as a guide; your situation could be different from what is shown. The IP Address shown will of course not apply as you will need the IP Address assigned to your unique situation.

5.1 Network

Networking configuration information for public and private networks.

LAN Interface Settings:	
IP Address:	10.10.3.1
Subnet Mask:	255.255.255.0
Enable VLAN support	
WAN Interface Settings:	
	□ DHCP
	Static IP Address
	© EVDO
	© <u>T1/E1</u>
IP Address:	143.101.120.218
Subnet Mask:	255.255.255.0
Network Settings:	
Default Gateway:	143.101.120.1
Note: In case of dynamic links, this DNS server ac server address obtained from the Servers. Default obtained from the server , if left blank .	ddress will override the DNS t value for dynamic links is
Primary DNS Server:	172.24.21.123
Secondary DNS Server:	172.24.21.48

Figure 19 Network Configuration Example

5.2 Firewall

Enable Firewall for WAN:

Basic WAN Firewall Settings:	
These setting apply to services that are running on the Sy	rstem.
Allow HTTP access through firewall:	V
Allow HTTPS access through firewall:	V
Allow TELNET access through firewall:	
Allow SSH access through firewall:	
Allow SNMP access through firewall:	
Allow TCP Port:	8080
Allow UDP Port:	
Trusted Management Addresses:	
Apply basic settings configuration only to the following addresses:	
^	Address can be host IP or network/mask, e.g. 10.10.10.1 or 10.10.10.0/24. To delete an entry,
←	highlight and delete it.
Forwarding WAN Firewall Settings:	
These settings apply to packets being forwarded to systemall.	ms running behind the
Enable Firewall Logging:	
Enable PPTP Server Pass-through:	
PPTP Server IP Address:	

Figure 20 Firewall Settings Example

5.3 NAT

NAT dynamic and static settings:

Dynamic NAT

Dynamic NAT allows a system with a private address to be mapped to a public address, allowing the system to access the public network.



Static NAT

Static NAT is a special form of NAT that allows the system to map public IP address and port pairs to a specific IP address and port running on the LAN. The public IP address can be either the system's WAN address or another IP address in the same subnet. For Static NAT to function, WAN NAT must be enabled.

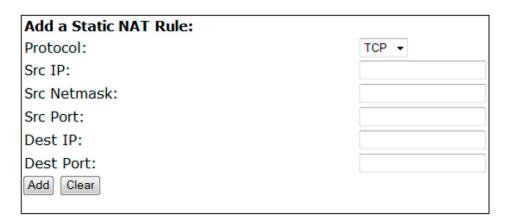


Figure 21 NAT Configuration Example

5.4 SIP Settings

SIP Protocol settings.

The SIP Server settings specify the address and port that all client traffic shall be forwarded to. 216.82.224.202 SIP Server Address: SIP Server Port: 5060 Use Custom Domain: SIP Server Domain: bandwidth.com List of SIP Servers: Create Enable Multi-homed Outbound Proxy Mode: Enable Transparent Proxy Mode: Limit Outbound to listed Proxies / SIP Servers: Limit Inbound to listed Proxies / SIP Servers: Stale Timer The stale timer, if set, is used to automatically delete SIP clients that have not registered within the given time period. Stale client time (m): 1440 Registration Rate-Pacing parameters are available on the Survivability page. Reset Submit

Figure 22 SIP Settings Example

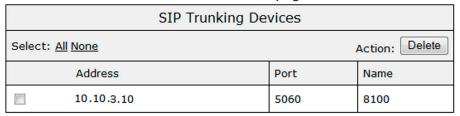
5.5 SIP Trunking

Configuration of SIP trunking devices.

SIP Trunking devices

A SIP trunking device can be a PSTN gateway, or similar device, that does not issue REGISTER messages. Calls will be forwarded to the device based on the dial-plan rules below.

If VLANS are enabled, the SIP trunking device needs to be in the same VLAN as defined in the VoIP ALG page.



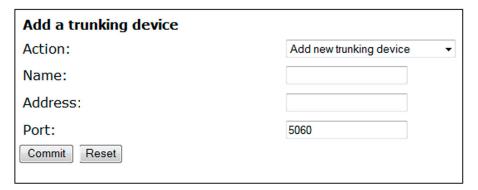


Figure 23 SIP Trunking Example

5.6 Rules

Rules are used to forward and/or modify incoming and outgoing calls. There are three types of rules:

- Inbound from server to trunking device
- Outbound from trunking device to server
- O Redirect from local telephone to trunking device (w/o routing to server)

Outbound rules can match against and/or modify either the calling or called number. Inbound and redirect rules operate on the called number only. Stripped and added digits always apply to the left-most digits of the DID.

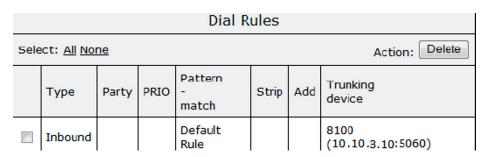


Figure 24 Dial Rules Example

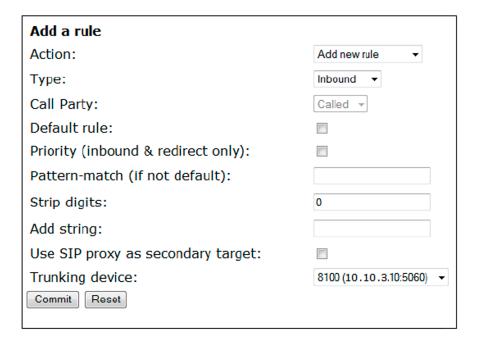


Figure 25 Add A Rule Example

5.7 Header Transformation

These header transformation rules are applied to all SIP trunking devices. They define how specified SIP headers should be transformed when forwarding to the SIP Server.

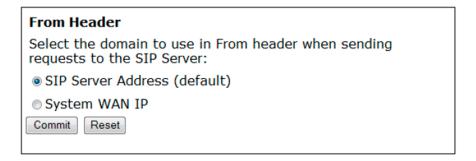


Figure 26 From Header Example

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