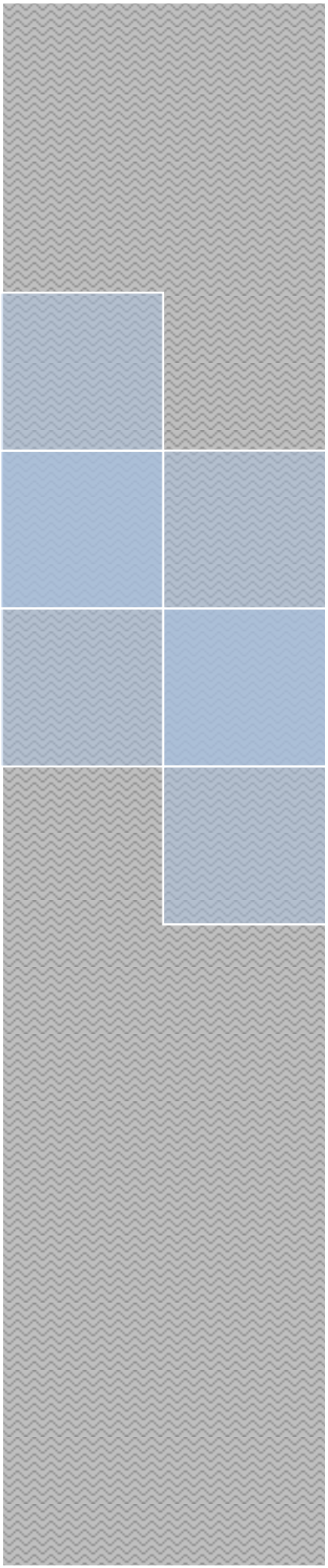

Netborder SS7 to VoIP Gateway

User Manual

Date: Nov 8 2012: Version: 1.12

Revision History

Document Revision	Date	Description of Changes
1.12	Nov 08 2012	Theory section and minor updates
1.11	Sep 23 2012	Quickstart section 5, Layout change
1.10	Sep 14 2012	Added MG Status, VLAN auto startup on eth config
1.09	Sep 12 2012	Updated network setup overview, snmp and monitoring
1.08	Sep 11 2012	Updated channel map, added more background info
1.07	Sep 09 2012	Added T38_Fax option in Media Gateway profile.
1.06	Sep 05 2012	Added rtpip on megaco profile
1.05	Aug 31 2012	Cosmetic Changes A.O, Added Megaco Overivew, VLAN routes, Reload
1.04	Aug 23 2012	USB CLI, Static Routes, Alarms, Improved instructions
1.03	Aug 22 2012	Pinout label
1.02	Aug 22 2012	VLAN, Factory Reset, Static Routes, Eth Options, usb console, DC PSU info
1.01	Aug 19 2012	Added extra diagrams, Media, SIP, Relay, Dialplan, Update, Cables, Appendix
1.00	Aug 2012	Initial revision of the document.

Conventions
This font indicates screen menus and parameters.
<> indicates keyboard keys (<Enter>, <q>, <s>).

NOTE
Notes inform the user of additional but essential information or features.

CAUTION
Cautions inform the user of potential damage, malfunction, or disruption to equipment, software, or environment.

Sangoma Technologies provides technical support for this product.
Tech-support e-mail: support@sangoma.com

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sangoma.com

Sangoma

Netborder SS7 to VoIP GW User Manual

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1 Product Overview

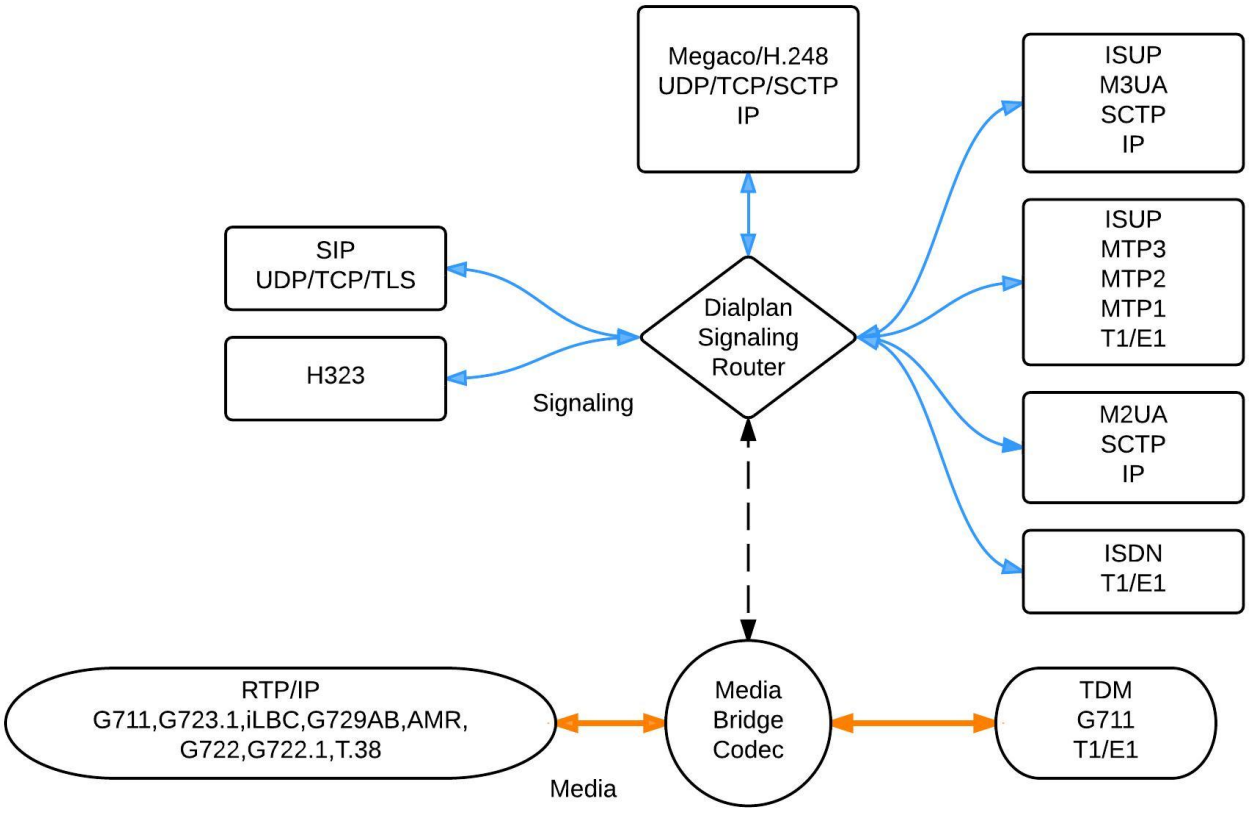
The NetBorder SS7 to VoIP Gateway is Sangoma's Carrier Class TDM to SIP VoIP Gateway product. For short, it is often referred to as NSG.

1.1 Features / Advantages

- Any to any switching gateway.
  - Ability to run all endpoints/protocols on single software image and appliance
  - SS7, Sigtran, SIP, H.323, Megaco Media Gateway, Signaling Gateway
  - Flexible dial plan to route from any endpoint to any endpoint
- Scalable and very high density
  - Up to 32 E1 per appliance appliance
  - Can scale up to 288 E1s in relay mode where multiple systems act as one
  - Transcoding available on all channels
- Extensive VoIP Signaling
  - SIP, H.323, Megaco/H.248
- Full featured SS7/Sigtran Signaling
  - SS7 ISUP Signaling with several national variants
  - ITU, ANSI, Bellcore, France, UK, China, India and Russia
  - Sigtran, M3UA, M2UA
  - Sigtran signaling gateway
- ISDN signaling
  - Q.931, QSIG,
- Faxing and Media Support
  - Pass-through
  - T.38
- Wide range of narrowband and wideband codecs supported
  For any-to-any codec transcoding
  - G.711, G.729, AMR
- Robust implementation with distribution

1.1.1 Any to Any Signaling and Media Gateway

- Route any signaling traffic from eny signaling endpoint.
- All protocols and signalling suppored from single gateway image.
  - Ability to change from Megaco GW to SIP gateway via config change.
- Route media with transcoding/dtmf/T.38 to/from end media endpoint.



NOTE:

- Limitations exist when running specific signaling combinations at same time.
  - Eg: M2UA SG cannot run at the same time as ISUP+MTP3+MTP2
  - Some codes such as AMR will reduce session capacity.
  - No reduction of capacity for G711, G729, iLBC

1.2 TDM T1/E1 Interfaces

- Electrical G.703.6/G.704 balanced
- Minimum 4 T1/E1
- Maximum 32 T1/E1 (960 ds0/sessions) per appliance
- Transcoding supported on all channels
- Extend capacity over 960 ports via ISUP relay feature and multiple appliances.

1.3 Ethernet Network Interfaces

- Two Gigabit network interfaces

1.4 VoIP Protocols

1.4.1 SIP

- SIP V2 / RFC 3261 RFC 3261 Session Initiate Protocol
- RFC 2976 SIP INFO Method
- RFC 3398 ISUP-SIP Mapping
- RFC 3515 Refer Method
- RFC 2327 Session Description Protocol
- RFC 3581 An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing
- RFC 3892 Referred-By Mechanism
- RFC 3891 "Replaces" Header
- RFC 3551: RTP/AVP
- RFC 3515: REFER
- RFC 2617: HTTP Digest Authentication
- SDP Bypass
- NSG exports all SS7 parameters via SIP custom X headers.

1.4.2 Megaco/H.248 & MGCP

- MEGACO Protocol Version 1.0, Internet RFC3525
- H.248.1 Version 1 Implementors' Guide, 13 April, 2006
- H.248 Sub-series Implementors' Guide, 13 April, 2006
- ITU-T recommendation H.248.1 Version 3 (09/2005): "Gateway control protocol"
- SDP : Session Description Protocol, Internet RFC 2327 & RFC 4566

- H.248.2 – Fax etal Package
- H.248.14 – Inactivity Timer Package
- Augmented BNF for Syntax Specifications: ABNF, Internet RFC 2324
- DTMF support
    - RFC 2833/4733 - "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals"
    - In-band DTMF detection/generation

1.4.3 H.323

Call Handling

- H.225.0 : Call signaling protocols and media stream packetization for packet-based multimedia communication systems
- H.245 : Control protocol for multimedia communication
- H.235, H.450, H.460

DTMF support

- RFC 2833/4733 - "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals"
- In-band DTMF detection/generation

1.5 TDM Protocols

1.5.1 SS7

- ISUP, MTP3, MTP2, MTP1, M3UA (RFC 3332), M2UA (RFC 3331), Relay
- Variants
    - ITU, ANSI, Bellcore, UK, China, France Spirou, India and Russian
- MTP2
    - ITU 88 & 92, ANSI 88 & 92, Peoples Republic of China
- MTP3
    - ITU 88 & 92 & ETSI, ANSI 88 & 92, 96 & Telcordia (including ANSI MTP3-B), China
- ISUP
    - ITU 88, 92 & 97, 2000, Telcordia 97, ANSI 88, 92, 95 and ETSI v2,v3
    - SPIROU, China, UK, Russia, India
- SCTP (RFC 2960)

1.5.2 ISDN

- CCITT 88, User & Network Side PRI/BRI
- AT&T 4ESS User Side - PRI, Network Side - PRI
- 5ESS User Side - PRI/BRI, Network Side - PRI/BRI
- DMS-100 User & Network Side - PRI/BRI
- ETSI User & Network Side - PRI/BRI
- Australian Telecom User Side - PRI/BRI and Network Side - PRI
- National ISDN-1 User Side - BRI
- NTT User & Network Side - PRI/BRI
- National ISDN-2 User & Network Side - PRI
- Q.SIG (PRI)
- LAPD & TEI Management

1.6 Call Routing

Configurable and extendable XML-based dial plan and routing rules XML Dialplan can be used to create complex routing scenarios between SIP and TDM.

- Call routing based on any call parameter present in a SIP or SS7 IAM message.
- Deep integration with signaling stacks
- Ability to use external applications to build complex routing logic*

1.7 Media Processing & Transcoding

Wide range of codecs supported for any to any codec negotiation.

- G.711
- G.723.1
- G.726
- iLBC
- G.729AB
- GSM
- G.722
- AMR
- G.722.1

1.8 Echo Cancellation & VQE

Telco grade hardware based echo canceling and Voice processing

- G.168-2002 with 128ms tail
- Noise cancellation
- DTMF Removal
- DTMF Detection
- FAX Detection
- Automatic Gain Control

1.9 DTMF Detection and Generation

Sangoma NSG gateway supports multiple DTMF internetworking scenarios.

- RFC 2833 Tone Relay
- In-band
- SIP INFO
- Hardware and software DTMF detection and generation

1.10 Management and Configuration

Sangoma NSG configuration, operation and troubleshooting are designed to be flexible.

- Web GUI
- Command line interface via ssh and usb to serial
- Call detail records in XML format
- Detailed logs with user configurable file size and auto rotation

1.11 Monitoring

- SNMP v1, 2, 3
- RTCP

1.12 Accounting

- Radius

1.13 Shipping Options

SKU	DESCRIPTION
SS7-NSG-AP04	Up to 4 E1/T1, ISUP to SIP, codec support, 4 signaling links, up to 12 point codes
SS7-NSG-AP08	Up to 8 E1/T1, ISUP to SIP, codec support, 8 signaling links, up to 12 point codes
SS7-NSG-AP16	Up to 16 E1/T1, ISUP to SIP, codec support, 16 signaling links, up to 12 point codes
SS7-NSG-AP32	Up to 32 E1/T1, ISUP to SIP, codec support, 32 signaling links, up to 12 point codes

1.14 Support and Professional Services

Sangoma Engineers are here to support your success. Whether you need technical support and software maintenance, training, consultation and installation services, Sangoma can help you. Please contact your Sales representative for more information.

2 NSG Product Information

2.1 NetBorder SS7 to VoIP Gateway Appliance

Fully integrated Industrial grade telco appliance running a customized OS, Netborder SS7 to VoIP application and TDM interfaces configured and installed by Sangoma.
NSG Appliance provides a full-featured, carrier-class VoIP deployment while leveraging the flexibility and cost effectiveness of standard computing platforms.

2.1.1 Hardware Specifications

- Industrial grade telecom appliance
- Size: 1U and 2U - 19'' Rackmount
- Min Capacity: 4 T1/E1 (1U)
- Max Capacity: 32 T1/E1 (2U)
- Power: AC, DC, Redundant
- AC Power Supply (Single)
  -
- DC Power Supply (Redundant)
  - The Input Current for -48VDC, is 12.0A (RMS).
  - With Inrush Current of 20.0A MAX.
- Depth: 20''
- Weight: 36lb
- Full Spec on Sangoma Site

2.2 NSG Shipping Box Contents

The first three tasks for installing and operating the Netborder SS7 to VOIP Gateway are

- Unpack
- Inspect
- Power up.

Carefully inspect the NSG Appliance for any damage that might have occurred in shipment.

If damage is suspected, file a claim immediately with the carrier, keep the original packaging for damage verification and/or returning the unit, and contact Sangoma Customer Service.

2.2.1 What is included in the box

- Netborder SS7 to VoIP Appliance
  - Appliance can be 1U or 2U depending on model ordered
- Power Cable
  - AC cable in case of AC PSU (black cable)
  - DC cable in case of DC PSU (RED & Black cable)
- Mounting Brackets
- Quickstart user guide

2.2.2 What is not included

- Appliance Rails
  Appliance Rails are made based on industry standard. They can be purchased from any third party equipment vendor. For example: General Devices http://www.generaldevices.com/.

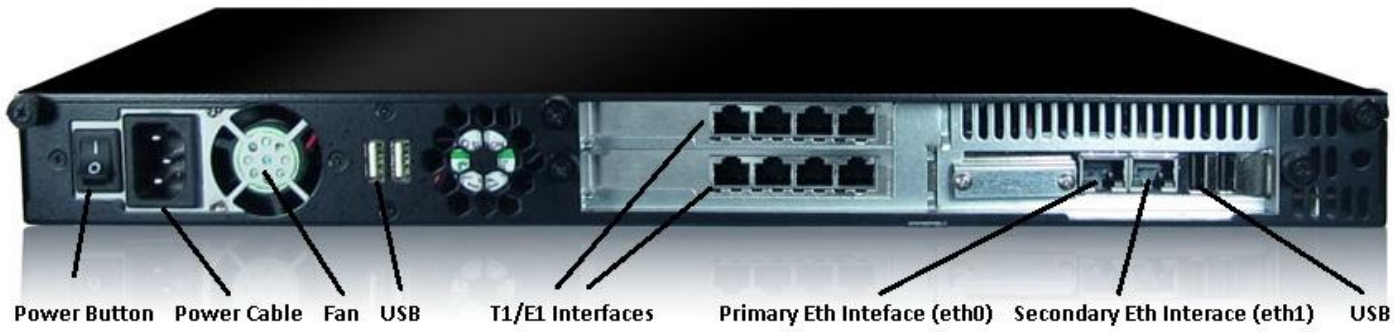
2.2.3 *Front Panel*

- Front Panel Reset/Power button is used for:
  - Factory Reset
    - Press 10 to 15 times every second.
  - Soft Reboot
    - Press 5 times every 3 seconds
  - Power on/off
    - Hold for 10 seconds
  - Nothing will happen if pressed once
    - To avoid accidental restart.
    - Caution: From NSG SW release 5.0
  - Refer to Factory Reset section.

- USB Ports can be used for Serial Console
  - Refer to Serial Console section.

- RAID1 SSD
  - The RAID1 is NOT Hot Plug
  - NSG appliances use industrial grade SSD
  - One must power down the machine in order to change SSD/HDD
  - Contact Sangoma Support for part replacement.

2.2.4 Rear Panel 1U



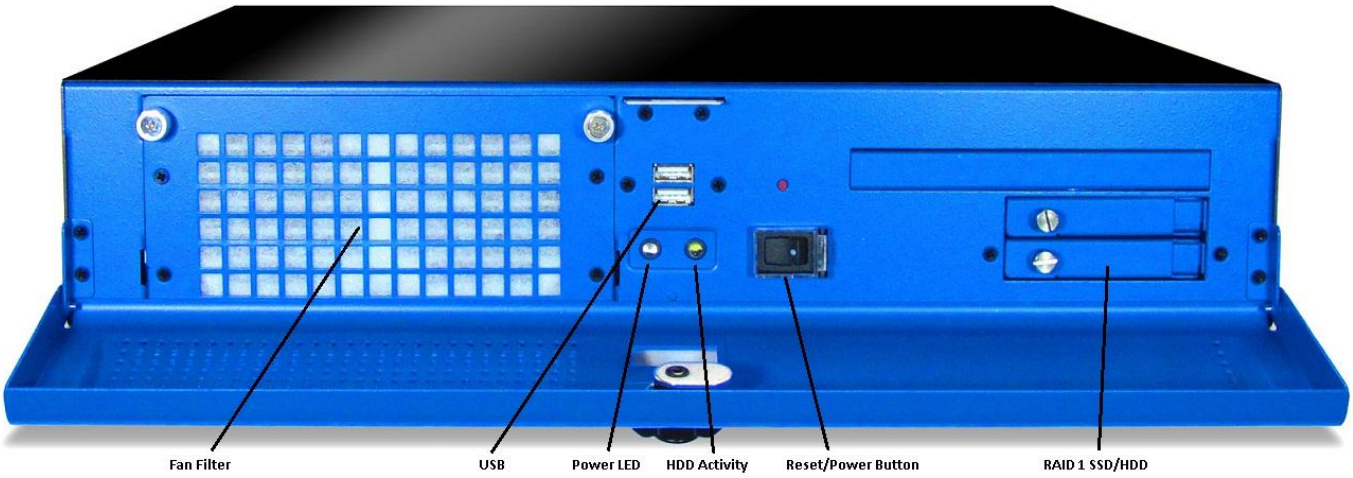
- Power button
  - Used to turn off the power supply
  - Not for Factory Reset

- USB Ports can be used for Serial Console
  - Refer to Serial Console section.

- PSTN T1/E1 Interfaces
  - RJ45 Connections

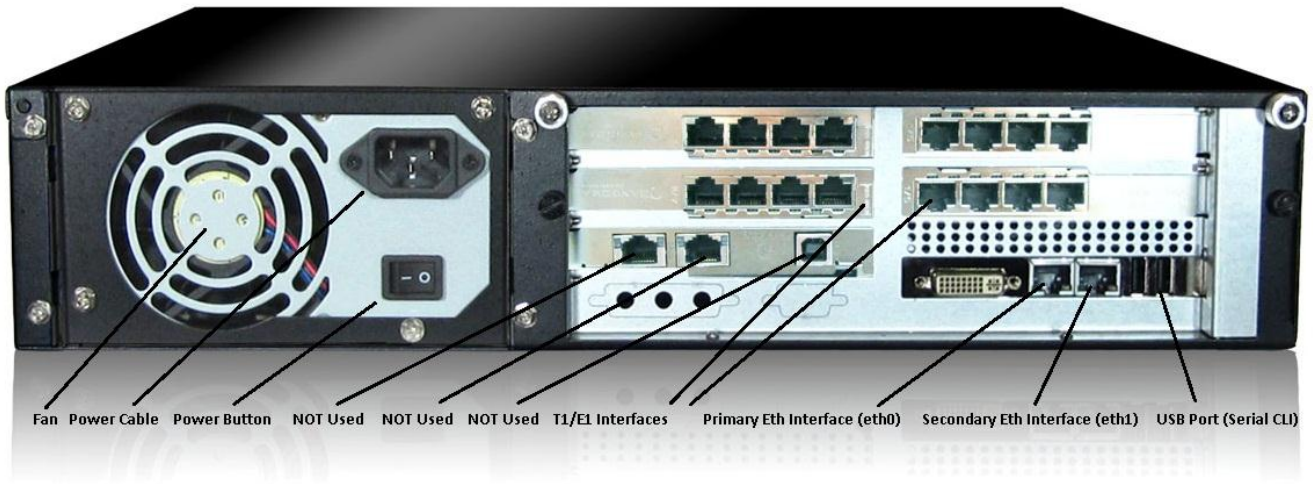
- Primary Eth Interface (eth0): Gig Ethernet Port
  - This adapter must be plugged into the LAN
  - SIP Signaling and RTP Media will flow through this device.
  - WebUI identifies this device as "eth0"
- Secondary Eth Interface (eth1): Gig Ethernet Port
  - This adapter is optional
  - It can be used for Monitoring and Statistics
  - WebUI identifies this device as "eth1"
- USB Ports
  - Used for Serial Console
  - Can be used re-flash the appliance
  - Future use: active/standby redundancy*

2.2.5 Front Panel 2u



- Fan Filter
- USB
  - Used for Serial CLI
  - Refer to the Serial CLI Section
- Power LED
- HDD Activity LED
- Front Panel Reset/Power button is used for:
  - Factory Reset
    - Press 10 to 15 times every second.
  - Soft Reboot
    - Press 5 times every 3 seconds
  - Power on/off
    - Hold for 10 seconds
  - Nothing will happen if pressed once
    - To avoid accidental restart.
    - Caution: From NSG SW release 5.0
  - Refer to Factory Reset section.
- RAID1 SSD
  - The RAID1 is NOT Hot Plug
  - NSG appliances use industrial grade SSD
  - One must power down the machine in order to change SSD/HDD
  - Contact Sangoma Support for part replacement.

2.2.6 Rear Panel 2U



2.2.6.1 Rear Panel Description

- Fan
- Internal Power supply
  - Default AC, non-redundant
  - Option: DC or AC Redundant
- Power Button
  - Used to turn off the machine
  - Not used for Factory Reset.
- Unused 2x Gig Ethernet Port
  - Not used at this time. Should NOT be plugged into the LAN.
- Primary Eth Interface (eth0): Gig Ethernet Port
  - This adapter must be plugged into the LAN
  - SIP Signaling and RTP Media will flow through this device.
  - WebUI identifies this device as "eth0"
- Secondary Eth Interface (eth1): Gig Ethernet Port
  - This adapter is optional
  - It can be used for Monitoring and Statistics
  - WebUI identifies this device as "eth1"
- USB Ports
  - Used for Serial Console
  - Can be used re-flash the appliance
  - Future use: active/standby redundancy*

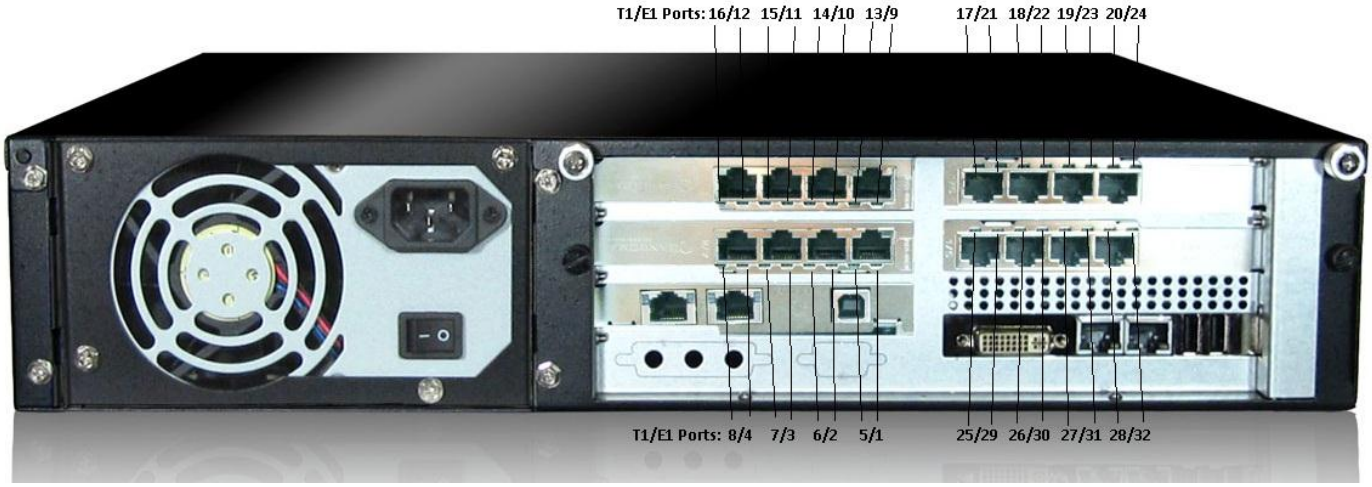
2.3 NSG T1/E1 Port Identification

Sangoma T1/E1 Interface boards come with two types of RJ45 Connections

- Low density Interface Boards
  - Single Port Interface Board
  - Dual Port Interface Board
  - Quad Ports Interface Board
  - RJ 45 Connector
    - Each RJ45 Connector connects to a single T1/E1 line.
  - Cable Type
    - Standard Cat5/Cat6 straight cable.

- High density Interface Boards
  - Eight Port Interface Board
  - RJ45 Connector
    - Each RJ45 Connector connects to two (2) T1/E1 lines.
  - Cable Type
    - A special Y cable is needed to connect 2 T1/E1 lines into a single RJ45 port.
    - If a standard Cat5/6 cable is used, only lower ports of the 8 port interface board will be used/connected.

- Board Type Identification
  - The number of LED on the T1/E1 Interface boards indicates the number of T1/E1 ports supported.
  - In case of 8 port T1/E1 board, there will be 2 LED per T1/E1 port.



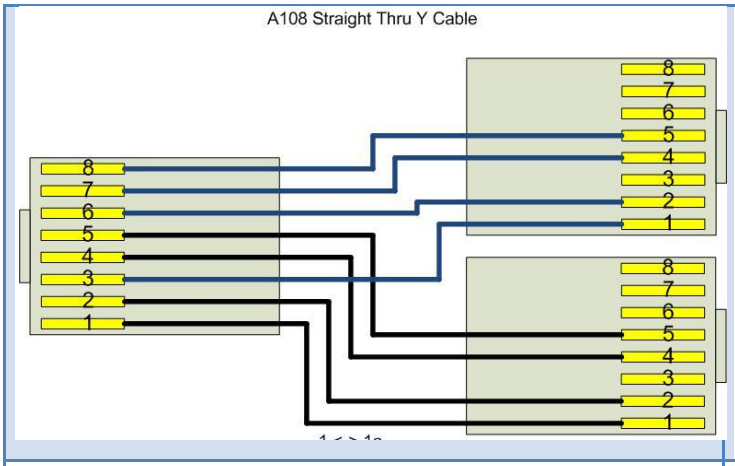
2.3.1 Cable Pinouts: T1/E1

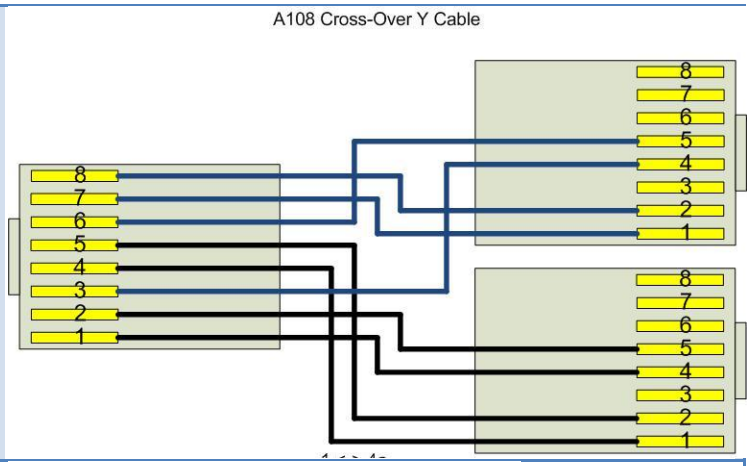
NSG Appliance utilizes Sangoma TDM T1/E1 digital board adapters.

- A101DE – 1-port E1/T1 board
- A102DE – 2-port E1/T1 board
- A104DE – 4-port E1/T1 board
- A108DE – 8-port E1/T1 board*

Eight Port Board Information	
The A108D board has dual purpose RJ45 connector, as it provides access to two T1/E1 ports from a single RJ45 Female connector. NOTE There are **two** LED per RJ45 connector.	

Eight Port Board *Straight Cable*	*Eight Port Board* *Cross Over – Back-to-Back Cable*
Y Cable for A108 connects 2 separate T1/E1 (straight). This is to connect the A108 board RJ45 ports to Telco Lines.	Y Cable for A108 connects 2 separate T1/E1 (cross). This is to connect the A108 against another T1/E1 card in back to back mode.
A = port N; B = port N + 4 1 <-> 1A [Rx ring] 2 <-> 2A [Rx tip] 3 <-> 1B 4 <-> 4A [Tx ring] 5 <-> 5A [Tx tip] 6 <-> 2B 7 <-> 4B 8 <-> 5B	A = port N; B = port N + 4 1 <-> 4A 2 <-> 5A 3 <-> 4B 4 <-> 1A 5 <-> 2A 6 <-> 5B 7 <-> 1B 8 <-> 2B





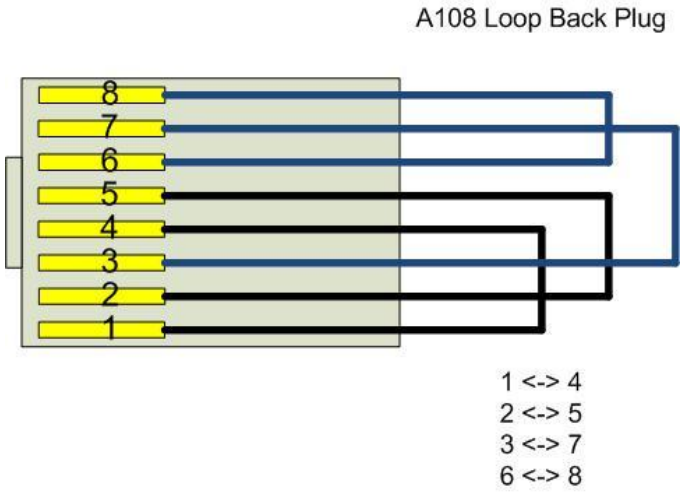
T1/E1 "Portsplitter" Cable
T1/E1 Split Cable for the Eight Port Board
Standard | ROHS: Yes | Length: 6'

SKU: CABL-630

A108D Loop Back Cable

This is to connect an A108 port in loopback mode

1 <-> 4
2 <-> 5
3 <-> 7
4 <-> 1
5 <-> 2
6 <-> 8
7 <-> 3
8 <-> 6



2.4 NSG Appliance Default Configuration

By default the NSG appliance gets shipped with following configuration.

- Static IP **192.168.168.2 / 255.255.255.0**
- Static IP Port eth0 (Primary Ethernet Interface Port)

- WebUI URL **http://192.168.168.2:81**
- Username **root**
- Password **sangoma**

3 User Interface

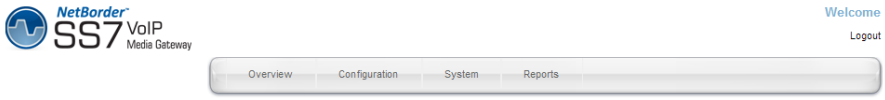
Netborder SS7 to VoIP media gateway provides the user with two interfaces

- WebGUI
  - Web GUI is preferred for almost all operations
  - Configuration, Operations, Statistics, Reports

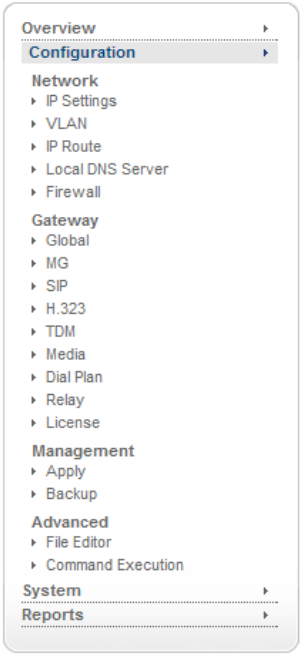
- Console via ssh or usb-serial
  - For power users familiar with Linux operating system, ssh or usb-serial console provides advanced and flexible interface for troubleshooting and automation.

3.1 WebGUI

- WebGUI resides on the port **81**
- Interface provides two identical menus for easy access to all options
  - Top Horizontal Menu



  - Side Vertical Menu



3.1.1 WebGUI Structure

3.1.1.1 Overview

- Control Panel
  - Used to control the gateway operations: start, stop
- TDM Status
  - Provides full overview of gateway utilization and states
- SIP Status
  - Provides full SIP statistics, call count
- MG Status
  - Megaco detail call status report per Profile
- VLAN Status
  - Provides full VLAN statistics, VLAN ID, IP, Netmask for each VLAN.

3.1.1.2 Configuration

- Network
  - Allows network configuration such as IP, Static IP Routes, VLAN, DNS and Firewall
- Gateway
  - Core product configuration
  - Provides configuration of all Signaling and Media Protocols
    - SIP, RTP,H.323, Media Processing, Megaco(MG), SS7/Sigtran (TDM)
  - Routing Logic / Dialplan
    - XML based dialplan
- Management
  - Apply
    - Write all configurations changed and set in Gateway section.
  - Backup
    - Backup all system configurations into a zip file.
    - Recover a system from a backup file
- Advanced
  - File Editor
    - Allows custom file editing for custom configuration
    - Troubleshooting
  - Command Execution
    - Instead of logging into a shell
    - Execute any system command via the WebGUI.

3.1.1.3 System

- Settings
  - Date
    - Set date time and sync to time server
  - Password
    - Change password
  - Shutdown

        - Shutdown or reboot a system
    - Update
        - Software and patch update system
- Resources
    - Processes
        - List of currently running process
    - Services
        - List of all available services
        - SSH service start/stop
- Hardware
    - Self-Test
        - Allow for system software and hw components test.
    - Firmware Update
        - Allows for firmware updates
        - Sangoma TDM boards
        - Sangoma Media processing boards
- Help
    - About
        - Shows system version and version of all important packages.
    - PBX Integration
        - Help documentation

3.1.1.4 Reports

- Dashboard
    - Overview
        - Overview of network interfaces
- Network
    - Network Report
        - Long term usage charts for each network device
    - Protocol Capture
        - PCAP packet capture with filter support for any network interface
- System
    - Gateway Logs
        - Specific gateway logs used to quickly trouble shoot gateway issues
        - Allows for log download
    - Advanced Logs
        - Full system wide logs with filters
    - Hardware Report
        - Full hardware overview and description
        - HDD, Memory and system usage
        - Device enumeration
    - Resource Report
        - Long term statistics

3.2 Console Structure

- Console access via ssh
- Console access via usb-serial
- Shell Commands via WebUI – Command Execution
- Gateway CLI Commands via WebUI – Command Execution
- Operating system is Linux based. Therefore Linux expertise is mandatory.
- WARNING
  - Working in shell is very powerful and flexible, but also dangerous
  - A system can be corrupted, formatted, erased if user makes a mistake.

3.2.1 Connect via SSH

Use default SSH clients on any desktop

- Windows – putty
- Linux – native ssh

On login prompt

- Username: root
- Password: <your custom password>

3.2.2 Connect via USB Serial

- usb to serial cable
  - One must use usb to serial cable + null modem cable
  - If Laptop does not have a serial port then use two usb to serial cables plus null modem cable per diagram below.
- Connect to any usb port on NSG appliance
  - All NSG appliances have usb port on rear panel
  - 2U NSG appliances have usb port in front panel as well.
- Configure Terminal Client on Laptop
  - Windows HyperTerminal
  - Linux – mincomm
- Serial Settings
  - 115200, N, 8, 1 vt100
- Press enter a few times until a login prompt appears.
  - Login via: username: **root**, password: **<your personal password>**

3.2.3 Bash Shell

Once successfully logged into the system, either via ssh or usb serial, user will be offered a bash prompt.

- NSG system is based on Linux
- The initial console after login will be a **bash** shell

3.2.3.1 System Commands

System commands are based on Linux operating systems.
Listed here are some most useful debugging commands.

- tcpdump
  - Provides network capture to a pcap file
  - Can be analyzed using wireshark on Desktop or Laptop.
- ethtool
  - Provides detail network interface information, like Ethernet link status.
  - Run: ethtool <enter> for all the options
  - Eg: ethtool eth0 - show Ethernet status
- Ifconfig
  - Network interface statistics tool
  - Shows error counters on Ethernet and TDM interfaces.
  - Notice the error and overrun counters on wanpipe w1g1 interfaces.
- wanpipemon
  - Sangoma TDM troubleshooting tool
  - T1/E1 alarms
    - wanpipemon –i w1g1 –c Ta

Refer to the appendix for all System Commands

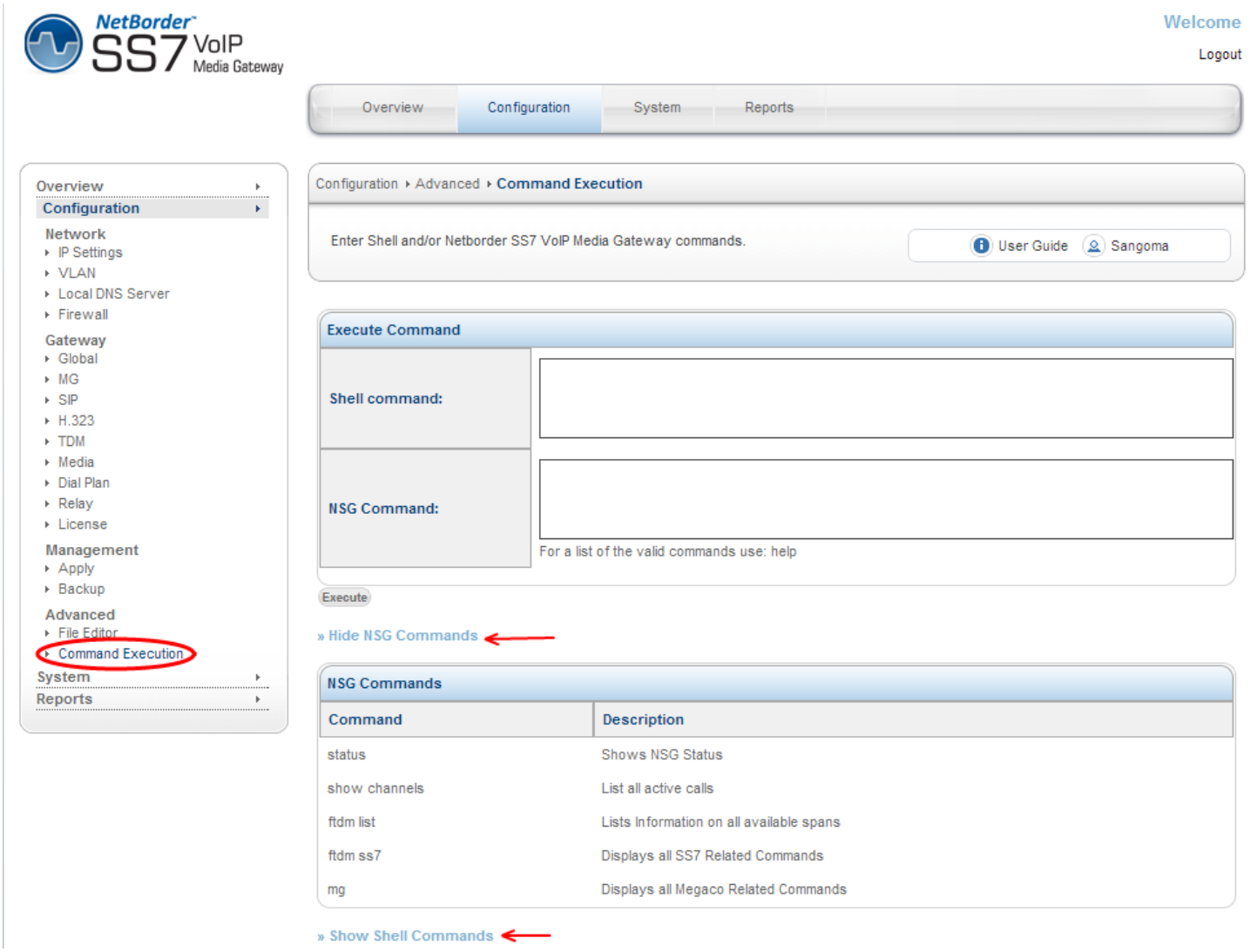
3.2.4 Gateway CLI – nsg_cli

- First log into the System Console (bash)
- Once on bash prompt run
  - **nsg_cli**
- NOTE
  The NSG gateway must be running and started in Control Pannel.

Command	Description
status	Shows NSG Status
show channels	List all active calls
ftdm list	Lists Information on all available spans
ftdm ss7	Displays all SS7 Related Commands
mg	Displays all Megaco Related Commands
log [debug|error|crit]	Set log level to debug loglevel critical

3.3 Shell/CLI from GUI

- Select **Command Execution** from side/top **Configuration** Menu
- Specify a shell or CLI command. Refer to guide below.



Warning

Do not run shell commands that run indefinitely. Such as “ping <ip>”. In such case the webgui will get stuck forever executing the command. In such case, user must login via CLI and kill the process.

In case of ping command one can limit number of pings to perform. eg: ping –c 10 <ip>

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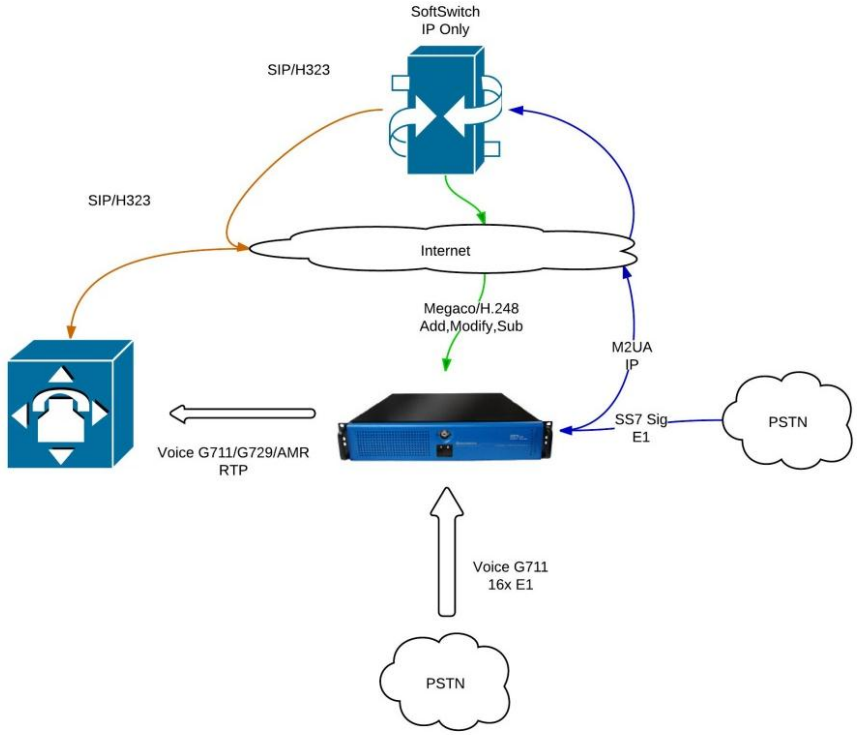
4 Usage Scenarios

4.1 Signaling Gateway: M2UA

- Pass through signaling from TDM to IP
  - MTP2 -> M2UA
- Pass through signaling from IP to TDM
  - M2UA -> MTP2

4.2 Megaco/H.248 Media Gateway: MG + SG

- Third part Softswitch/MGC controlling Netborder SS7 Media Gateway using Megaco/H.248 protocol.
  - Bridge RTP media to TDM Voice 64kb G.711 channels
  - Bridge TDM Voice 64kb G.711 channels to RTP media ports
- Media specific functions
  - Transcoding
  - DTMF
  - T.38 Faxing



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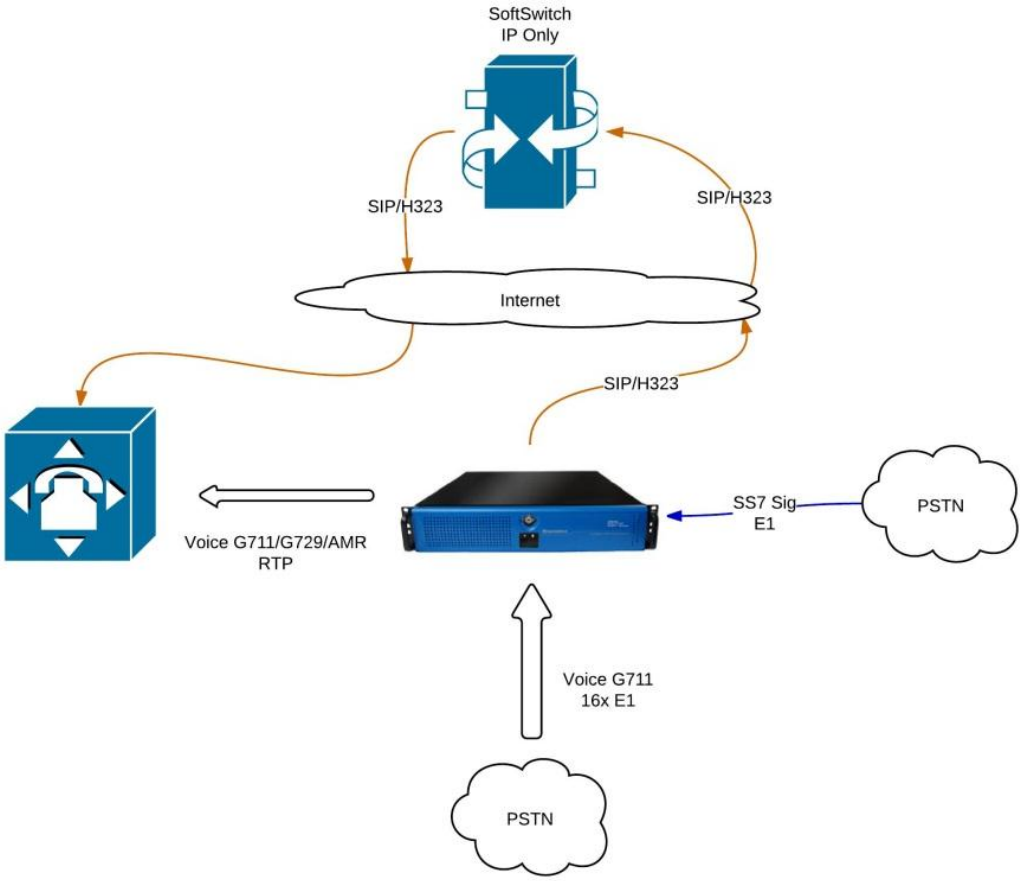
4.2.1 Megaco Quick Configuration

In order to configure the system for Megaco Operation

- Perform the First Boot/Initial Setup
  - Section 5
  - Connect and Power up the system
  - Change password
- Perform the Network Connection
  - Section 6
  - Setup IP, VLAN and Routes
- Perform Megaco Configuration
  - Section 8
  - Create Megaco Profile
    - Configuration -> MG Menu
  - Setup TDM interfaces and bind to Megaco Profile
    - Configuration -> TDM Menu
  - Create Sigtran M2UA Gateway (optional)
    - Configuration -> TDM Menu
- Perform Media Transcoding Configuration
  - Section 11
  - Specify supported codecs.
- Apply configuration
  - Section 12
- Start Gateway
  - Section 17

4.3 SIP/H323 to SS7 ISUP

- Bridge signaling sessions from H.323 to SS7 ISUP
  - Bridge RTP media to TDM Voice 64kb G.711 channels
- Bridge signaling session from SS7 ISUP to H.323
  - Bridge TDM Voice 64kb G.711 channels to RTP media ports
- Media specific functions
  - Transcoding
  - DTMF
  - T.38 Faxing



4.3.1 H323 to SS7 ISUP Quick Start Guide

In order to configure the system for Megaco Operation

- Perform the First Boot/Initial Setup
  - Section 5
  - Connect and Power up the system
  - Change password
- Perform the Network Connection
  - Section 6
  - Setup IP, VLAN and Routes
- Perform Initial Gateway Configuration
  - Section 7
- Perform SS7 ISUP Configuration
  - Section 9
  - Create SS7 ISUP Profile
    - Configuration -> TDM Menu
  - Setup TDM interfaces and bind to SS7 ISUP Profile
    - Configuration -> TDM Menu
- Perform Media Transcoding Configuration
  - Section 11
  - Specify supported codecs.
- Apply configuration
  - Section 12
- Dial Plan
  - Section 13
- Start Gateway
  - Section 17

4.4 Any to Any Signaling and Media Gateway

- Route any signaling traffic from eny signaling endpoint simultaneously.
- Ability to run all protocols together at the same time.
- Route media with transcoding/dtmf/T.38 to/from end media endpoint.

5 First Boot/Initial Setup

- Unpack the NSG shipping box
- Connect the NSG appliance to a power source
- Connect the NSG appliance to LAN
- Connect to NSG appliance via Laptop Browser
- Provision the Appliance
    - Change Password
    - Change Hostname & IP
    - Date Time
    - Self Test
- Initial Provision Done
- Next step is to configure the Gateway.
    - Please refer to usage scenarios in section 5.

5.1 Power Connection

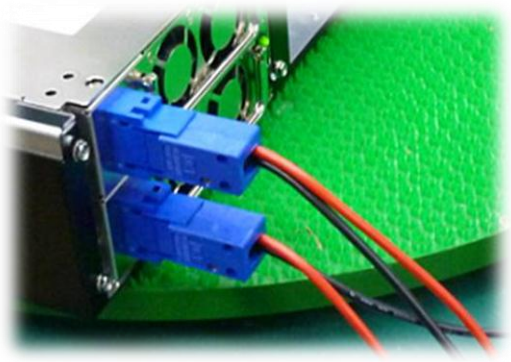
Sangoma NSG comes with three types of power supplies

- AC PSU
    - AC Single PSU (Default)
    - AC Dual-Redundant PSU
- DC PSU
    - DC Dual-Redundant PSU

5.1.1 PSU Connection

- Standard 110V or 220V, 50-60Hz connection.
- Optional Dual-Redundant AC 110V or 220V, 50-60Hz connection.
- Optional Dual-Redundant DC -48V

5.1.2 DC PSU Connection



Connecting cables to a power supply depends on the remote power source.

Power Source Type	Black Wire	Red Wire
If power source -48V	-48V	0V (Ground)
If power source +48V	0V (Ground)	+48V

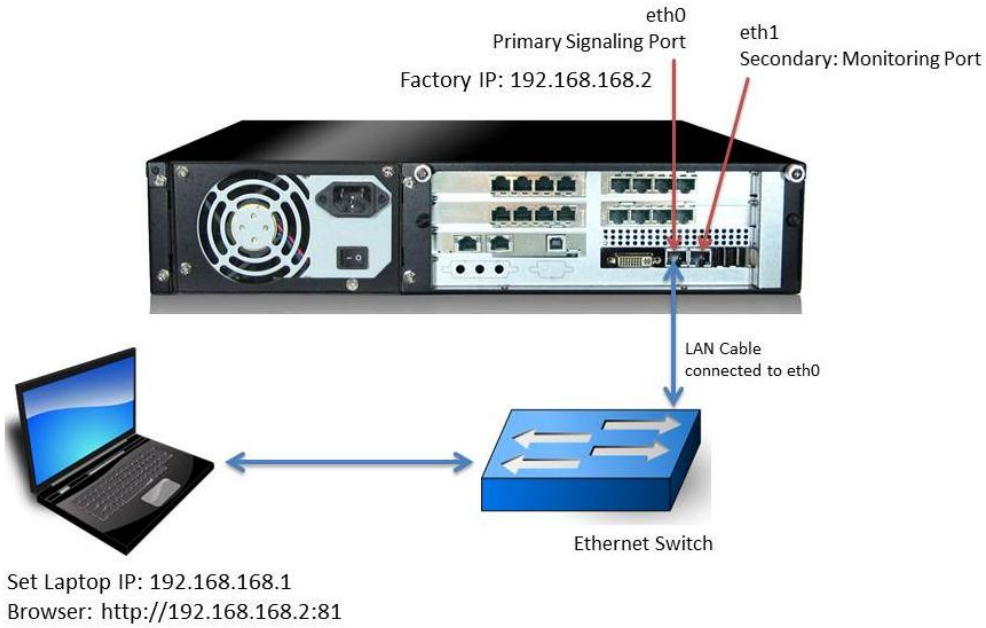
- The PSU **has** voltage reverse protection.
  If the red and black wires are connected the wrong way, the system will not power up. But there will be **no** damage to the PSU or the system.

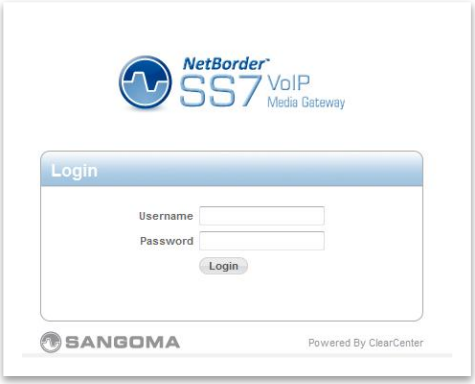
VOLTAGE	**DC -36V ~ -72V**
INPUT CURRENT:	12.0A (RMS). FOR -48 VDC
INRUSH CURRENT	20A (Max)
DC OUTPUT	400W (Max)

5.2 Establishing Initial WebGUI Connection

NSG factory settings are not very useful, as the Primary Ethernet port:eth0 is set to a static IP address. Proceed to connect to the NSG Appliance via Laptop's web browser.

- Connect the Primary Signaling Port: eth0 to a LAN Switch
- Connect Laptop to LAN Switch
- Configure Laptop to IP address: 192.168.168.1/24
- Using Laptop web browser go to URL: http://192.168.168.2:81
- Login via
    - Username: **root**, Password: **sangoma**



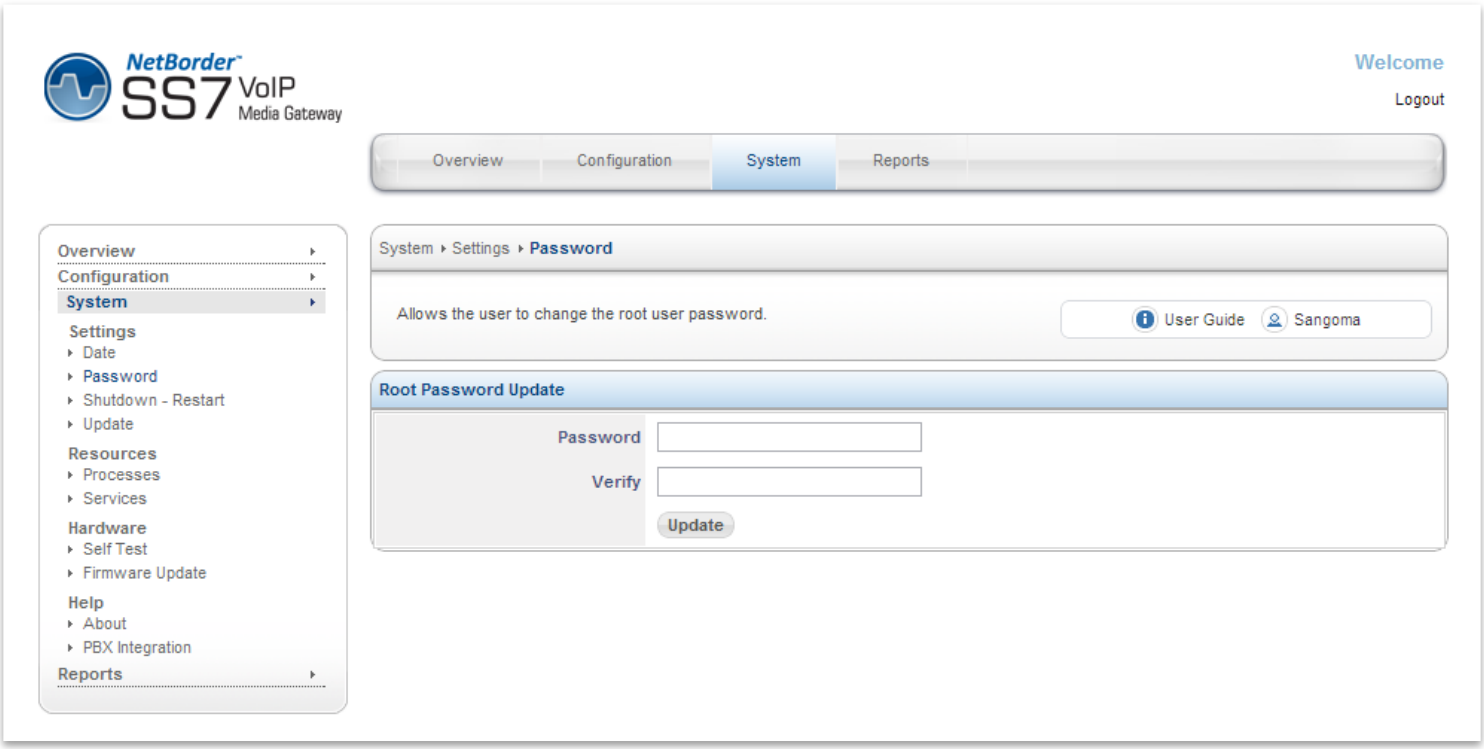


5.3 Change Password

After successful Login, please proceed to change the default password.

Sangoma NSG appliance comes with default password.
For security reasons please change the password.

- Select **Password** page from side/top **System** menu
- Enter your new password
- Press update to save

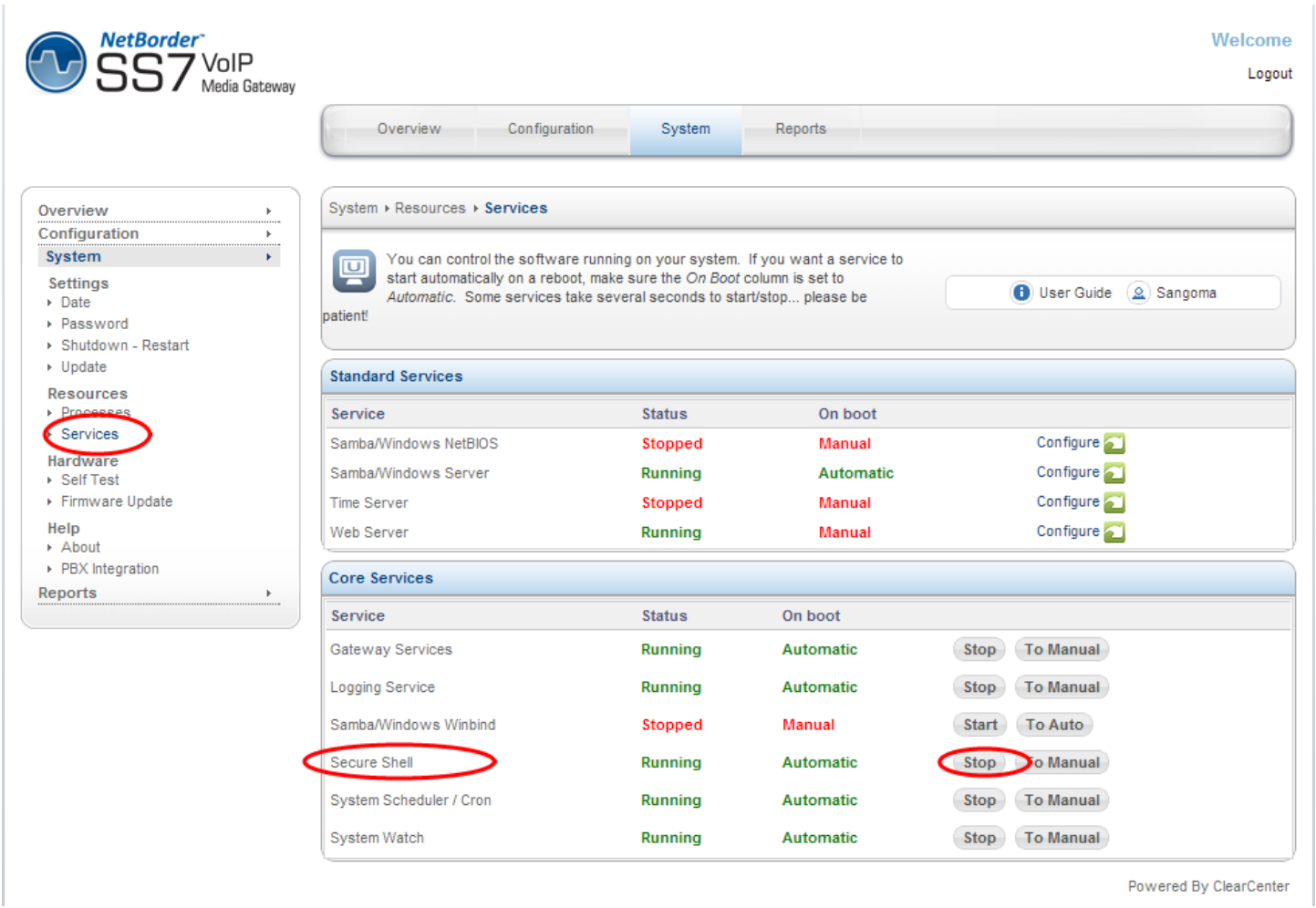


5.4 Console SSH Configuration

By default NSG systems come with SSH **enabled**.

To configure ssh service

- Select **Services** from side/top System Menu
- Enable or disable **Secure Shell** service



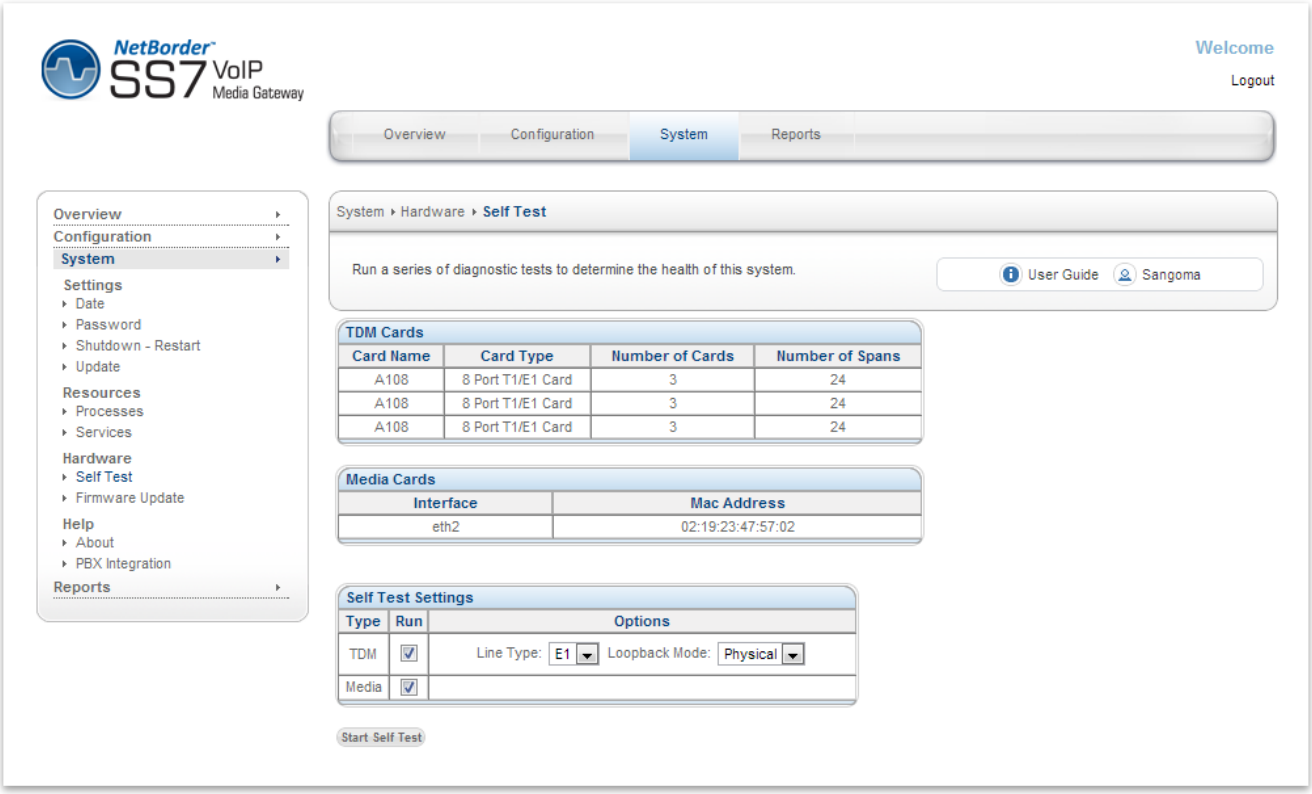
Service	Description	Status
Samba/Windows NetBIOS	Windows NetBIOS server	Not used / Not required
MySQL	MySQL database	Not used / Not required
Samba/Windows Server	Windows File server	Not used / Not required
Time Server	Network Time Protocol	Should be configured and enabled. Note: There must be internet access to reach the NTP service.
Web Server	web/httpd server	Not used / Not required
Gateway Service	NSG VoIP to SS7 gateway	Do not configure it here Use Control Panel
Logging Services	Syslog, logging service	Should be configured and enabled.
Samba/Windows Winband		Not used/ Not required
Secure Shell	SSH server	Should be configured and enabled.
System Scheduler/Cron	System scheduler	Should be configured and enabled
System Watch	System watch	Should be configured and enabled

5.5 Self Test

Self-Test page must be run on initial installation or on any hardware upgrade. It will run a battery of tests on Sangoma TDM and Transcoding hardware.

5.5.1 Running Self-Test

- Select Self Test from side/top System Menu
- If in North America select T1
- If not in North America select E1
- Select Media Transcoding Hardware if present.
- Click Start Self-Test
    - Refer to warning section below



WARNING:

- All services during the Self-Test will be stopped.
- The existing configuration will be restored after Self Test.
- Do not run Self-Test in production!
- Only run Self-Test during on initial setup or during a maintenance window.

The Self-Test can be used to detect:

- Defective TDM hardware
- Defective Media Transcoding hardware
- Miss-configured system device drivers
- PCI Interrupt errors
- Motherboard System issues

5.6 NSG License

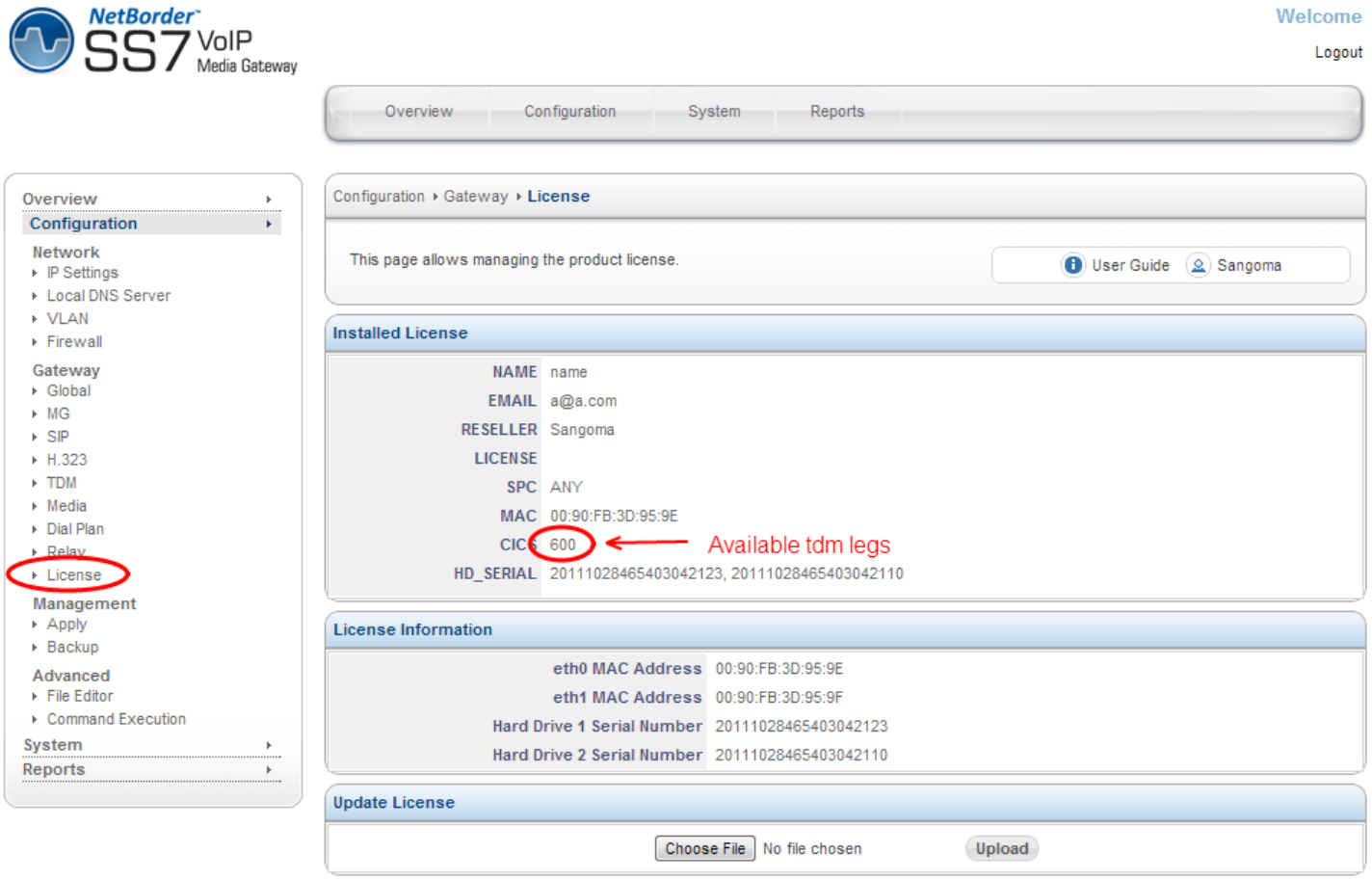
Each NSG appliance comes with pre-installed license.
In case of upgrades, of expansions please contact Sangoma Sales.

To update NSG license

- Select **License** from side/top **Configuration** Menu

- Obtain NSG License from Sangoma Support
- Upload the License into the NSG Gateway via the **Upload** Button

The License page offers the detailed license overview.



License Variables	Description
Name	Customer Name
Email	Customer Email
Reseller	Reseller Name
License	NA
SPC	SPC stands for: self point code It's used to bind a specific set of point codes to the license. ANY: is a special value which allows use of an SPC value.
MAC	System's MAC address. License code checks the MAC address and confirmes if MAC is correct. One can check vs License Information section.
CICS	Number of TDM channels allowed by the license. From example above CICs = 600 For RTP to TDM calls: License allows 600 calls For TDM to TDM calls: License allows 300 calls

6 Network Configuration

Network configuration section only applies to Physical Network Interfaces: eth0 and eth1. It does not apply to VLAN IP and route configuration.

Network Setup

- Physical network interfaces: eth0, eth1 are configured in the section **Configuration-> Settings-> IP Settings**.
  This section can only be used to modify/configure IP, Host, DNS information for Physical Network interfaces eth0 and eth1.

Default Route/Gateway

- To configure a system default route through the IP Settings section, the appropriate interface role type to use is "**External**". The External interfaces get associated to the default system route.

  CAUTION:
  - There can only be ONE External network interface.
  - There can only be ONE system default route.

Static Routes

- Static routes that apply to physical network interfaces eth0, eth1 should be configured in **Configuration-> Network -> IP Route** section.

  CAUTION:
  - Do not try to configure VLAN routes in this section. .
  - route configuration files are only meant to be used for eth0,eth1 interfaces.

Media Ethernet Interface: Transcoding

- NSG comes with optional, media/codec transcoding hardware. The media transcoding hardware network interface is: eth2. The media transcoding network interface comes preconfigured with a 10.x.x.x ip address.

  Configuration of the eth2 device should be performed in **Configuration->Settings->Media**.

  CAUTION:
  One should take this into account when assigning IP addresses to eth0,eth1 or VLAN interfaces. Confirm that ip address range set does not conflict with eth2 media transcoding network interface.

VLAN Config IP & Routes

- VLAN's can be configured in section **Configuration-> VLAN**
- VLAN can be configured on top of eth0 and eth1 network interface only.

- All VLAN related configuration such as IP address, VLAN ID and VLAN routes must be configured in VLAN configuration section only.

  CAUTION:
  - Do not use Static IP Route section to create a VLAN routes.
  - Static IP Route section is only for physical interfaces eth0 and eth1.

VLAN Default Route

- If a system default route needs to be configured via VLAN interface.
- Configure the system default route in **Configuration-> VLAN** section.
- Refer to the VLAN section below.

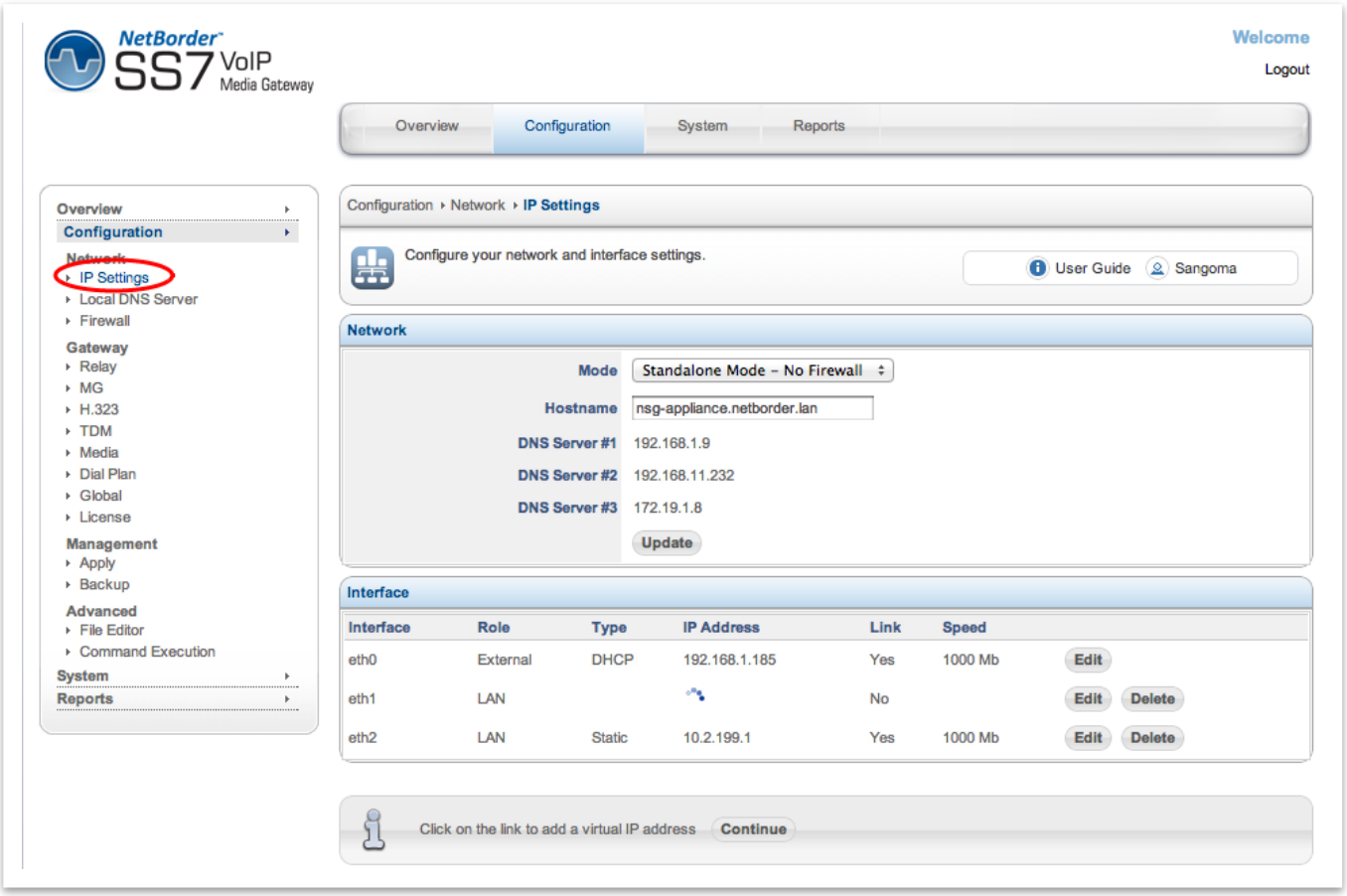
  CAUTION:

  - Make sure that all physical network interfaces in **IP Settings** section are configured for role "LAN". No physical network interface eth0, eth1 should be configured for role "External". This would result in multiple system default routes.

6.1 Physical Network Interface Configuration

By default the NSG appliance pre-configured with **192.168.168.2/24** address on Primary Port (eth0). The IP address can be changed based as follows

- Select **IP Settings** from side/top **Configuration** menu
- Specify Firewall Mode and Hostname
- Select **Edit** under eth0 and eth1 device and configure



NOTE

- **eth2** device is a Sangoma Transcoding device and should be modified.
- **eth2** device is configured in **Configuration -> Media** section of the GUI will configure this device.

6.2 Appliance Network Interfaces

- eth0
  - Primary Signaling Port
  - By default provisioned as static 192.168.168.2
  - By default allows access to ssh and management http
- eth1
  - Secondary Signaling or Management Port
  - By default provisioned as static no IP address
  - By default allows access to ssh and management http
- eth2
  - Sangoma transcoding DSP board
  - Provisioned using Media page. Do not modify in this section.

6.3 Selecting Default Route

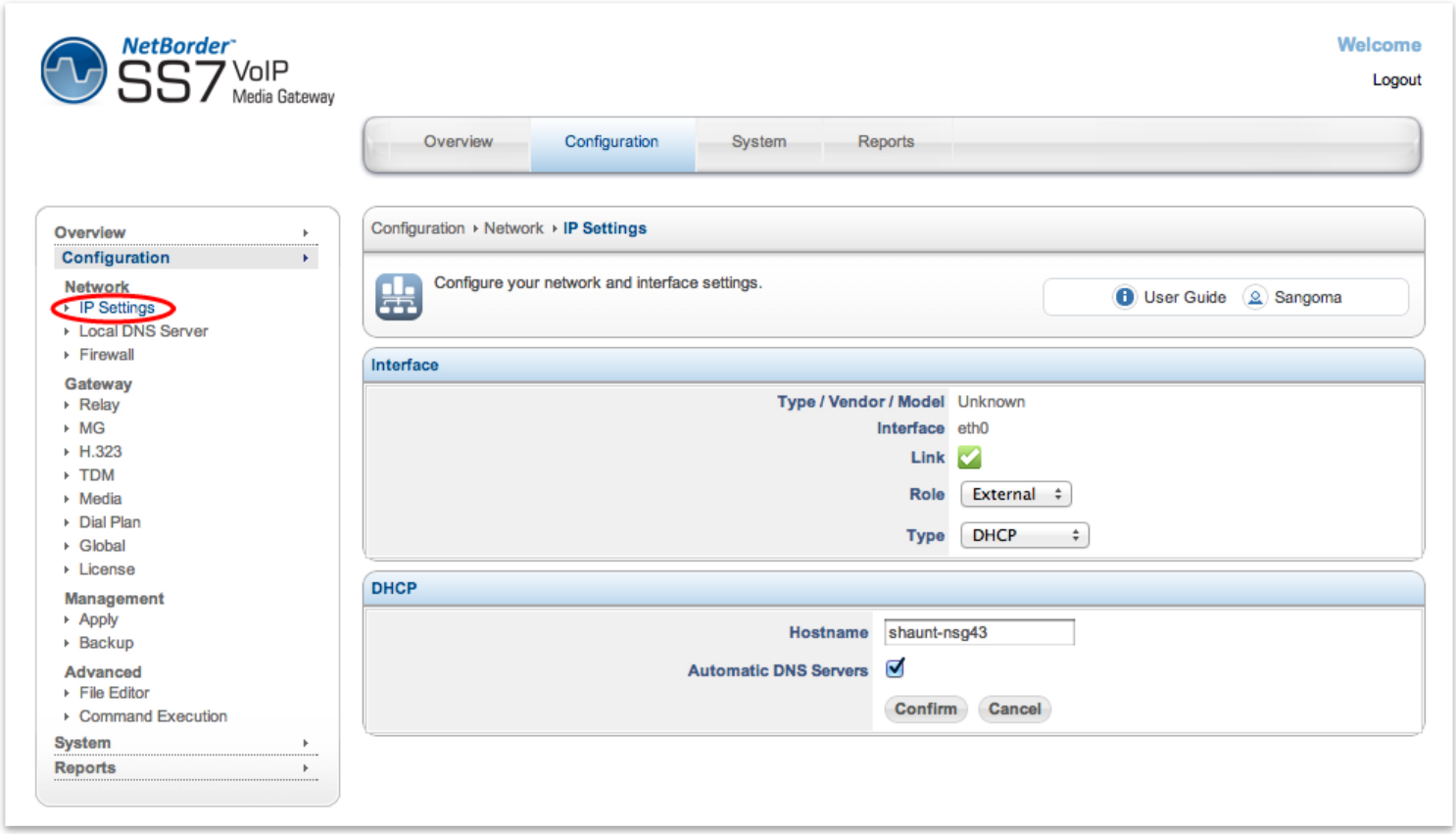
NSG appliance should have a single default route.
The default route is used to access Internet.
To configure a default route on eth0

- Set the eth0 interface mode to **External**.
- Refer to section below.

6.4 Network Section

Variable Name	Input Options	Description
Mode	Standalone – No Firewall	Firewall Disabled
	Standalone	Firewall Enabled Warning: All active service ports must be explicitly enabled
Hostname	String	A hostname is the full name of your system. If you have your own domain, you can use a hostname like nsg.example.com Alternatively, you can also make one up: gateway.lan, mail.lan. The hostname does require at least one period (.)
Name/DNS Servers	Domain Name or IP address eg. 8.8.8.8	On DHCP and DSL/PPPoE connections, the DNS servers will be configured automatically for your IP Settings. In these two types of connections there is no reason to set your DNS servers. Users with static IP addresses should use the DNS servers provided by your Internet Service Provider (ISP). If you are using Multi-WAN, please review the documentation on the topic of DNS servers.

6.5 Interface Section



6.5.1 Network Role

When configuring a network interface, the first thing you need to consider is the network role in IP Settings. Will this network card be used to connect to the Internet, for a local network, for a network with just server systems? The following network roles in IP Settings are supported in NSG and are described in further detail in the next sections:

- External - network interface with direct or indirect access to the Internet
- LAN - local area network
- Hot LAN - local area network for untrusted systems
- DMZ - de-militarized zone for a public network

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Option	Description
External	Network interface with direct or indirect access to the Internet External interface is used as the system default route. WARNING: You should have only ONE external network interface. Usually eth0 is the external interface
LAN	Connection to your local network Usually eth1 is the LAN interface
Hot LAN	Hot LAN (or "Hotspot Mode") allows you to create a separate LAN network for untrusted systems. Typically, a Hot LAN is used for: • Servers open to the Internet (web server, mail server) • Guest networks • Wireless networks A Hot LAN is able to access the Internet, but is not able to access any systems on a LAN. As an example, a Hot LAN can be configured in an office meeting room used by non-employees. Users in the meeting room could access the Internet and each other, but not the LAN used by company employees.
DMZ	In NSG, a DMZ interface is for managing a block of public Internet IP addresses. If you do not have a block of public IP addresses, then use the Hot LAN role of your IP Settings. A typical DMZ setup looks like: • WAN: An IP addresses for connecting to the Internet • LAN: A private network on 192.168.x.x • DMZ: A block of Internet IPs (e.g from 216.138.245.17 to 216.138.245.31) NSG GUI has a DMZ firewall configuration page to manage firewall policies on the DMZ network.

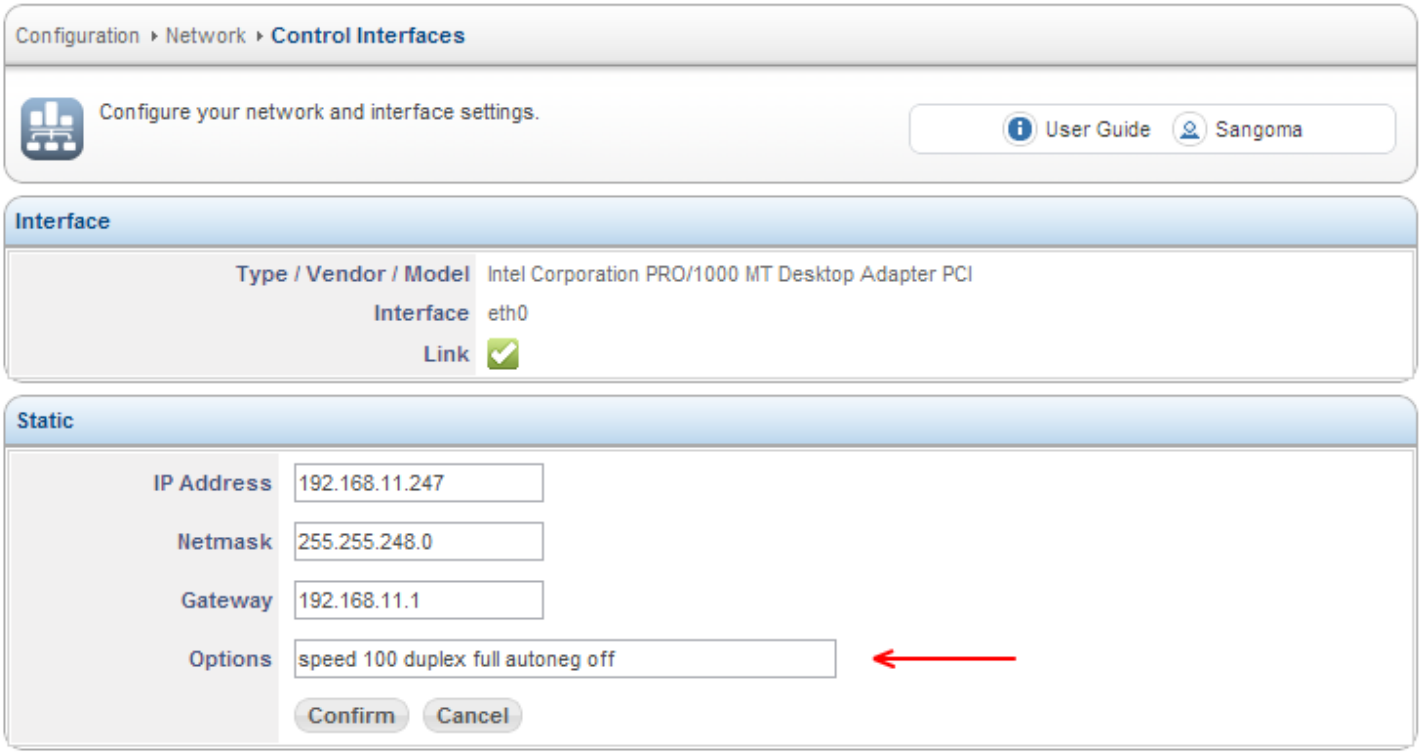
6.5.2 Types

Option	Description
DHCP	For most cable and Ethernet networks, DHCP is used to connect to the Internet. In addition, your system will have the DNS servers automatically configured by your ISP when the Automatic DNS Servers checkbox is set.
Static	If you have a static IP, you will need to set the following parameters: • IP • Netmask (e.g. 255.255.255.0) • Gateway (typically ends in 1 or 254) • Ethernet Options (able to force 100MB or 1000mb)
PPPoE DSL	For PPPoE DSL connections, you will need the username and password provided by your ISP. In addition, your system will have the DNS servers automatically configured by your ISP when the Automatic DNS Servers checkbox is set.

6.5.3 Ethernet Options

Setting custom Ethernet options such as disabling auto negotiation is done as part of the IP Settings.

- Select **IP Settings** from side/top **Configuration** Menu



Specify **Options** field in order to add special configuration to this interface.
Options are any device-specific options supported by ethtool.
In above example the Ethernet device is set for 100Mb with negotiation disabled.

Options	[speed 10|100|1000|2500|10000] [duplex half|full] [port tp|aui|bnc|mii|fibre] [autoneg on|off] [advertise %%x] [phyad %%d] [xcvr internal|external] [wol p|u|m|b|a|g|s|d...] [sopass %%x:%%x:%%x:%%x:%%x:%%x] [msglvl %%d]

6.6 Virtual IP's

NSG supports virtual IPs. To add a virtual IP address, click on the link to configure a virtual IP address and add specify the IP Address and Netmask. You will also need to create advanced firewall rules if the virtual IP is on the Internet.

6.7 IP Troubleshooting

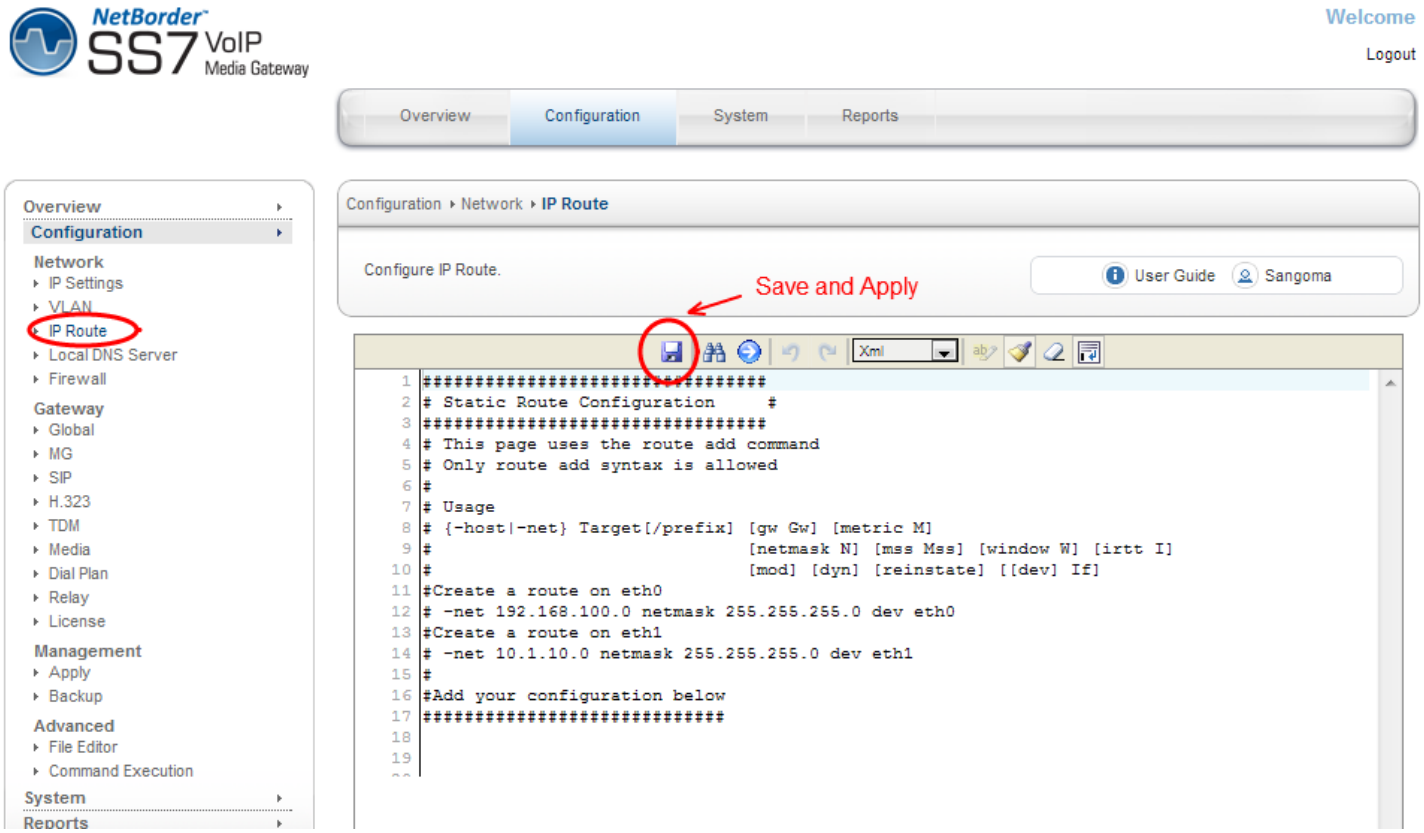
In most installs, the network cards and IP settings will work straight out of the box. However, getting the network up the first time can be an exercise in frustration in some circumstances. Issues include;

- Network card compatibility
- Invalid networks settings (username, password, default gateway)
- Cable/DSL modems that cache network card hardware information

6.8 Static Routes

In some cases a static route must be defined for a specific network interface: eth0 or eth1.
The static route support is done via File Editor

- Select **IP Route** from side/top **Configuration** Menu
- Add a custom route command
- Save and Apply



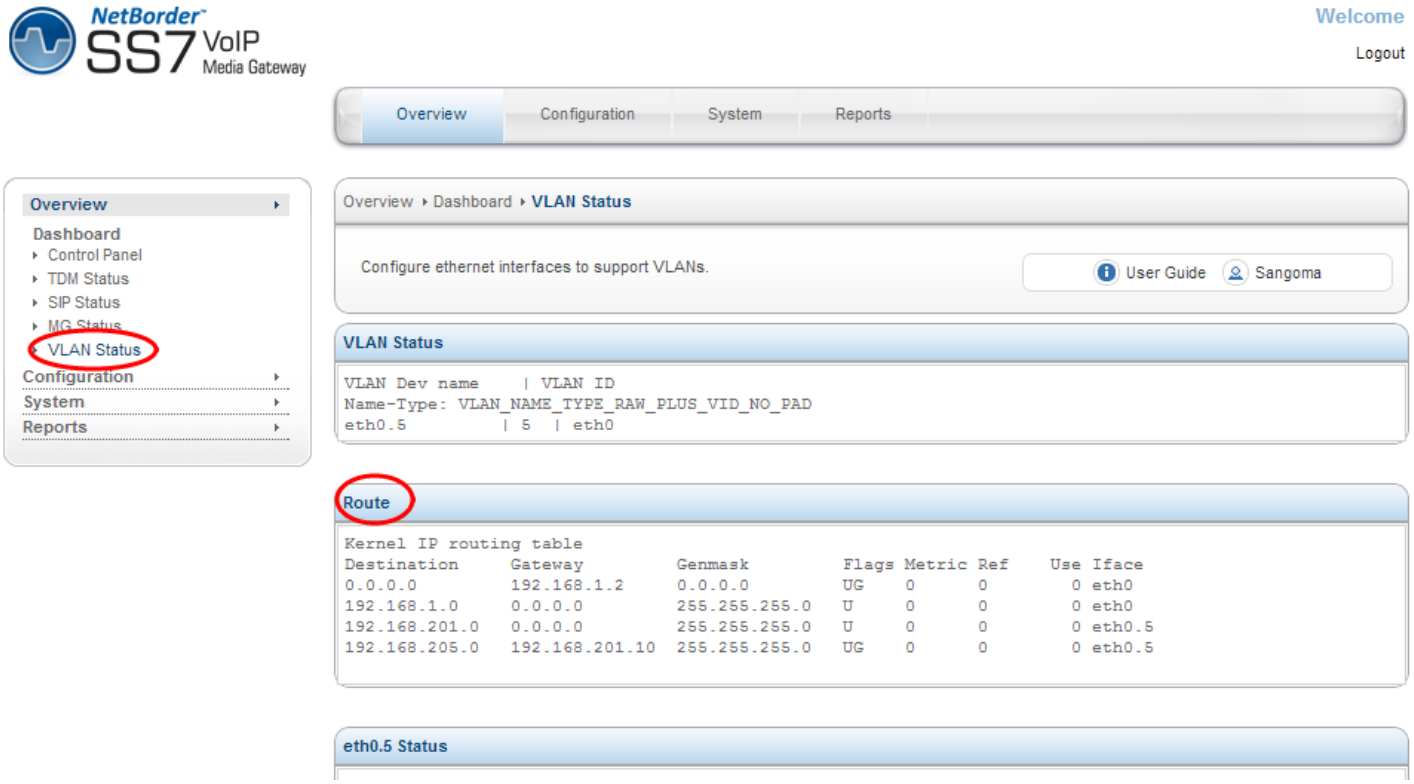
NOTE

- The IP Route section only allows route add command syntax

Route File Name	*Description*
Usage	Use to create static routes for Primary Signaling Ethernet Port:eth0 Usage: {-host|-net} Target[/prefix] [gw Gw] [metric M] [netmask N] [mss Mss] [window W] [irtt I] [mod] [dyn] [reinstate] [[dev] If] Example: #Route a class C network 10.133.20.0 via gw IP -net 10.133.20.0 netmask 255.255.255.0 gw 10.132.30.1 #Route a class B network 10.133.0.0 via gw IP -net 10.133.0.0 netmask 255.255.0.0 gw 10.132.30.1 #Route a class B network 10.133.0.0 via device eth0 -net 10.133.0.0 netmask 255.255.0.0 dev eth0

6.8.1 Routing Table Status

- Select **VLAN Status** from side/top **Overview** Menu
- Second table shows full system routing table.



6.9 VLAN

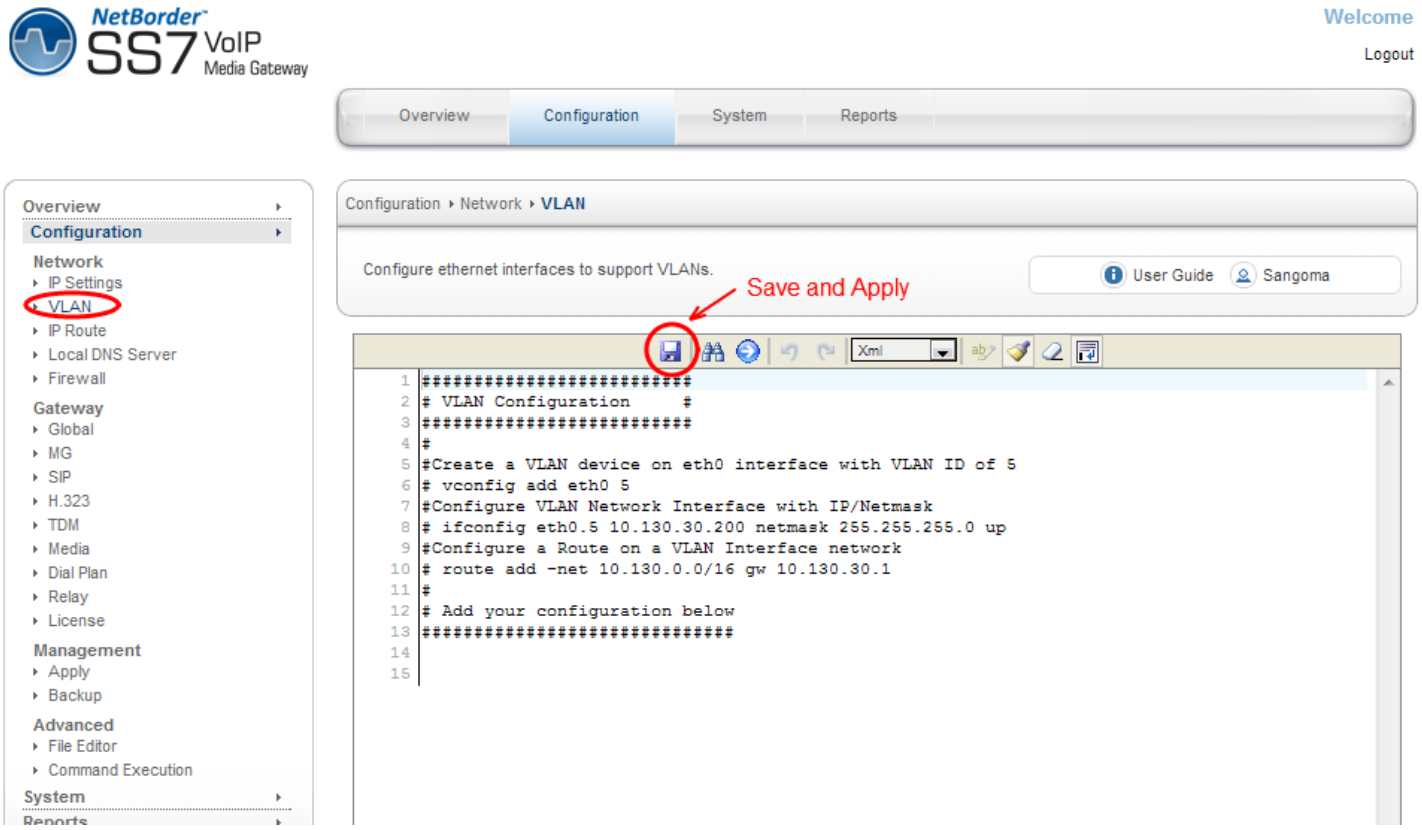
Virtual local area network, virtual LAN or VLAN is a concept of partitioning a physical network, so that distinct broadcast domains are created. NSG mark's packets through tagging, so that a single interconnect (trunk) may be used to transport data for various VLANs.

A VLAN has the same attributes as a physical local area network (LAN), but it allows for end stations to be grouped together more easily even if not on the same network switch. VLAN membership can be configured through software instead of physically relocating devices or connections. Most enterprise-level networks today use the concept of virtual LANs(VLAN). Without VLANs, a switch considers all interfaces on the switch to be in the same broadcast domain.

6.9.1 VLAN Configuration

Currently NSG only supports VLAN configuration via GUI

- Select **VLAN** from side/top **Configuration** Menu
- Copy in the VLAN configuration script below into the file editor
- Save
    - On save the VLAN configuration will be applied
    - Proceed to VLAN Status confirm VLAN configuration.



NOTE

- The VLAN network interfaces are created over physical network interface. Make sure that the physical network interface eth0 or eth1 are configured in IP Settings, before attempting to configure VLAN on top of them eth0 or eth1.
- The Save/Apply post processing will display VLAN configuration status.

Example of sample script that could be copied into the VLAN config startup script:

```
#Create a VLAN device on eth0 interface with VLAN ID of 5
vconfig add eth0 5

#configure VLAN device with IP/Net mask
ifconfig eth0.5 192.168.1.100 netmask 255.255.255.0 broadcast 192.168.1.255 up

#configure a default route within a vlan
route add –net 192.168.1.0/24 gw 192.168.1.1

#if system default route needs to go through VLAN
#Note that there can only be ONE system default route.
#Make sure all interfaces in IP Settings are set to LAN (not External)
route add default gw 192.168.1.1 eth0.5
```

In the example above, a single VLAN was created

- on top of the Primary Signaling Ethernet Port:eth0 with
- VLAN ID=5 and
- IP =192.168.1.100/24.

.

6.9.2 VLAN Routes

An optional route can be created to point to a gateway within a VLAN network.

NOTE
Only routes related to VLAN interfaces are allowed in the VLAN configuration section.

CAUTION
If a system default route needs to go through a VLAN

- Confirm that IP Settings interfaces are all set to **LAN** role.
- As there can be only ONE system default route.

6.9.3 Additional VLAN

If more VLAN's are needed, proceed to repeat the above steps for all VLANs.

When **Save** button is pressed

- The VLAN configuration will be applied
- The script above will be executed line by line.
- Status window will pop up with VLAN config status. If one of the lines fails, the pop up will report it.
- Proceed to **Overview -> VLAN status** below to confirm VLAN and Route configuration

6.9.4 vconfig help

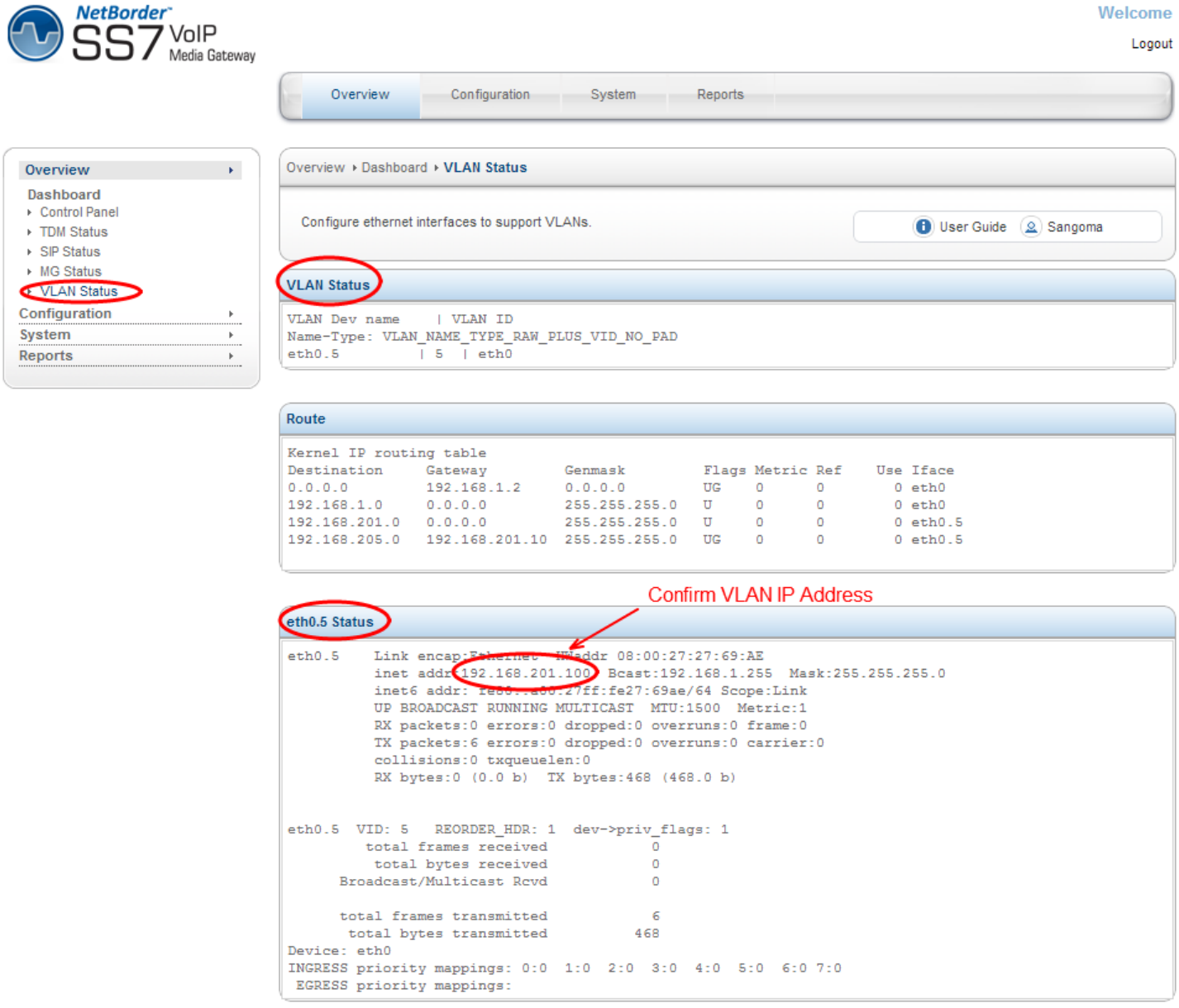
```
# vconfig
Expecting argc to be 3-5, inclusive.  Was: 1

Usage: add            [interface-name] [vlan_id]
       rem            [vlan-name]
       set_flag       [interface-name] [flag-num]       [0 | 1]
       set_egress_map  [vlan-name]      [skb_priority]   [vlan_qos]
       set_ingress_map [vlan-name]      [skb_priority]   [vlan_qos]
       set_name_type   [name-type]

* The [interface-name] is the name of the ethernet card that hosts
  the VLAN you are talking about.
* The vlan_id is the identifier (0-4095) of the VLAN you are operating on.
* skb_priority is the priority in the socket buffer (sk_buff).
* vlan_qos is the 3 bit priority in the VLAN header
* name-type:  VLAN_PLUS_VID (vlan0005), VLAN_PLUS_VID_NO_PAD (vlan5),
              DEV_PLUS_VID (eth0.0005), DEV_PLUS_VID_NO_PAD (eth0.5)
* bind-type:  PER_DEVICE  # Allows vlan 5 on eth0 and eth1 to be unique.
              PER_KERNEL  # Forces vlan 5 to be unique across all devices.
* FLAGS:  1 REORDER_HDR  When this is set, the VLAN device will move the
           ethernet header around to make it look exactly like a real
           ethernet device.  This may help programs such as DHCPd which
           read the raw ethernet packet and make assumptions about the
           location of bytes.  If you don't need it, don't turn it on, because
           there will be at least a small performance degradation.  Default
           is OFF.
```

6.9.5 VLAN Status

- Select **VLAN** Status from side/top **Overview** Menu
- This page shows
  - All configured VLANs
  - System Routing table
  - Individual VLAN configuration
  - Individual VLAN IP information



NOTE

- Confirm that VLAN Interface contains the correct IP address.
- If the IP address is not set, the VLAN configuration has not been set properly.

6.10 Date & Time Service Config

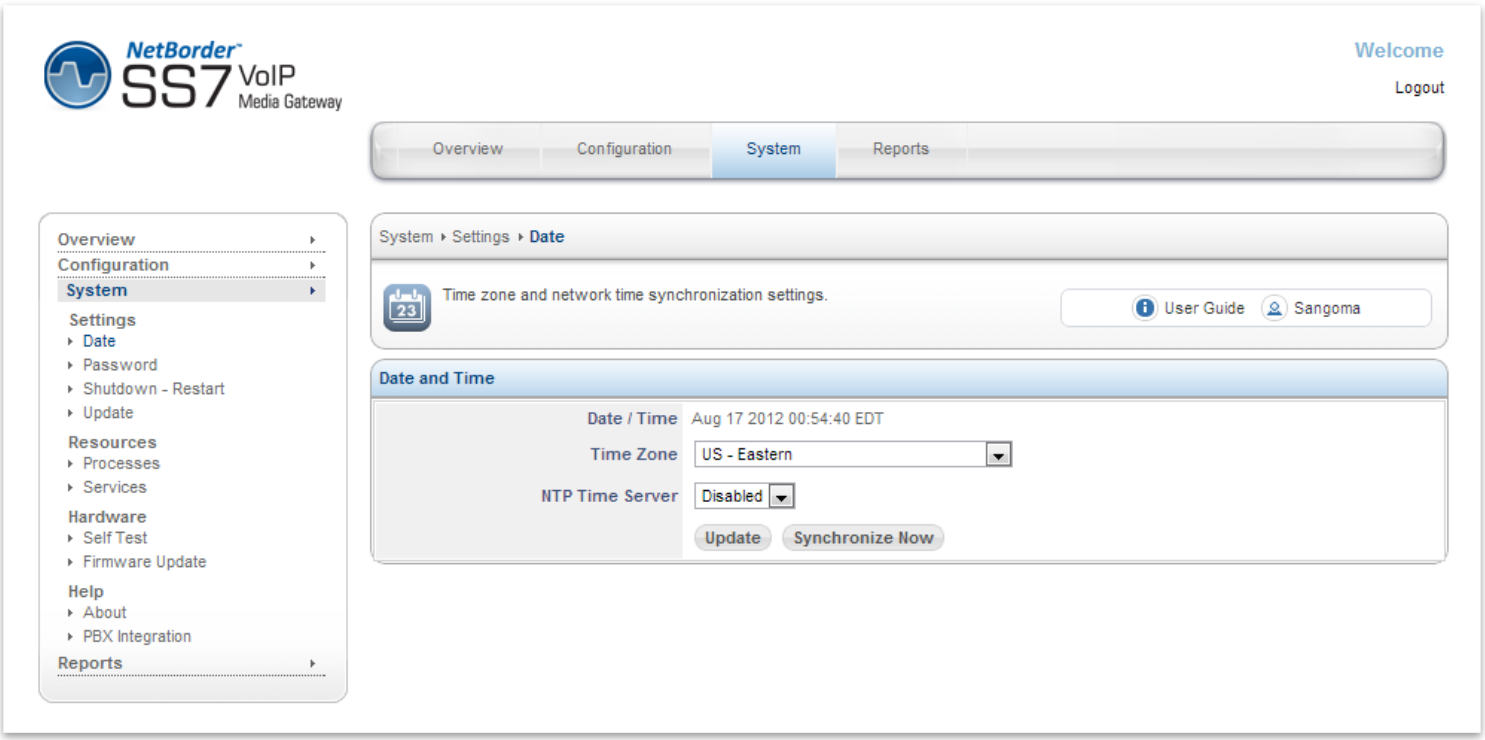
The Date/Time configuration tool allows you to:

- Select your time zone
- Synchronize your clock with network time servers
- Enable/disable a local time server for your network

Note that you need to configure your IP address and default route in order to be able to use a default time server that is located on the internet.

To configure

- Select **Date** from side/top **System** menu
- Refer below to all available options.



Option	Description
Date/Time	The system date, time and time zone information is displayed for informational purposes. Please make sure it is accurate since it is not unusual to have computer clocks improperly set on a new installation.
Time Zone	It is important to have the correct time zone configured on your system. Some software (notably, mail server software) depends on this information for proper time handling.
NTP Time Server	An NTP Time Server is built into NSG.
Time Synchronization	Hitting the Synchronize Now button will synchronize the system's clock with network time servers.

7 Initial Gateway Configuration

NSG by default contains following VoIP/TDM Sections

- Global Gateway Config
  - Configured in Global gateway section.
  - Used to configure SIP, H323, RTP, RADIUS options.

- SIP/RTP
  - Configured in Global Gateway section
  - SIP profile is always started

- MG
  - Configured in MG gateway section
  - MG Termination ID's are mapped to TDM channels in TDM gateway section.
  - For full MG configuration one must configure MG and TDM sections.

- H323
  - Single H323 profile, configured in H323 gateway section
  - H323 profile is started only if H323 gateway section is saved.

- SS7
  - Configured in TDM gateway section
  - ISUP Termination
  - M2UA Signaling Gateway

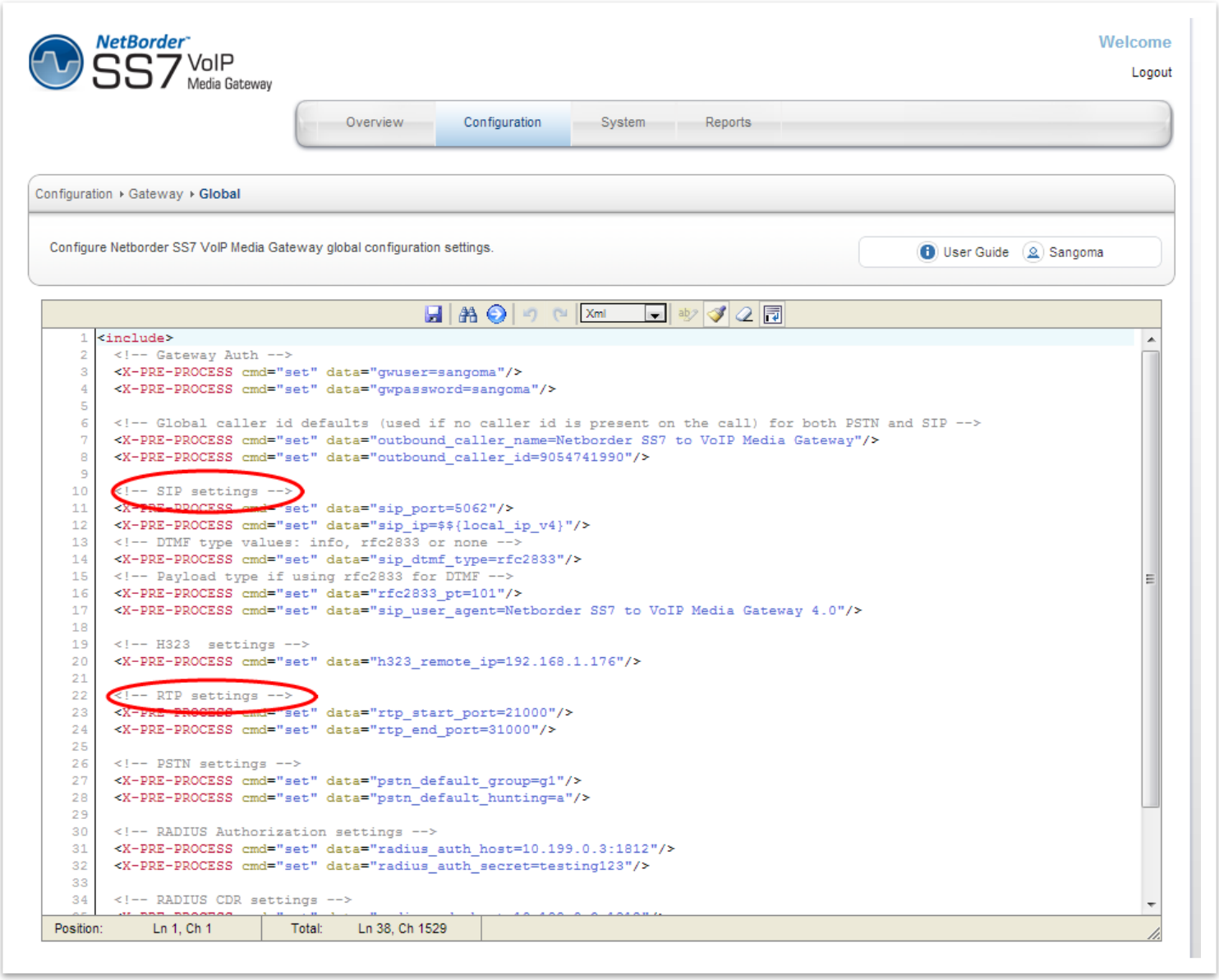
- Media/Transcoding
  - Configured in Media gateway section
  - Enable and select hw codec support
  - Note: HW transcoding is an optional feature.

- Dialplan
  - Used for SIP to TDM and H323 to TDM mode
  - **Note**: Dialplan is not used in MG/Megaco/H.248 mode.

- Apply
  - All configuration files are saved to disk at this step.
  - Above configuration sections only save information in local database.
  - NSG Gateway can be started in **Control Panel** after this step
  - **TDM Status** can be used to monitor Gateway Status.

7.1 Global Gateway Configuration

- Select **Global** from side/top **Configuration** Menu
- Change a SIP global variable and Click on Save (Disk Icon)
- Proceed to Control Panel and Restart the VoIP Gateway.



Field Name	Possible Values	Default Value	Description
gwuser	Any string	Sangoma	NSG SIP incoming registration authentication user name.
gwpassword	Any string	Sangoma	NSG SIP incoming registration authentication password
outbound_caller_name	Any string	Netborder SS7 to VoIP Media Gateway	Global caller id name defaults (used if no caller id name is present on the call) for both PSTN and SIP
outbound_caller_id	Any digits	9054741990	Global caller id defaults (used if no caller id is present on the call) for both PSTN and SIP
sip_port	Any port number	5062	SIP service local port number.
sip_ip	Any ip address	System IP	SIP service, local IP address. By default a local system eth0 address is taken as default ip address.
sip_dtmf_type	rfc2833 info none	rfc2833	rfc2833 - DTMF passed via RTP oob message info - DTMF passed via SIP INFO message none - DTMF passed via inband media
rfc2833_pt	Any number	101	rfc2833 rtp payload type override. Ability to set the RTP payload type for rfc2833. Use d edge cases where remote equipment is not per spec.
sip_user_agent	Any string	Netborder SS7 to VoIP Media Gateway 4.0	SIP INVITE user agent name string.
rtp_start_port	Any port	21000	RTP port starting range value. NSG will pick RTP ports for each call within this range.
rtp_end_port	Any port	31000	RTP port stop range value. NSG will pick RTP ports for each call within this range
pstn_default_group	g1,g2,g3,g4 ….	g1	Default pstn dial group number, in case the group is not specified in the dial string.
radius_auth_host	Any ip address:port	10.199.0.3:1812	Location of the Radius server, that will be used to authenticate incoming calls.
radius_auth_secret	Any string	testing123	Password of the remote Radius server.
radius_cdr_host	Any ip address:port	10.199.0.3:1812	Location of the Radius server, that will be used to keep track of billing via CDRs.
radius_auth_secret	Any string	testing123	Password of the remote Radius server.

8 Megaco/H.248 Media Gateway Configuration

8.1 Overview

H.248 or Megaco or Gateway Control Protocol is a recommendation from ITU which defines protocols that are used between elements of a physically decomposed multimedia gateway. It is an implementation of the Media Gateway Control Protocol Architecture (RFC 2805). H.248 is also called Megaco or in IETF domain. It is now known as Gateway Control Protocol.

H.248/Megaco is standard protocol for controlling the elements of a physically decomposed multimedia gateway, which enables separation of call control from media conversion. H.248/Megaco is a master/slave protocol used to separate the call control logic from the media processing logic in a gateway.

The H.248/Megaco model describes a connection model that contains the logical entities, or objects, within the Media Gateways (MGs) that can be controlled by the Media Gateway Controller. The main entities are Contexts and Terminations.

8.1.1 Terminations

These source or sink one or more media streams or control streams. Terminations may be physical or ephemeral.

Physical Terminations represent physical entities that have a semi-permanent existence. For example, a Termination representing ports on the gateway, such as TDM channel or DS0 might exist for as long as it is provisioned in the gateway. Ephemeral Terminations represent Connections or data flows, such as RTP streams, or MP3 streams, and usually exist only for the duration of their use in a particular Context.

Terminations have properties, such as the maximum size of a jitter buffer, which can be inspected and modified by the MGC. A termination is given a name, or Termination ID, by the MG.

8.1.2 Contexts

These are star connections created by associating multiple terminations. A Context is a logical entity on an MG that is an association between a collection of Terminations. A NULL context contains all non-associated terminations. A Context is a logical entity on an MG that is an association between a collection of Terminations. A ContextID identifies a Context.

The normal, "active" context might have a physical termination (say, one DS0 in a DS3) and one ephemeral one (the RTP stream connecting the gateway to the network). Contexts are created and released by the MG under command of the MGC. A context is created by adding the first termination, and it is released by removing (subtracting) the last termination.

A termination may have more than one stream, and therefore a context may be a multistream context. Audio, video, and data streams may exist in a context among several terminations.

8.2 Commands

The commands defined by megaco are very simple, since they can be heavily extended using packages.

8.2.1 Sent from controller to gateway

Add
- Used to add a termination to a context

Modify
- Used to modify an existing termination

Substract:
- Used to remove a termination from a context

Move:
- used to move a termination to another context (call-waiting is achieved by moving it to the NULL context, which keeps it opened).

AuditValue
- Returns the current values of properties, signals and statistics

AuditCapabilities:
- Returns metadata on the current termination (the possible values for all elements)

8.2.2 Sent from gateway to controller

Notify
- Carries an event defined in one of the packages [P1]

ServiceChange:
- Notifies the controller that the gateway is going out of service / back in service. [P1]

A MEGACO-configured NSG starts by sending a Service Change command to its MGC. When an MGC accepts the NSG registration, the session can start. Subsequently, the NSG responds to MGC commands. Event notifications are sent only if the MGC requests them specifically.

8.3 Packages

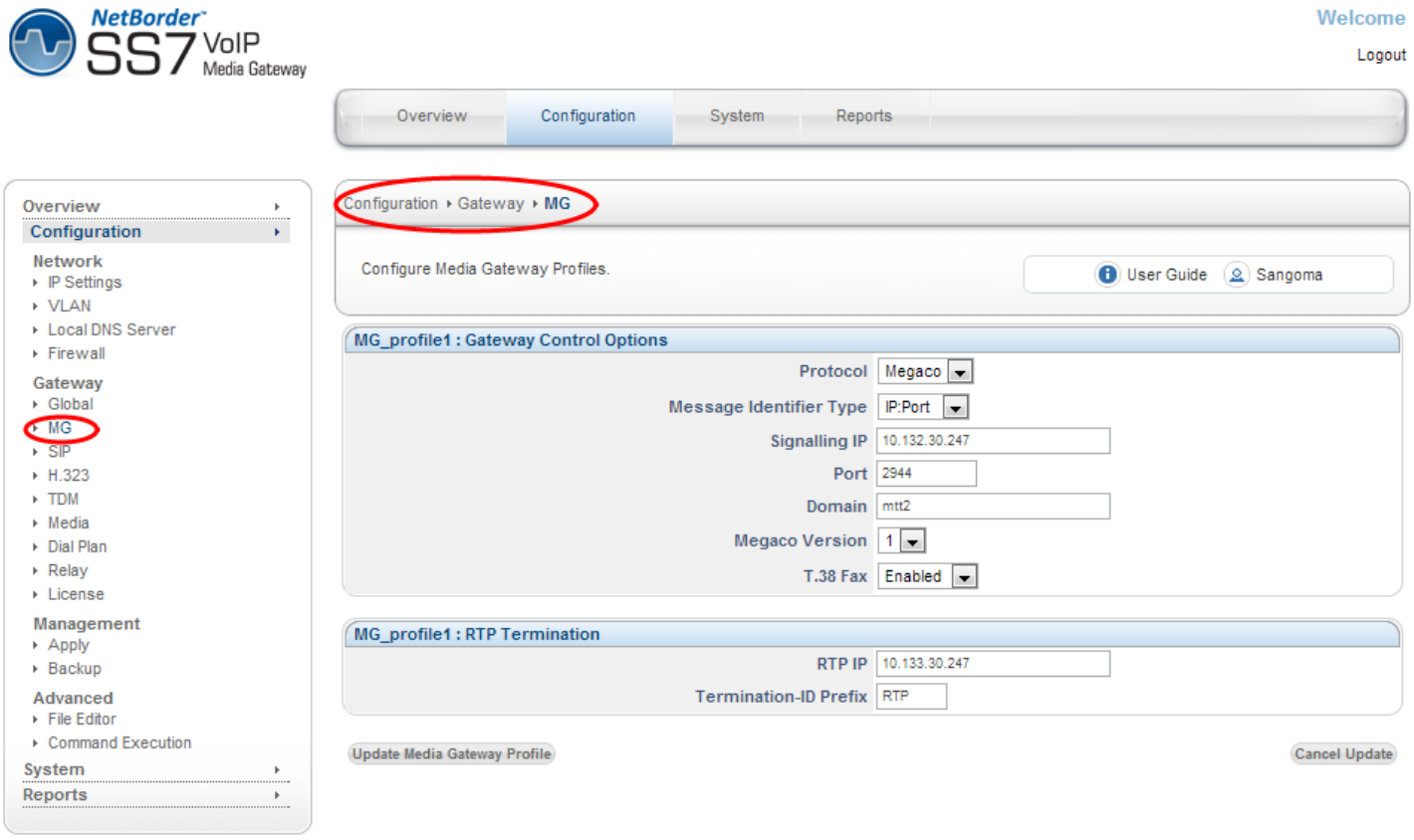
Additional features are provided in packages, which define additional properties, events and signals that are included in the descriptors used in the protocol's commands. Packages follow an inheritance model similar to object oriented programming, with some of those defined as "to be extended only" providing only an indicative structure for proprietary implementation.

Some properties are read-only and others are read-write, for more information refer to H.248.1 Appendix E.

8.4 Create MG Profile

Media gateway profile will contains all the required configuration parameters to bring up the Media gateway stack.

- Select **MG** from the side/top Configuration menu
- Select **Add New Profile**
  - Use default profile name, or specify one
- Select **Create Media Gateway Profile**
- Configure the MG Profile based on information received from our provider.
- Select **Update Media Gateway** Profile to save



Followings are the fields, that need to be configured.

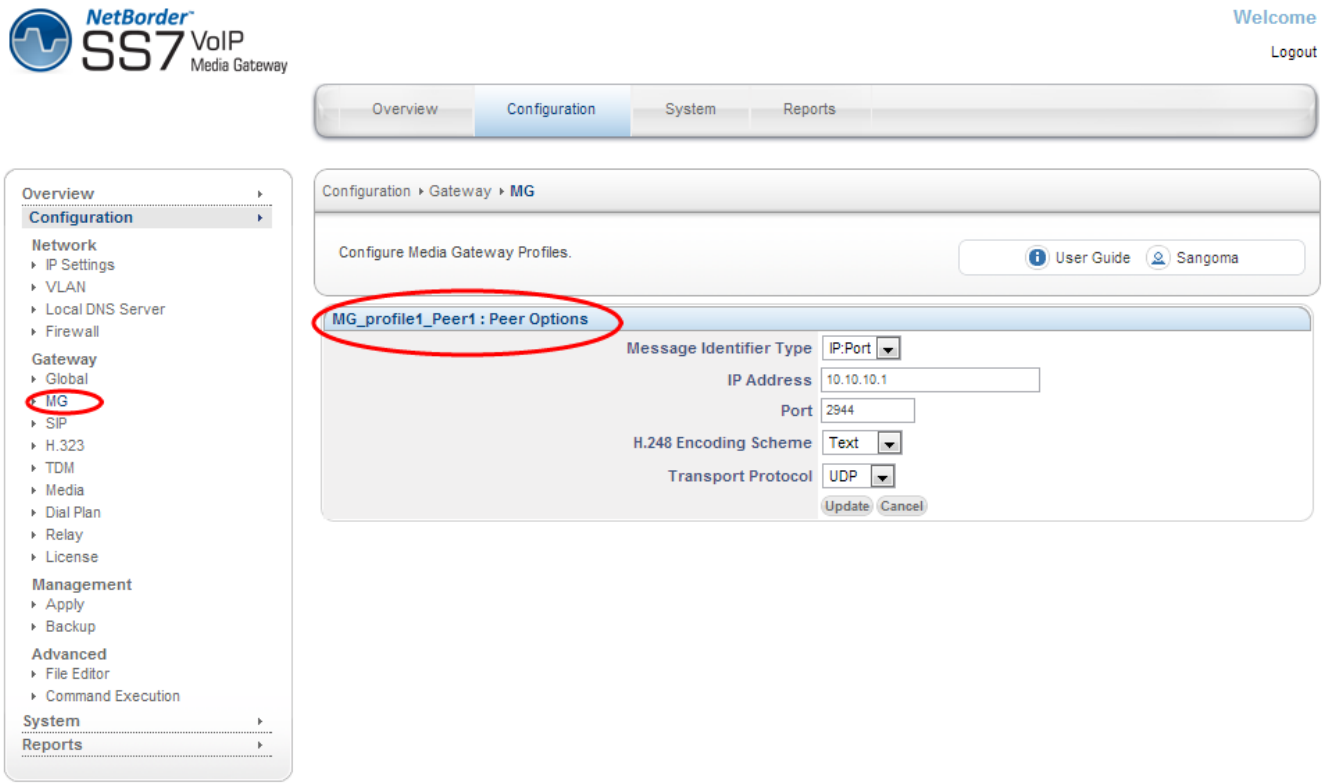
Field Name	Possible values	Default Values	Description
Protocol	MEGACO MGCP	MEGACO	Type of protocol Media Gateway is going to use. NOTE: Currently Media Gateway supports only MEGACO
Message Type Identifier	IP-PORT IP DOMAIN	IP-PORT	Media gateway message identifier (MID) type field will be used to build the message identifier field which Media Gateway will use in all the originating messages. For example: If MID type is IP-PORT then Message identifier format will be "[IP-Address]:Port" If MID type is DOMAIN then message identifier format will "<Domain>". Refer to Domain section below. If MID type is IP then message identifier format will "[IP-Address]" Note: IP-Address, Port and Domain values will be as defined above.
Signaling IP	any ipv4 addr	NA	Media Gateway, Megaco, source IP address.
Port	1 - 65000	NA	Media Gateway source Port.
Domain	(a string value)	NA	Media Gateway domain name. Used as MID Type, when MID Type is set to DOMAIN. Ignored if MID Type is not Domain. Default to system domain name.
Megaco Version	1 2 3	1	Megaco protocol version which Media Gateway will use while communicating with Media Gateway Controller
T.38 Fax	Enable/Disable	Enable	If enable MG will configure to detect and send CNG/CED Fax notify events to MGC. This will prompt MGC to modify the RTP stream to T.38. If disable MG will not notify MGC about CNG/CED, thus disabling T.38 faxing. Fax will go

			through as G711 stream.
RTP IP	any ipv4 addr	Same as Signaling IP.	Megaco RTP source IP address. By default it should be set to SIgnaling IP address, this way both signaling and media originate from single IP address. In VLAN scenarios it's possible to use separate IP addresses for Signaling and RTP.
Termination-ID Prefix	any number starting from 1	NA	RTP termination id prefix which Media Gateway will use while allocating RTP terminations. This variable is used as a name of RTP termination. Eg: RTP/1, RTP/2 …

8.5 Create MG Peer Profile

Each Media gateway profile will associate with one or multiple peers.
NOTE: As of now NSG supports only “one peer per MG profile”.

- Select **Add Peer** in MG Section
- Fill in the peer information
- Select **Update** to Save



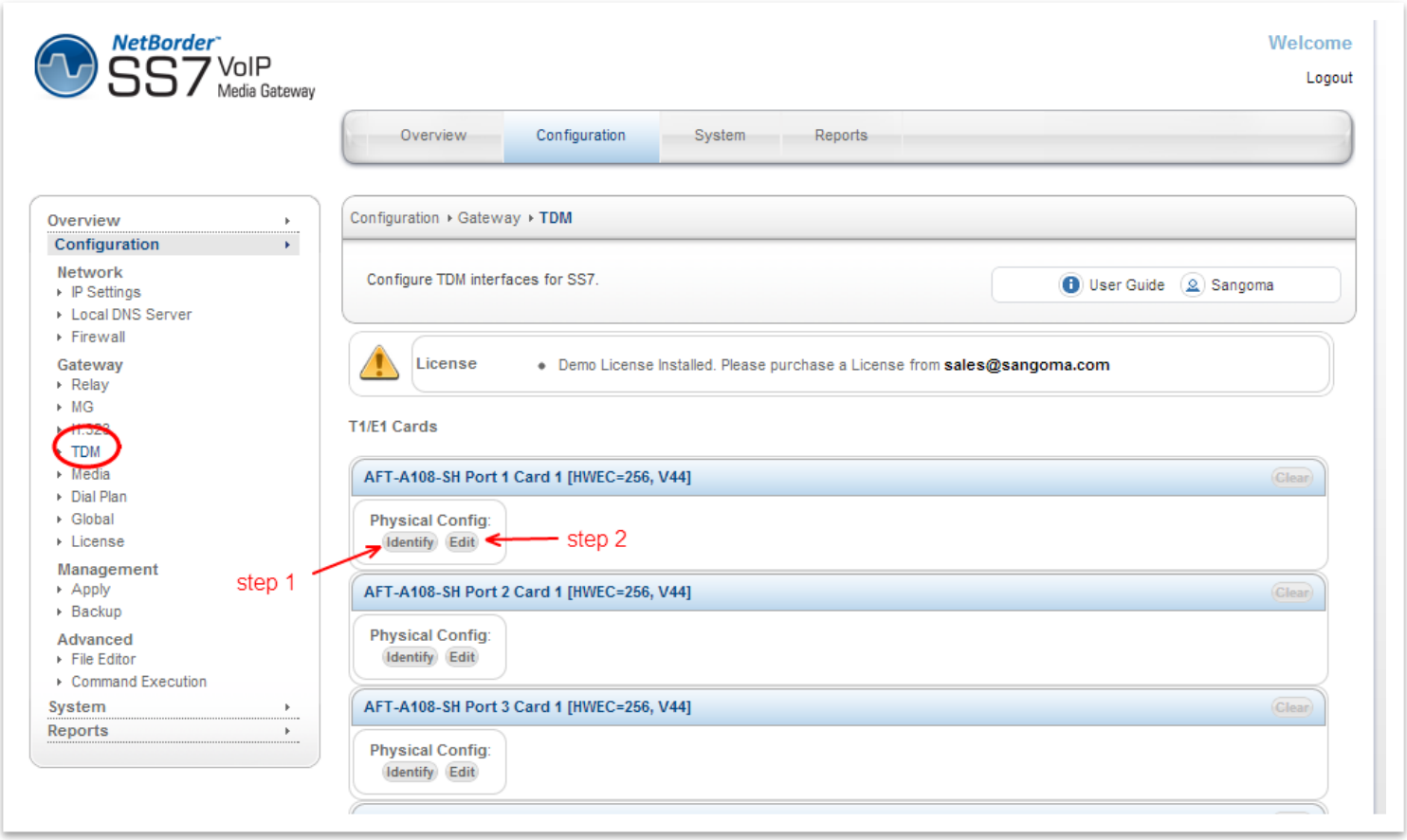
Followings are the fields which need to be configured.

Field Name	Possible values	Default Values	Description
Message Identifier Type	IP-PORT IP	IP-PORT	Media gateway Controller message identifier (MID) type field will be used by Media Gateway to identify the peer. Message identifier value will be built based on MID type field. For example: If MID type is IP-PORT then Message identifier format will be "[IP-Address]:Port" If MID type is IP then message identifier format will "[IP-Address]" Note: IP-Address and Port values will be as defined above.
IP Address	NA	NA	Media Gateway Controller IP address.
Port	NA	2944	Media Gateway Controller Port number Default: 2944
H.248 Encoding Scheme	TEXT BINARY	TEXT	Encoding scheme of MEGACO protocol which will be used by Media Gateway while encoding/decoding the H.248 messages.
Transport Protocol	UDP TCP SCTP	UDP	Media Gateway will use the transport type field to decide which transport to use for transmitting/receiving MEGACO messages. NOTE: currently we are supporting only UDP/TCP.

- Once the **Media Peer** is configured the Megaco configuration section is complete.
- Proceed to **TDM Termination for Media Gateway**

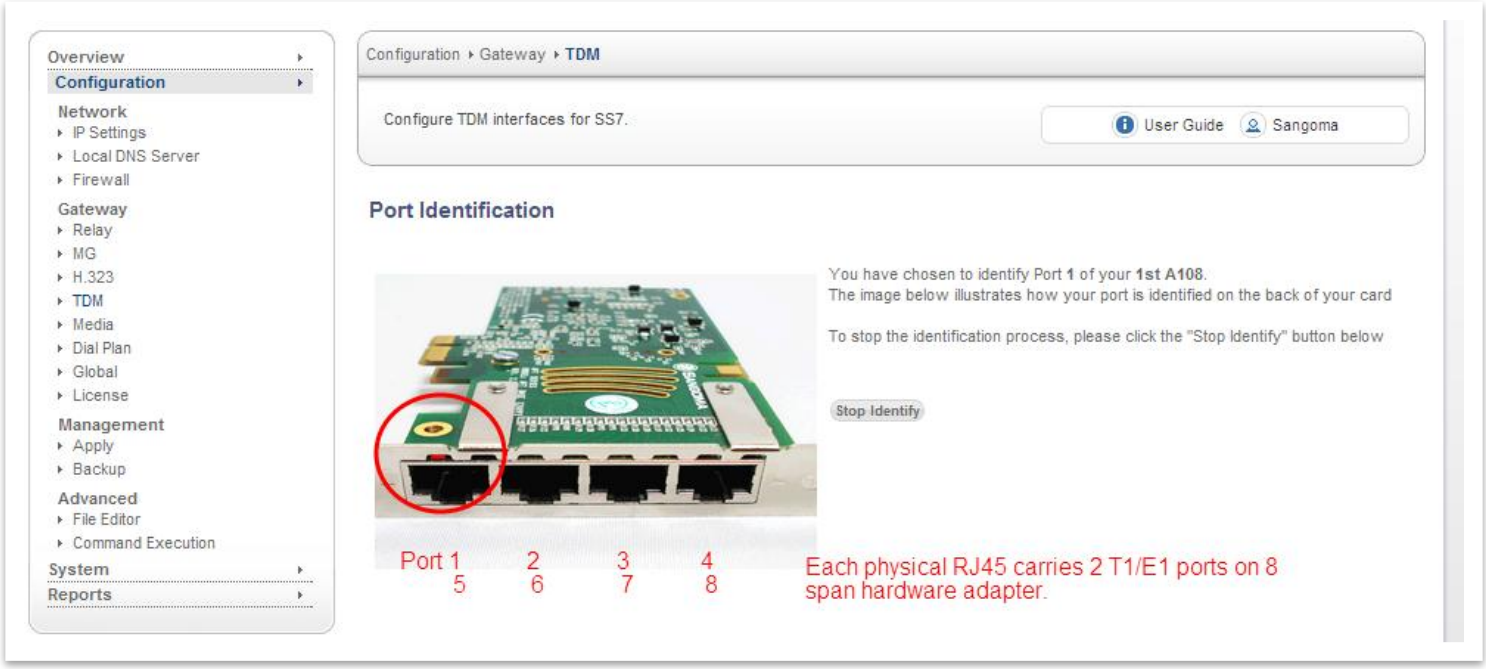
8.6 TDM Termination for Media Gateway

- Select **TDM** from side/top **Configuration** menu
- The TDM section will display all installed TDM Spans/Ports.



8.6.1 Identify

- In order to determine which physical T1/E1 port is: Port 1 Card 1
- Select **Identify** button for Port 1 Card 1
- The LED light will start flashing on a rear RJ45 T1/E1 port: rear panel.
- Look at the rear panel of the appliance and plug in RJ45 cable to the blinking RJ45 T1/E1 port.
- Once the Port 1 Card 1 is identified, the subsequent ports for that board are labeled.
- Or alternatively keep using the Identify feature for each port.

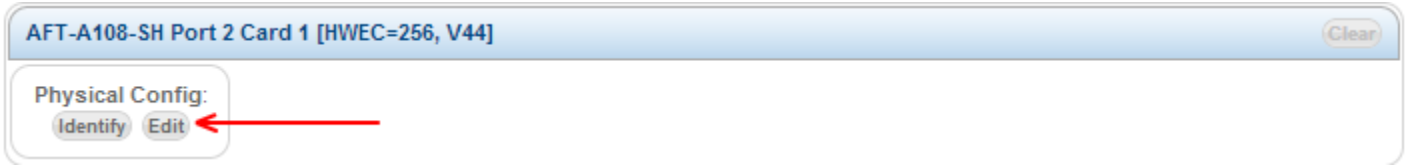


NOTE

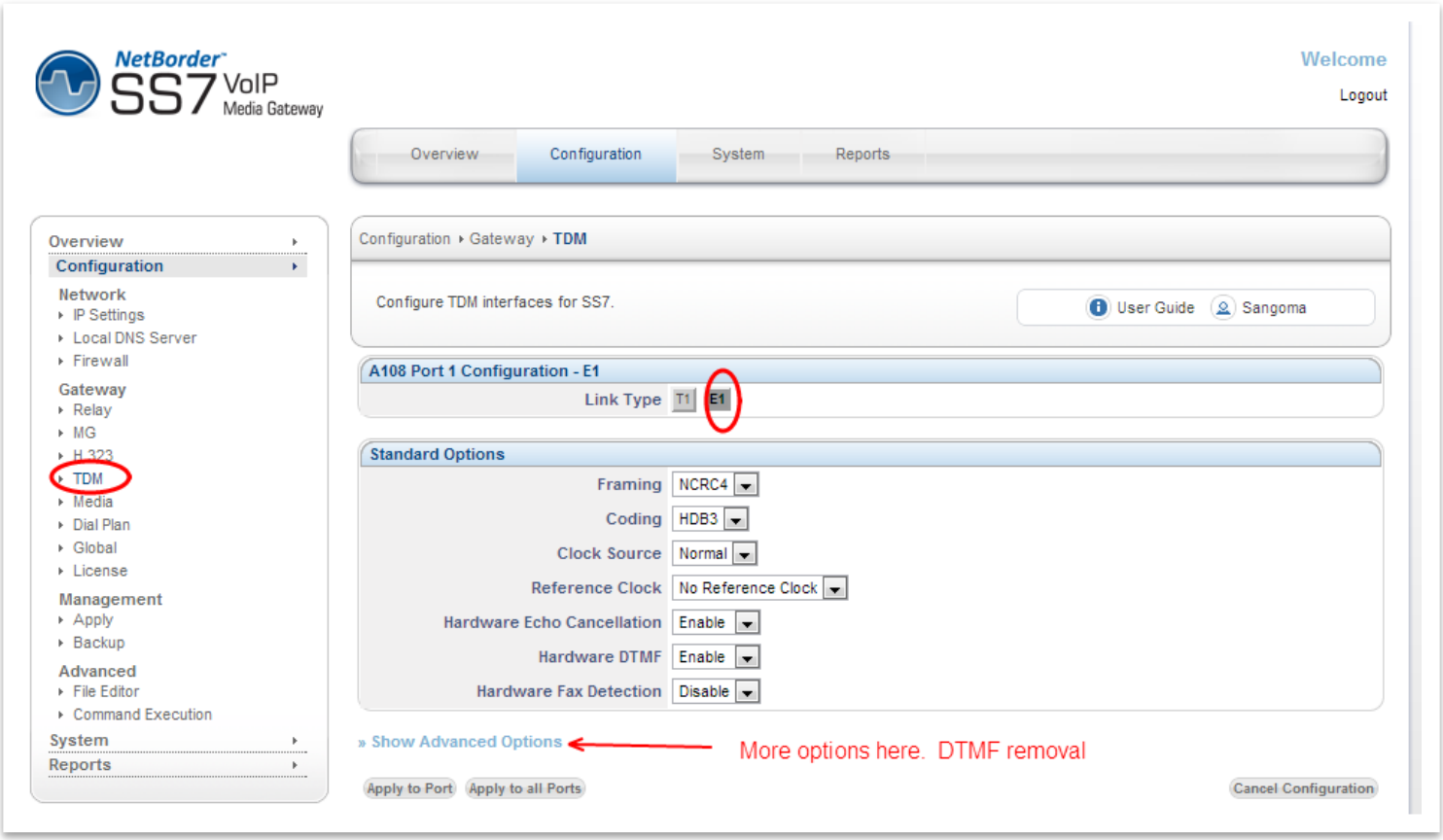
- Identify picture of the device is always set to A108D – 8 T1/E1 card. The LED will always bling port 1. The image is not meant to reflect the real hardware image, nor real port location. User should always view the rear panel for the flashing LED.
- All Sangoma TDM T1/E1 cards Port 1 is closest to the PCI slot.

8.6.2 Edit T1/E1 Config

- Once the port has been identified and plugged into the T1/E1 network.
- Select **Edit** button for Port 1 Card 1 to configure the physical T1/E1 parameters.
- Select the port configuration type: T1 or E1
  - T1: North American Market and Japan
  - E1: Europe and the world
- Fill in Physical Configuration T1 or E1 parameters
  - Fill in the T1/E1 parameters based on the provider provision document.

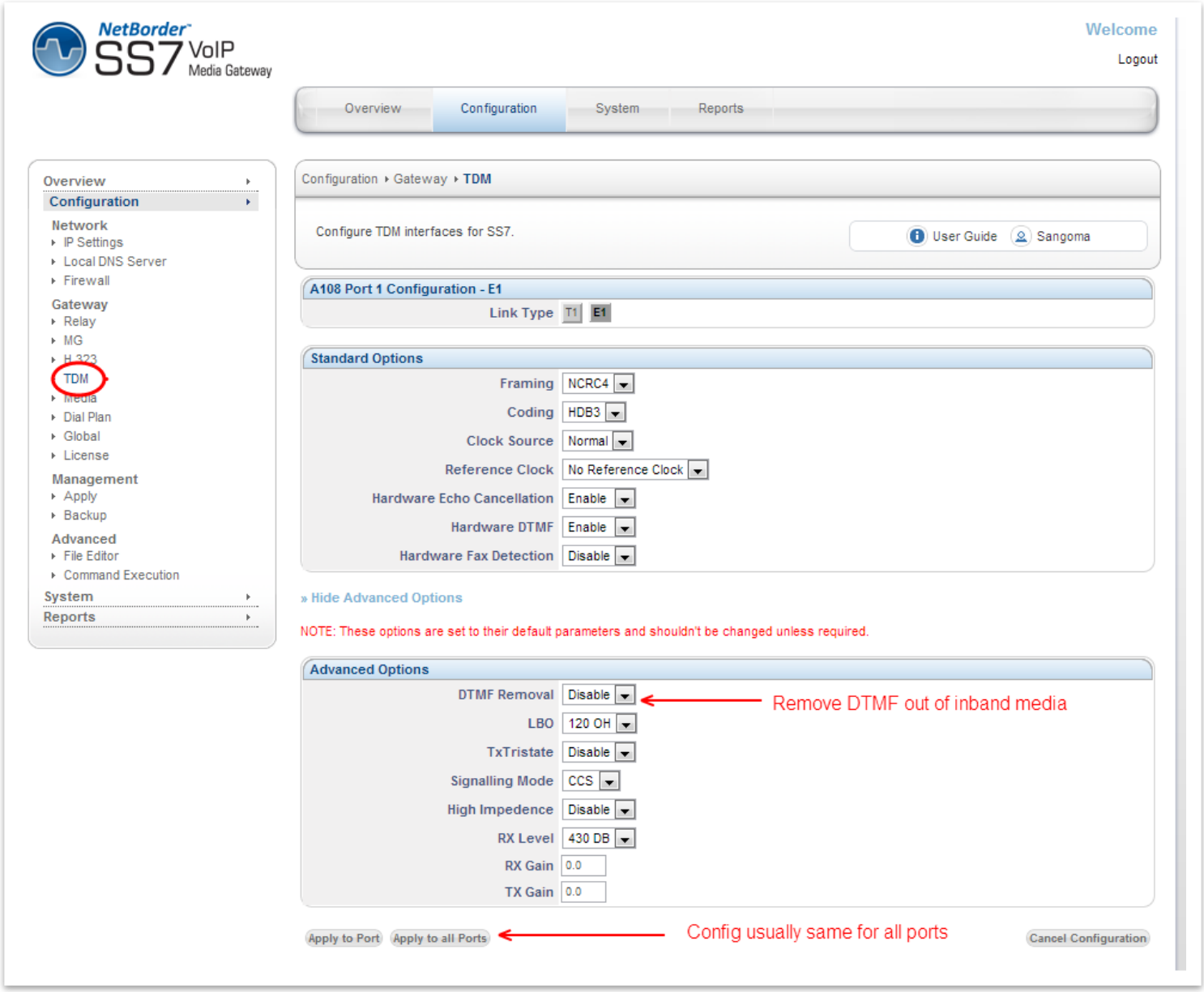


8.6.2.1 Standard T1/E1 Parameters



- In case advanced parameters are not necessary proceed
- Apply to Port
    - Applies the configuration for a single T1/E1 port
    - (The one that is currently being edited)
- Apply to all Ports
    - Apply to all T1/E1 ports on a board.
    - Bulk config feature
    - (This feature saves time as T1/E1 ports are usually provisioned the same)

8.6.2.2 Advanced T1/E1 Parameters



NOTE
After T1/E1 configuration, the NSG wizard will request **Link Type** Configuration.

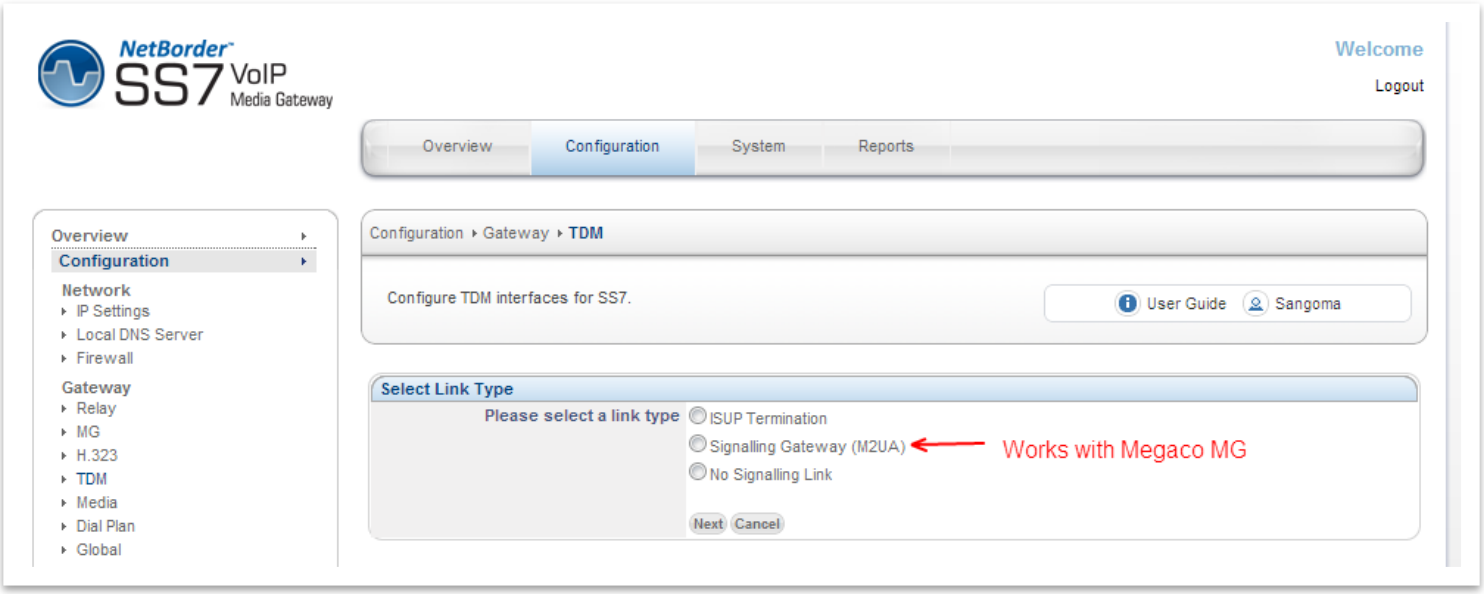
8.7 Span Link Type

When configuring TDM Terminations for Megaco Media Gateway there are two possibilities

- Voice Mode
  - All TDM channels are used for Voice 64kbs G.711
  - Example: All channels 1-31 on an E1 line are used for voice
  - Link Type = Voice Only

- Mix Mode
  - Voice 64kbs G.711 channels and SS7 signaling channels.
  - Example: Channel 16 is used for SS7 signaling, 1-15,17-31 are used for voice.
  - Link Type = Signaling Gateway (M2UA)

- If configuring for **Voice Mode** select **No Signaling Link**
- If configuring for **Mixed Mode** select **Signaling Gateway (M2UA)**



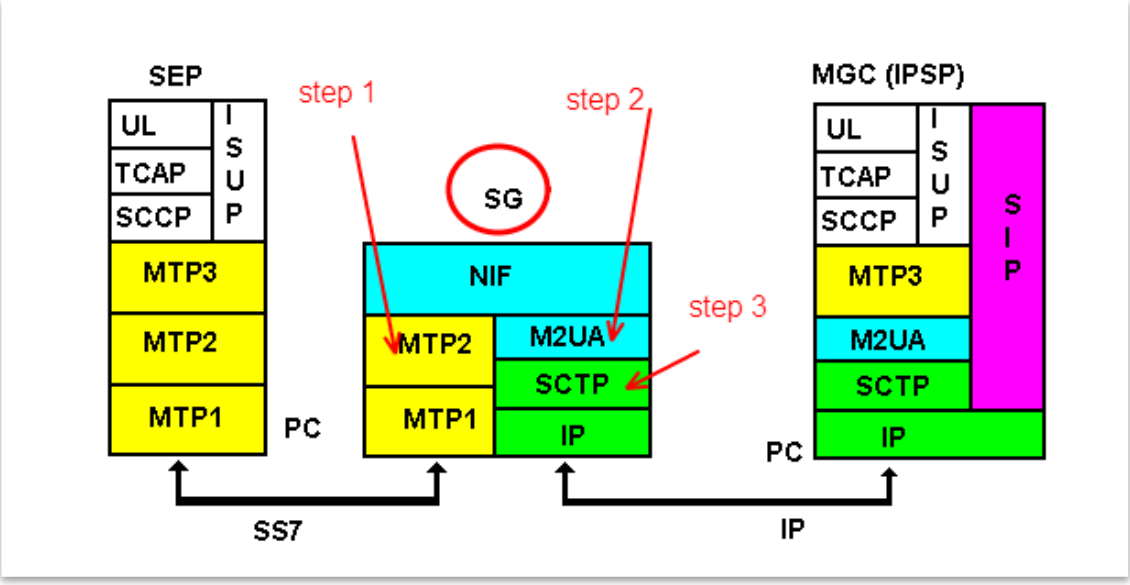
NOTE
The rest of this section will continue to document the **Signaling Gateway (M2UA)** option.
Next page will introduce the Signaling Gateway Overview, followed by the next config section in the WebGUI.

8.8 Signaling Gateway Overview

NSG supports Signaling Gateway operation mode.

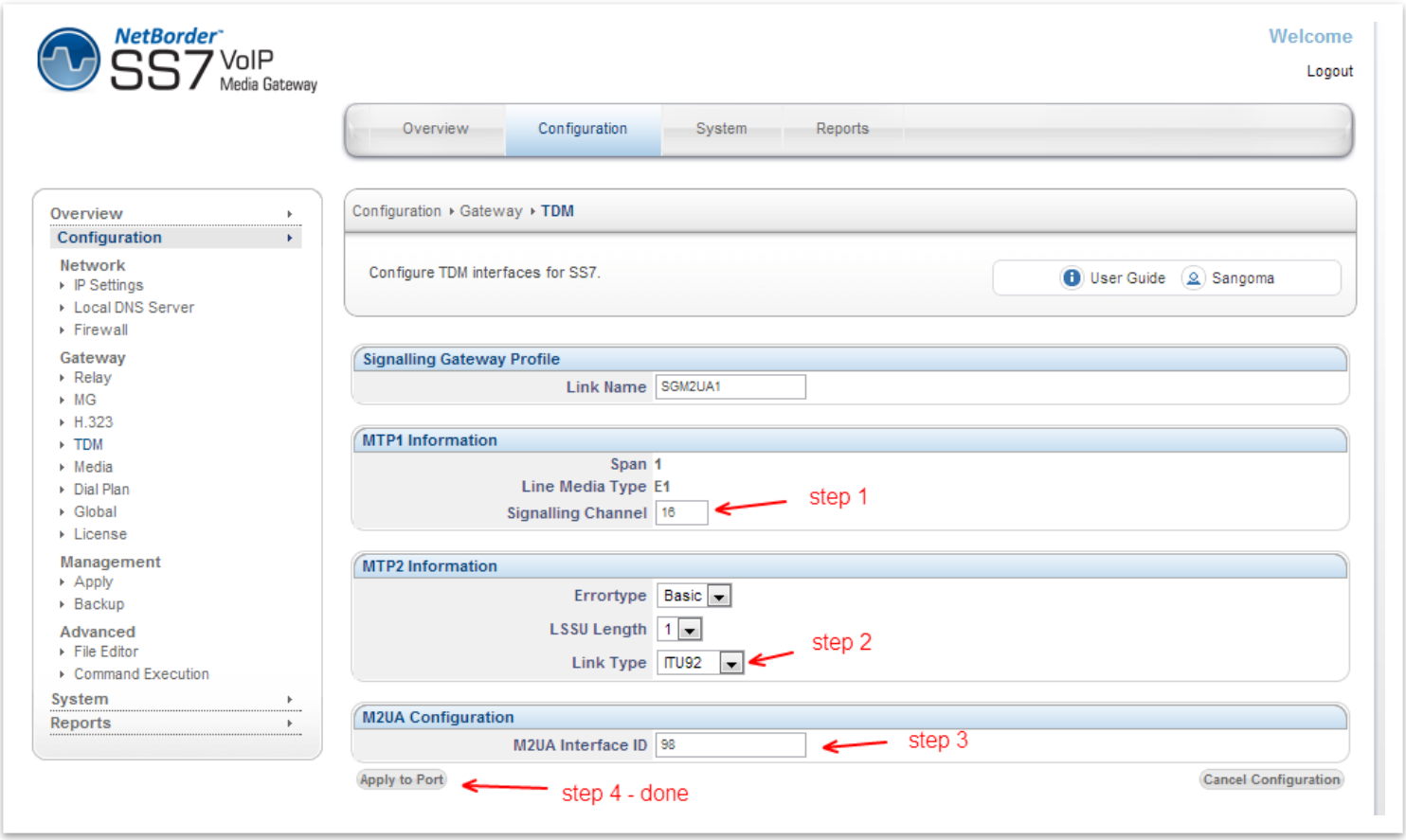
In Signaling gateway mode, NSG will bridge T1/E1 SS7 signaling link to IP and pass it transparently to the MGC/Softswitch, via M2UA protocol. Looking at the diagram below, NSG Signaling Gateway will configure:

- MTP1 & MTP2 protocols over the TDM port
- M2UA/SCTP protocol over IP network
- NIF (Network interworking function) to bridge the two



8.8.1 MTP1/2 Link Configuration

- Specify MTP1/2 information based on provider provision document
- Step1: Identify which channel on T1/E1 line will carry signaling
- Step2: Specify MTP2 signaling information based on provision document
- Step3: Specify M2UA Interface ID based on provision document
- **Apply to Port** to save configuration



Field Name	*Possible Values*	*Default Value*	*Description*
Link Name	NA	NA	M2UA Profile name
Span	NA	NA	Span number which is going to associated with this M2UA profile.
Line Media Type	E1/T1	E1	Media type
Signaling channel	NA	NA	Signaling channel of the span which will carry the M2UA signaling messages.
ErrorType	Basic/PCR	Basic	MTP2 error type.
LSSU length	1/2	1	LSSU length
Link Type	ITU92 ITU88 ANSI96 ANSI92 ANSI88 ETSI	ITU92	SS7 link variant.
M2UA Interface ID	NA	NA	M2UA Interface identifier which will map to this particular signaling span/channel and uniquely identify the link between M2UA SG and MGC.

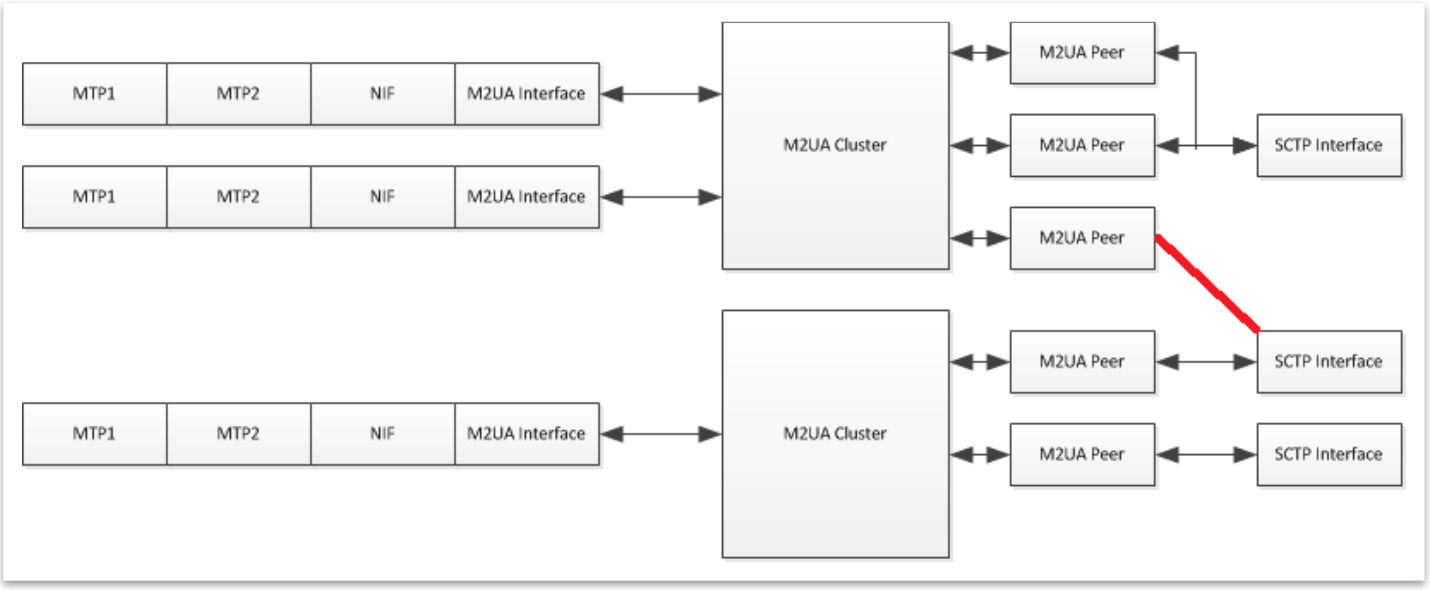
NOTE
Next section in WebUI will relate to M2UA configuration. Before we proceed however, the M2UA interface architecture will be introduced in order to provide a big picture to the user.

8.8.2 M2UA Interface

This section provides in-depth overview on how the M2UA interface is constructed. It should help the user better understand the WebUI configuration objects for M2UA protocol.

WebUI for M2UA contains 3 sections: Cluster, Peer and SCTP

- SCTP interfaces are standalone objects on which a peer bind to (regardless of its cluster).
  - 1 SCTP binds to 1 or more peers
  - 1 peer binds to 1 SCTP
  - Thus SCTP are shared across all peers
  - SCTP cannot be deleted if used by any peer (even from another cluster).
  - Deleting a peer or a cluster does not delete SCTP.
- Peers are bound to cluster.
  - 1 peer binds to 1 cluster
  - 1 cluster binds to 1 or more peer
  - Deleting a cluster will delete peers.
- Cluster are bound to MTP2 through M2UA binding and nif interface
  - 1 cluster binds to 1 or many MTP2 (through M2UA->NIF relationship)
  - 1 MTP2 binds to 1 cluster through NIF interface binding



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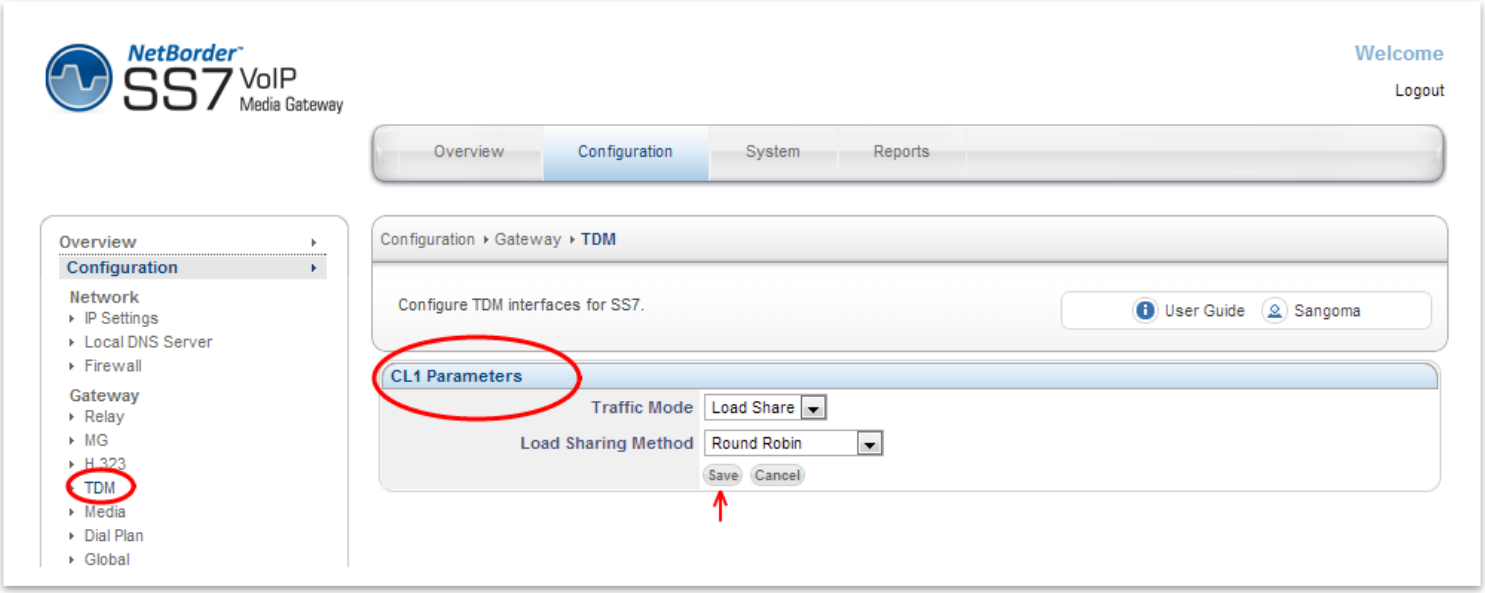
8.8.3 M2UA Cluster Creation

M2UA Cluster is a group of peers to which M2UA SG will communicate

- Select Create Cluster
- Leave the Cluster values default unless the provider specifies otherwise.
- Select **Save** to Continue

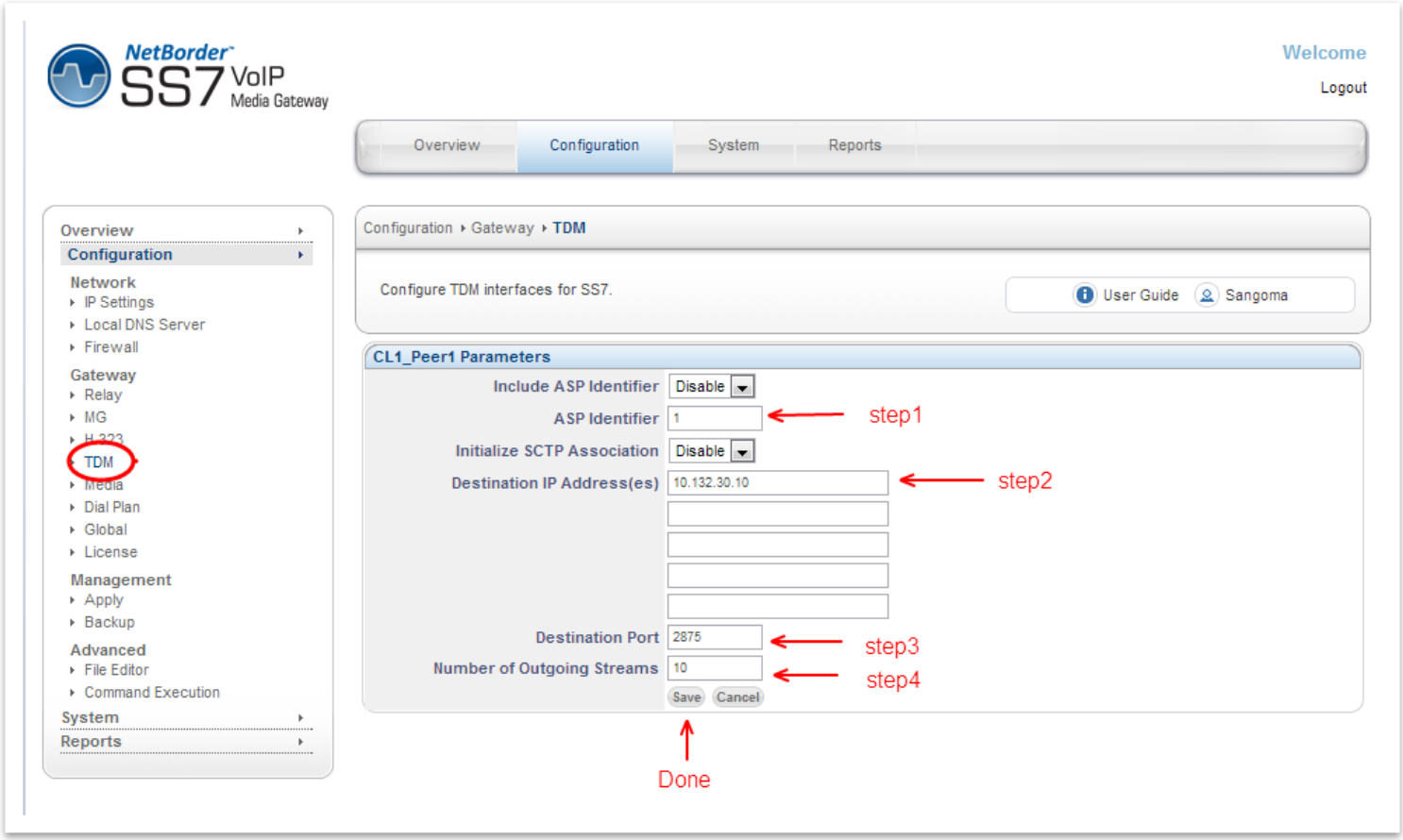


Field Name	*Possible Values*	*Default Value*	*Description*
Traffic Mode	Load Share Override Broadcast	Load Share	This parameter defines the mode in which this Cluster is supposed to work.
Load Sharing Method	Round Robin Link Specified Customer Specified	Round Robin	This parameter defines the load share algorithm which is used to distribute the traffic

8.8.4 M2UA Cluster Peers

M2UA Peers will be configured under the M2UA clusters

- Select **Add** under Cluster Peers Profile
- Select **Create** Cluster Peer Profile
- Specify the Cluster Peer parameters based on provider provision document

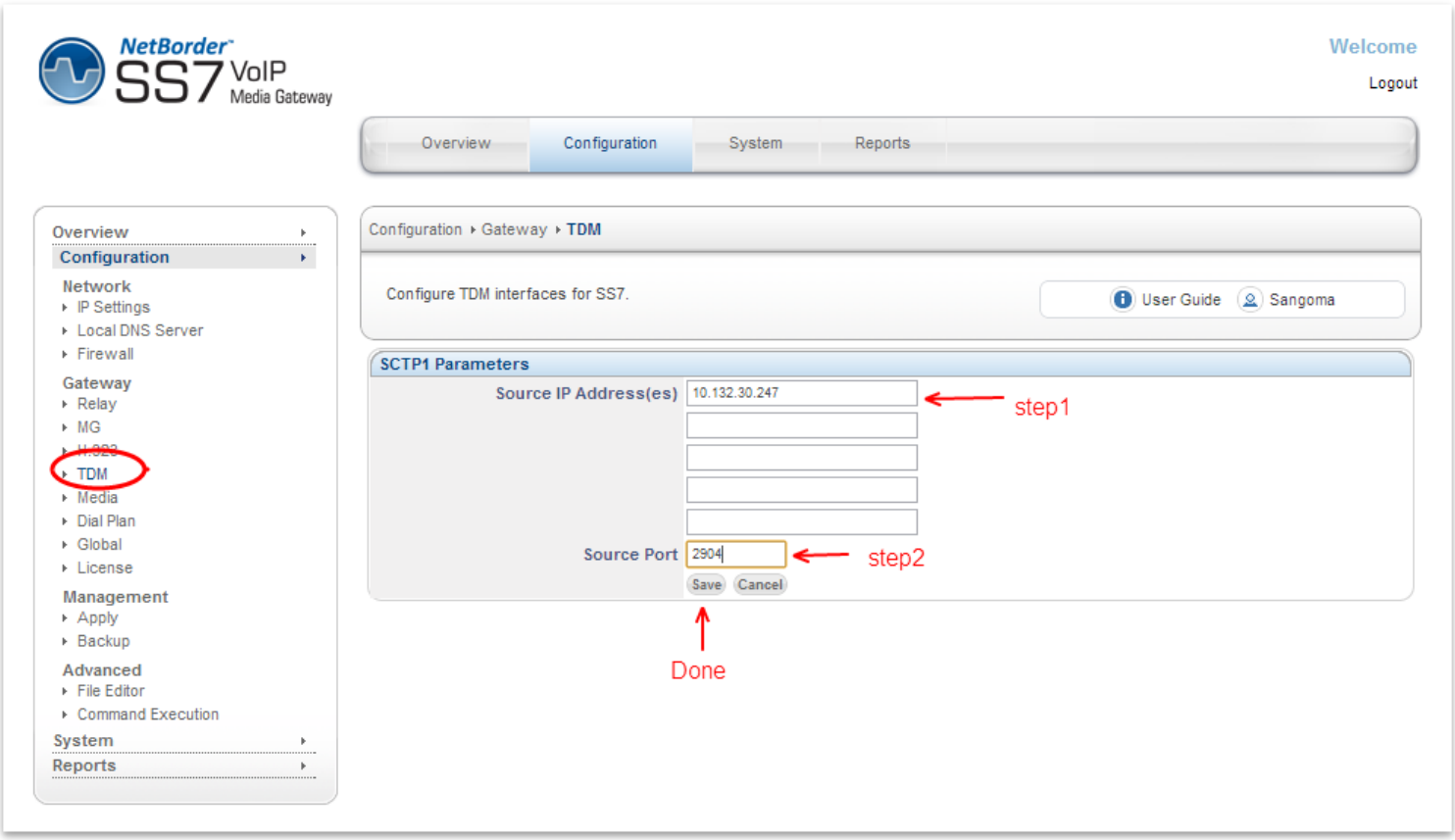


Field Name	Possible Values	Default Value	Description

Include ASP Identifier	Disable Enable	Disable	Flag used to indicate whether include the ASP ID in the ASP UP message
ASP Identifier	NA	NA	ASP identifier for this ASP node. Set to 1 in case ASP is Disabled
Initialize SCTP Association	Disable Enable	Disable	Flag used to indicate if M2UA SG has to start SCTP association or not. If Disable means M2UA SG will wait for SCTP association request from MGC. If Enable that means M2UA SG will initiate the SCTP association request towards MGC.
Destination IP Address(es)	NA	NA	Destination IP address
Destination port	NA	2904	Destination ASP Port Default M2UA ASP port: 2904
Number of Outgoing Streams	NA	10	Number of outgoing streams supported by this association. Default 10

8.8.5 SCTP Interface

- Select Add SCTP Interface
- Select Create SCTP Interface
- Specify SCTP Information based on provider provision document



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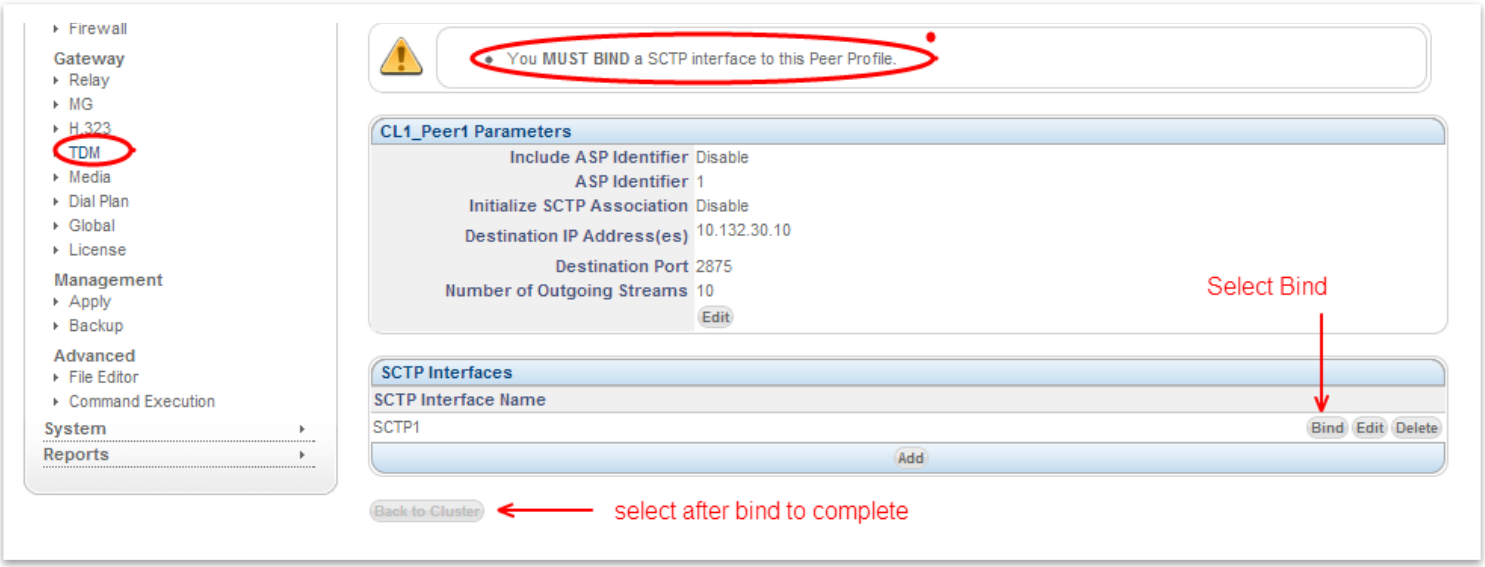
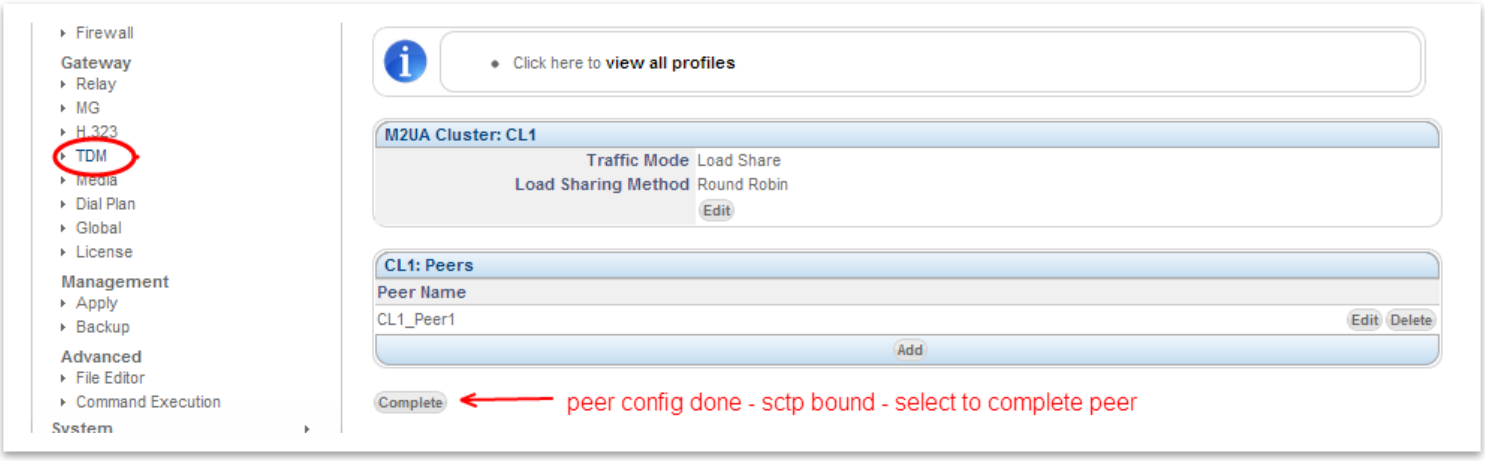
8.8.6 Binding all components

- All components have been created
  - M2UA Cluster
  - M2UA Peer
  - SCTP Interface

- Next step is to Bind / Connect them together
  - SCTP interface into M2UA Peer
  - M2UA peer into M2UA Cluster





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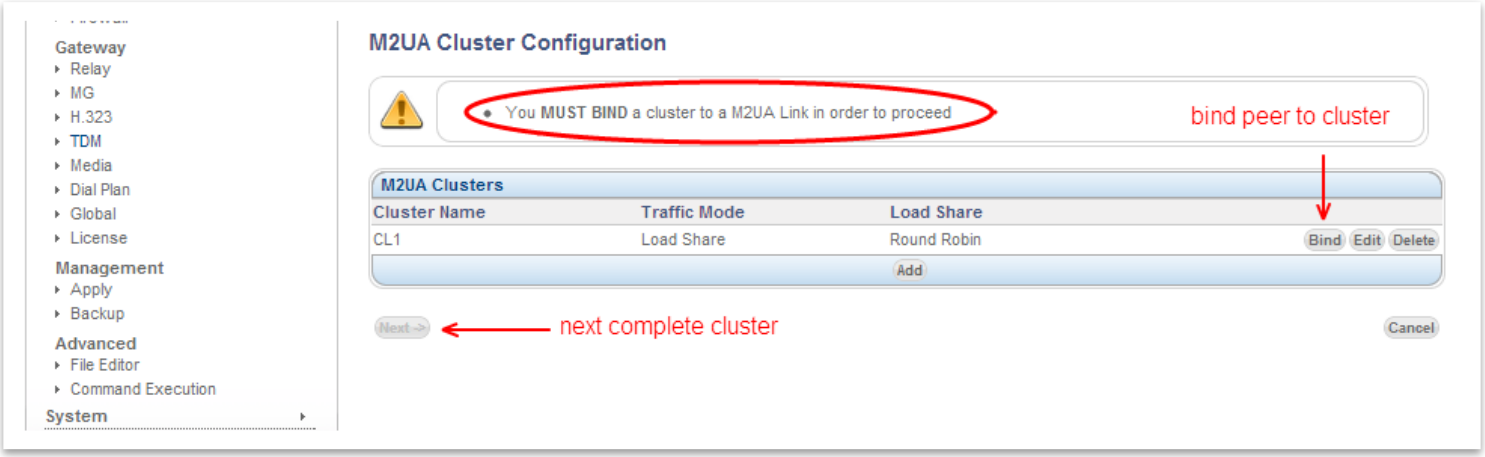
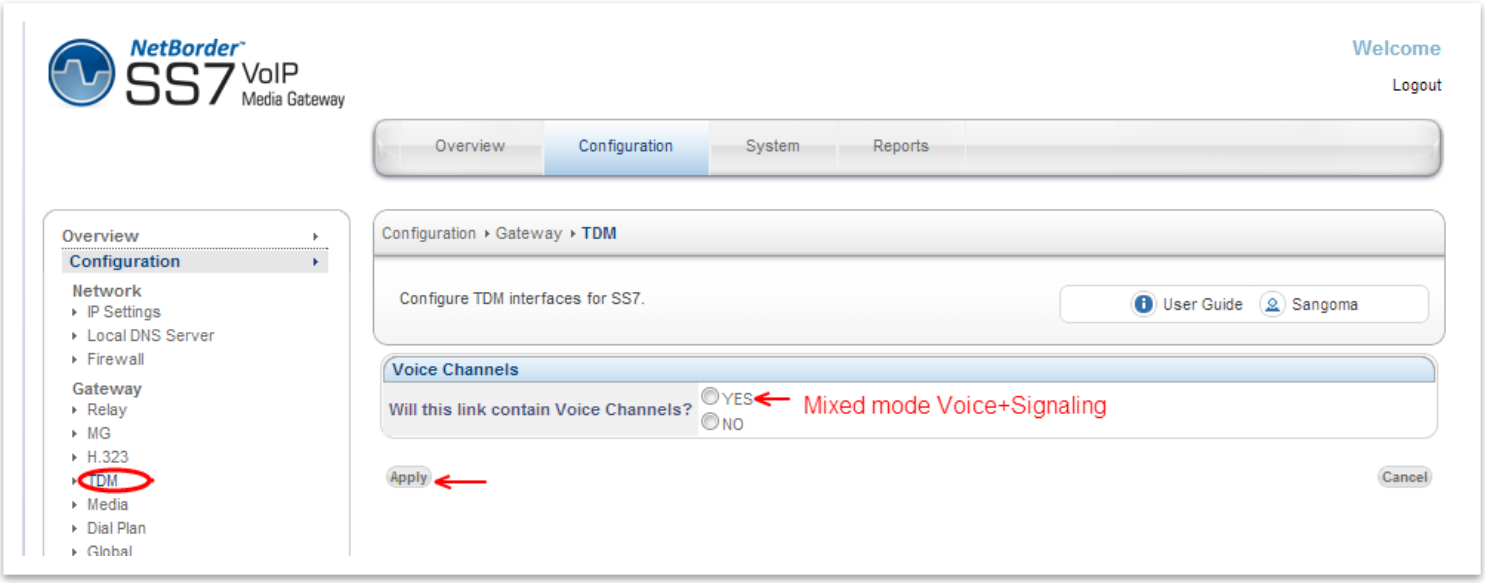


8.8.7 *Mixed Mode Configuration*

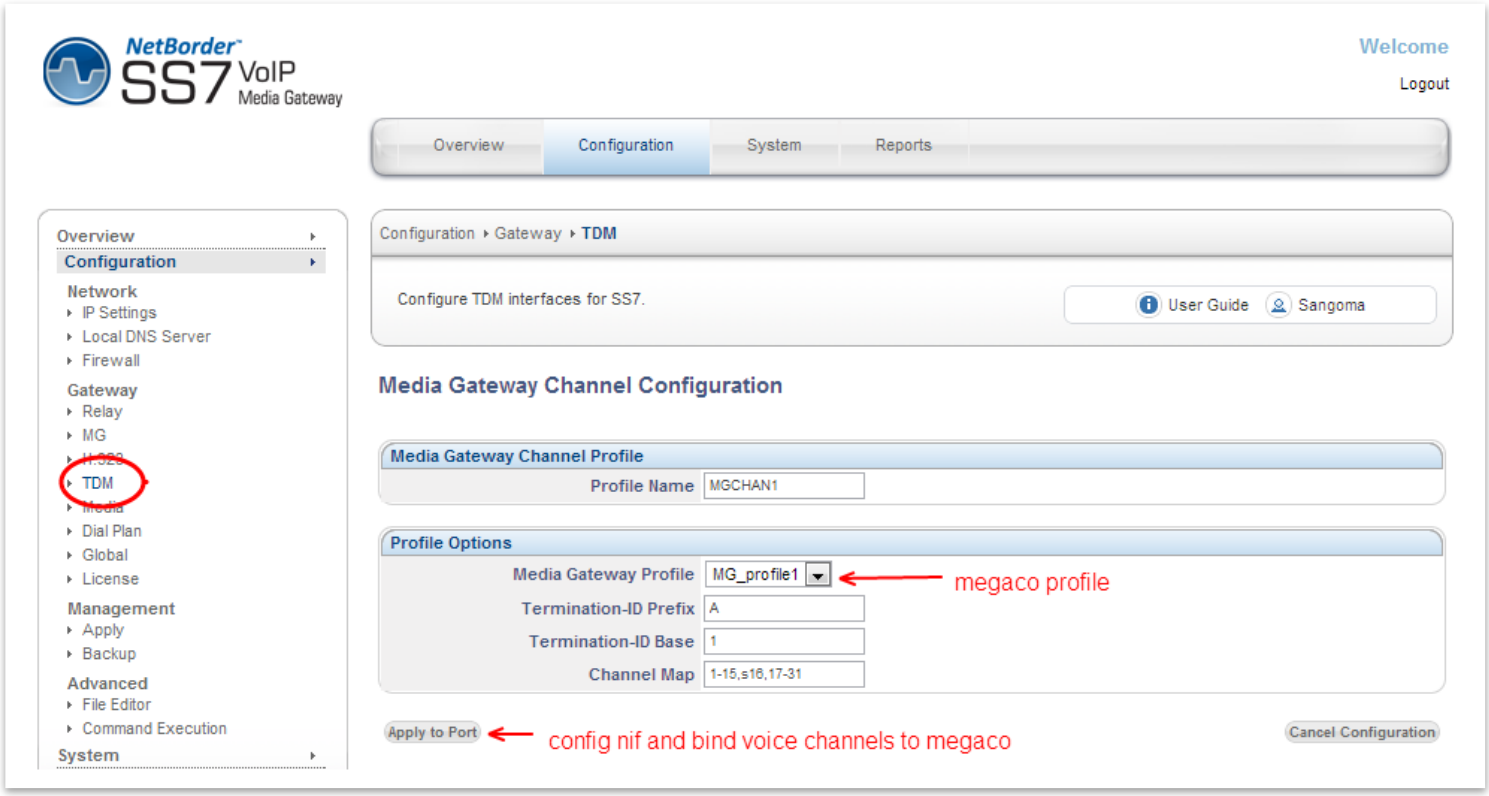
- Signaling is bridged by M2UA to the MGC/Soft switch
- Voice is controlled by Megaco/H.248
- Specify that Voice is part of this TDM Span



NOTE
Rest of this section will document the **Mixed Mode Configuration**

8.8.8 Bind Megaco to TDM

The last step of the configuration is to bind the TDM voice channels to Megaco Profile.

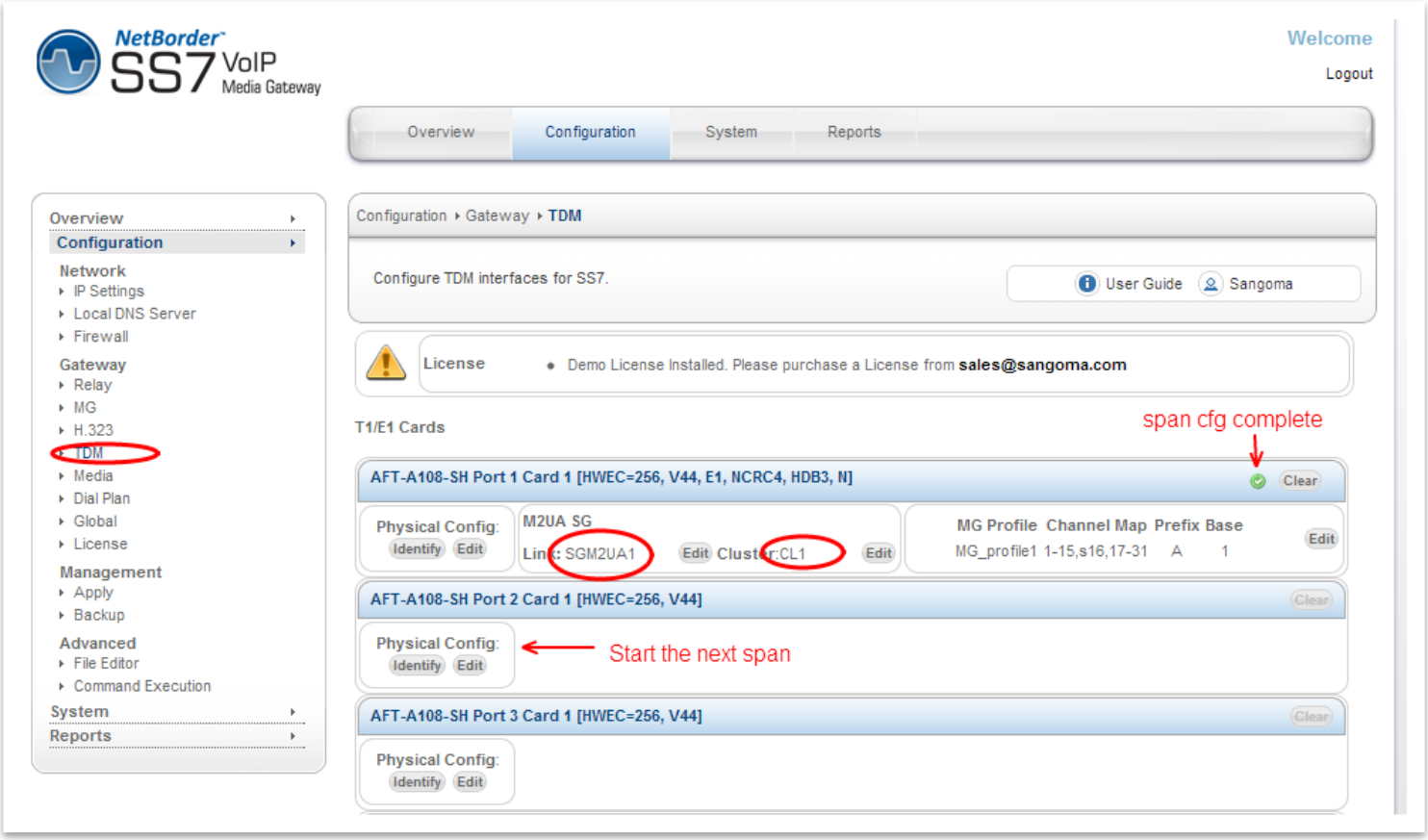


Field Name	*Possible Values*	*Default Value*	*Description*
Media Gateway Profile	List of Gateways	First in the List	Select Megaco Profile that will be used to control the TDM channels for this span.
Termination ID Prefix	NA	NA	Usually a letter A-Z. This prefix is defined by MGC. Please refer to MGC configuration.
Termination ID Base	NA	NA	Usually a number starting from 1. This value is defined by MGC. Please refer to MGC configuration.
Channel Map	NA	NA	List of channels to be controlled by Megaco Example: 1-15,s16,17-31 Channels 1-15 and 17-31 are used for Voice and should be controlled by Megaco Channel 16 (prefixed by letters) indicates that channel 16 carries signaling channel. Megaco will ignore this channel as it's not voice. Prefix Letters to signaling channel: **s**: megaco id not used, id mapped to signaling channel **g**: megaco id is used, id mapped to next available voice channel. The bind between megaco and TDM would be as follows **Channel Map: 1—31 (no signaling channel)** A1: channel 1 A2: channel 2 … A16: channel 16 … A30: channel 30 A31; channel 31

			Channel Map: 1-15,s16,17-31 (signaling on ch 16) A1: channel 1 A2: channel 2 … A15: channel 15 … A16: not used – A16 points to signaling channel 16 A17: channel 17 A18: channel 18 … A31: channel 31 **Channel Map: 1-15,g16,17-31 (signaling on ch 16)** A1: channel 1 A2: channel 2 A15: channel 15 A16: channel 17 - A16 is used and it points to ch 17. A17: channel 18 … A30: channel 31

8.8.9 TDM Termination Complete

- A span has been configured and bound to a Megaco Profile.
- Configuration for this span is done
  - Confirmed in WebUI by a green checkmark.



- Next step is to repeat the process for the rest of the spans.
- In typical configurations there is one or two spans (T1/E1 ports) that contain signaling channels. The rest of the spans are usually voice only.
- In voice only config, there is no Signaling Gateway configuration.
  - The configuration jumps directly to “Bind TDM to Megaco” section of the WebUI.

NOTE
The changes made in the Configuration section of the WebUI are only stored one the scratch disk. User MUST proceed to **Apply** page in the **Management Section** to save new configuration.

9 SS7 ISUP

SS7 is a signaling protocol, it is used to carry call control information such as call start, call progress, call hangup etc. The SS7 call control information is used to control arbitrary number of voice channels that are carried using T1/E1 spans.

In a typical SS7 setup the telco will provide you with SS7 information that will be used to map T1/E1 physical spans and channels into SS7 call control information.

The NSG TDM SS7 configuration page has been designed as bottom up SS7 configuration approach.

1. Identify T1/E1 spans on your system
2. For each T1/E1 span on your system:
    a. Determine which T1/E1 spans will carry SS7 Link channels
    b. T1/E1 Span can either carry an
        i. SS7 Link in one of its channels or
        ii. All T1/E1 channels can be used to carry voice.
    c. Configure T1/E1 physical configuration parameters
    d. Identify if T1/E1 span carries SS7 link or is Voice Only

    e. If T1/E1 span has an SS7 link associate with it:
        i. Create a new SS7 Link
        ii. Next step is to bind the new SS7 Link to an SS7 Linkset.
        iii. If an SS7 Link set does not exist, Create a new SS7 Link Set
        iv. Then bind the SS7 Link to an existing or new SS7 Link Set
        v. Next step is to bind the SS7 Linkset into an SS7 Route.
        vi. If an SS7 Route does not exist, Create a new SS7 Route
        vii. Then bind the SS7 Linkset to an existing or new SS7 Route
        viii. Next step is to bind the SS7 Route into an SS7 ISUP Interface
        ix. If an SS7 ISUP Interface does not exist, Create a new SS7 ISUP Interface
        x. Then bind the SS7 Route to an existing or new SS7 ISUP Interface

    f. The Last step is to assign CIC values to each physical T1/E1 timeslot in the span.

        Whether the Span carries only voice or it contains the SS7 Link, each timeslot must be associated with a SS7 CIC value.

        This way when an incoming SS7 Call Start message arrives with an arbitrary CIC value. The NSG system can open the appropriate physical voice channel associated with the CIC value.

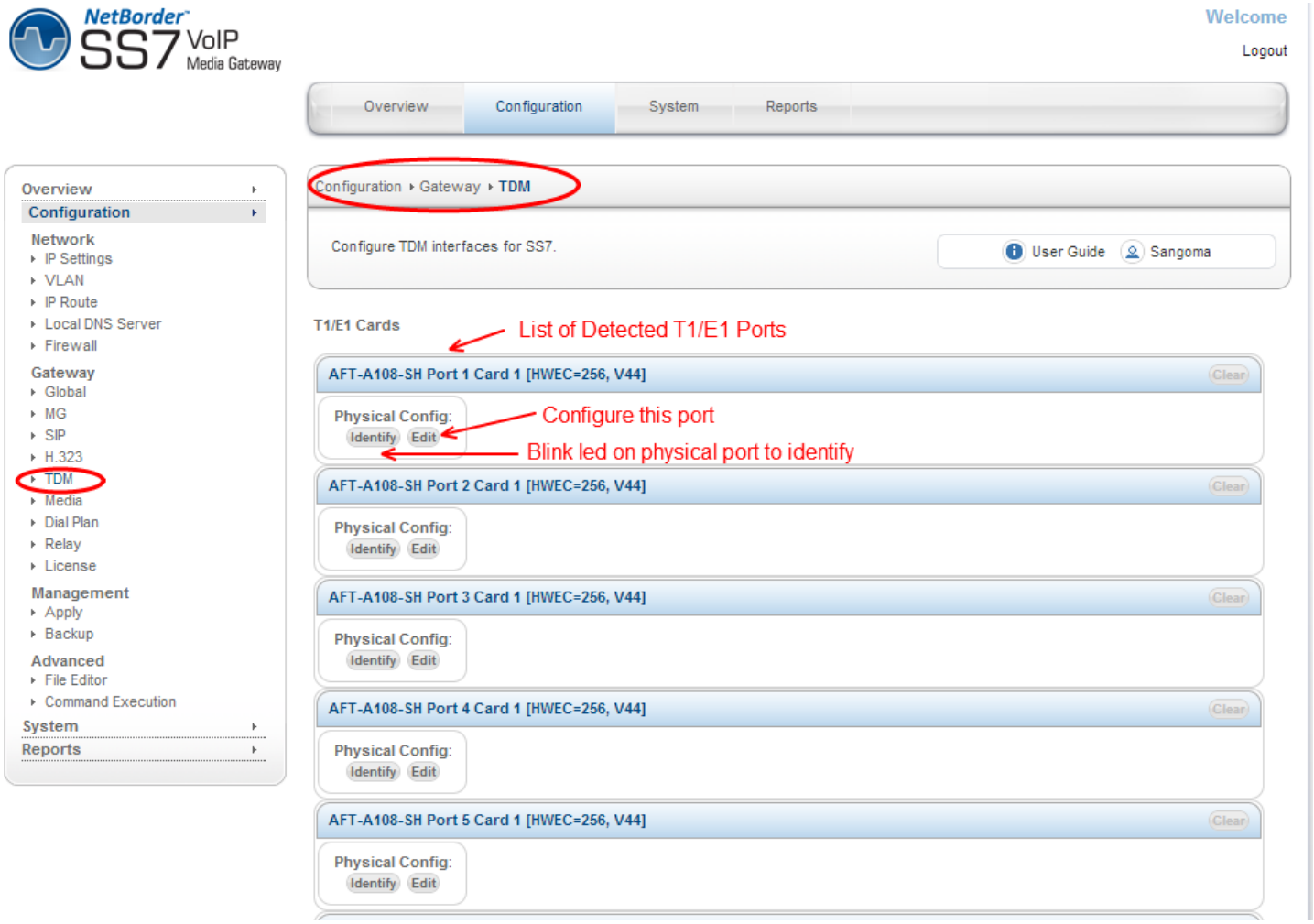
3. Once all T1/E1 spans are configured you need to **Apply** the configuration files.
   Note that this step does not start the NSG gateway. It just writes the appropriate configuration files.

4. Proceed to the Control Panel to start the NSG SS7 to VoIP Gateway.

9.1 TDM SS7 Configuration Page

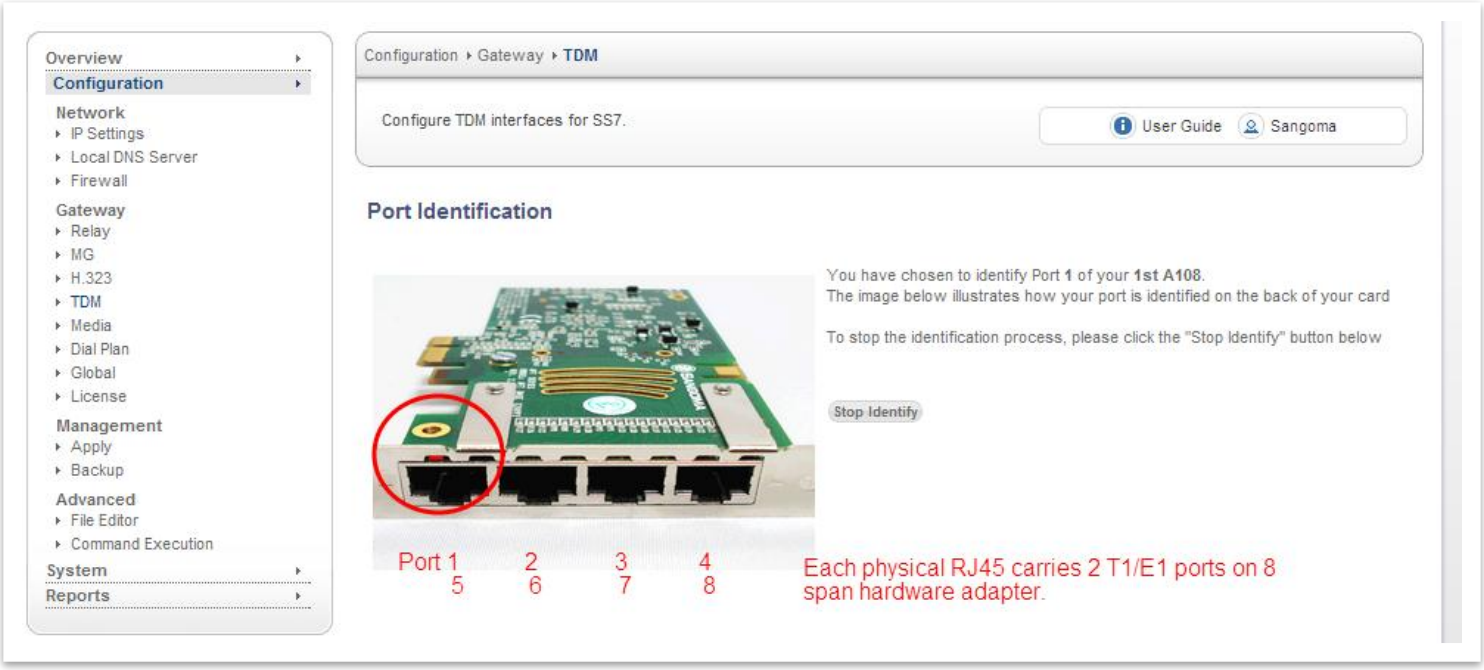
- Select **TDM** from side/top **Configuration** menu
- The TDM section will display all installed TDM Spans/Ports.

The TDM Configuration page will display to the user every T1/E1 card detected by NSG.
Each card is logically separated into ports, which initially displays the firmware version and the Echo Cancellation security chip ID. If the echo cancellation security chip ID is 0, then the card installed does not have echo cancellation. If there is a alert image next to the firmware version, that means the firmware on the system is out of date, and must be updated in order to have the most up to date and efficient firmware running.



9.2 Port Identification

- In order to determine which physical T1/E1 port is: Port 1 Card 1
- Select **Identify** button for Port 1 Card 1
- The LED light will start flashing on a rear RJ45 T1/E1 port: rear panel.
- Look at the rear panel of the appliance and plug in RJ45 cable to the blinking RJ45 T1/E1 port.
- Once the Port 1 Card 1 is identified, the subsequent ports for that board are labeled.
- Or alternatively keep using the Identify feature for each port.

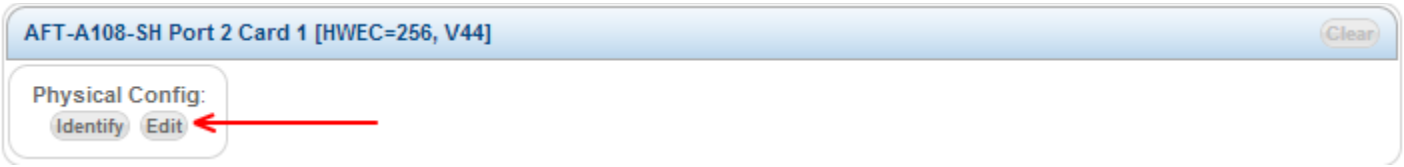


NOTE

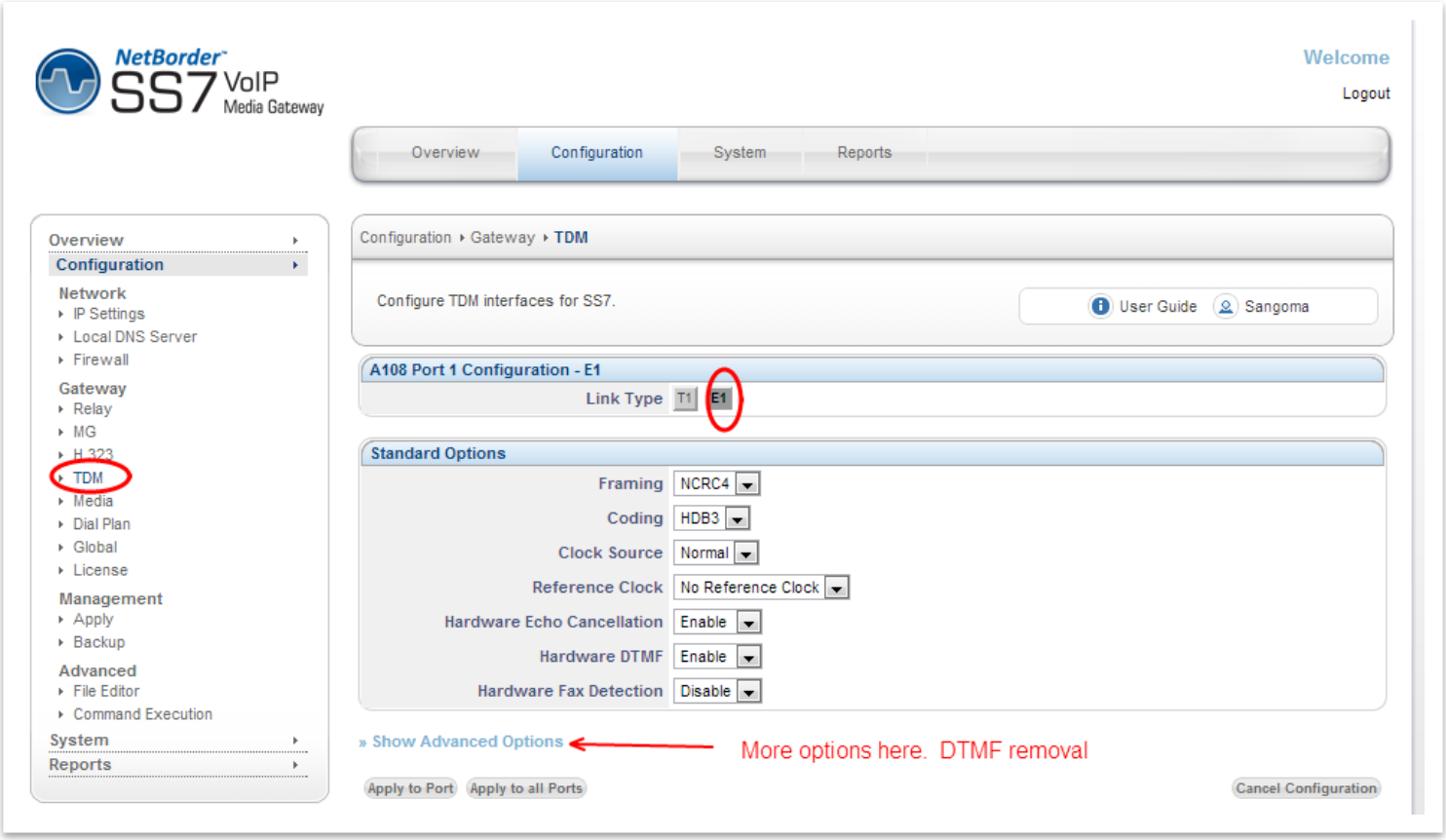
- Identify picture of the device is always set to A108D – 8 T1/E1 card. The LED will always bling port 1. The image is not meant to reflect the real hardware image, nor real port location. User should always view the rear panel for the flashing LED.
- All Sangoma TDM T1/E1 cards Port 1 is closest to the PCI slot.

9.3 Edit T1/E1 Config

- Once the port has been identified and plugged into the T1/E1 network.
- Select **Edit** button for Port 1 Card 1 to configure the physical T1/E1 parameters.
- Select the port configuration type: T1 or E1
  - T1: North American Market and Japan
  - E1: Europe and the world
- Fill in Physical Configuration T1 or E1 parameters
  - Fill in the T1/E1 parameters based on the provider provision document.



9.3.1 Standard T1/E1 Parameters



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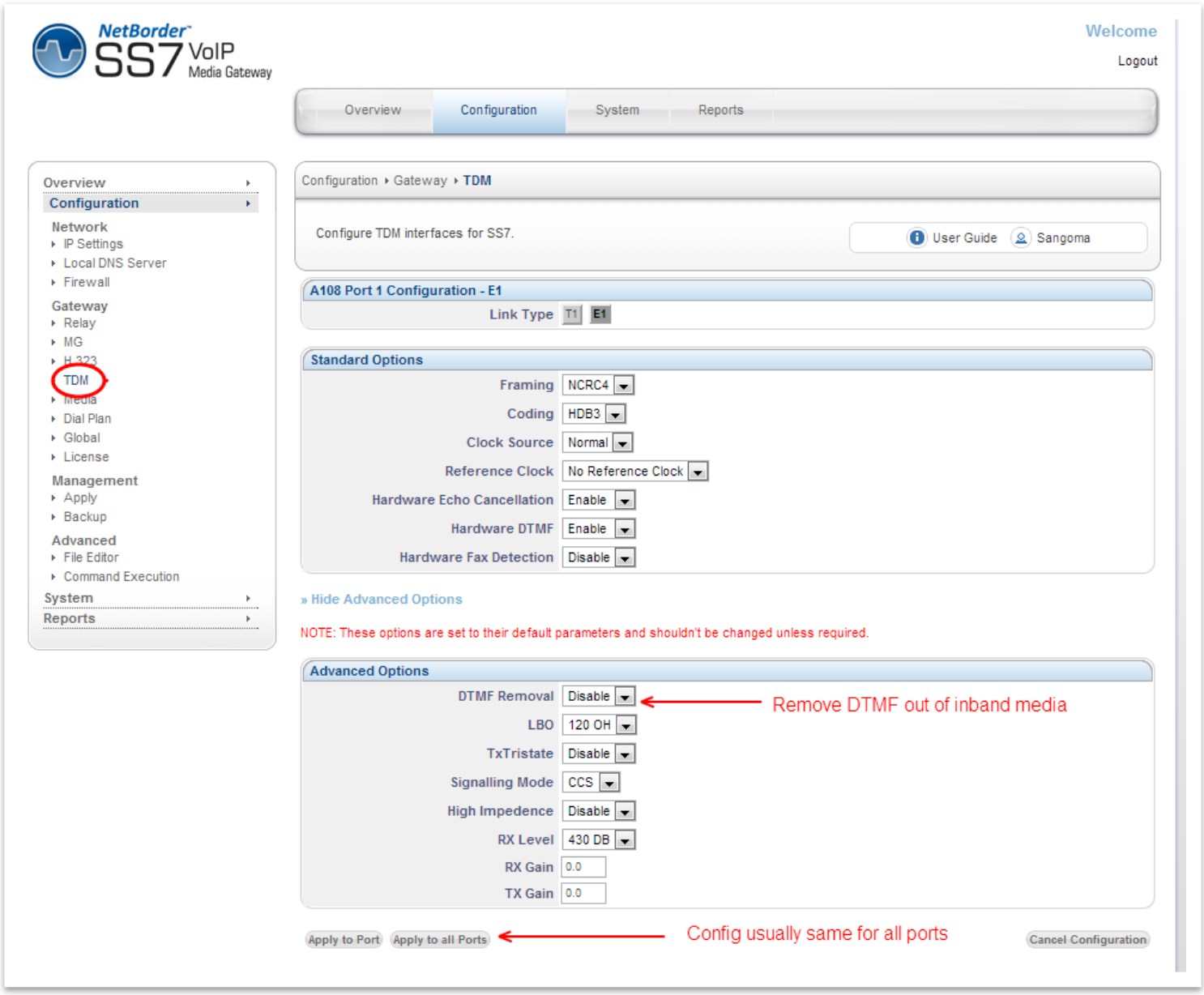
- In case advanced parameters are not necessary proceed
- Apply to Port
  - Applies the configuration for a single T1/E1 port
  - (The one that is currently being edited)
- Apply to all Ports
  - Apply to all T1/E1 ports on a board.
  - Bulk config feature
  - (This feature saves time as T1/E1 ports are usually provisioned the same)

9.3.2 Advanced T1/E1 Parameters



NOTE
After T1/E1 configuration, the NSG wizard will request **Link Type** Configuration.

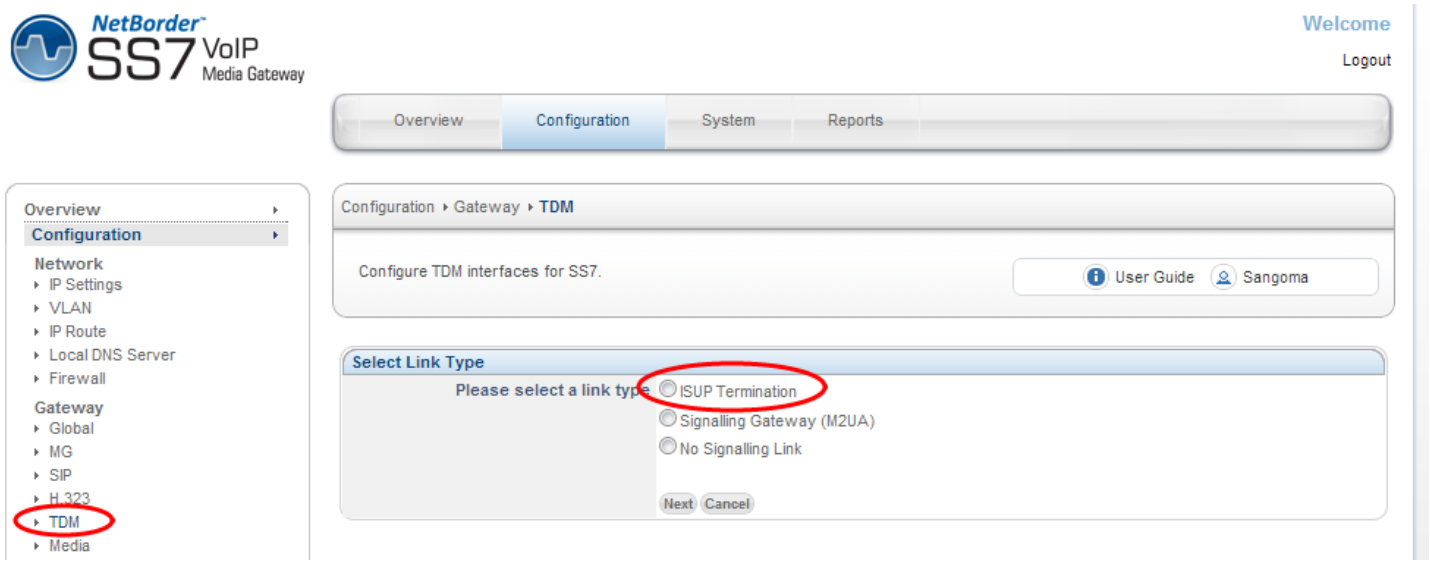
9.4 Span Link Type

When configuring TDM Terminations for Megaco Media Gateway there are two possibilities

- Voice Mode
  - All TDM channels are used for Voice 64kbs G.711
  - Example: All channels 1-31 on an E1 line are used for voice
  - Link Type = Voice Only

- Mix Mode
  - Voice 64kbs G.711 channels and SS7 signaling channels.
  - Example: Channel 16 is used for SS7 signaling, 1-15,17-31 are used for voice.
  - Link Type = ISUP Termination

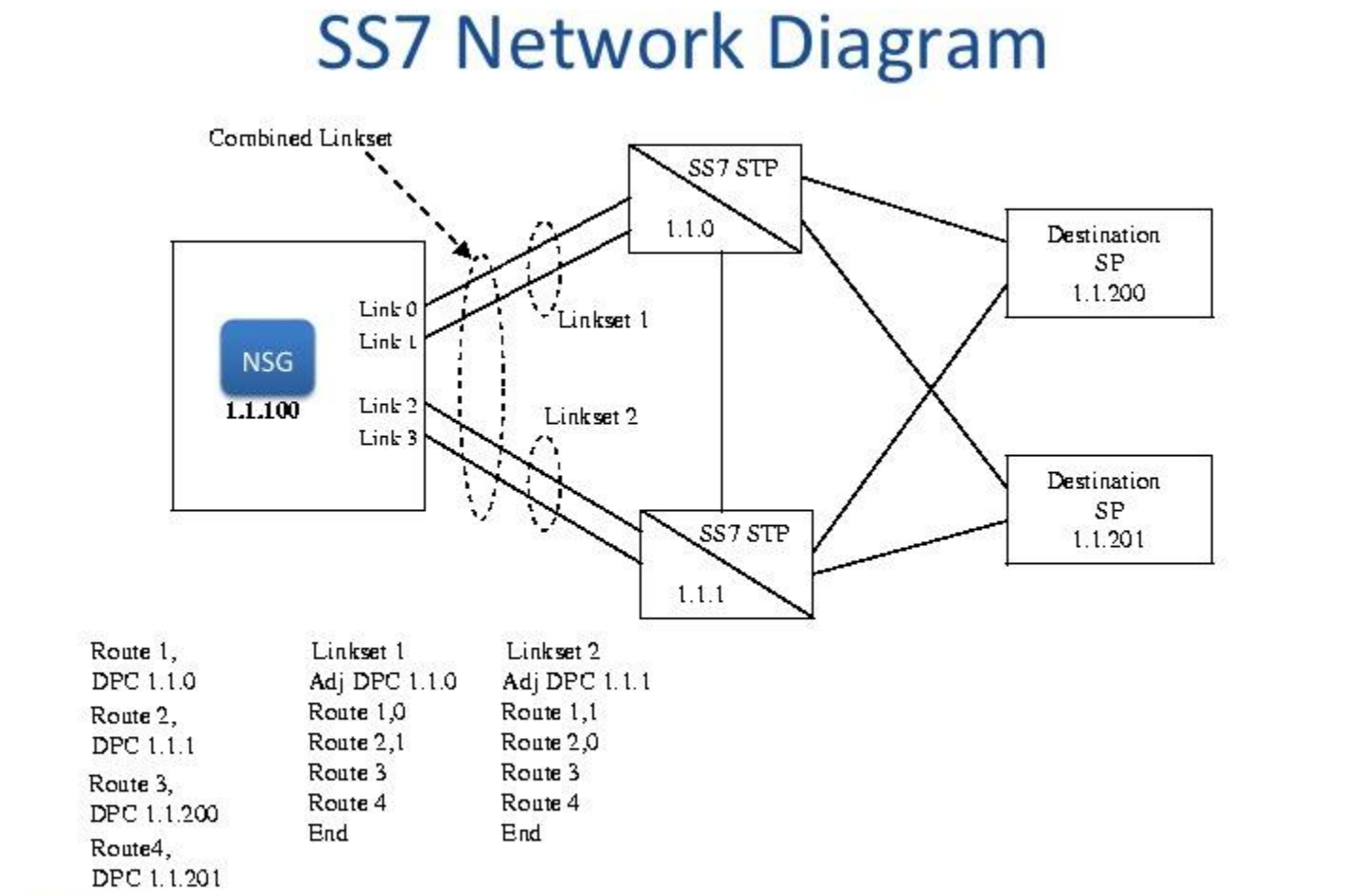
- If configuring for **Voice Mode** select **No Signaling Link**
- If configuring for **Mixed Mode** select **ISUP Termination**



NOTE

- The rest of this section will continue to document the **ISUP Termination** option.
- In case of **Voice Mode** – the GUI will skip the ISUP configuration and proceed directly to Channel Map Section below.

9.5 SS7 Network Overview



9.5.1 Links

- physical signaling links between the TX board and the adjacent signaling points. One link configuration must be performed for each physical signaling link. The attributes of a link include the point code of the adjacent signaling point, protocol variant employed on the link (ITU-T or ANSI), point code length, maximum packet length, various timer values, membership in a linkset, and others.

9.5.2 Linksets

- are groups of from one to 16 links that directly connect two signaling points. Although a linkset usually contains all parallel signaling links between 2 SPs, it is possible to define parallel link sets. Each signaling link defined is assigned membership in exactly one link set.

9.5.3 Routes

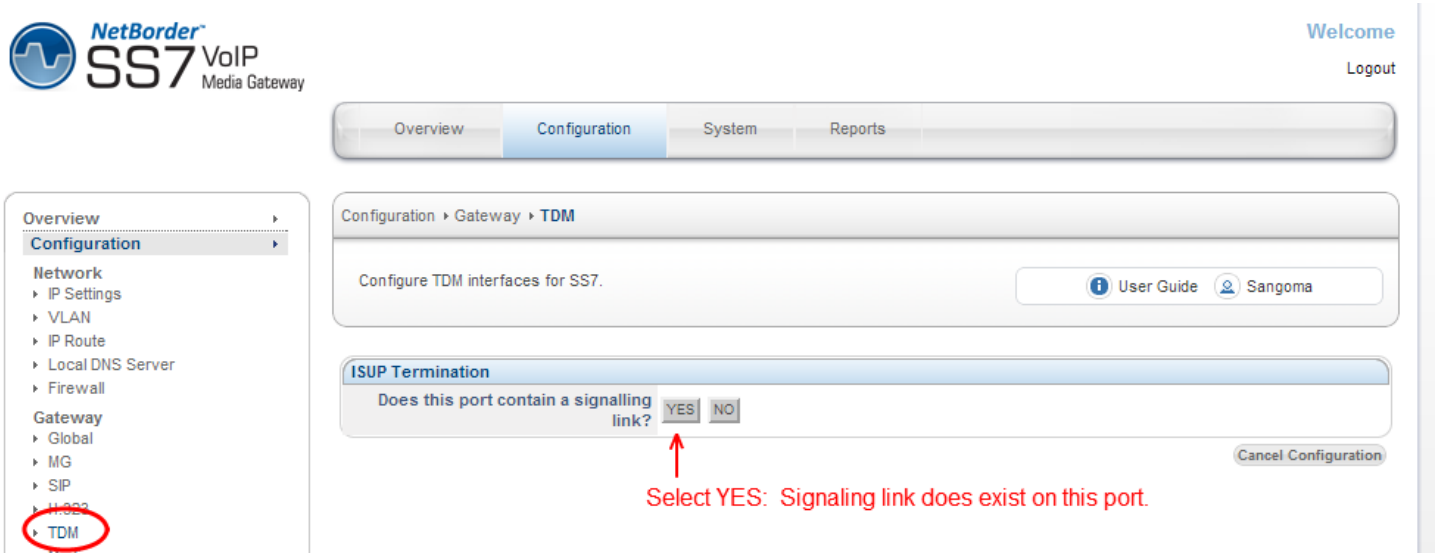
- specify the destination signaling points (or sub-networks (clusters) when route masks are employed) that are accessible from the target node. Each route is assigned a direction - up or down. One up route is required for the actual point code assigned to the signaling point being configured and for each point code that is to be emulated. Up routes are used to identify incoming messages that are to be routed up to the applications/user parts. One down route is required for each remote signaling point/network/cluster that is to be accessible from the SP being configured.

9.6 MTP2 Link Configuration

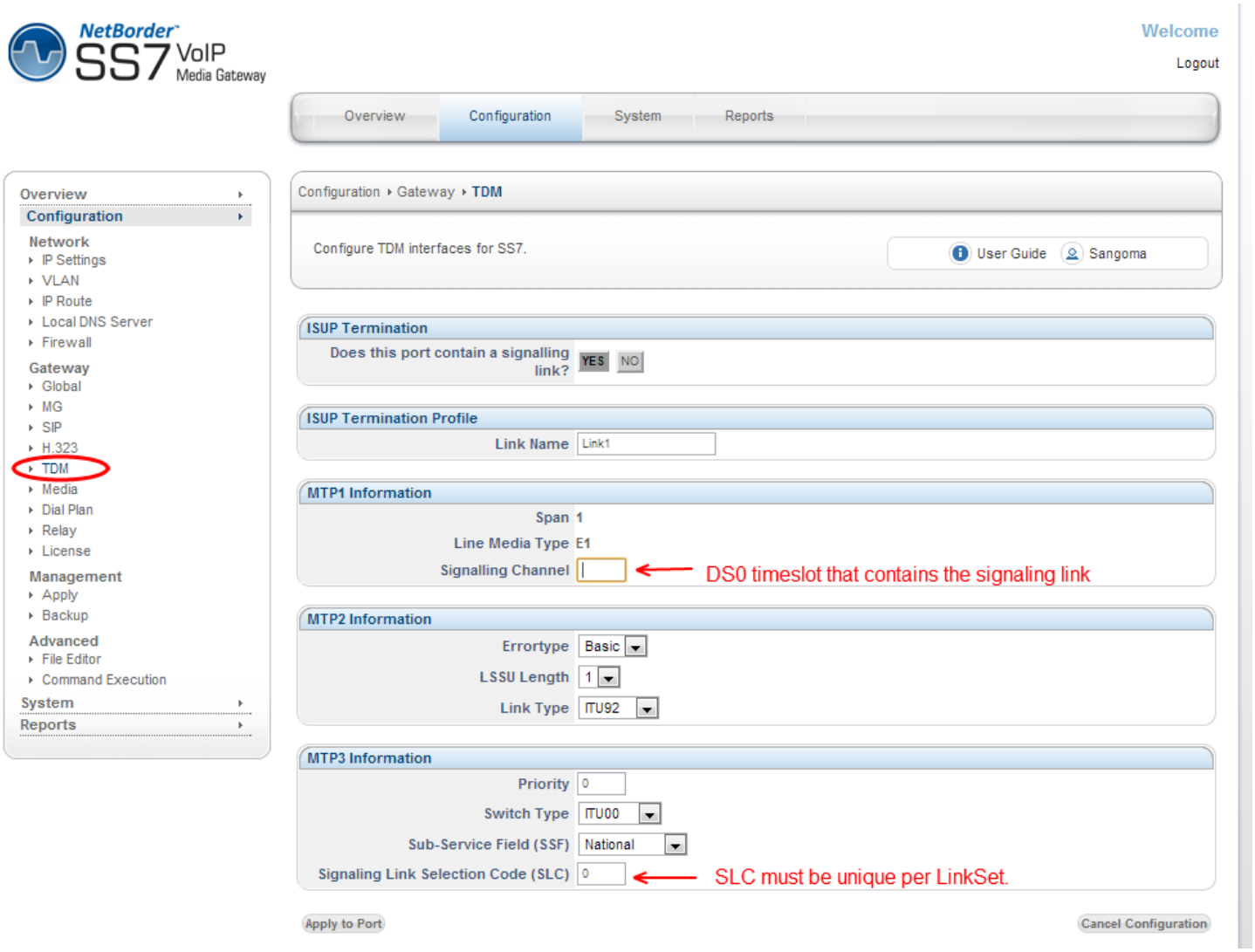
Proceed to configure the SS7 ISUP link that exists on a DS0 timeslot of a T1/E1 port.
The information required for the SS7 Link configuration must be provided by the Telco.

Next screen will confirm if the T1/E1 port contains a signaling link.

- Please select YES if the SS7 signaling link exists on current T1/E1 port.
- By selecting NO this T1/E1 port would not contain a signaling link, but the voice channels would still be controlled by the ISUP signaling. Thus channel mapping would still apply.



The following screen will configure the MTP1 and MTP2 protocol configuration of the SS7 Link.



CAUTION

- The SLC configuration value MUST be unique for each SS7 Link, in case all SS7 Links belong to same Link Set.

Click on Apply to Port button to proceed to next configuration section

Field Name	*Possible Values*	*Default Value*	*Description*
Link Name	Any String	Link1	Name to identify the SS7 Link. By default the GUI will select a unique name. However it is sometimes useful to specify a SS7 Link name that relates to the remote destination.
Span			This is readonly information field. Provides the user with span number information.
Line Media Type			This is readonly information field. Provide the user with T1/E1 link type that has previously been configured.
Signaling Channel	Single Digit 1-31		User must specify the DS0 location of the SS7 signaling channel. The timeslot number relates to physical DS0 channel. Valid options are E1: 1 to 31 T1: 1 to 24 A usual location of a SS7 signaling channel is 1 or 16.
Error Type	Basic PCR	Basic	MTP2 error correction type Two forms of error correction are defined for an SS7 signaling link at MTP2: the basic method and the PCR method. Default: Basic The basic method is generally applied to configurations in which the one-way propagation delay is less than 40 ms, Optional: PCR PCR is applied on intercontinental signaling links in which the one-way propagation delay is greater than 40 ms and on all signaling links established via satellite. The maximum supported signaling link loop (round trip) delay is 670 ms (the time between the sending of a message signal unit [MSU] and the reception of the acknowledgment for this MSU in undisturbed operation).
LSSU Length	1 or 2	1	1- or 2-byte link status signal unit (LSSU) format
Link Type	ITU92 ITU88 ANSI96 ANSI92 ANSI88 ETSI	ITU92	MTP2 protocol supports different variants Outside North America • ITU and ETSI standards are used In North America • ANSI standards are used.

MTP3 Priority	Digit	0	Default traffic priority for this link.
Switch Type	ITU00 ITU97 ITU92 ITU88 ETSI V2 ETSI V3 UK RUSSIA INDIA ANSI92 ANSI95	ITU00	MTP3 protocol supports different variants Outside North America • ITU and ETSI standards are used In North America • ANSI standards are used.
Sub Service Filed (SSF)	National International Spare Reserved	National	Please confirm with your provider which value to use.
Signaling Link Selection Code (SLC)	Digit 0-X	0	SLC can normally be set to 0 by default. Except when there are multiple SS7 Links in a Link Set. In such case SLC must be unique for each SS7 Link. In such case • For each SS7 Link in a LinkSet increment the value of SLC by one.

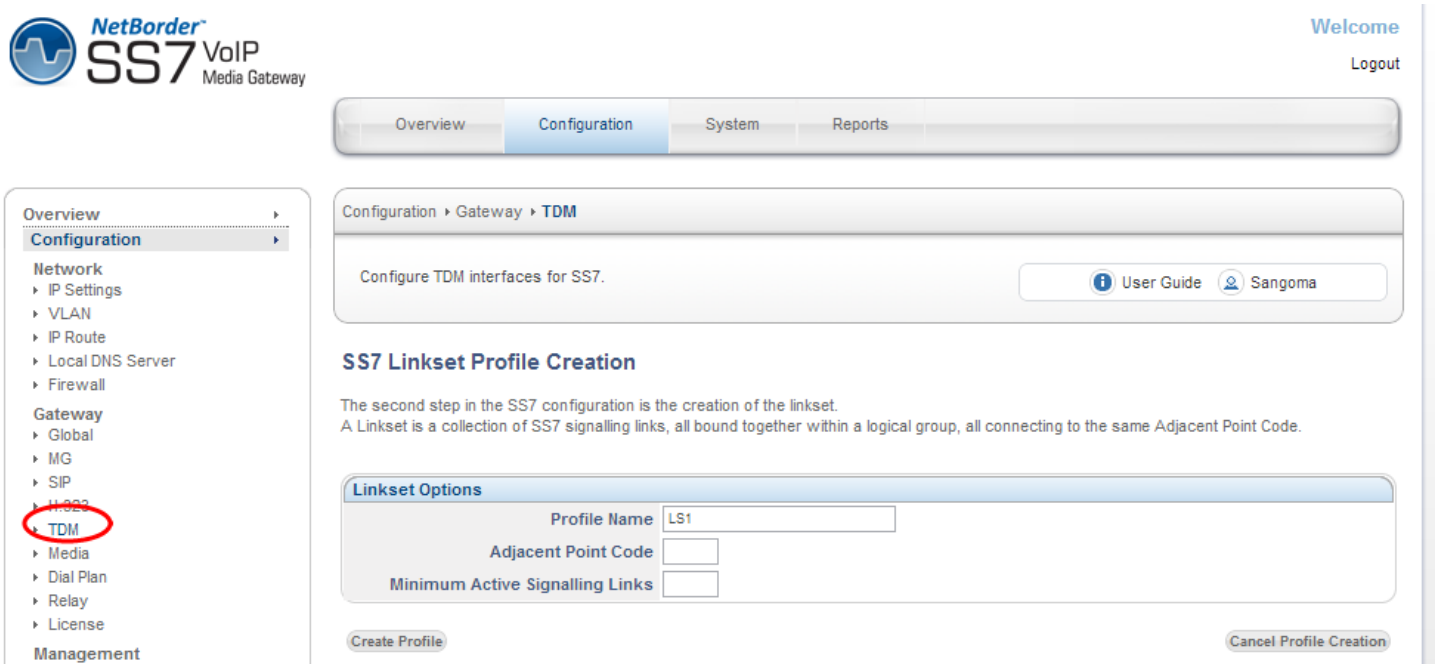
9.7 MTP3 Linkset Configuration

A number of links can be grouped into a linkset that connects to an adjacent point. Each signaling link is provided with a unique code called a signaling link code (SLC). Traffic is load-shared across this linkset. The signaling links within a linkset also provide a redundant transport mechanism. Therefore the more links there are to a linkset the higher the transport bandwidth is and the higher the redundancy.

Linkset configuration on NSG GUI is based on Linkset profiles. It is designed so that multiple SS7 signaling links can use the same SS7 Linkset Profile. The term used when attaching links to linksets in NSG is BIND. You have to bind a link to a linkset in order to proceed.

NOTE

- If no Linkset profile exists, user will be directed to the Linkset profile creation page.
- If Linkset profile already exists, user will be directed to Link profile list page. Where user will be able create a Linkset profile or edit existing Linkset profile.



Click on Create Profile once the configuration is completed.

NOTE

- On very fist Linkset profile, the Link will automatically be BINDED to the Linkset.

Field Name	*Possible Values*	*Default Value*	*Description*
Profile Name	Any String	LS1	Name to identify the SS7 Linkset. By default the GUI will select a unique name. However it is sometimes useful to specify a SS7 Linkset name that relates to the remote destination.
Adjacent Point Code	If ITU integer: 1 to X If ANSI three integers separated by dash		Point-code is an SS7 address for an element in the SS7 network. The Adjacent point is the SS7 equipment which the signaling links terminate on. This equipment will also have a unique point code. This equipment may be either STP equipment or SSP equipment depending on type of interconnect

			If ITU • Single integer number: eg 500 If ANSI • Three integers separated by dash: eg 100-200-400 Please refer to your Telco provider for this information.
Minimum Active Signaling Links	Integer 1-X		A Linkset can contain number of SS7 Links. This field defines how data should be distributed across links in a linkset. For Round Robin – make the value equal number of links in a linkset • This mode will use all links equally. **Recommended** For Active Standby –make the value 1 or less than total number of links. • This mode will use the first link until it gets saturated. • And only use another link if necessary

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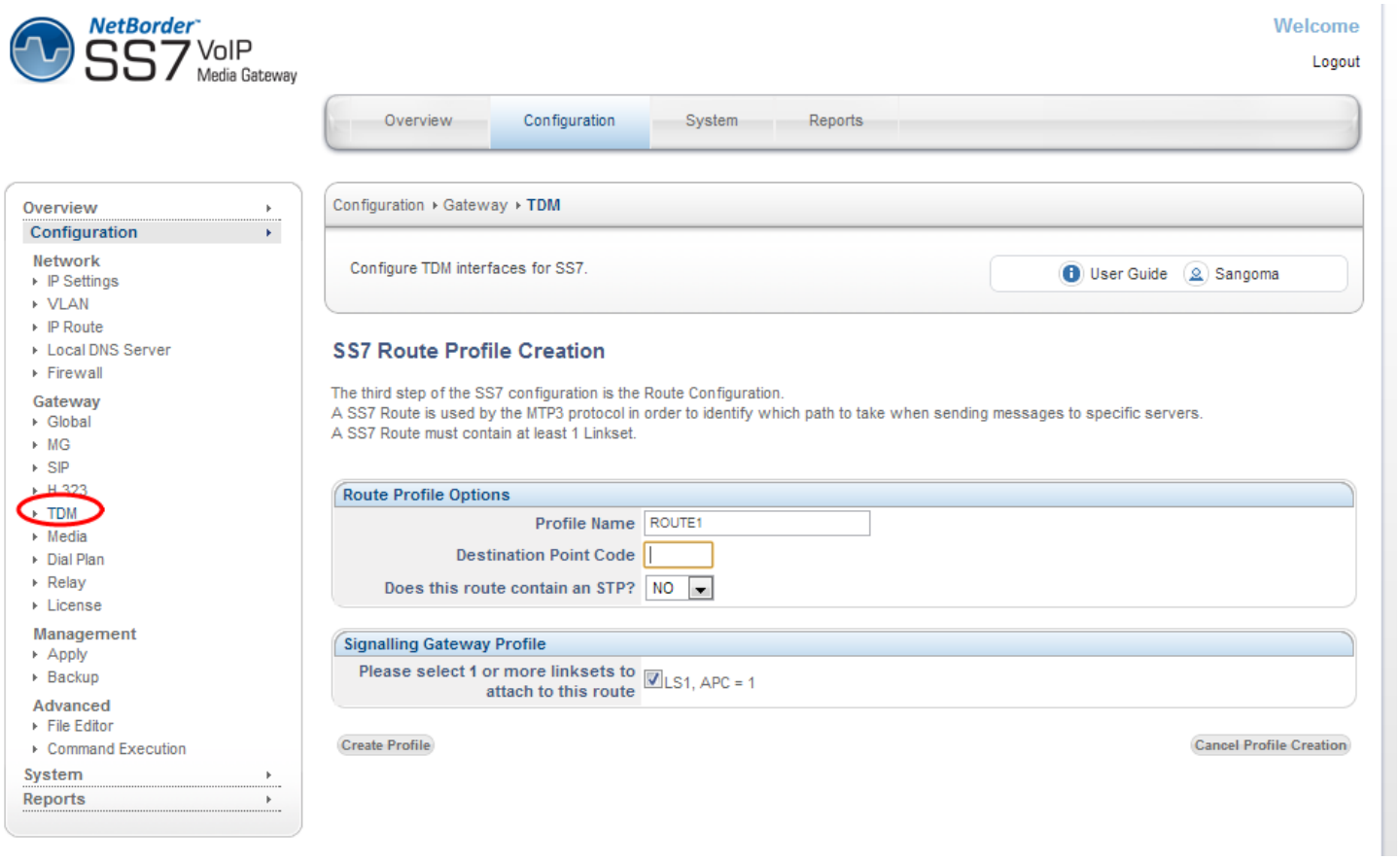
9.8 MTP3 SS7 Route

Route is a collection of linksets to reach a particular destination. A linkset can belong to more than one route. Service Provider personnel statically maintain signaling endpoint routing tables. The routing table identifies the links, linksets, primary routes, and alternate routes for each DPC. All links in the linkset share the traffic load equally.

After a successful Linkset configuration, NSG GUI will present a user with Route Configuration screen.

- If no Route profiles exist, user will be presented with Route create page.
- If a Route profile already exists, user will be presented with Route profile list. Where user will be able to either create new Route or edit existing Route profile.



NOTE

- If a new linkset needs to be attached to a route, the user must edit the route, then add the new linkset to that route.

- The user will only need to edit a route if a new linkset is created on the system. If no new linksets are created, the user will proceed directly to the channel map and CIC map configuration

Field Name	*Possible Values*	*Default Value*	*Description*
Profile Name	Any String	ROUTE1	Name to identify the SS7 Route. By default the GUI will select a unique name. However it is sometimes useful to specify a SS7 Route name that relates to the remote destination.
Destination Point Code	If ITU integer: 1 to X If ANSI three integers separated by dash		Point-code is an SS7 address for an element in the SS7 network. The Destination Point of the SS7 network defines the switching equipment within the PSTN network which terminates the TDM interfaces of this interconnect. This point is also allocated a unique point-code within the SS7 network. If the adjacent point is a SSP or MSC interconnect the destination point will be the same as the adjacent point. Eg: A-Link = APC differs from DPC F-Link = APC is equal to DPC If ITU (outside North America) • Single integer number: eg 500 • Default link type – F link If ANSI (North America) • Three integers separated by dash: eg 100-200-400 • Default link type – A link Please refer to your Telco provider for this information.
Does route contain STP?	Yes or No	No	
Signaling Gateway Profile List			List of existing Linksets that can be bound to a Route profile. There has to be at least a single Linkset bound to a route. In theory there can be a number of Linkset profiles bound a a single route.

9.9 ISUP Interface Configuration

ISUP connects, manages, and disconnects all voice and data calls in the PSTN. ISUP sets up and tears down the circuits used to connect PSTN voice and data subscribers.. ISUP is used in cellular or mobile networks for trunking connections.

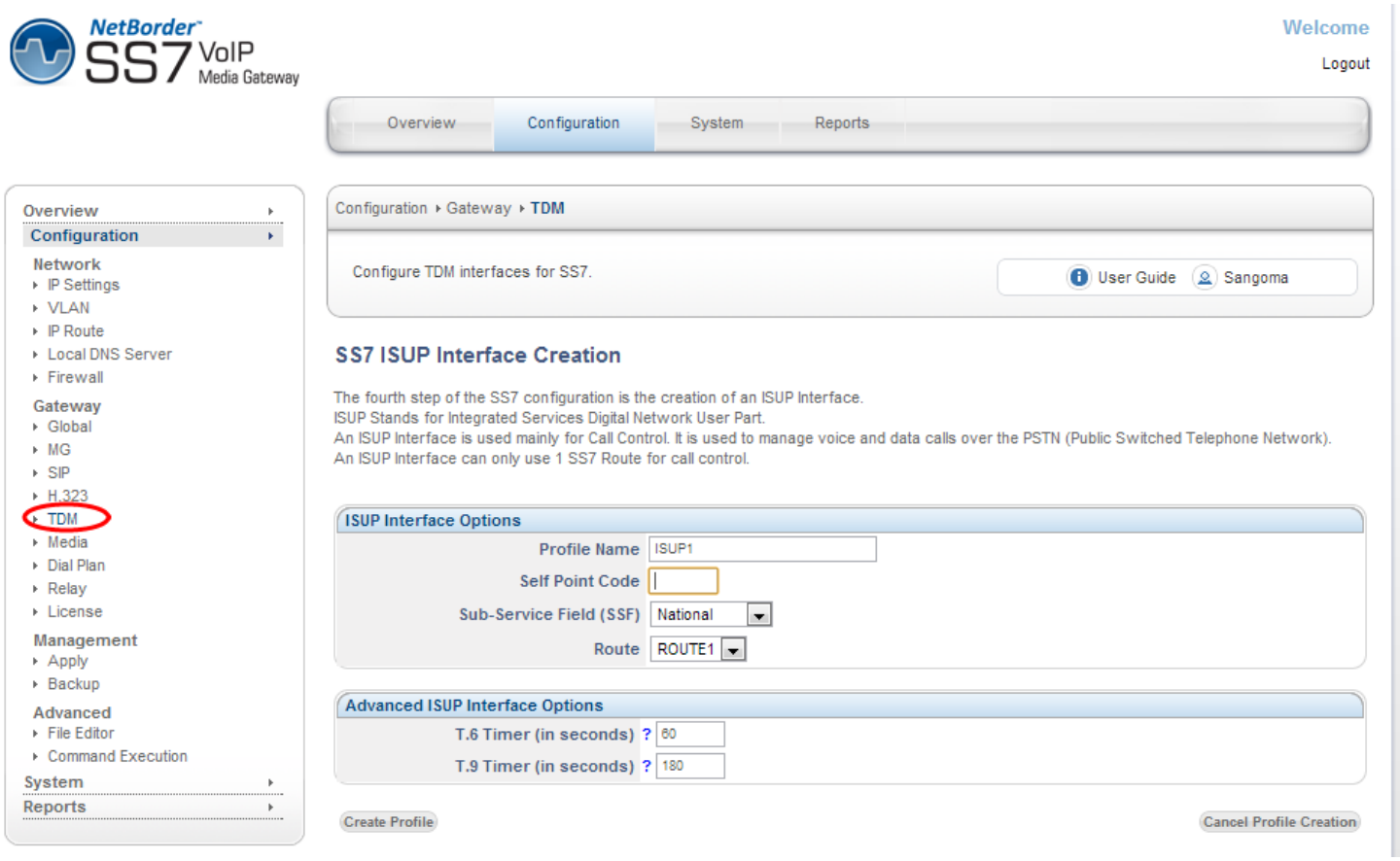
ISUP information is transferred in MTP3 messages similar to the other L4 protocols. The ISUP section covers the following topics:

- ISUP ServicesBasic and Supplementary
- End-to-end SignalingPass-along and SCCP
- Call Setup and Teardown
- ISUP Message Format
- ISUP Call Control Messages

Like the linkset configuration and route configuration profiles, the ISUP Interface configuration is also configured as profiles. It is setup so that 1 SS7 route can be attached to 1 ISUP Interface.

After a successful Route configuration, NSG GUI will present a user with Route Configuration screen.

- If no ISUP profiles exist, user will be presented with ISUP create page.
- If an ISUP profile already exists, user will be presented with ISUP profile list. Where user will be able to either create new ISUP Interface Profile or edit existing ISUP Interface profile.



Field Name	*Possible Values*	*Default Value*	*Description*
Profile Name	Any String	ISUP1	Name to identify the SS7 ISUP Interface profile. By default the GUI will select a unique name. However it is sometimes useful to specify a SS7 ISUP Interface name that relates to the remote destination.
Self Point Code	If ITU integer: 1 to X If ANSI three integers separated by dash		Point-code is an SS7 address for an element in the SS7 network. The Self Point Code /Originating Point describes the equipment that is interconnecting into the SS7 network. The originating point will be provided with a unique point-code by the network provider allowing for identification of this point with in the SS7 network.

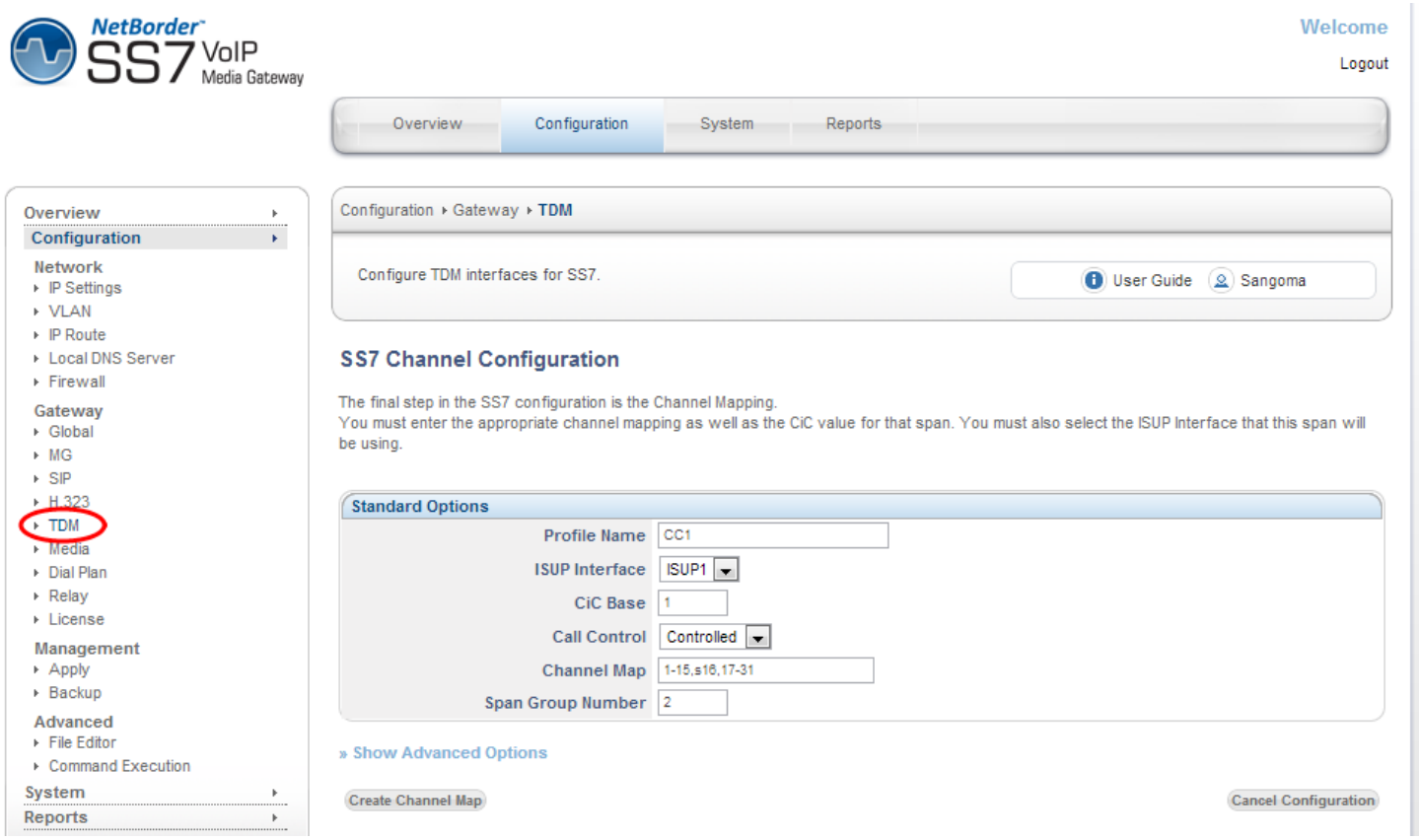
			Sefl Point Code is the address of the NSG SS7 Gateway in the SS7 network. If ITU (outside North America) • Single integer number: eg 500 If ANSI (North America) • Three integers separated by dash: eg 100-200-400 Please refer to your Telco provider for this information.
Sub Service Field SSF	National International Spare Reserved	National	Please confirm with your provider which value to use.
Route			List of existing Route profiles that can be bound to a Route profile. There has to be a single Route bound to an ISUP Interface profile.

ISUP Timer	*Spec Value (s) (ITU Q.764)*	*Default Value (s)*	*Timer Name*	*XML Tag for Manual Control*
T1	15-60	15	isup.t1	isup_interface
T2	180	180	isup.t2	isup_interface
T3	120	120	isup.t3	isup_interface
T4	300-900	300	isup.t4	isup_interface
T5	300-900	300	isup.t5	isup_interface
T6	60-120	60	isup.t6	isup_interface
T7	20-30	20	isup.t7	isup_interface
T8	10-15	10	isup.t8	isup_interface
T9	90-180	180	isup.t9	isup_interface
T10	4-6	4	isup.t10	isup_interface
T12	15-60 (ITU) 4-15(ANSI)	150	isup.t12	isup_interface
T13	300-900	300	isup.t13	isup_interface
T14	15-60	15	isup.t14	isup_interface
T15	300-900	300	isup.t15	isup_interface
T16	15-60	15	isup.t16	isup_interface
T17	300-900	300	isup.t17	isup_interface
T27	240	240	isup.t27	isup_interface
T31	360	360	isup.t31	isup_interface

T33	12-15	12	isup.t33	isup_interface
T34	2-4	4	isup.t34	isup_interface
T35	15-20	15	isup.t35	isup_interface
T36	10-15	12	isup.t36	isup_interface

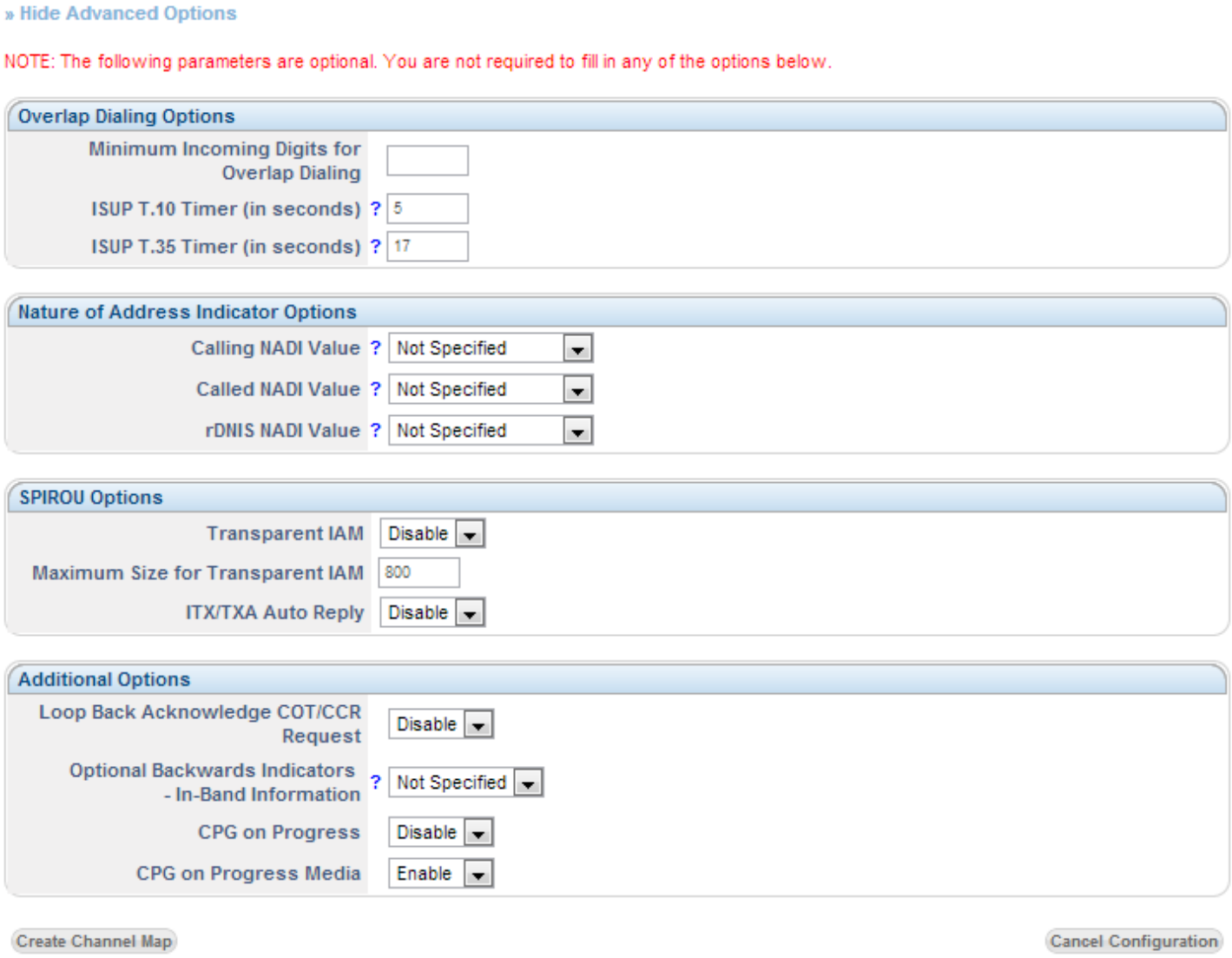
9.10 ISUP CIC Channel Mapping

The last step of the configuration is to bind the TDM voice channels to ISUP Profile and map ISUP CIC's to the TDM timeslots.



Field Name	*Possible Values*	*Default Value*	*Description*
Profile Name	Any String	CC1	Name to identify the SS7 Call Control profile. By default the GUI will select a unique name. However it is sometimes useful to specify a SS7 ISUP Interface name that relates to the remote destination.
ISUP Interface	List of existing ISUP Interface profiles	Current Profile	ISUP Interface points to the list of currently defined ISUP Interface profiles. Each ISUP profile defines its own Self-Point-Code/Origination Code. With multiple ISUP profiles, one can configure a system with multiple Self-Point-Codes. Selected ISUP Interface Profile will be used to control the physical TDM T1/E1 DS0 channels.
CIC Base	Integer 1 to Any	1	Start of the ISUP CIC numbers. ISUP CIC numbers are logical representations of the physical DS0 channels. The mapping between CIC and DS0 channels is one to one. This information is provided by the Telco. **CAUTION** • Improper mapping between CIC and Physical T1/E1 DS0 can result in one way or no audio. Even though the call completes successfully on SS7 signaling.
Call Control	Controlled Controlling Bothway Incoming Outgoing	Controlled	Refer to Telco information.
Channel Map			List of channels to be controlled by ISUP Interface Example: 1-15,s16,17-31 Channels 1-15 and 17-31 are used for Voice and should be controlled by ISUP Interface Channel 16 (prefixed by letters) indicates that channel 16 carries signaling channel. ISUP Interface will ignore this channel as it's not voice.

			Prefix Letters to signaling channel: **s**: ISUP CIC id not used, id mapped to signaling channel **g**: ISUP CIC id is used, id mapped to next available voice channel. The bind between ISUP and TDM would be as follows **Channel Map: 1—31 (no signaling channel)** CIC 1: channel 1 CIC 2: channel 2 … CIC 16: channel 16 … CIC 30: channel 30 CIC 31; channel 31 **Channel Map: 1-15,s16,17-31 (signaling on ch 16)** CIC 1: channel 1 CIC 2: channel 2 … CIC 15: channel 15 … CIC 16: not used – A16 points to signaling channel 16 CIC 17: channel 17 CIC 18: channel 18 … CC 31: channel 31 **Channel Map: 1-15,g16,17-31 (signaling on ch 16)** CIC 1: channel 1 CIC 2: channel 2 CIC 15: channel 15 CIC 16: channel 17 - A16 is used and it points to ch 17. CIC 17: channel 18 … CIC 30: channel 31
Span Group Number	Integer	1	Default group number used to dial out over a trunk group. Usually the group number will correspond to the trunk group.



Field Name	*Possible Values*	*Default Value*	*Description*
Minimum Incoming Overlap Dialing	Integer		Enables overlap dialing in ISUP.
ISUP Interface	List of existing ISUP Interface profiles	Current Profile	ISUP Interface points to the list of currently defined ISUP Interface profiles.

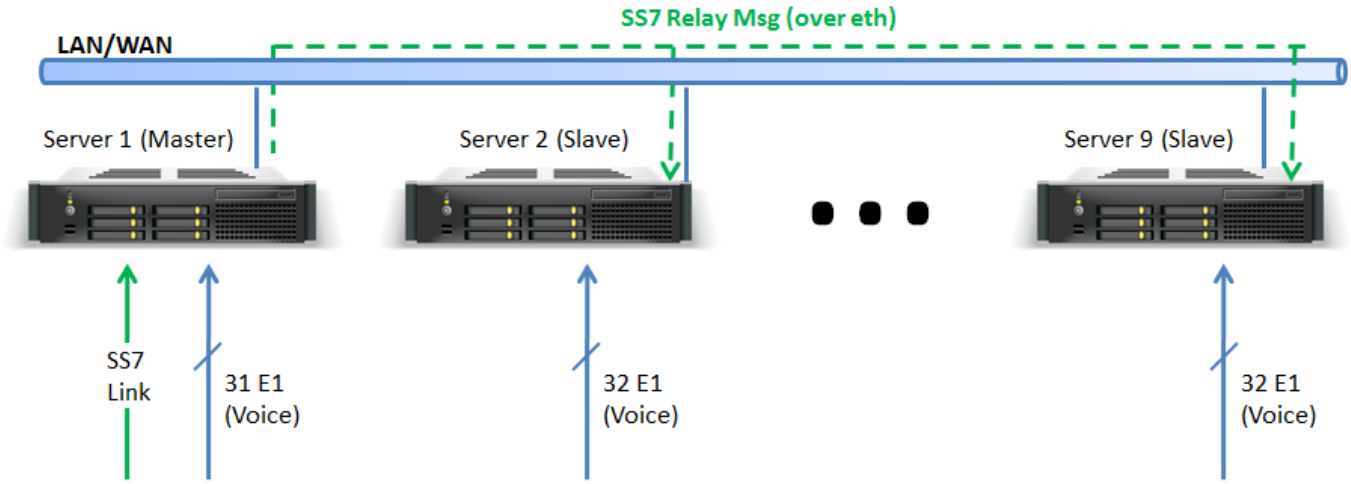
			Each ISUP profile defines its own Self-Point-Code/Origination Code. With multiple ISUP profiles, one can configure a system with multiple Self-Point-Codes. Selected ISUP Interface Profile will be used to control the physical TDM T1/E1 DS0 channels.
CIC Base	Integer 1 to Any	1	Start of the ISUP CIC numbers. ISUP CIC numbers are logical representations of the physical DS0 channels. The mapping between CIC and DS0 channels is one to one. This information is provided by the Telco. **CAUTION** • Improper mapping between CIC and Physical T1/E1 DS0 can result in one way or no audio. Even though the call completes successfully on SS7 signaling.
Call Control	Controlled Controlling Bothway Incoming Outgoing	Controlled	Refer to Telco information.
Channel Map			List of channels to be controlled by ISUP Interface Example: 1-15,s16,17-31 Channels 1-15 and 17-31 are used for Voice and should be controlled by ISUP Interface Channel 16 (prefixed by letters) indicates that channel 16 carries signaling channel. ISUP Interface will ignore this channel as it's not voice. Prefix Letters to signaling channel: **s**: ISUP CIC id not used, id mapped to signaling channel **g**: ISUP CIC id is used, id mapped to next available voice channel. The bind between ISUP and TDM would be as follows **Channel Map: 1—31 (no signaling channel)** CIC 1: channel 1 CIC 2: channel 2 … CIC 16: channel 16 … CIC 30: channel 30 CIC 31; channel 31

			Channel Map: 1-15,s16,17-31 (signaling on ch 16) CIC 1: channel 1 CIC 2: channel 2 … CIC 15: channel 15 … CIC 16: not used – A16 points to signaling channel 16 CIC 17: channel 17 CIC 18: channel 18 … CC 31: channel 31 **Channel Map: 1-15,g16,17-31 (signaling on ch 16)** CIC 1: channel 1 CIC 2: channel 2 CIC 15: channel 15 CIC 16: channel 17 - A16 is used and it points to ch 17. CIC 17: channel 18 … CIC 30: channel 31
Span Group Number	Integer	1	Default group number used to dial out over a trunk group. Usually the group number will correspond to the trunk group.

10 Relay: SS7

NSG SS7 relay enables a single NSG gateway (master) to control multiple NSG gateways (slaves) with as little as 1 signaling link connected to the master.

You can have up to 8 slave machines that are controlled by a single master gateway. Signaling messages (MTP2 traffic) are passed over the IP network to the slave machines.



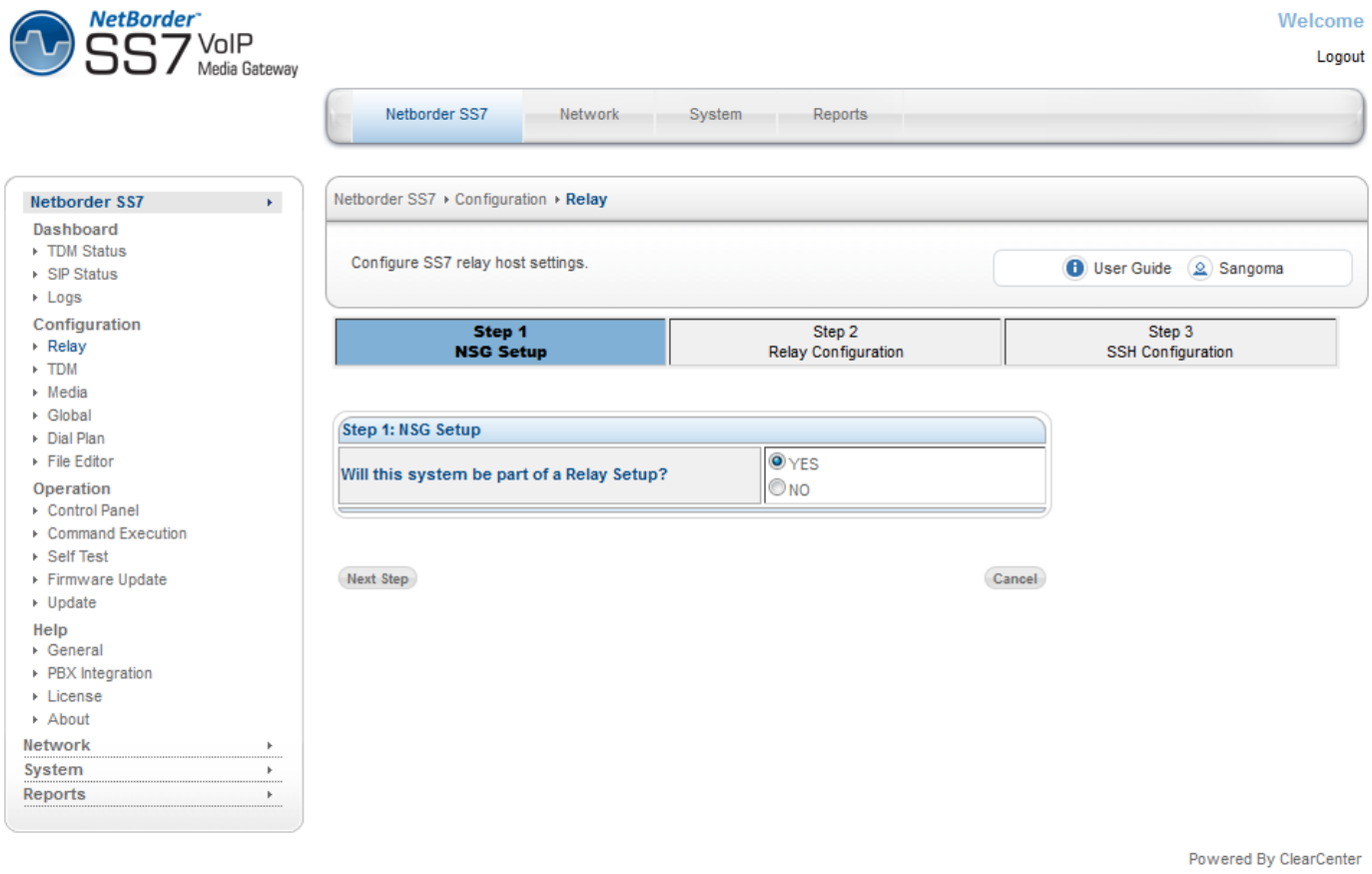
Having to configure up to 8 machines individually would be a tedious task from an operations perspective. In order to simplify the configuration process of this distributed system, the relay option enables the Master gateway to configure all the slaves machine from its web UI and pushing the configurations to the slave gateways over SSH.

This following section will guide you through the configuration of the Relay mode to enable remote control of the Slave gateways.

10.1 Relay Configuration

To access the Relay: SS7 configuration section

1. Select **Relay** from side/top **Configuration** Menu

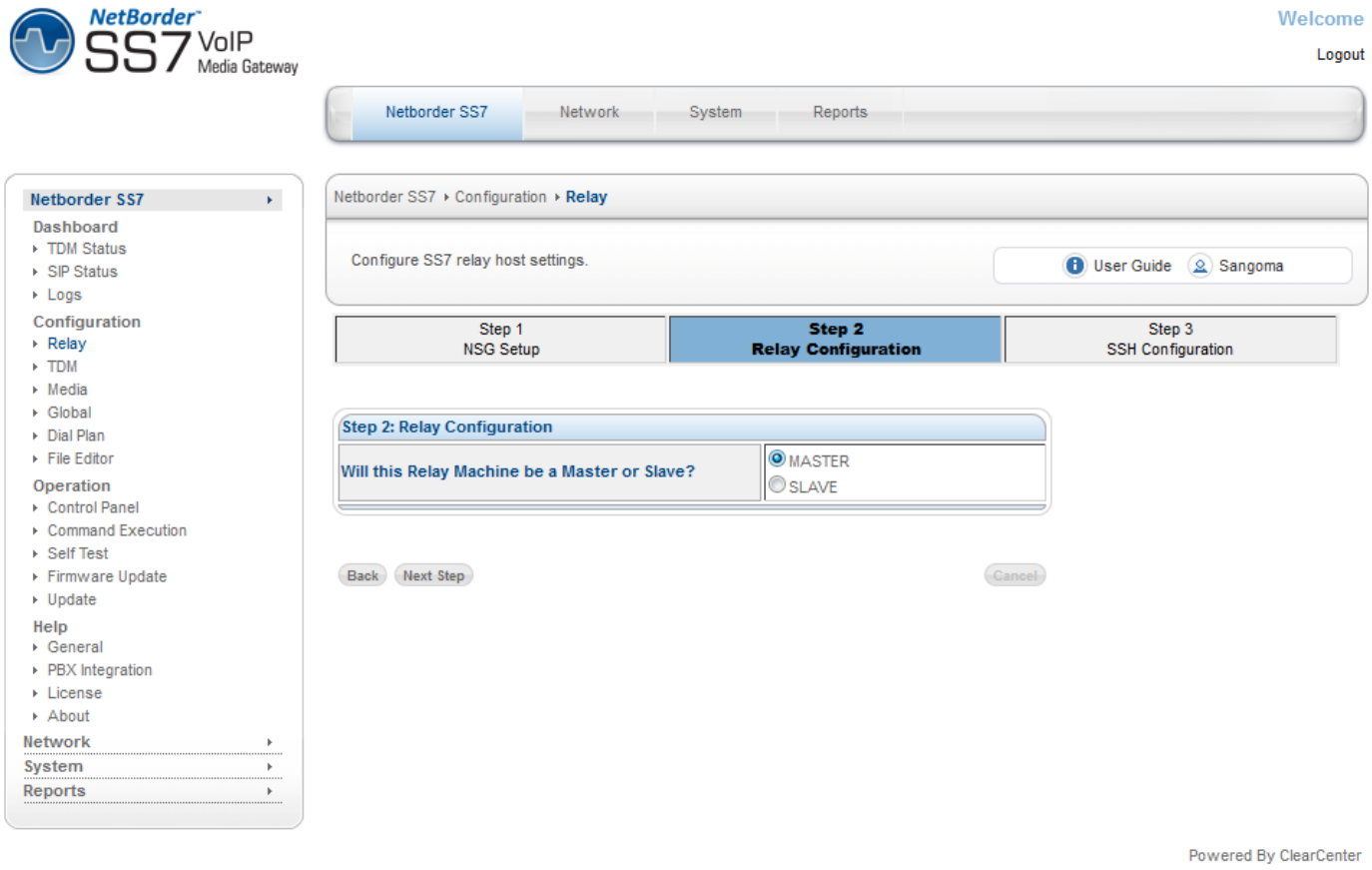


- Select **NO** if you do not want to enable Relay mode in your installation and proceed to the next section to resume SS7 configuration.

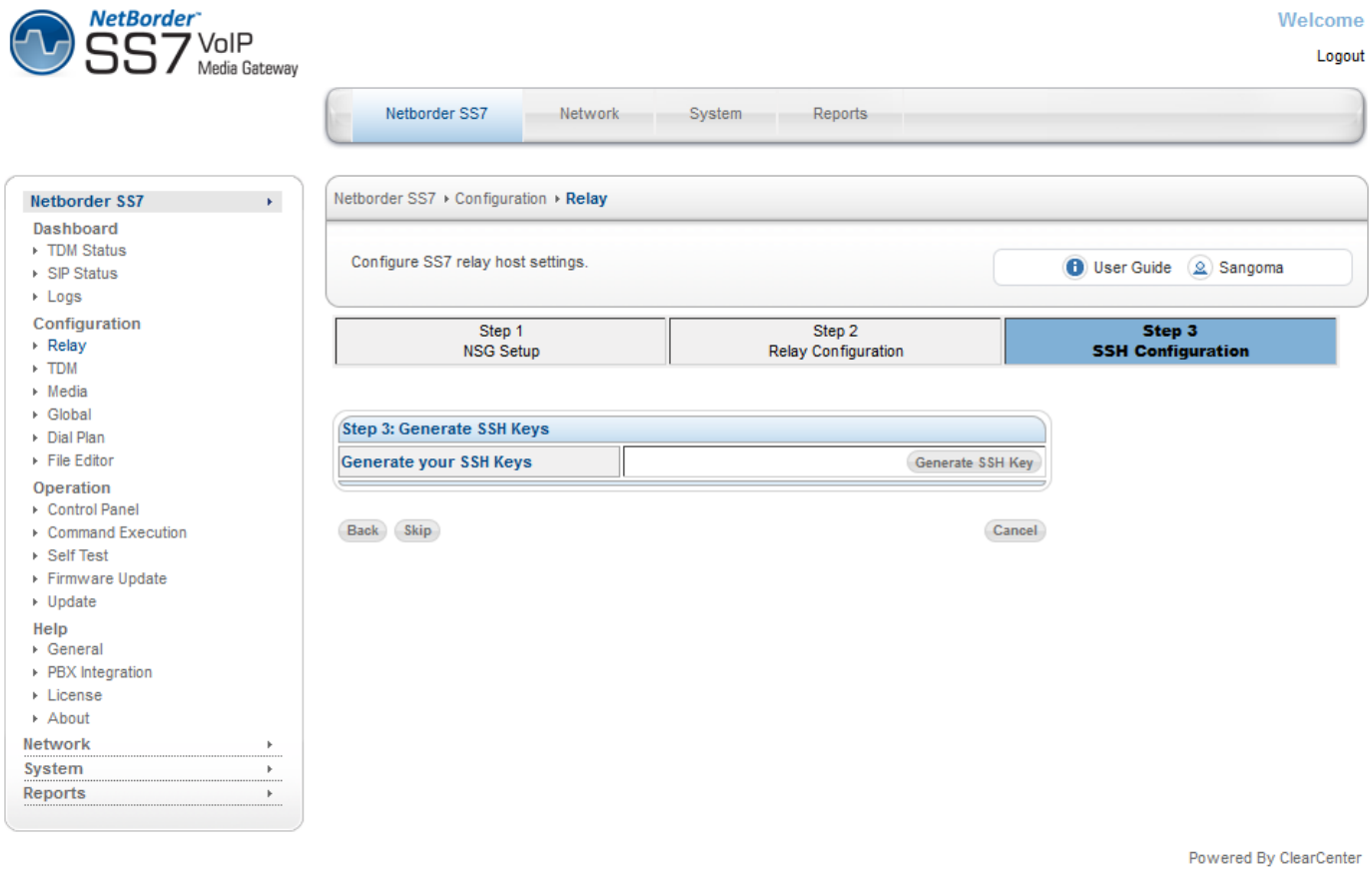
- Select **YES** to activate the relay Mode

10.1.1 Configuring the master gateway

We will start by configuring the master machine first.

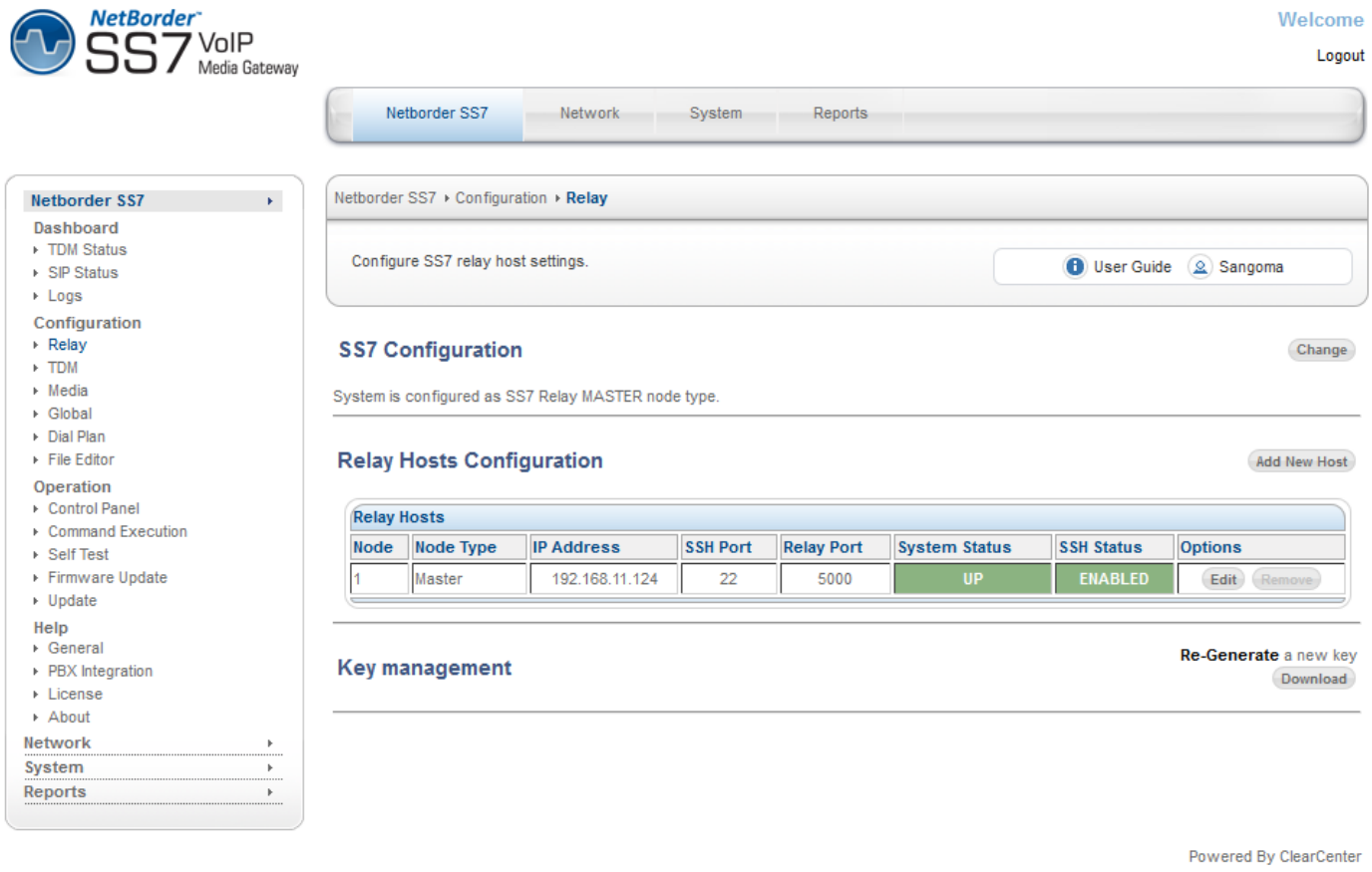


Select the Master option in step 2 and click "Next Step" to continue.



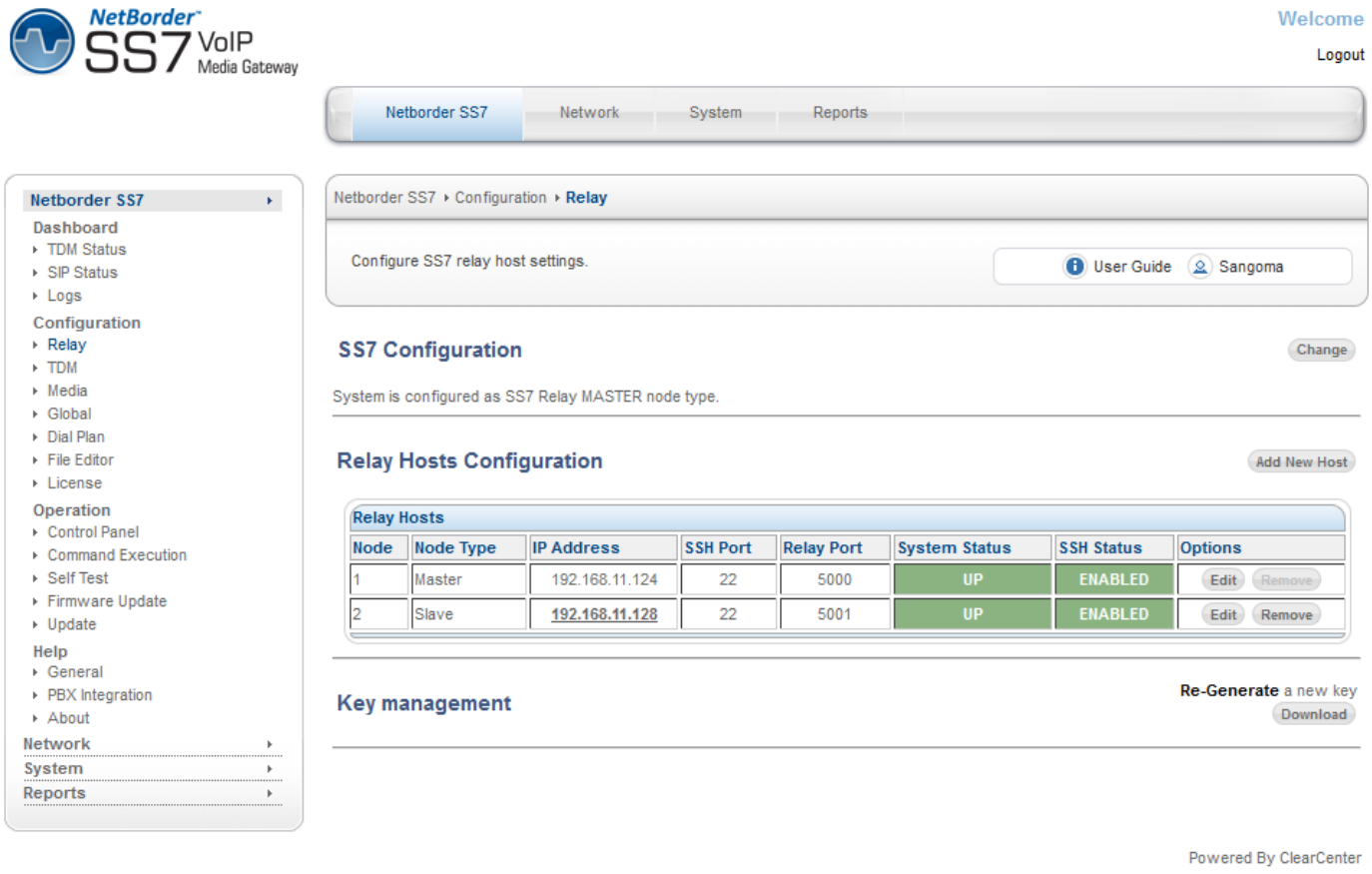
In Step 3, you will generate an SSH key and download the public key that will be uploaded to all the slave gateways. This key will enable a secure SSH connection between the master and the slave machines to push the configurations.

The Relay Master will listen for incoming relay traffic on port 5000.



Once the SSH key has been generated you will need to click on the "Add New Host" button to add 1 or more slave gateways to the relay configuration.

The listening relay port for all subsequent slave instances will increase by 1 port. Slave on node 2 will listen on port 5001, Slave on node 3 will listen on port 5002, etc...

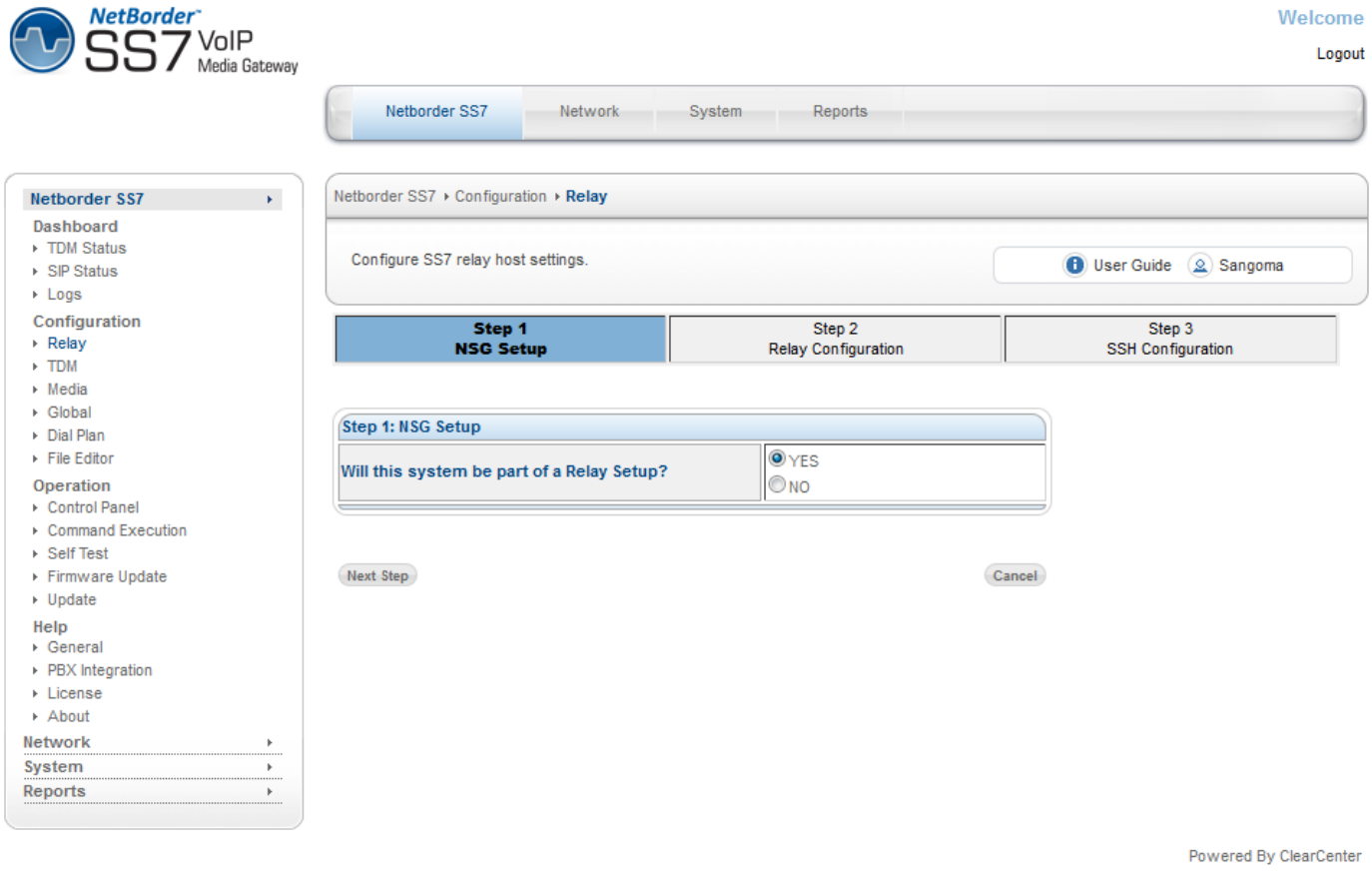


Once you have configured all your slave hosts, you can now configure your slave machine(s)

10.1.2 Configuring the slave gateway

To access the Relay: SS7 configuration section

1. Select **Relay** from side/top **Configuration** Menu



Select **YES** in step 1 to enable Relay mode.

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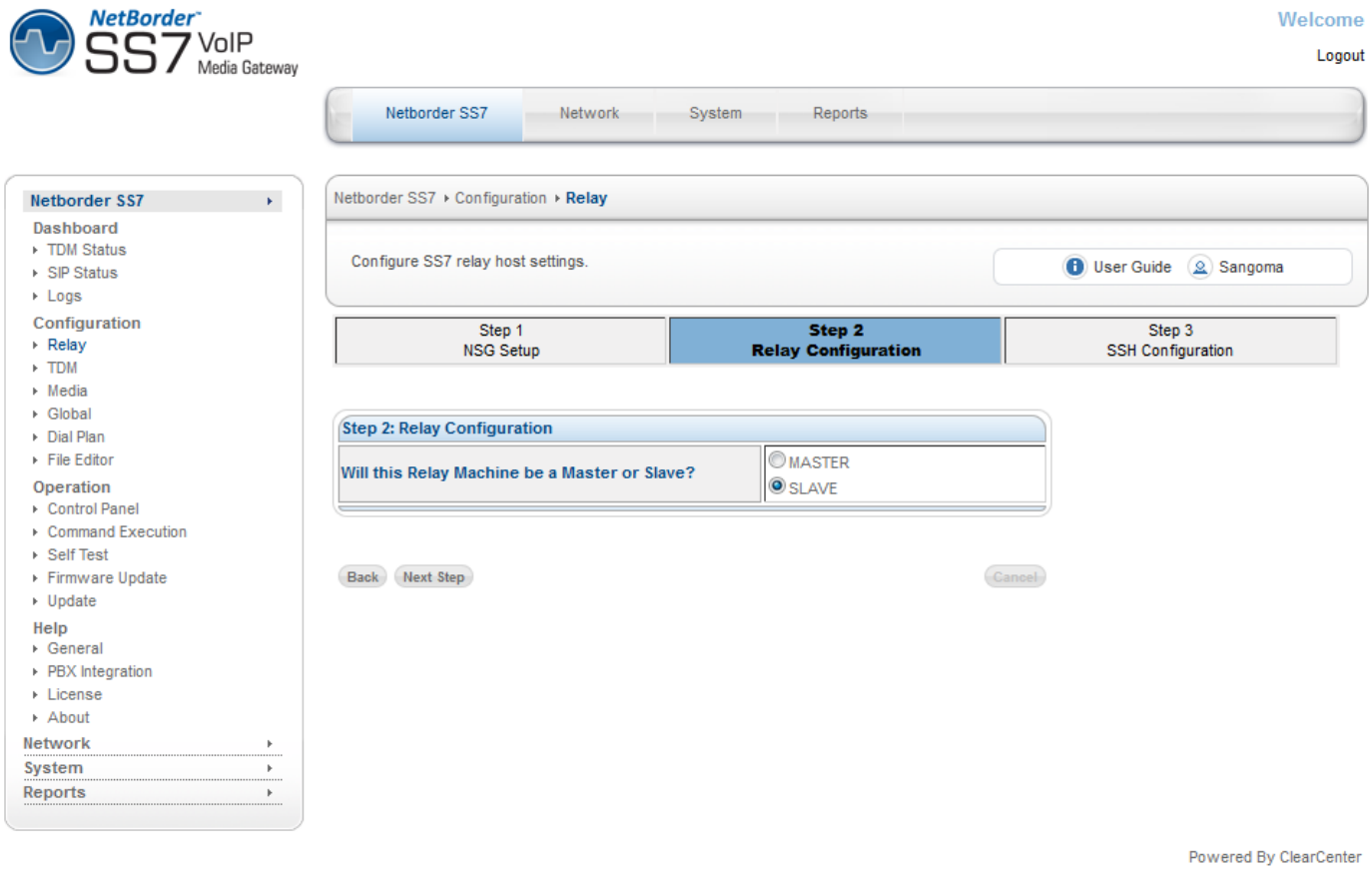
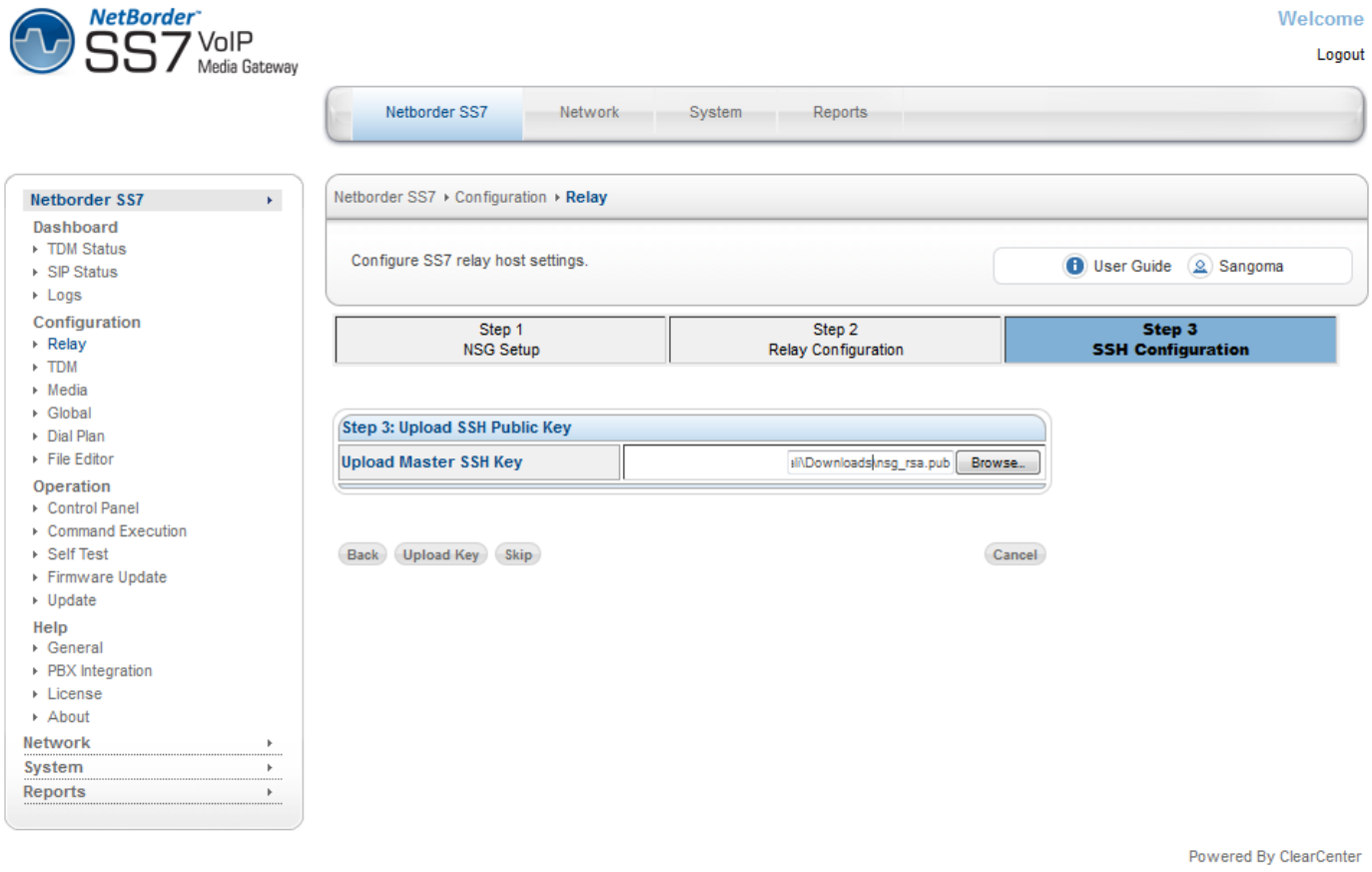
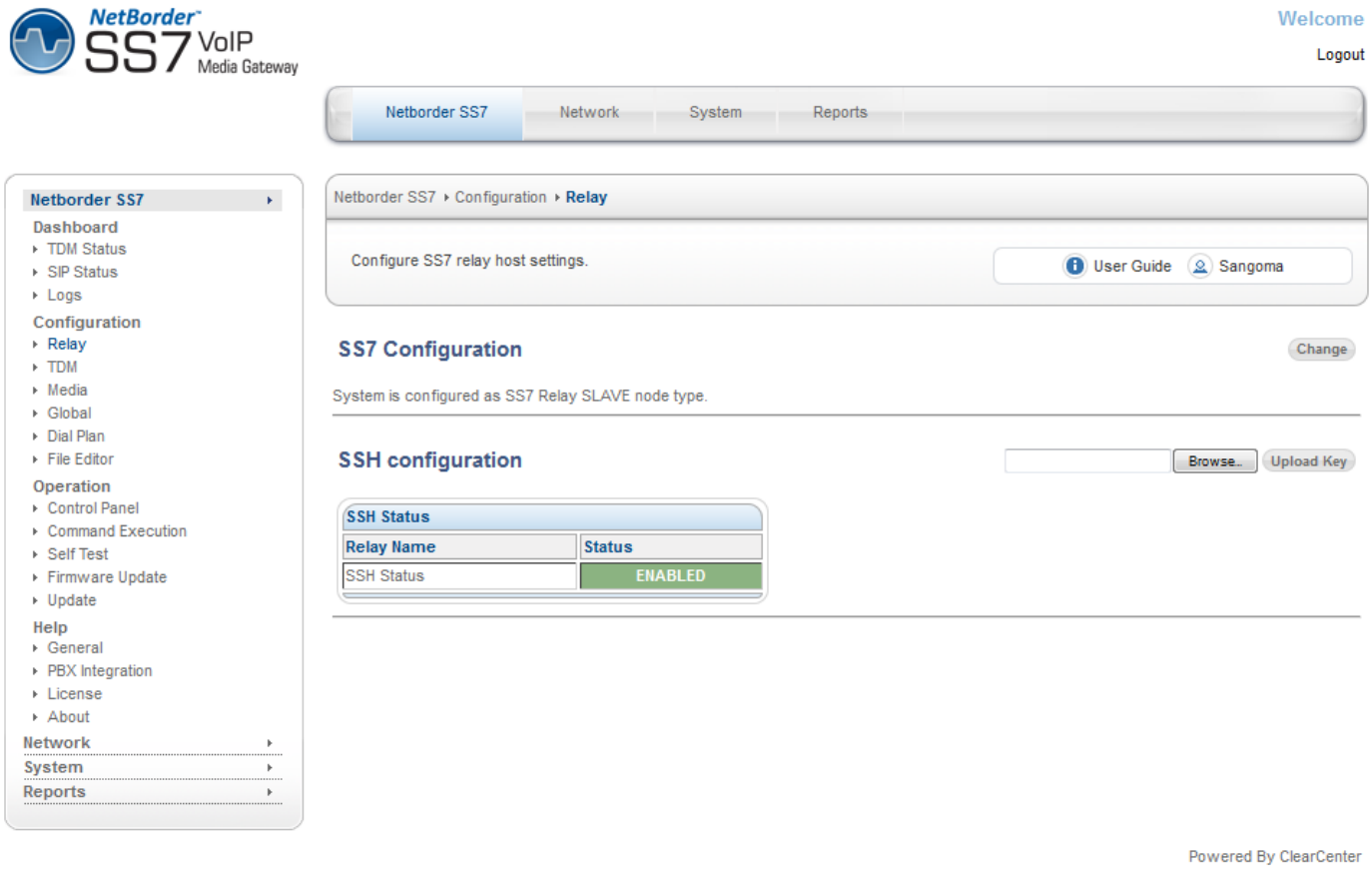
sangoma.com



Select the **SLAVE** option in step 2 and click "Next Step" to continue.



Upload the public key that you downloaded and saved when you configured the master gateway earlier.



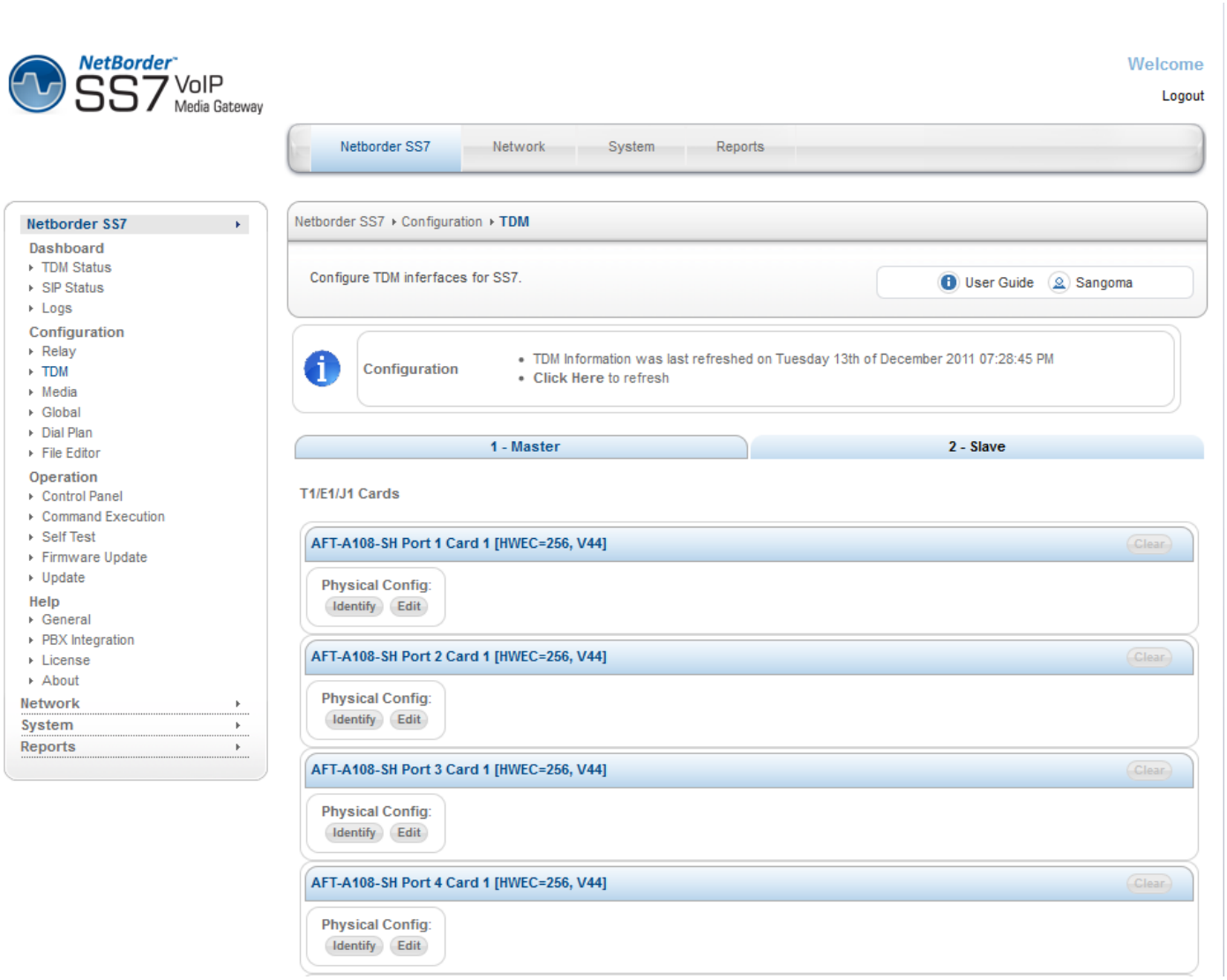
Once the key has been uploaded, the SSH link will have been enabled.

Repeat these steps for all the slave machines and return to the master WebUI when you are finished.

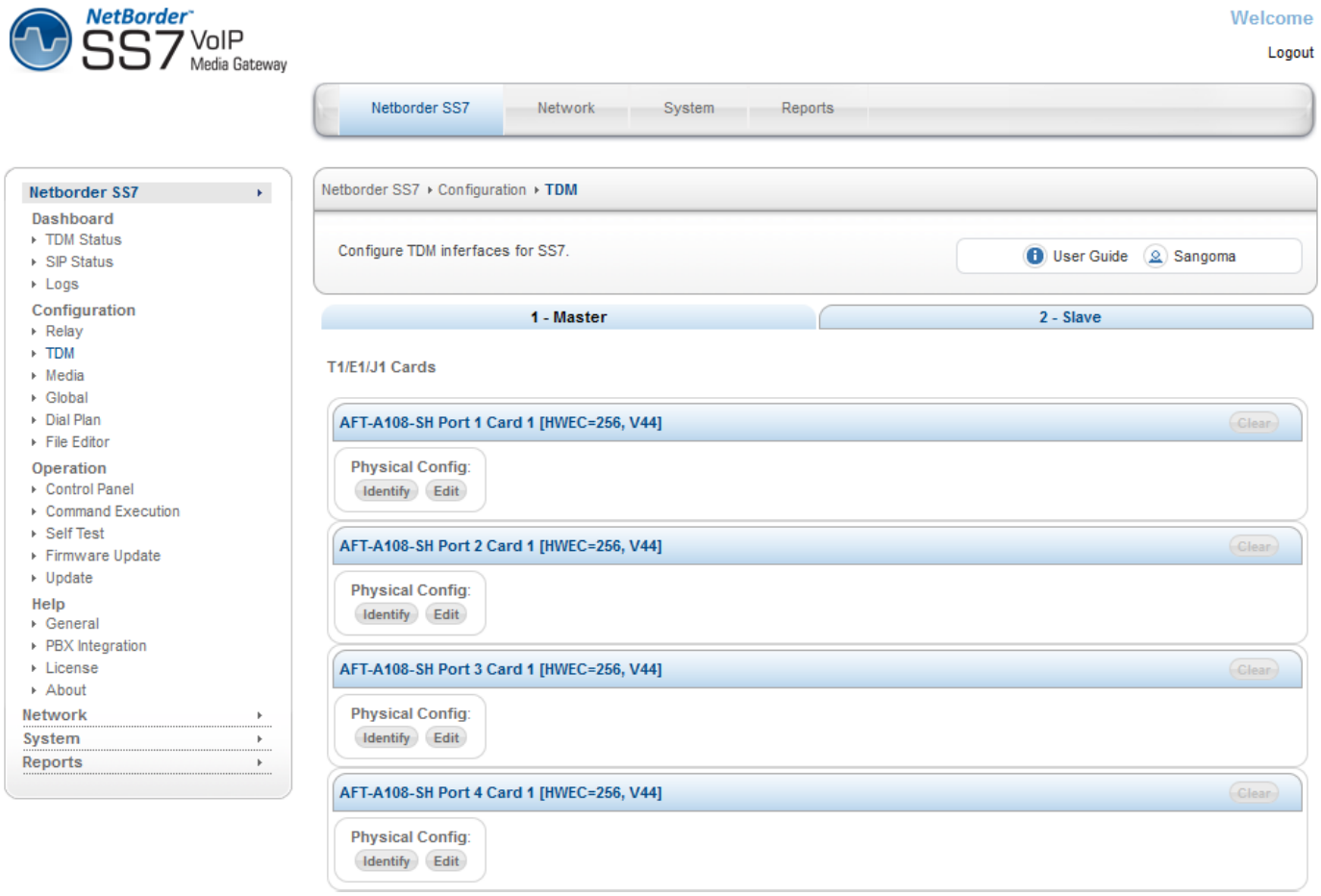
10.1.3 Configuring the slave TDM configurations from the master gateway

Open the master WebUI in your browser.

1. Select **TDM** from side/top Configuration Menu



The TDM configuration is presented in a tabbed pane, each tab represents a machine to configure. Select the **Slave** tab to configure the slave gateway.



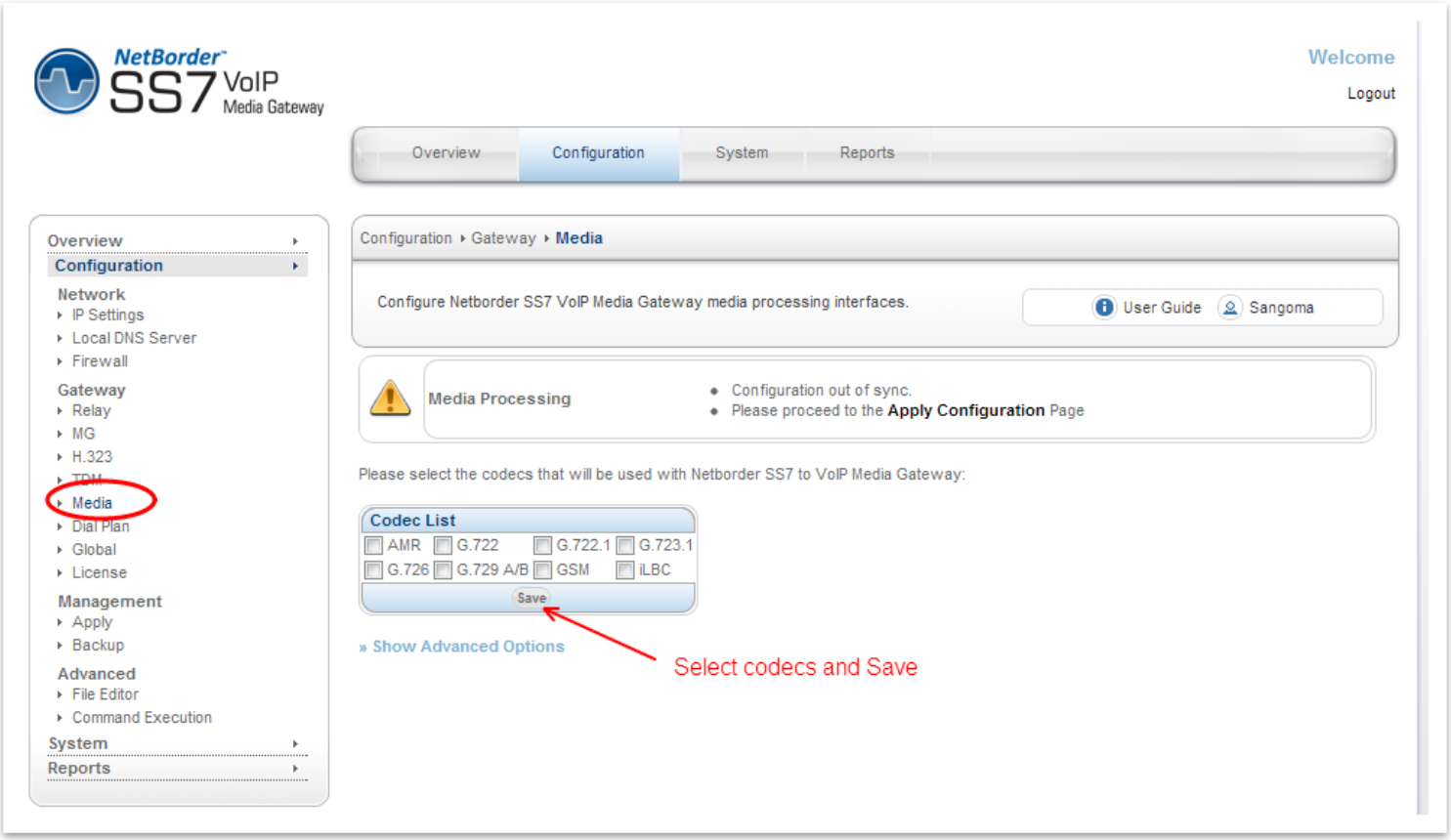
Once you have completed configuring the master and slave(s) TDM configurations, you will click on the "Generate config" button that will push the configuration to each slave over a secure SSH connection. All this is done from the convenience of the master server's WebUIgateway's web gui, removing the need to log on to each slave server's WebUIgateway's individually.

11 Media Transcoding Configuration

NSG will enable ALL Media Codec's by default. There is no extra configuration needed.
Use this configuration page in case you want to limit which codecs should be enabled, or disable media codec support.

To access NSG Media Transcoding Configuration

- Select Media from side/top Configuration Menu
- Select any or all supported/listed codecs
- Once done press **Save**

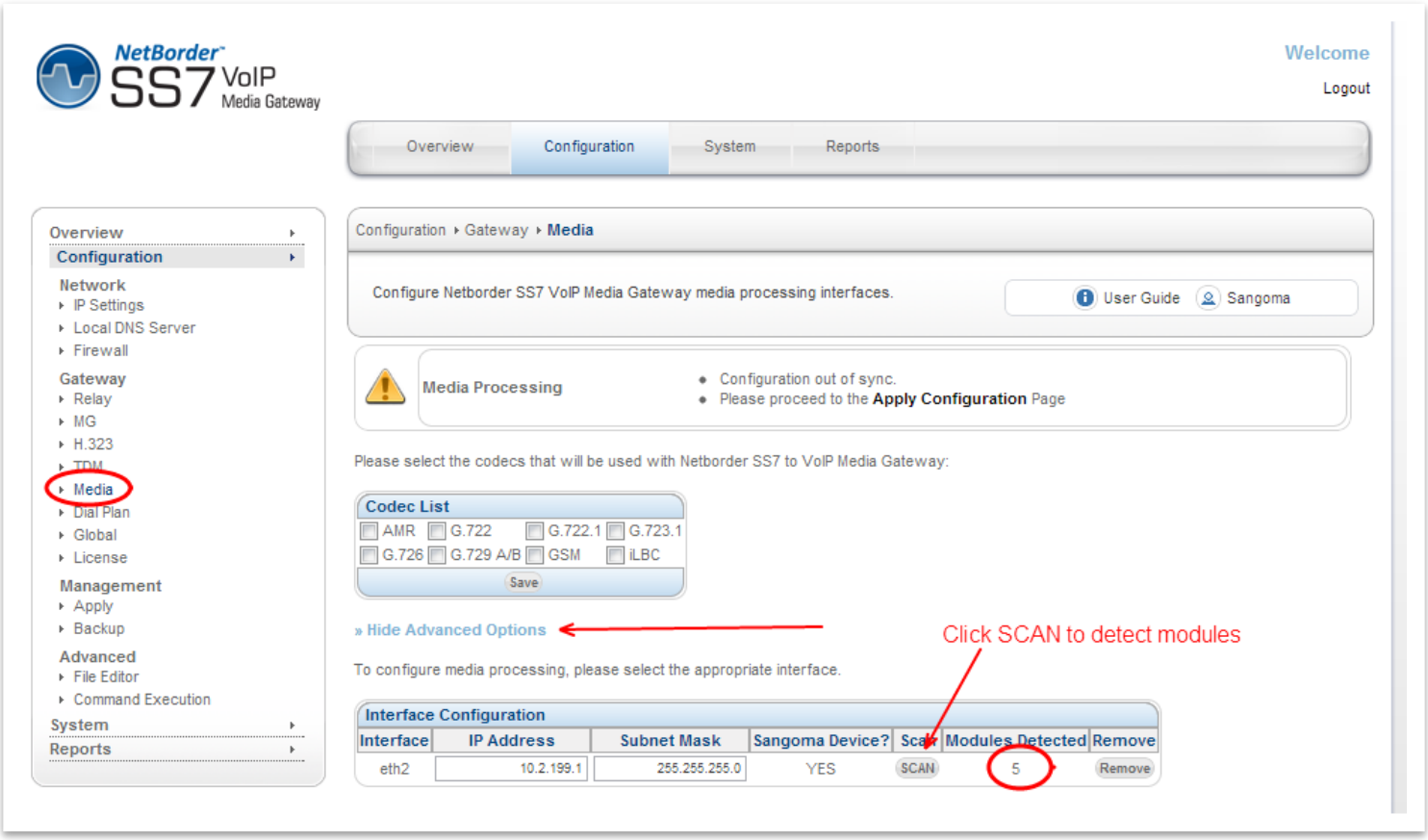


NOTE
At this point the codec selection is over. One can proceed to Media hardware discovery in the Advanced Options of the Media page.

11.1 Media Hardware

Once Codec selection has been made, proceed to Advanced Options section of the Media page.

- Select SCAN
  - This step will auto-detect all NSG transcoding resources
- Confirm that GUI detected exact number of transcoding resources as installed.
- User has an option of changing the assigned Local IP address of the Media device.



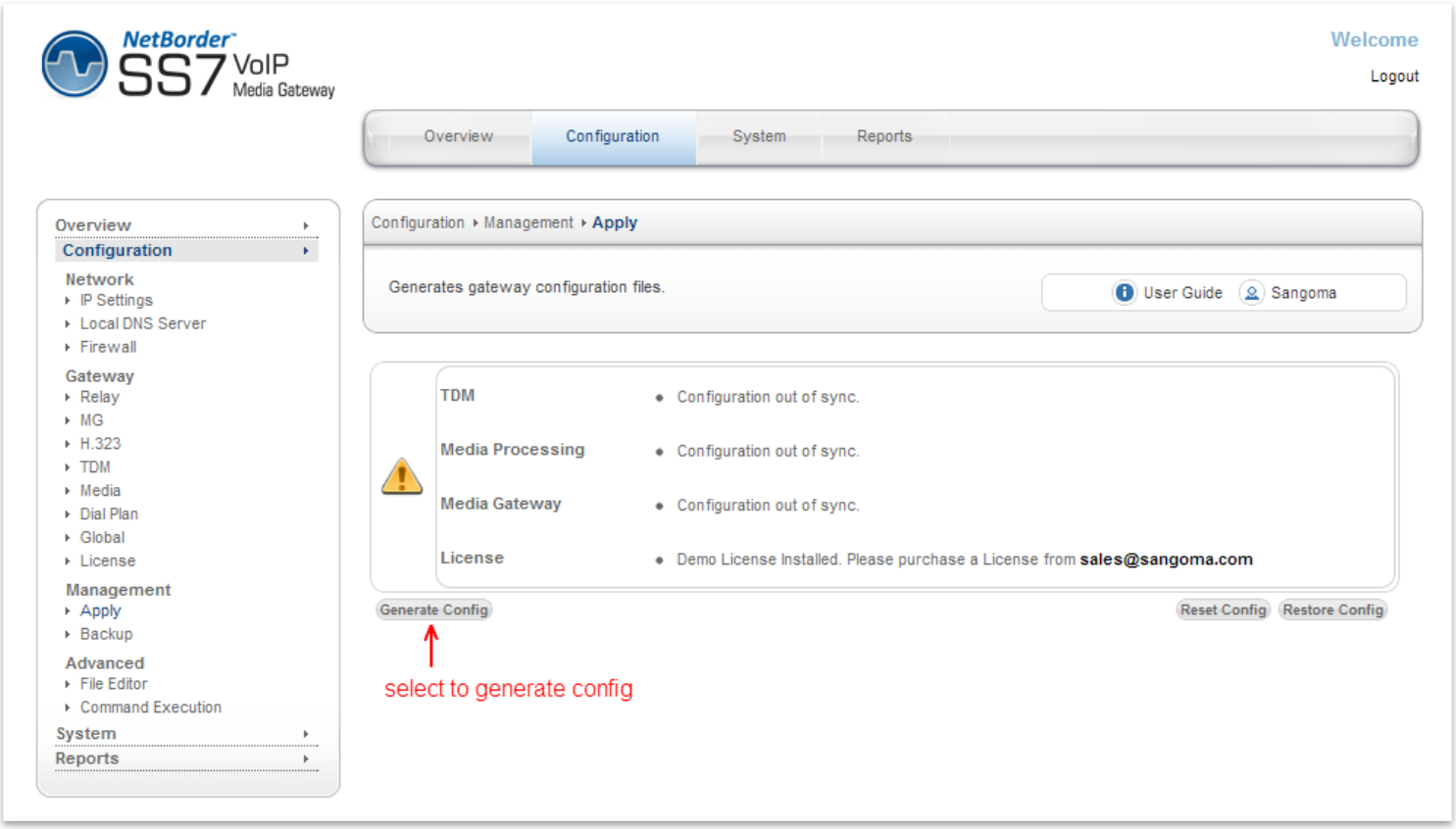
NOTE
At this point the Media configuration is complete.

- Proceed to the next section, or
- If finished all gateway configuration, proceed to Apply to generate configs.

12 Applying Configuration

The changes made in the **Configuration** section of the WebUI are only stored one the scratch disk. User MUST proceed to Apply page in the Management Section to save new configuration.

- Select **Apply** from side/top **Configuration** Menu
- Visually confirm the warnings
  - License warning need to be resolved with Sales
- Select **Generate Config** to apply the configuration to file/disk.
  - Generate Config will generate all necessary NSG SS7 VoIP Gateway configuration files needed to successful start the NSG gateway.



CAUTION:

- The generate config option will not be offered in case NSG gateway is started. Confirm that NSG is fully stopped in Control Panel before Applying configuration.

NOTE

- After configuring the NSG endpoint/protocol configuration, proceed to **Dialplan** to configure the routing rules.

13 Dialplan

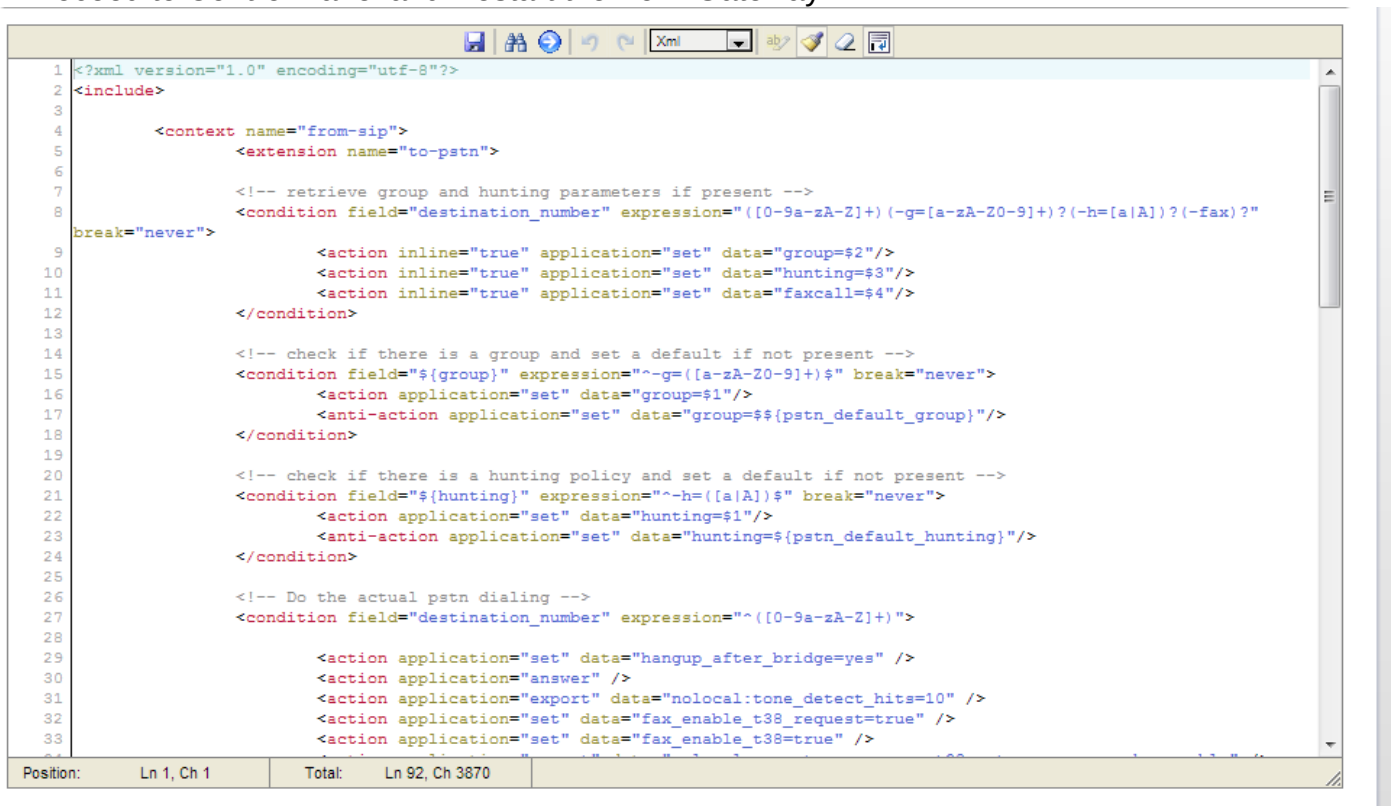
When a call is received in the NetBorder SS7 Gateway, from SIP,H232 or SS7 the dialplan is fetched to retrieve the route information to find the outgoing call location.

Note: Dialplan is **not** used in MG/Megaco/H.248 mode: MGC performs the routing.

- PSTN to SIP Dialplan
- SIP to PSTN Dialplan
- References

To access Dialplan configuration section

- Select **Dialplan** from side/top **Configuration** Menu
- Change a variable and Click on Save (Disk Icon)
- Proceed to Control Panel and Restart the VoIP Gateway.



```
<?xml version="1.0" encoding="utf-8"?>
<include>

        <context name="from-sip">
                <extension name="to-pstn">

                <!-- retrieve group and hunting parameters if present -->
                <condition field="destination_number" expression="([0-9a-zA-Z]+)(-g=[a-zA-Z0-9]+)?(-h=[a|A])?(-fax)?"
break="never">
                        <action inline="true" application="set" data="group=$2"/>
                        <action inline="true" application="set" data="hunting=$3"/>
                        <action inline="true" application="set" data="faxcall=$4"/>
                </condition>

                <!-- check if there is a group and set a default if not present -->
                <condition field="${group}" expression="^-g=([a-zA-Z0-9]+)$" break="never">
                        <action application="set" data="group=$1"/>
                        <anti-action application="set" data="group=$${pstn_default_group}"/>
                </condition>

                <!-- check if there is a hunting policy and set a default if not present -->
                <condition field="${hunting}" expression="^-h=([a|A])$" break="never">
                        <action application="set" data="hunting=$1"/>
                        <anti-action application="set" data="hunting=${pstn_default_hunting}"/>
                </condition>

                <!-- Do the actual pstn dialing -->
                <condition field="destination_number" expression="^([0-9a-zA-Z]+)">

                        <action application="set" data="hangup_after_bridge=yes" />
                        <action application="answer" />
                        <action application="export" data="nolocal:tone_detect_hits=10" />
                        <action application="set" data="fax_enable_t38_request=true" />
                        <action application="set" data="fax_enable_t38=true" />
```

Dialplan is pre-configured for

- SIP to TDM and TDM to SIP Bridging.
  Section "from-sip" routes calls from SIP to PSTN/SS7
  Section "from-pstn" routes calls from PSTN/SS7 to SIP.

- H.323 to TDM and TDM to H.323 Bridging
  Section “from-h323” routes calls from H.323 to PSTN

13.1 Dialplan Reload/Apply

Note that Dialplan can be modified in real time without the need to restart the gateway.
Once you Save the Dialplan, you will be prompted to Reload the gateway which will apply the changes without any service interrupt. All the currently established calls will not be affected. Only the newly established calls will start using the new dialplan rules.

13.2 PSTN to SIP Dialplan

By default NSG is setup to send an call to a SIP IP address. The remote SIP address must be configured in **Configuration -> Global** section.

```
<context name="from-pstn-to-sip">
      <extension name="to-sip">
            <!-- handle the case where there might not be destination number at all -->
            <condition field="destination_number" expression="^(.{1,})$" break="never">
                  <action application="set" data="destnumber=$1"/>
                  <anti-action application="set" data="destnumber=unknown"/>
            </condition>

            <!-- Dial to the gateway user (it may ring multiple registrations, first answer wins) -->
            <condition field="destination_number" expression="^(.*)$">
                  <action application="set" data="hangup_after_bridge=yes" />
                  <action application="set" data="tone_detect_hits=1" />
                  <action application="export" data="fax_enable_t38_request=true" />
                  <action application="export" data="fax_enable_t38=true" />
                  <action application="tone_detect" data="faxdisable 1100 r +5000 disable_ec 1"/>
                  <action application="export" data="execute_on_answer=tone_detect fax_disable_ec 2100 r +5000
t38_gateway 'self nocng'" />
                  <action application="set" data="sip_contact_user_replacement=${destnumber}"/>
                  <action application="set" data="hangup_after_bridge=yes"/>
                  <action application="bridge" data="sofia/internal/${destnumber}@${sip_dest_ip}:${sip_dest_port}"/>
                  <action application="hangup" data="${originate_disposition}"/>

            </condition>
      </extension>
</context>
```

13.3 PSTN to H323 Dialplan

By default NSG is setup to send a call to an H323 IP address. The remote H323 address must be configured in **Configuration -> Global** section.

```
<context name="from-pstn-to-h323">
	<extension name="to-h323">
		<!-- handle the case where there might not be destination number at all -->
		<condition field="destination_number" expression="^(.{1,})$" break="never">
			<action application="set" data="destnumber=$1"/>
			<anti-action application="set" data="destnumber=unknown"/>
		</condition>

		<!-- Dial to the gateway user (it may ring multiple registrations, first answer wins) -->
		<condition field="destination_number" expression="^(.*)$">
			<action application="set" data="hangup_after_bridge=yes" />
			<action application="set" data="tone_detect_hits=1" />
			<action application="export" data="fax_enable_t38_request=true" />
			<action application="export" data="fax_enable_t38=true" />
			<action application="tone_detect" data="faxdisable 1100 r +5000 disable_ec 1"/>
			<action application="export" data="execute_on_answer=tone_detect fax_disable_ec 2100 r +5000
t38_gateway 'self nocng'" />
			<action application="set" data="sip_contact_user_replacement=${destnumber}"/>
			<action application="set" data="hangup_after_bridge=yes"/>
			<action application="bridge" data="opal/h323:${destination_number}@${h323_remote_ip}"/>
			<action application="hangup" data="${originate_disposition}"/>
		</condition>
	</extension>
</context>

</include>
```

13.4 SIP/H323 to PSTN Dialplan

Note that both SIP and H323 profiles share the same “from-sip” context name name. The from-sip context will pass all calls to TDM interfaces.

```
<?xml version="1.0" encoding="utf-8"?>
<include>

	<context name="from-sip">
		<extension name="to-pstn">

		<!-- retrieve group and hunting parameters if present -->
		<condition field="destination_number" expression="([0-9a-zA-Z]+)(-g=[a-zA-Z0-9]+)?(-h=[a|A])?(-fax)?"
break="never">
			<action inline="true" application="set" data="group=$2"/>
			<action inline="true" application="set" data="hunting=$3"/>
			<action inline="true" application="set" data="faxcall=$4"/>
		</condition>

		<!-- check if there is a group and set a default if not present -->
		<condition field="${group}" expression="^-g=([a-zA-Z0-9]+)$" break="never">
			<action application="set" data="group=$1"/>
			<anti-action application="set" data="group=$${pstn_default_group}"/>
		</condition>

		<!-- check if there is a hunting policy and set a default if not present -->
		<condition field="${hunting}" expression="^-h=([a|A])$" break="never">
			<action application="set" data="hunting=$1"/>
			<anti-action application="set" data="hunting=${pstn_default_hunting}"/>
		</condition>

		<!-- Do the actual pstn dialing -->
		<condition field="destination_number" expression="^([0-9a-zA-Z]+)">

			<action application="set" data="hangup_after_bridge=yes" />
			<action application="answer" />
			<action application="export" data="nolocal:tone_detect_hits=1" />
			<action application="set" data="fax_enable_t38_request=true" />
			<action application="set" data="fax_enable_t38=true" />
			<action application="export" data="nolocal:execute_on_answer_1=tone_detect fax_disable_ec
2100 r +5000 disable_ec 1" />
			<action application="export" data="nolocal:execute_on_answer_2=t38_gateway peer
ced_preamble" />

			<action application="set" data="hangup_after_bridge=yes"/>
			<action application="bridge" data="freetdm/${group}/${hunting}/$1"/>
			<action application="hangup" data="${originate_disposition}"/>
		</condition>
	</extension>
</context>
```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13.5 Dialplan Syntax

There are several elements used to build an XML dialplan. In general, the dialplan groups logically similar functions and calling activities into a 'context'. Within a context are extensions, each with 'condition' rules and associated 'actions' to perform when the condition rules match.

The following is a sample dialplan to illustrate these concepts. We have left out the XML "wrapper" to help make the basic concepts more clear:

```
<context name="example">
  <extension name="500">
    <condition field="destination_number" expression="^500$">
      <action application="bridge" data="user/500"/>
    </condition>
  </extension>

  <extension name="501">
    <condition field="destination_number" expression="^501$">
      <action application="bridge" data="user/501"/>
      <action application="answer"/>
      <action application="sleep" data="1000"/>
      <action application="bridge" data="loopback/app=voicemail:default ${domain_name} ${dialed_extension}"/>
    </condition>
  </extension>
</context>
```

Each rule is processed in order until you reach the action tag which tells NSG what action to perform. You are not limited to only one condition or action tag for a given extension.

In our above example, a call to extension 501 rings the extensions. If the user does not answer, the second action answers the call, and following actions delay for 1000 milliseconds (which is 1 second) and connect the call to the voicemail system.

13.5.1 Context

Contexts are a logical grouping of extensions. You may have multiple extensions contained within a single context.

The context tag has a required parameter of 'name'. There is one reserved name, any, which matches any context. The name is used by incoming call handlers (like the [Sofia] SIP driver) to select the dialplan that runs when it needs to route a call. There is often more than one context in a dialplan.

A fully qualified context definition is shown below. Typically you'll not need all the trimmings, but they are shown here for completeness.

```
<?xml version="1.0"?>
  <document type="freeswitch/xml">
    <section name="dialplan" description="Regex/XML Dialplan">
    <!-- the default context is a safe start -->
      <context name="default">
        <!-- one or more extension tags -->
      </context>
      <!-- more optional contexts -->
    </section>
  </document>
```

13.5.2 Extensions

Extensions are destinations for a call. This is the meat of NSG routing dialed numbers. They are given a name and contain a group of conditions, that if met, will execute certain actions.

A 'name' parameter is required: It must be a unique name assigned to an extension for identification and later use.

For example:

```
<extension name="Your extension name here">
   <condition(s)...
      <action(s) .../>
   </condition>
</extension>
```

NOTE: Typically when an extension is matched in your dialplan, the corresponding actions are performed and dialplan processing stops. An optional `continue` parameter allows your dialplan to continue running.

```
<extension name="500" continue="true">
```

13.5.3 Conditions

Dialplan conditions are typically used to match a destination number to an extension. They have, however, much more power than may appear on the surface.
NSG has a set of built-in variables used for testing. In this example, the built-in variable destination_number is compared against the regular expression ^500$.
This comparison is 'true' if <destination_number> is set to 500.

```
<extension name="500">
   <condition field="destination_number" expression="^500$">
      <action application="bridge" data="user/500"/>
   </condition>
</extension>
```

Each condition is parsed with the Perl Compatible Regular Expression library. (go here for PCRE syntax information).

If a regular expression contains any terms wrapped in parentheses, and the expression matches, the variables $1,$2..$N will be set to the matching contents within the parenthesis, and may be used in subsequent action tags within this extension's block.

For example, this simple expression matches a four digit extension number, and captures the last two digits into $1.

```
<condition field="destination_number" expression="^\d\d(\d\d)$">
   <action application="bridge" data="sofia/internal/$1@example.com"/>
</condition>
```

A destination number of 3425 would set $1 to 25 and then bridge the call to the phone at 25@example.com

13.5.4 Multiple Conditions (Logical AND)

You can emulate the logical AND operation available in many programming languages using multiple conditions. When you place more than one condition in an extension, *all* conditions must match before the actions will be executed. For example, this block will only execute the actions if the destination number is 500 *AND* it is Sunday.

```
<condition field="destination_number" expression="^500$"/>
<condition wday="1">
    action(s)...
</condition>
```

Keep in mind that you must observe correct XML syntax when using this structure. Be sure to close all conditions *except the last one* with `/>`. The last condition contains the final actions to be run, and is closed on the line after the last action.
By default, if any condition is false, NSG will move on to the anti-actions or the next extension without even evaluating any more conditions.

13.5.5 Multiple Conditions (Logical OR, XOR)

It is possible to emulate the logical OR operation available in many programming languages, using multiple conditions. In this situation, if one of the conditions matches, the actions are executed.
For example, this block executes its actions if the destination number is 501 *OR* the destination number is 502.

```
<condition field="destination_number" expression="^501|502$">
   action(s)...
</condition>
```

This method works well if your OR condition is for the same field. However, if you need to use two or more different fields then use the new **regex** syntax

```
<extension name="Regex OR example 1" continue="true">
  <condition regex="any">
    <!-- If either of these is true then the subsequent actions are added to execute list -->
    <regex field="caller_id_name" expression="Some User"/>
    <regex field="caller_id_number" expression="^1001$"/>
    <action application="log" data="INFO At least one of the conditions matched!"/>
    <!-- If *none* of the regexes is true then the anti-actions are added to the execute list -->
    <anti-action application="log" data="WARNING None of the conditions matched!"/>
  </condition>
</extension>
```

Using this method it becomes easier to match the caller's name OR caller ID number and execute actions whether either is true.
A slightly more advanced use of this method is demonstrated here:

```
<extension name="Regex OR example 2" continue="true">
  <condition regex="any" break="never">
    <regex field="caller_id_name" expression="^Michael\s*S?\s*Collins"/>
    <regex field="caller_id_number" expression="^1001|3757|2816$"/>
    <action application="set" data="calling_user=mercutioviz" inline="true"/>
    <anti-action application="set" data="calling_user=loser" inline="true"/>
  </condition>

  <condition>
    <action application="answer"/>
```

```
    <action application="sleep" data="500"/>
    <action application="playback" data="ivr/ivr-welcome_to_freeswitch.wav"/>
    <action application="sleep" data="500"/>
  </condition>

  <condition field="${calling_user}" expression="^loser$">
    <action application="playback" data="ivr/ivr-dude_you_suck.wav"/>
    <anti-action application="playback" data="ivr/ivr-dude_you_rock.wav"/>
  </condition>
</extension>
<extension name="Regex XOR example 3" continue="true">
  <condition regex="xor">
    <!-- If only one of these is true then the subsequent actions are added to execute list -->
    <regex field="caller_id_name" expression="Some User"/>
    <regex field="caller_id_number" expression="^1001$"/>
    <action application="log" data="INFO Only one of the conditions matched!"/>
    <!-- If *none* of the regexes is true then the anti-actions are added to the execute list -->
    <anti-action application="log" data="WARNING None of the conditions matched!"/>
  </condition>
</extension>
```

Basically, for this new syntax you can have a condition to have a "regex" attr instead of "field" and "expression" etc. When there is a "regex" attr, that means you plan to have one or more <regex> tags that are similar to the condition tag itself that it has field and expression in it.

The value of the "regex" attr is either "all" or "any" or "xor indicating if all expressions must match or just any expression or only one must match(xor) . If it's set to "any" it will stop testing the regex tags as soon as it finds one match, if it is set to "all", it will stop as soon as it finds one failure.

From there it will behave like a normal condition tag either executing the actions or anti-actions and breaking based on the "break" attr.

The basic difference here is once there is a "regex" attr, the <regex> tags parsed for "all" or "any" take the place of the single "field" and "condition"

NOTE: Also, if any captures are done in the "expression" attrs of a <regex> tag, only the data from the newest capture encountered will be considered in the $n expansion or FIELD_DATA creation. In addition, you can set DP_REGEX_MATCH_1 .. DP_REGEX_MATCH_N to preserve captures into arrays.

```
<extension name="Inbound_external">
```

```
  <condition regex="any">
    <regex field="${sip_from_host}" expression="domainA"/>
    <regex field="${sip_from_uri}" expression="1234567890@domainB"/>
    <regex field="${sip_from_uri}" expression="user@domainC"/>
    <regex field="caller_id_name" expression="^(John Smith)$"/>
    <regex field="caller_id_number" expression="^(55512341)|(55512342)|(55512343)$"/>

    <action application="set" data="domain_name=domainZ"/>
    <action application="transfer" data="${destination_number} XML domainZ"/>
  </condition>
 </extension>
```

This is another example to show that all regex conditions must be true, then the action will get executed; otherwise, the anti-action will. This is the same logic as follows:

```
IF (cond1 AND cond2 AND cond3) THEN
 do actions
ELSE
 do other actions
ENDIF
```

Basically, the <condition regex="all"> tells the parser, "Hey, execute the <action>'s only if all regexes PASS, otherwise execute any <anti-action>'s".

```
  <condition regex="all">
  <regex field="${sip_gateway}" expression="^${default_provider}$"/>
  <regex field="${emergency_call}" expression="^true$"/>
  <regex field="${db(select/emergency/autoanswer)}" expression="^1$"/>

  <!-- the following actions get executed if all regexes PASS -->
   <action application="set" data="call_timeout=60"/>
   <action application="set" data="effective_caller_id_name=${regex(${caller_id_name}|^Emerg(_.*)$|Auto%1)}"/>
   <action application="set" data="autoanswered=true"/>
   <action application="bridge" data="user/1000@${domain_name},sofia/gateway/1006_7217/${mobile_number}"/>

  <!-- the following anti-actions are executed if any of the regexes FAIL -->
   <anti-action application="set" data="effective_caller_id_name=${regex(${caller_id_name}|^Emerg(_.*)$|NotAuto%1)}"/>
   <anti-action application="set" data="call_timeout=30"/>
   <anti-action application="set" data="autoanswered=false"/>
```

```
  <anti-action application="bridge" data="user/1000@${domain_name},sofia/gateway/1006_7217/${mobile_number}"/>
</condition>
```

13.5.6 Complex Condition/Action Rules

Here is a more complex example, performing time-based routing for a support organization. The user dials extension 1100. The actual support extension is 1105 and is staffed every day from 8am to 10pm, except Friday, when it is staffed between 8am and 1pm. At all other times, calls to 1100 are sent to the support after-hours mailbox.

```
<extension name="Time-of-day-tod">
   <!--if this is false, FreeSWITCH skips to the next *extension*.-->
   <condition field="destination_number" expression="^1100$" break="on-false"/>

   <!--Don't bother evaluating the next condition set if this is true.-->
   <condition wday="6" hour="8-12" break="on-true">    <!--Fri, 8am-12:59pm-->
      <action application="transfer" data="1105 XML default"/>
   </condition>

   <condition wday="1-5" hour="8-21" break="on-true">   <!--Sunday-Thursday, 8am-9:59pm-->
      <action application="transfer" data="1105 XML default"/>
   </condition>

   <condition> <!--this is a catch all, sending the call to voicemail at all other times. -->
      <action application="voicemail" data="default ${domain} 1105"/>
   </condition>
</extension>
```

In this example, we use the `break=never` parameter to cause the first condition to 'fall-through' to the next condition no matter if the first condition is true or false. This is useful to set certain flags as part

of extension processing. This example sets the variable `begins_with_one` if the destination number begins with 1.

```
<extension name="break-demo">
 <!-- because break=never is set, even when the destination does not begin
      with 1, we skip the action and keep going -->
 <condition field="destination_number" expression="^1(\d+)$" break="never">
   <action application="set" data="begins_with_one=true"/>
 </condition>

 <condition field="destination_number" expression="^(\d+)$">
    ...other actions that may query begins_with_one...
 </condition>
</extension>
```

13.5.7 Variables

Condition statements can match against channel variables, or against an array of built in variables.

13.5.7.1 Built-In Variables

The following variables, called 'caller profile fields', can be accessed from condition statements directly:

- **context** Why can we use the context as a field? Give us examples of usages please.
- **rdnis** Redirected Number, the directory number to which the call was last presented.
- **destination_number** Called Number, the number this call is trying to reach (within a given context)
- **dialplan** Name of the dialplan module that are used, the name is provided by each dialplan module. Example: XML
- **caller_id_name** Name of the caller (provided by the User Agent that has called us).
- **caller_id_number** Directory Number of the party who called (caller) -- can be masked (hidden)
- **ani** Automatic Number Identification, the number of the calling party (caller) -- cannot be masked
- **aniii** The type of device placing the call ANI2
- **uuid** Unique identifier of the current call? (looks like a GUID)
- **source** Name of the FreeSWITCH module that received the call (e.g. PortAudio)
- **chan_name** Name of the current channel (Example: PortAudio/1234). Give us examples when this one can be used.
- **network_addr** IP address of the signaling source for a VoIP call.
- **year** Calendar year, 0-9999
- **yday** Day of year, 1-366
- **mon** Month, 1-12 (Jan = 1, etc.)
- **mday** Day of month, 1-31
- **week** Week of year, 1-53
- **mweek** Week of month, 1-6
- **wday** Day of week, 1-7 (Sun = 1, Mon = 2, etc.) or "sun", "mon", "tue", etc.
- **hour** Hour, 0-23
- **minute** Minute (of the hour), 0-59
- **minute-of-day** Minute of the day, (1-1440) (midnight = 1, 1am = 60, noon = 720, etc.)
- **time-of-day** Time range formatted: hh:mm[:ss]-hh:mm[:ss] (seconds optional) Example: "08:00-17:00"
- **date-time** Date/time range formatted: YYYY-MM-DD hh:mm[:ss]~YYYY-MM-DD hh:mm[:ss] (seconds optional, note tilde between dates) Example: 2010-10-01 00:00:01~2010-10-15 23:59:59

For example:

```
<condition field="network_addr" expression="^192\.168\.1\.1$"/>  <!-- network address=192.168.1.1 >
<condition mon="2">   <!-- month=February -->
```

13.5.7.2 Caller Profile Fields vs. Channel Variables

One thing that may seem confusing is the distinction between a caller profile field (the built-in variables) and a channel variable.
Caller profile fields are accessed like this:

```
<condition field="destination_number" attributes...>
```

While channel variables are accessed like this:

```
<condition field="${sip_has_crypto}" attributes...>
```

Please take note of the **${variable_name}** syntax. Channel variables may also be used in action statements. In addition, API functions can be called from inside a condition statement to provide dynamic data.

For example, you can use the **cond** API:

```
<condition field="${cond(${my_var} > 12 ? YES : NO)}" expression="^YES$">
   <action application="log" data="INFO ${my_var} is indeed greater than 12"/>
</condition>
```

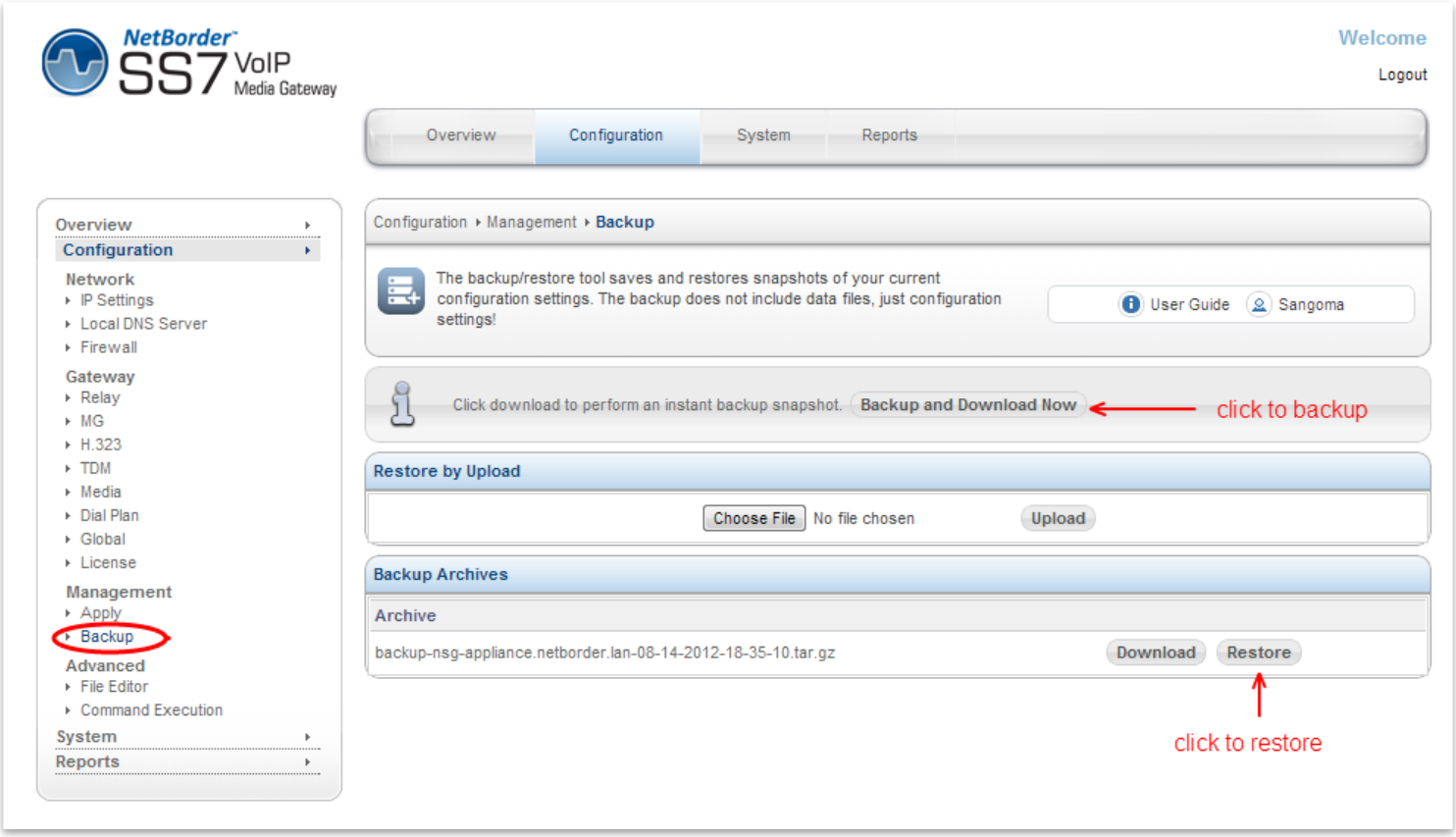
This example tests ${my_var}. If it is more than 12, "YES" is returned. Otherwise "NO" is returned. The condition tests the results for "YES" and logs the resulting message to the NSG log.

14 Backup Restore System

Appliance configuration can be backed up to a zipped file.
Appliance can be restored from a same file.

- Select **Backup** from side/top **Configuration** Menu

- Click on Backup and Download Now
  - Note that a backup will be offered for download as well as stored locally on the system.
  - Note the Backup Archive shows previous backups that can be used to restore the system.



14.1 Restore a System

The default scenario for system Restore is to

- recover an existing system from factory reset, or
- to recover to another system, due to system failure

CAUTION

- After a system has been restored via WebGUI a **reboot is mandatory**.

After a reboot

- Confirm the VLAN configuration -> Overview -> VLAN Status
- Confirm the IP route configuration -> Overview -> VLAN Status (Routing Rules)
- Confirm Gateway is status in Overview -> Control Pannel
- Confirm Gateway status in Overview -> TDM Status

14.2 Restore to a new System

It is possible to back-up a working system, and restore the configuration to another target system, with the intent of quickly provisioning a new target system.

However as backup will duplicate the current system, this is only useful in the case where original system failed and is being replaced.

Restore has not been designed to provision new systems.
The amount work necessary to change a restored new system to operation is equivalent to starting from scratch.

If using restores to provision a new system:

- License
  The license is going to be invalid on a new system. Thus user must update the system with correct license after the restore from the backup.
- IP Settings
  IP settings are going to be duplicated and most likely invalid if the original system is still functioning. Thus user must go into the IP Settings section and update the local IP settings.
- VLAN
  VLAN IP settings are going to be duplicated and most likely invalid if the original system is still functioning. Thus user must go into the VLAN Settings section and update to new values.
- Megaco/SIP/H323
  All IP settings will most likely have to change.
- TDM Spans
  Target system must have identical T1/E1 spans installed as the source system. If TDM installation is not identical there could be port mismatches or configuration errors, which will cause the system to fail.

If provisioning from backing is the goal then user would have to edit the backup files manually to update above settings before restoring to a target system.

This is not recommended and requires expert level understanding of the backup files and manual configuration files. Which defeats the purpose of the WebGUI.

NOTE
Sangoma has a product roadmap plan for mass system provisioning.
If this is of interest please contact Sales.

15 Factory Reset & Reboot

15.1 Factory Reset

- Find a power button in front of the NSG Appliance
- Press the power button repeatedly fast (every 1 sec) for 10 sec.
- On factory reset trigger
    - You will hear a loud high frequency beep for 10 seconds indicating that factory reset has been successful.
    - The system will be restored to factory settings and the system will reboot.

CAUTION

- If you do not hear the factory reset sound and system reboots, you have triggered a soft reboot sequence.
    - Once system comes back up, re-try the factory reset sequence.

15.2 Appliance Soft Reboot

- Find a power button in front of the NSG Server
- Press the power button three times with more than 2sec delay in between..
    - Press power button
    - Count to 3 (3 sec)
    - Press power button
    - Count to 3 (3 sec)
    - Press power button
- When there were 3 power button presses within 10sec and 3sec apart, the NSG System will do a soft reboot.

NOTE

- A soft reboot can be triggered via WebGUI or USB CLI
- WebGUI -> System -> Shutdown.
- USB CLI -> reboot command

15.3 Appliance Shutdown

- Find a power button in front of the NSG Appliance
- Press the power button and hold it until machine shutdown.

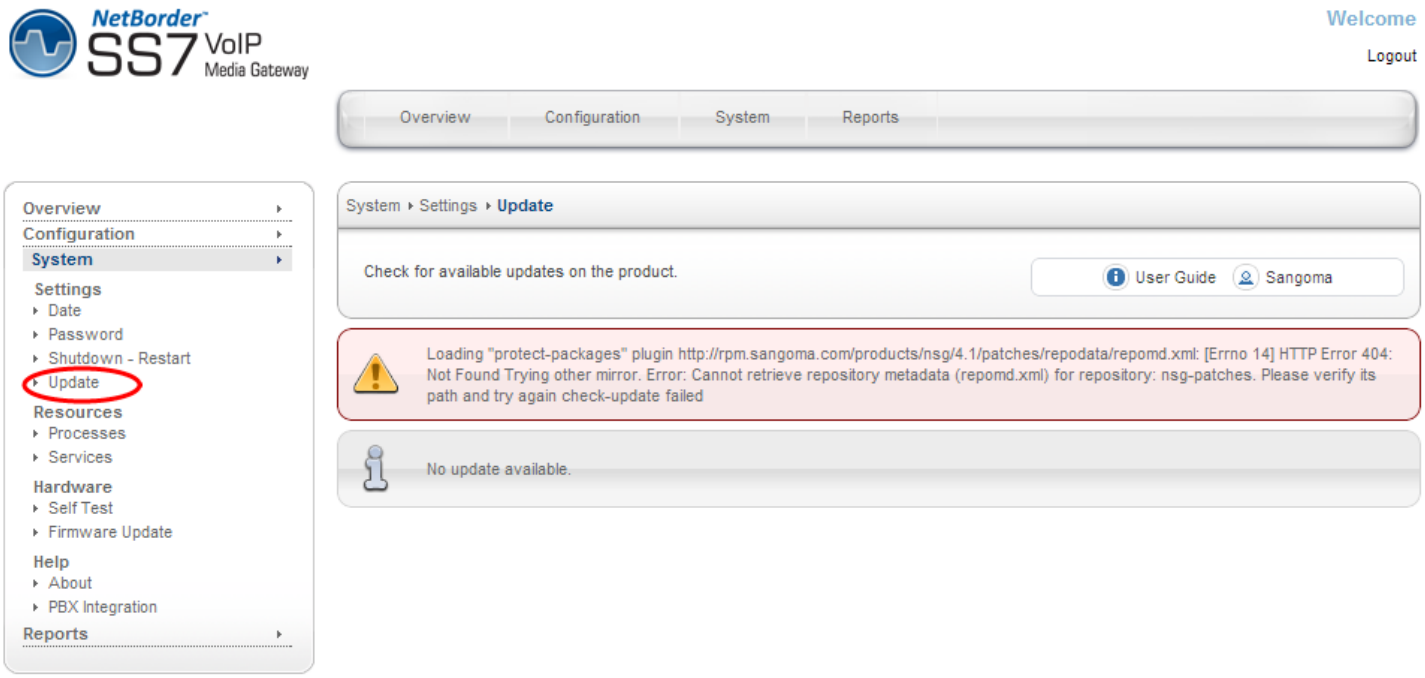
16 Upgrade

User has three choices when upgrading NSG system.

- Centralized Push Upgrade from NOC
- WebUI Update Page

16.1 WebUI System Update

- Select **Update** from side/top **System** Menu
- Review available packages for upgrade.
- Proceed with the upgrade process



16.2 Console SSH Update

NSG product uses Linux RPM as part of its package management system.

- Download new NSG RPM version

- Stop NSG services
  - User the GUI Control Panel
  - Alternatively run:
    - services nsg stop
    - services nsg-webui stop

- Install new package
  - rpm –Uvh nsg-4.3.1.rpm

- Restart NSG services
  - Use the GUI Control Panel
  - Alternatively run:
    - services nsg-webui start
    - services nsg start

NOTE

- Using NSG console to upgrade the system is very powerful, as the process can be scripted and centralized. This way all NSG appliances in the files can be upgraded from a single upgrade machine in the NOC.

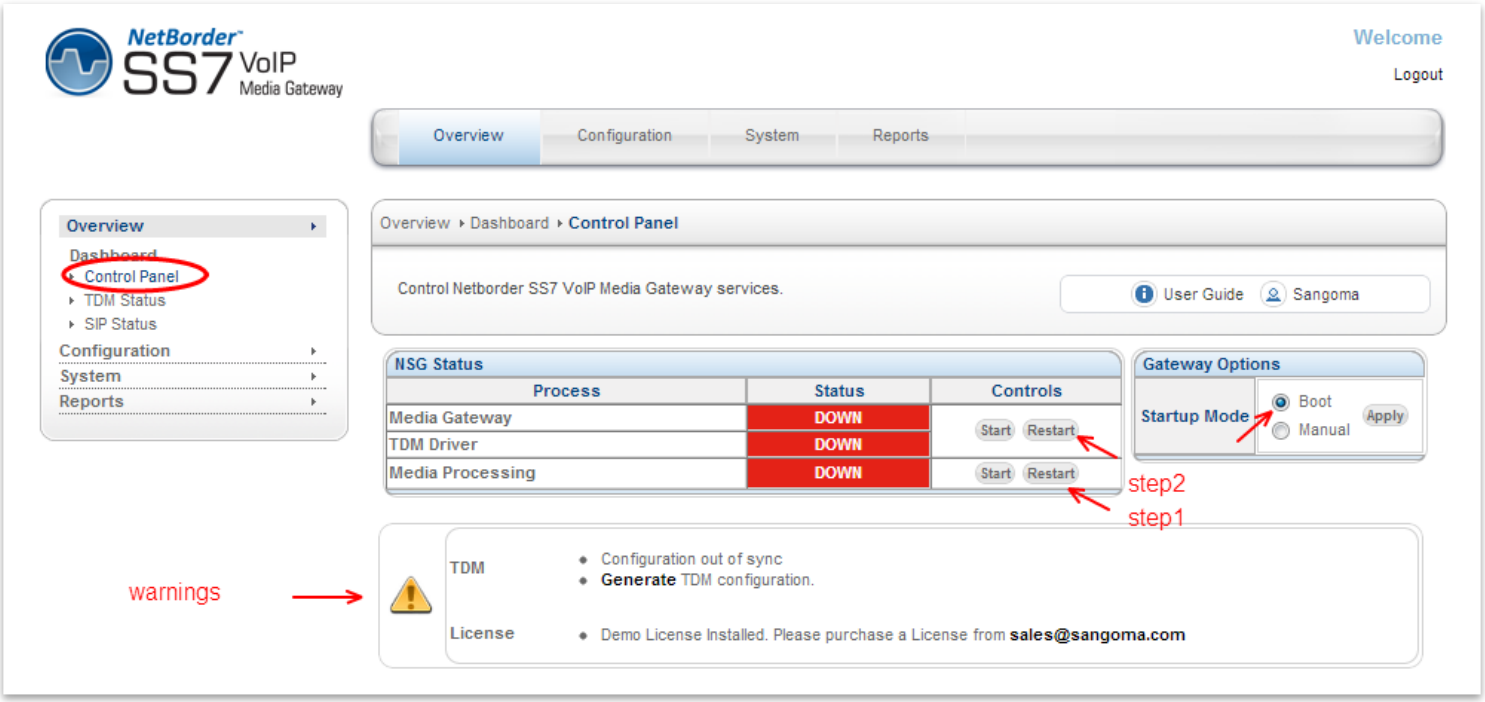
17 Operations

17.1 Starting the Gateway

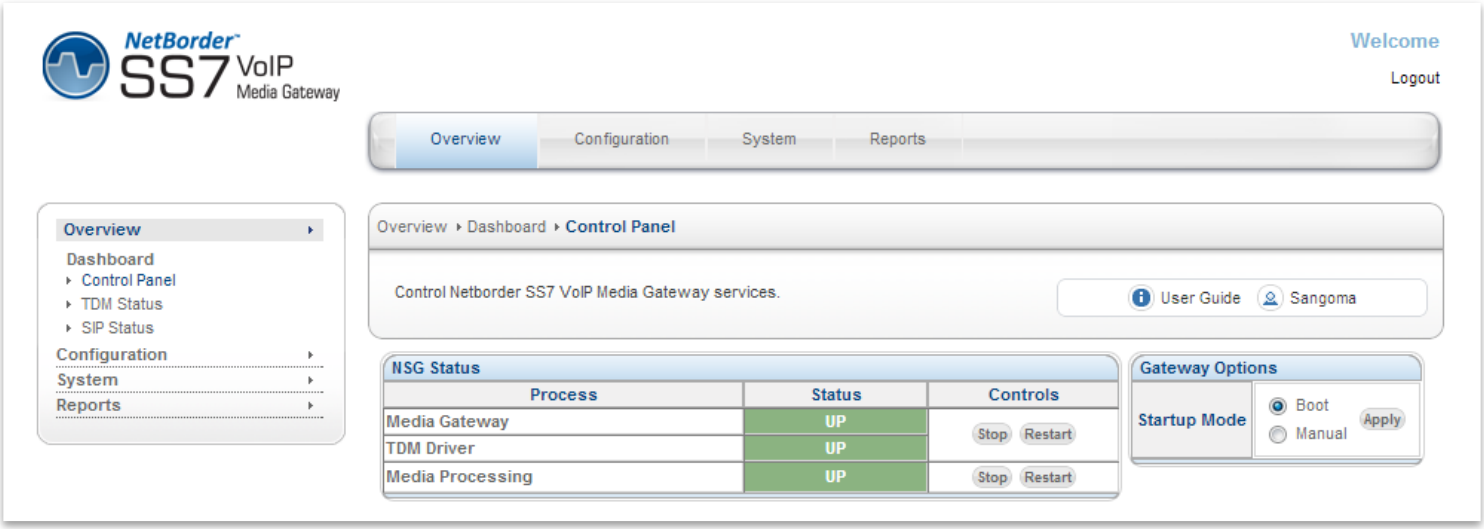
- Select **Control Panel** from side/top **Overview** Menu
- Confirm that warnings are clear
- Start the Media Processing First
  - Media Processing will start the Transcoding resources.
  - Note that Media Processing is optional

- Start the Media Gateway Second.
  - Media Gatway will start
  - TDM Hardware Spans (T1/E1 ports)
  - Netborder SS7 to VoIP Gateway Software

- Confirm that the **boot** button is selected.
  - This will confirm that gateway starts on reboot.



- When the Gateway starts successfully the green status bar will appear.
- System is now running.



NOTE

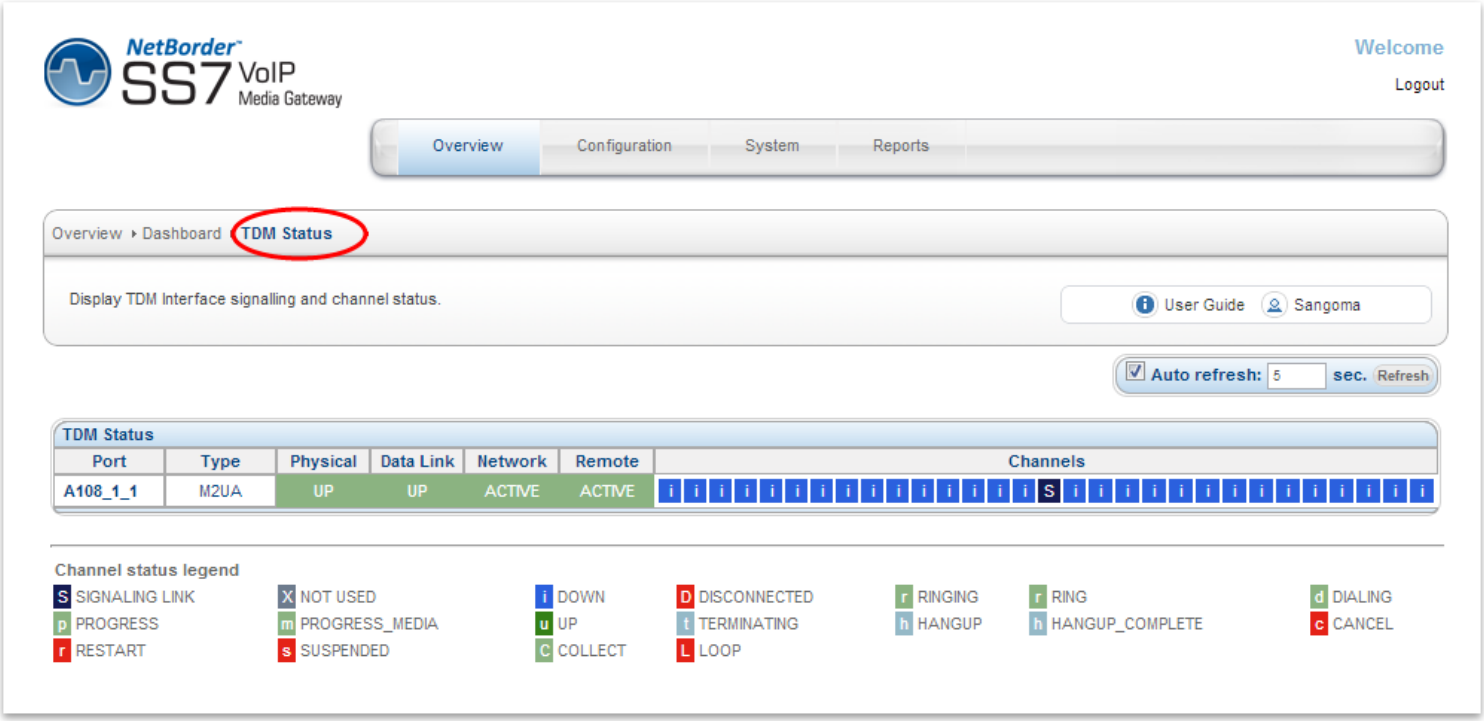
- Before attempting to pass traffic through the gateway, proceed to **TDM Status** to check the state of the NSG gateway. There is no point of attempting calls while the status of the gateway protocol is down.

17.2 Gateway Status

17.2.1 Megaco/M2UA TDM

The TDM Stats page displays the unified status of all NSG components

- Select **TDM Stats** from side/top **Overview** Menu



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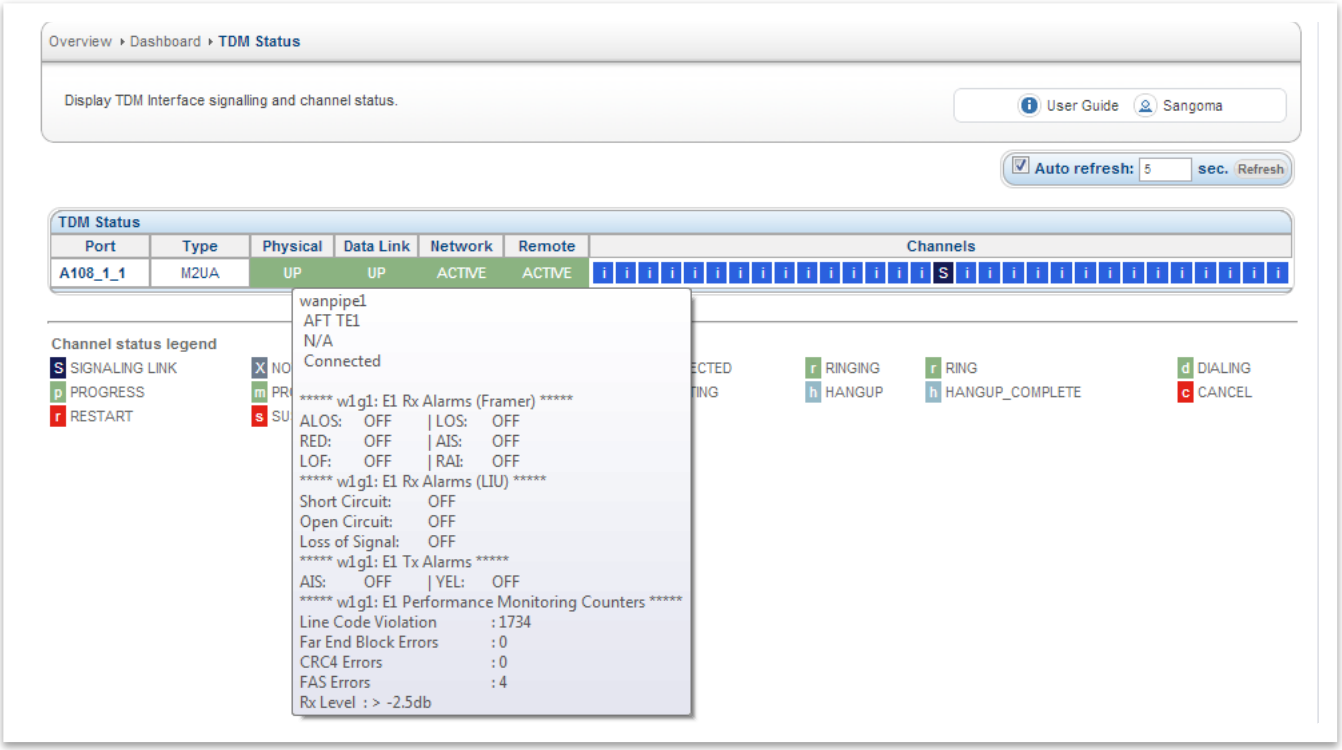
Field Name	*Description*
Port	Physical Port number. Identifies the hardware resource and T1/E1 port number. The T1/E1 port number relates to the T1/E1 board.
Type	Signaling Type In this example we see: M2UA
Physical	Physical T1/E1 layer status. Hover the mouse over the Physical status section (green) to display detailed T1/E1 alarms and status.
Data Link	MTP2 Link Layer status. Hover the mouse over the UP and a popup will display detailed MTP2 status
Network	M2UA Link Layer status Hover the mouse over the UP and a popup will display detailed M2UA status
Remote	Remote MGC Megaco Peer status. This indicates that MG is connected to the MGC Megaco profile. Hover the mouse over the UP and a popup will display detailed Megaco Peer status
Channels	If Megaco link state is IN-SERVICE Channel is blue - down If Megaco link state is OUT-OF-SERVICE Channel is red – down If channel is in use Channel is green – up Hover the mouse over each channel for more detailed data.

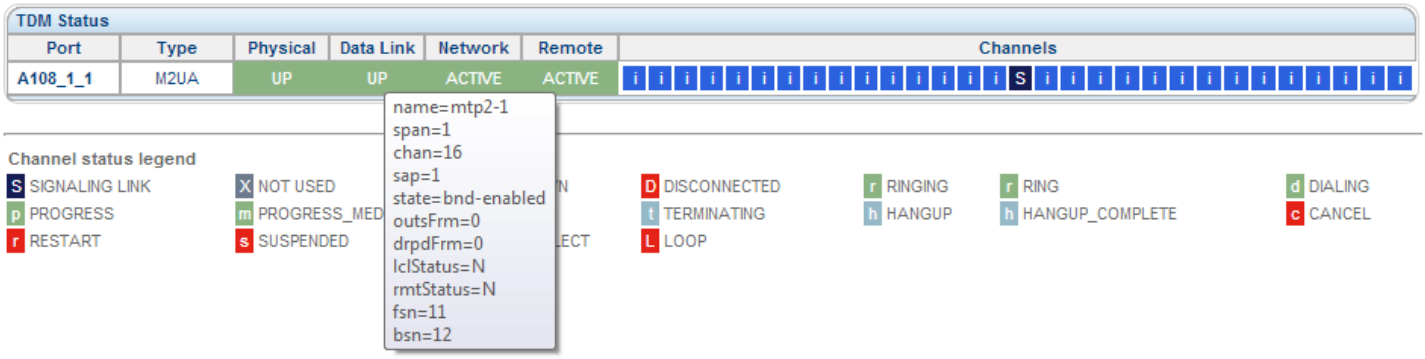
17.2.1.1 Physical T1/E1 Alarms

Hover the mouse over Physical Status Section.
For detailed information about Alarms refer to Troubleshooting Section 18.



17.2.1.2 Data Link MTP2 Alarms

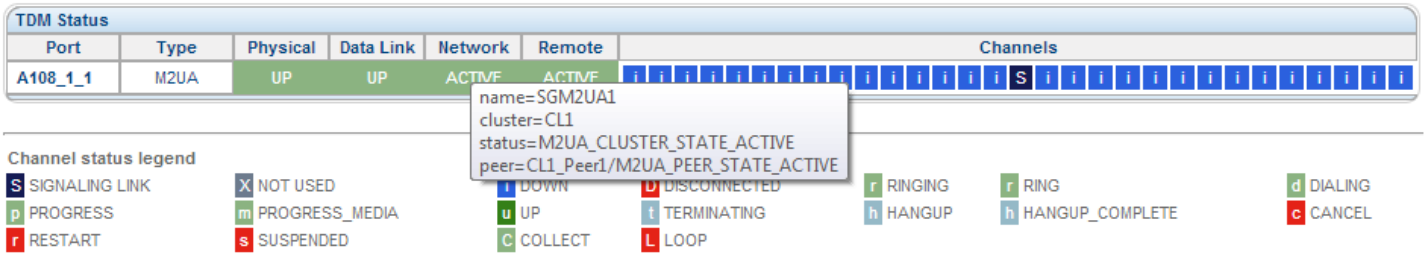
Hover the mouse over Data Link Section.



state	bnd-enable	MTP2 is connected
	bnd-disable	MTP2 is disconnected

17.2.1.3 Network M2UA Alarms

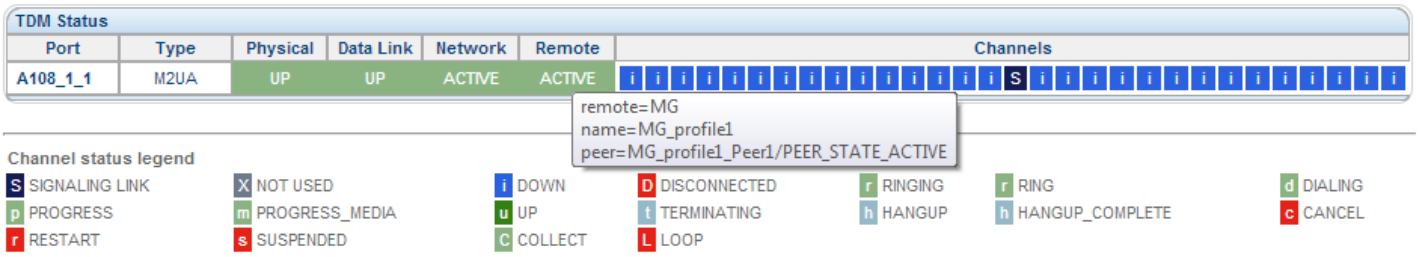
Hover the mouse over Network Section.



status	M2UA_CLUSTER_STATE_ACTIVE	Local state is connected.
peer	M2UA_PEER_STATE_ACTIVE	Remote MGC M2PA is in sync with local M2UA connection

17.2.1.4 Remote Megaco Alarms

Hover the mouse over Remote Section



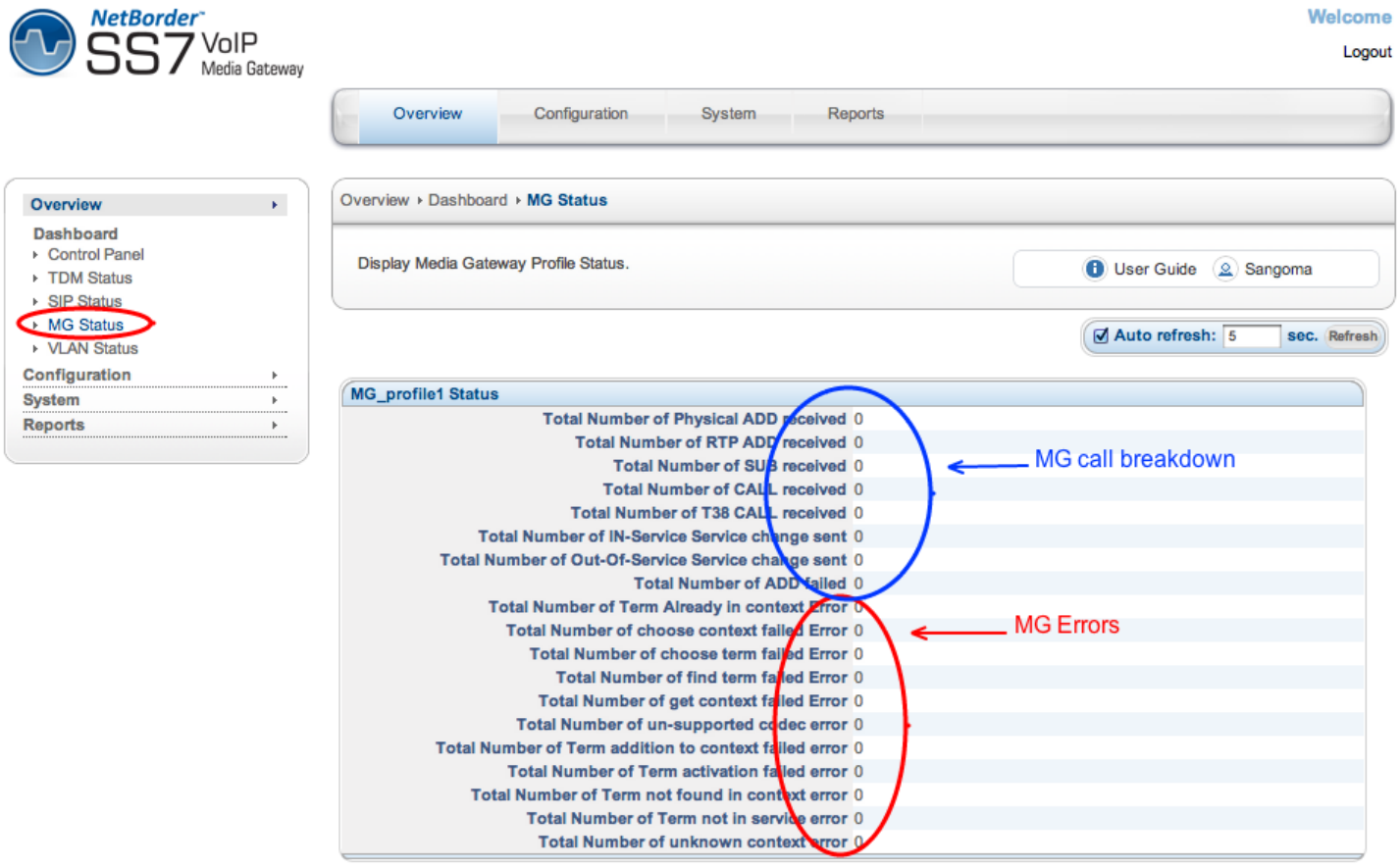
peer	PEER_STATE_ACTIVE	Remote MGC Megaco protocol is in sync with local Megaco profile.

NOTE
For more information on how to debug each section please refer to the Troubleshooting section.

17.3 Megaco Status

Megaco Status page provides detailed Megaco call statistics per Megaco Profile.

- Select **MG Status** from side/top **Overview** Menu



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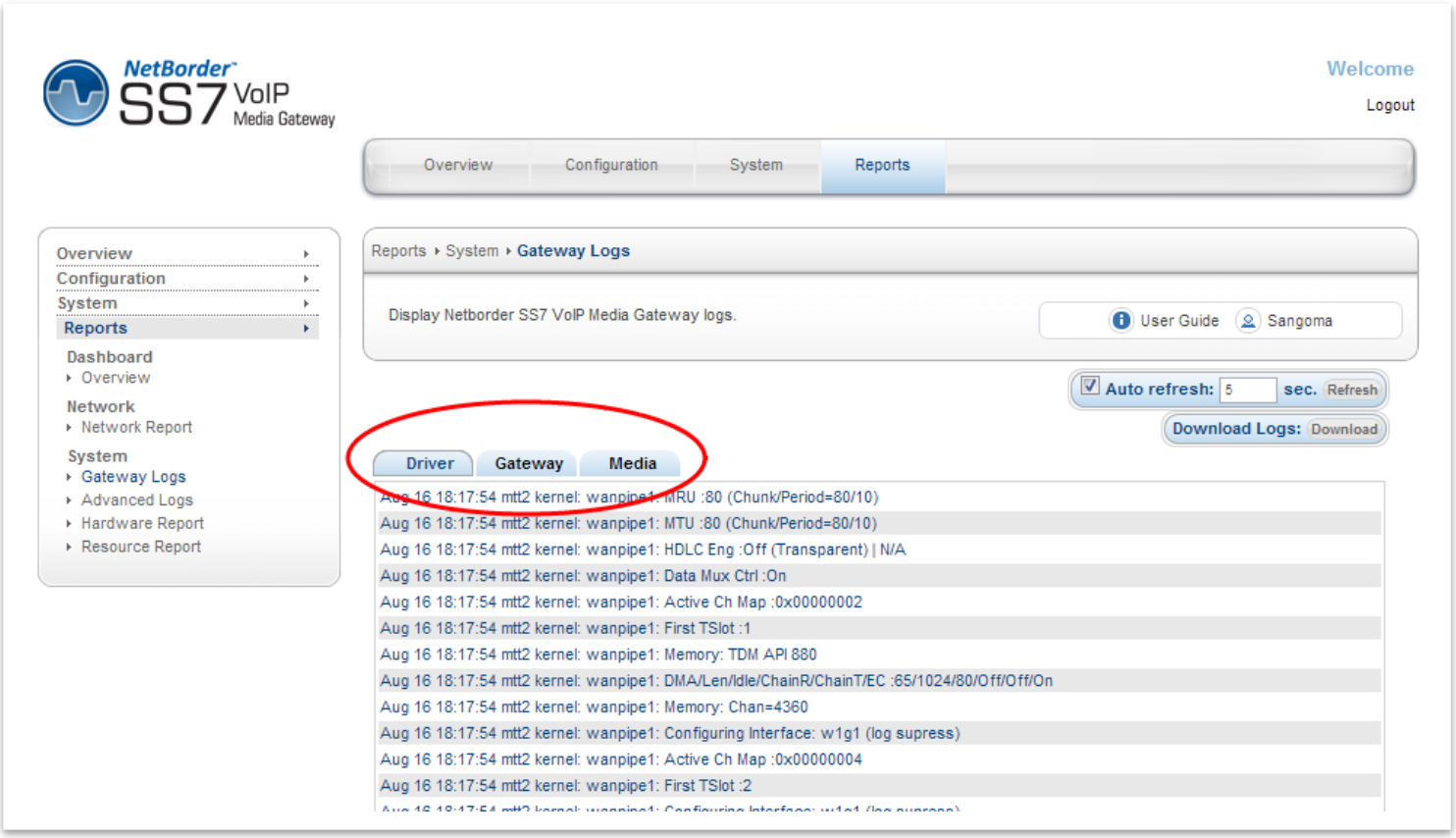
Reports

17.4 Gateway Logs

- Select **Gateway Logs** from side/top **Reports** Menu



NOTE
All error events will be displayed in RED for easy identification.

Log	Description
Driver	TDM device driver log. All errors will be identified in RED This log will show • TDM Driver startup sequence • TDM T1/E1 connection/disconnection • TDM Driver general errors • System errors • OS Errors
Gateway	SS7 to VoIP Gateway log All errors will be identified in RED This log will show • Gateway startup sequence • Gateway startup errors • Gateway run time errors and warnings
Media	Media Transcoding log All errors will be identified in RED This log will show • Media Transcoding server startup sequence • Media startup errors • Media transcoding run time errors and warnings

17.4.1 Gateway Log Download

When working with Sangoma support, you will be asked to download and submit the NSG logs.

- Select **Download Logs** Button
- Save the zipped file to your computer
- Send the zipped debug package to Sangoma Support

Download Logs contains

- All Gateway, Driver and Transcoding log files
- Full Gateway configuration

17.5 Advanced Logs

Detailed historical logs can be found in Advanced Logs Section. This page can be used to determine historical alarm, events and errors.

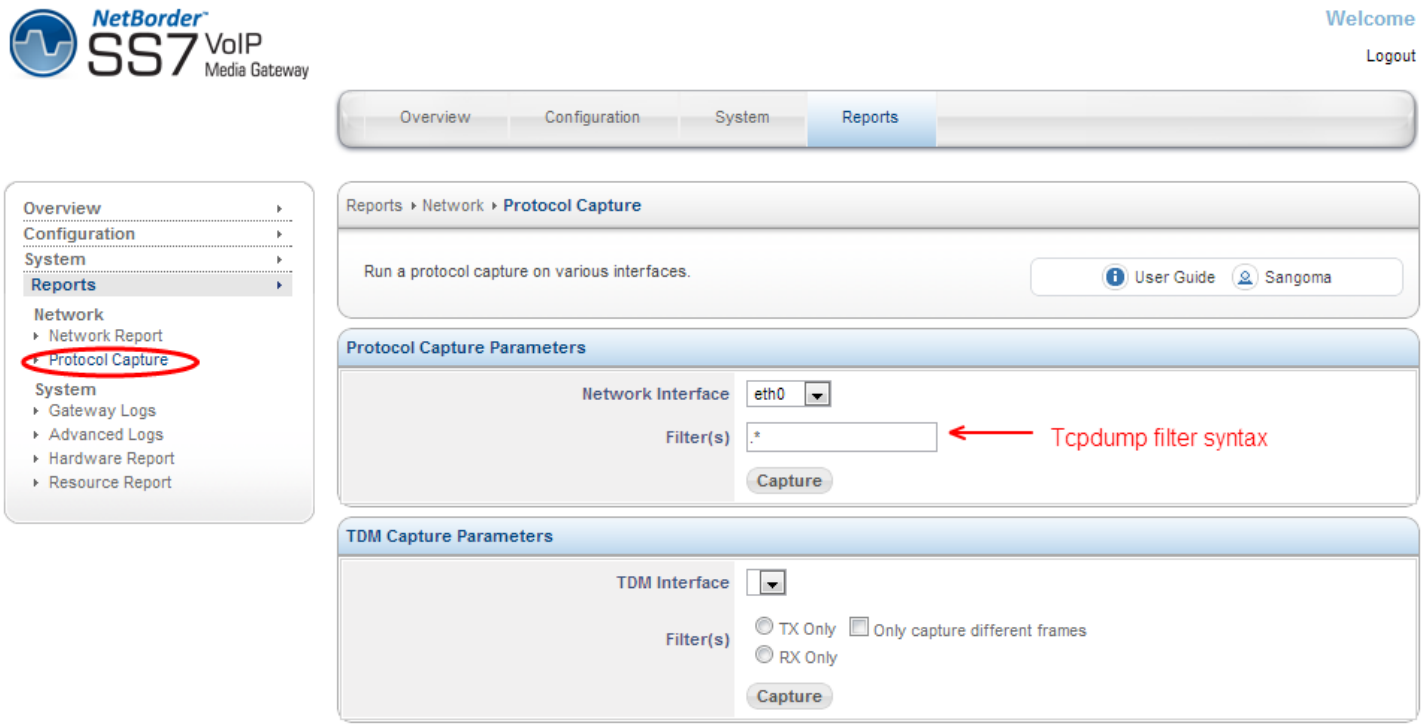
- Select **Advanced Logs** from side/top **Reports** Menu

Files	Description	
messages	Displays kernel and driver level messages. Including all T1/E1 status changes or error messages.	Filter: E1 — All E1 messages Filter: E1.*con — All E1 connected & disconnected messages Filter: : ON — All T1/E1 Alarms ON events Filter: : OFF — All T1/E1 Alarms OFF events Filter: Error — All Error messages Filter: wanpipe — All T1/E1 driver messages
nsg/sangomagw.log	Display all NSG gateway logs.	Filter: ERR — All Error Messages Filter: WARN — All Warning Messages
sngtc_server.log	Displays all Media Transcoding logs	Filter: ERR — All Error Messages Filter: WARN — All Warning Messages

17.6 Packet Capture

The packet capture page captures network traffic from Ethernet interface, TDM interface or both.

- Select **Packet Capture** from side/top **Reports** Menu
- Filter
  - Default filter will capture all packets on the Ethernet device
- Select Capture to start capturing
- Wait…
- Select Stop Capture when Capture done
- Download Link with capture pcap file is ready for download.


NetBorder SS7 VoIP Media Gateway
Welcome
Logout
Overview
Configuration
System
Reports
Overview
Configuration
System
Reports
Network
Network Report
Protocol Capture
System
Gateway Logs
Advanced Logs
Hardware Report
Resource Report
Reports ▸ Network ▸ Protocol Capture
Run a protocol capture on various interfaces.
User Guide
Sangoma
Protocol Capture Parameters
Network Interface
eth0
Filter(s)
.*
Tcpdump filter syntax
Capture
TDM Capture Parameters
TDM Interface
Filter(s)
TX Only
Only capture different frames
RX Only
Capture

17.6.1 Ethernet Capture Filter Options

host <ip>	True if either the IPv4/v6 source or destination of the packet is host.
dst host <ip>	True if the IPv4/v6 destination field of the packet is host, which may be either an address or a name
src host <ip>	True if the IPv4/v6 source field of the packet is host.
net <ip>	True if either the IPv4/v6 source or destination address of the packet has a network number of net.
port <port>	True if either the source or destination port of the packet is port.
dst port <port>	True if the packet is ip/tcp, ip/udp, ip6/tcp or ip6/udp and has a destination port value of port.
src port <port>	True if the packet has a source port value of port.
vlan <vlan_id>	True if the packet is an IEEE 802.1Q VLAN packet. If [vlan_id] is specified, only true if the packet has the specified vlan_id. For example: vlan 100 && vlan 200 filters on VLAN 200 encapsulated within VLAN 100, and vlan && vlan 300 && ip filters IPv4 protocols encapsulated in VLAN 300 encapsulated within any higher order VLAN.
tcp, udp, icmp	True if protocol matches
not <port> | not <ip>	Exclude a port/ip/protocol out of the trace

NOTE
Please refer to tcpdump documentation for more info.

18 Monitoring & Management

NSG Currently offers number of monitoring and management options

- SNMP
- Web GUI Status
- SSH CLI (Scripting)

18.1 SNMP

Simple Network Management Protocol (SNMP) is an "Internet-standard protocol for managing devices on IP networks." Devices that typically support SNMP include routers, switches, servers, workstations, printers, modem racks, and more." It is used mostly in network management systems to monitor network-attached devices for conditions that warrant administrative attention. SNMP is a component of the Internet Protocol Suite as defined by the Internet Engineering Task Force (IETF). It consists of a set of standards for network management, including an application layer protocol, a database schema, and a set of data objects

- NSG provides SNMP support version 1, 2, 3
    - Note that SNMP version 1,2 are mutually exclusive to version 3.

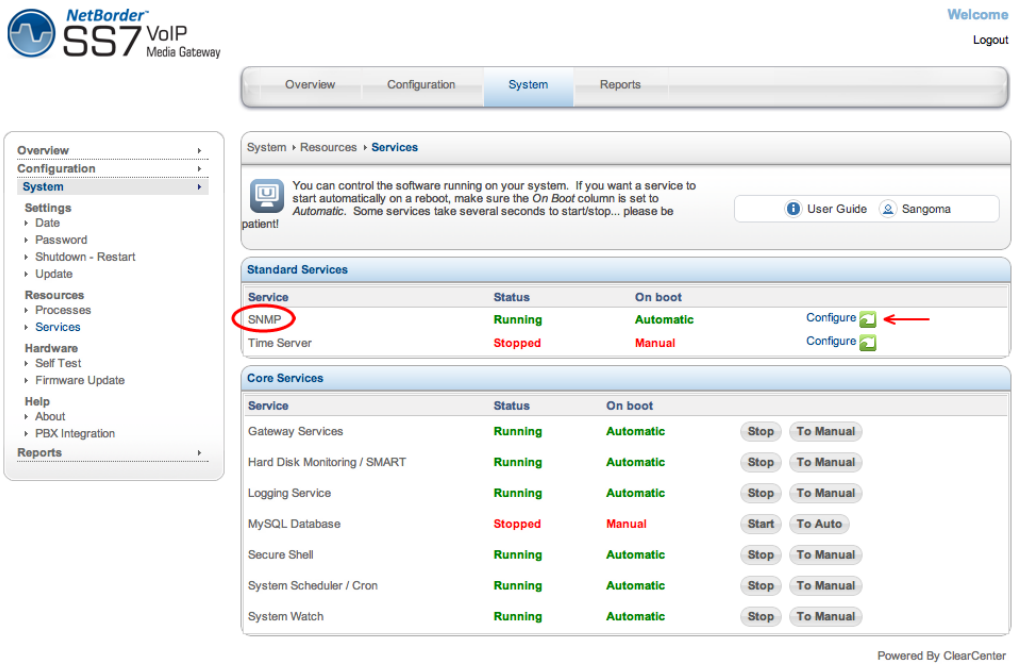
- SNMP Version3 requires user authentication, and is more secure than versions 1 & 2.

- By default NSG comes pre-configured with SNMP version 1 & 2 enabled.

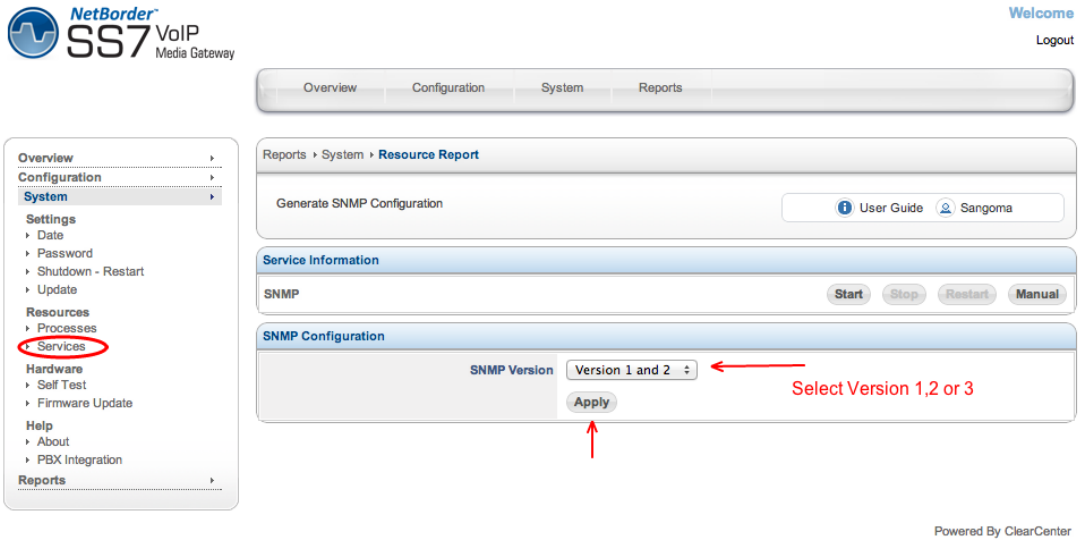
18.2 SNMP Configuration

To configure SNMP proceed to System -> Services from the side/top System menu.

- Select SNMP service **Configure** Button



NOTE: Before configuring SNMP service, the SNMP service must be stopped.



- Select SNMP Version 1&2 or 3
- SNMP Version 3 requires user authentication
  - Please specify a username and password
- Click **Apply** to save.

18.3 SNMP Test

In order to confirm NSG responds to SNMP requests, one can use number of standard snmp client tools to obtain system information.

```
snmpwalk  -c public -v 1 <nsg ip address or dns name>

or

snmpwalk  -c public -v2c  <nsg ip address or dns name>
```

This should show some basic information about the system including:

```
SNMPv2-MIB::sysDescr.0 = STRING: Linux nsg-nc-43.sangoma.local 2.6.39-4.sng2 #1 SMP Wed Dec 21 17:26:48 EST
2011 i686
SNMPv2-MIB::sysObjectID.0 = OID: NET-SNMP-MIB::netSnmpAgentOIDs.10
DISMAN-EVENT-MIB::sysUpTimeInstance = Timeticks: (176243) 0:29:22.43
SNMPv2-MIB::sysContact.0 = STRING: Root <root@localhost> (configure /etc/snmp/snmp.local.conf)
SNMPv2-MIB::sysName.0 = STRING: nsg-nc-43.sangoma.local
SNMPv2-MIB::sysLocation.0 = STRING: Unknown (edit /etc/snmp/snmpd.conf)
SNMPv2-MIB::sysORLastChange.0 = Timeticks: (0) 0:00:00.00
SNMPv2-MIB::sysORID.1 = OID: SNMPv2-MIB::snmpMIB
SNMPv2-MIB::sysORID.2 = OID: TCP-MIB::tcpMIB
SNMPv2-MIB::sysORID.3 = OID: IP-MIB::ip
SNMPv2-MIB::sysORID.4 = OID: UDP-MIB::udpMIB
SNMPv2-MIB::sysORID.5 = OID: SNMP-VIEW-BASED-ACM-MIB::vacmBasicGroup
SNMPv2-MIB::sysORID.6 = OID: SNMP-FRAMEWORK-MIB::snmpFrameworkMIBCompliance
SNMPv2-MIB::sysORID.7 = OID: SNMP-MPD-MIB::snmpMPDCompliance
SNMPv2-MIB::sysORID.8 = OID: SNMP-USER-BASED-SM-MIB::usmMIBCompliance
…
IF-MIB::ifDescr.2 = STRING: eth0                    (Primary Ethernet Port)
IF-MIB::ifDescr.3 = STRING: eth1                    (Secondary Ethernet Port)
```

```
IF-MIB::ifDescr.4 = STRING: eth2                (Media Transcoding Port)
IF-MIB::ifDescr.6 = STRING: eth1.1302           (VLAN)
IF-MIB::ifDescr.7 = STRING: eth1.1301           (VLAN)
IF-MIB::ifDescr.8 = STRING: eth1.1300           (VLAN)
IF-MIB::ifDescr.11 = STRING: w1g1               (T1/E1 TDM Port)
```

To determine the T1/E1 or Ethernet State

```
IF-MIB::ifAdminStatus.1 = INTEGER: up(1)
IF-MIB::ifAdminStatus.2 = INTEGER: up(1)
IF-MIB::ifAdminStatus.3 = INTEGER: up(1)
IF-MIB::ifAdminStatus.4 = INTEGER: up(1)
IF-MIB::ifAdminStatus.6 = INTEGER: up(1)
IF-MIB::ifAdminStatus.7 = INTEGER: up(1)
IF-MIB::ifAdminStatus.8 = INTEGER: up(1)
IF-MIB::ifAdminStatus.11 = INTEGER: up(1)
IF-MIB::ifOperStatus.1 = INTEGER: up(1)
IF-MIB::ifOperStatus.2 = INTEGER: up(1)         (Primary port eth0 status  – In this example eth0 link is up)
IF-MIB::ifOperStatus.3 = INTEGER: down(2)
IF-MIB::ifOperStatus.4 = INTEGER: up(1)
IF-MIB::ifOperStatus.6 = INTEGER: down(2)
IF-MIB::ifOperStatus.7 = INTEGER: down(2)
IF-MIB::ifOperStatus.8 = INTEGER: down(2)
IF-MIB::ifOperStatus.11 = INTEGER: down(2)      (T1/E1 TDM Port Status – In this example T1/E1link is down, in alarm)
```

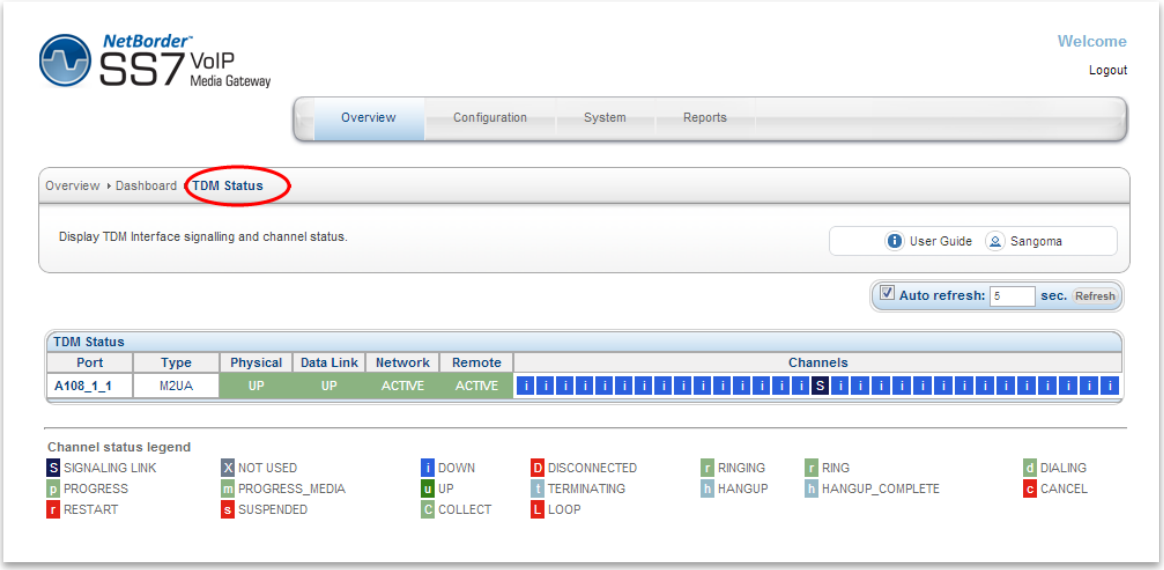
Note that all TDM interfaces/spans have the following nomenclature: “w<SPAN>g<CHAN>”

- w1g1 translates to SPAN 1
- w2g1 translates to SPAN 2
- w31g1 translates to SPAN 31

19 Troubleshooting

In any network troubleshooting it is best to start from the bottom.

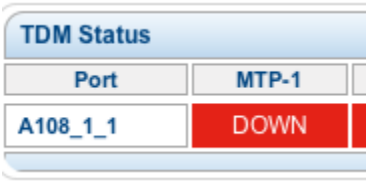
Physical Layer: T1/E1 Ports	T1/E1 Alarms and Statistics Command to read T1/E1 Alarms • wanpipemon -i w1g1 -c Ta
Data Link and Network Layers: ISUP Termination MTP2 Link MTP3 Link ISUP Link	Trace/Capture TDM Signaling channel From GUI: Reports -> Packet Capture • Open pcap file in Wireshark From SSH • wanpipemon -i w1g1 -c trd #hdlc decoded • wanpipemon -i w1g1 -c tr #raw • wanpipemon options ○ -rx rx only ○ -tx tx only ○ -diff different frames only
Data and Network Link Layers: Megaco MG + SG SCTP M2UA M3UA Megaco	Trace/Capture Ethernet Port From GUI: Reports -> Packet Capture • Open pcap file in Wireshark



19.1 Physical Layer

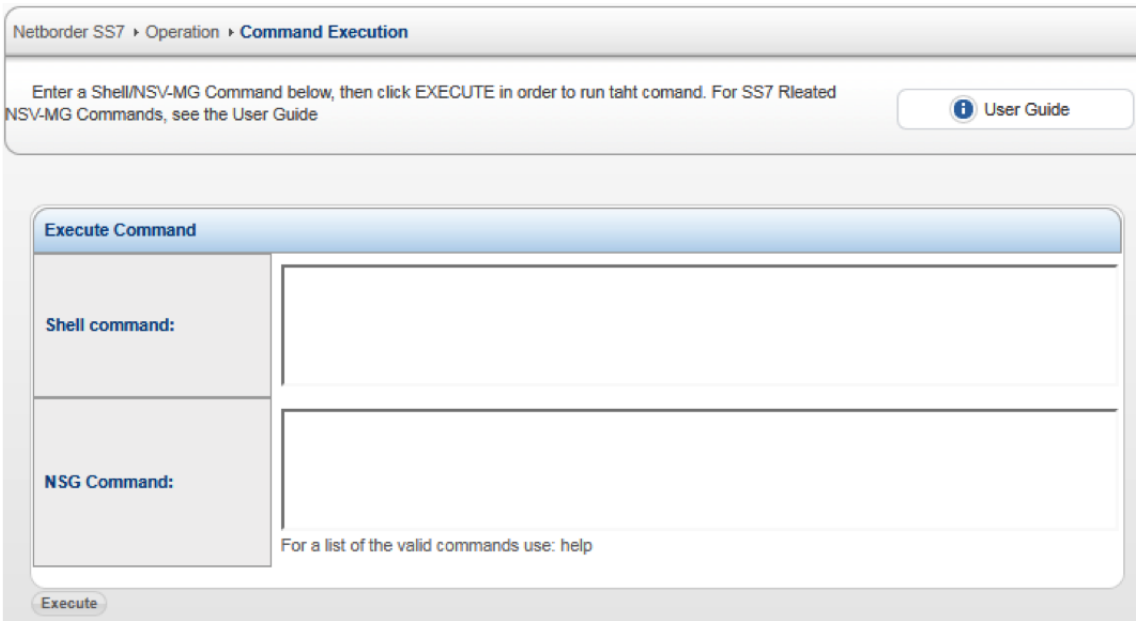
The first step in troubleshooting any connectivity issue is troubleshooting the physical layer. Identifying whether a user has a physical layer issue is by using the **TDM Status** page and checking the MTP-1/M2UA column.

If the column is listed as "DOWN" for that particular port, proceed with troubleshooting the physical layer.



When physical layer is down, all layers above the physical layer will also be in a "DOWN" or "TRYING" state.

In order to start troubleshooting, the user must proceed to the "Command Execution" page, which is located under the "Configuration" menu.



The best way to troubleshoot physical layer issues is through the **shell** command option. Below is a list of commands that can be run within the shell command section to help diagnose issues:

19.1.1 NSG TDM Driver related commands

- wanpipemon -i wXg1 -c Ta
  - where X is the span/port number in question.
  - Span number can be found in GUI -> TDM section for each physical T1/E1 port
  - Output low level T1/E1 Alarms
- wanrouter status
  - Output wanpipe physical status statistics

19.1.2 T1/E1 Port Status

The first step in debugging physical layer issues would be to check whether wanrouter status reports the line "Connected" or "Disconnected". To do this, within the "Shell Command" textbox, enter the command "wanrouter status". It will return a result like the one below:

-> **wanrouter status**

Shell Command

```
Devices currently active:
        wanpipe1


Wanpipe Config:

Device name | Protocol Map | Adapter   | IRQ | Slot/IO | If's | CLK | Baud rate |
wanpipe1    | N/A          | A101/1D/A102/2D/4/4D/8| 169 | 4       | 1    | N/A | 0         |

Wanrouter Status:

Device name | Protocol | Station | Status        |
wanpipe1    | AFT TE1  | N/A     | Disconnected  |
```

All the devices running on a NSG system will be listed as a "wanpipe" device. In this example, "wanpipe1" is being reported as "Disconnected", which tells us that the physical layer is in fact in a "DOWN" state.

19.1.3 T1/E1 Port Debugging

The next step would be to check where the issue lies.

To do this, the user would need to run the command

- wanpipemon -i wXg1 -c Ta
  (where X stands for the wanpipe number).

In this example, "wanpipe1" is in a disconnected state, therefore the interface name would be "w1g1".

The command returns an output similar to the one below:

-> wanpipemon -i w1g1 -c Ta

Shell Command

```
***** wlg1: E1 Rx Alarms (Framer) *****

ALOS:    OFF      | LOS:  ON
RED:     ON       | AIS:  OFF
LOF:     ON       | RAI:  OFF

***** wlg1: E1 Rx Alarms (LIU) *****

Short Circuit:  OFF
Open Circuit:   ON
Loss of Signal: ON

***** wlg1: E1 Tx Alarms *****

AIS:     OFF      | YEL:  ON


***** wlg1: E1 Performance Monitoring Counters *****

Line Code Violation     : 0
Far End Block Errors    : 0
CRC4 Errors             : 0
FAS Errors              : 0


Rx Level        : < -44db
```

Check for Short or Open Circuit	Possibly a bad cable • Try another cable Possibly a bad T1/E1 port on NSG • Unplug the E1 from NSG and run NSG self-test to confirm
Check Rx Level	if equal to -44db • No Cable • Circuit disconnected on Telco side • No power on the line If lower than -2.5db (-10db-20db) • Cable problem, bad cable, short • Low signal strength If equal to -2.5db • E1 signal strength is perfect

Check Alarms	

RED	Indicates the device is in alarm
LOF	(Loss of Framing). Raised after four consecutive frames with FAS error. If RAI and AIS alarms are not indicated, verify that you have selected the proper line framing (i.e T1: ESF, D4, E1:CRC4, NCRC4..etc)
LOS	(Loss Of frame Signal)
AIS	(Alarm Indication Signal): typically know as a BLUE Alarm. all-ones signal transmission to the receiving equipment to indicate that an upstream repeater (telco equipment) is in alarm, due to upstream transmission fault, either from another repeater or from the telco itself. If ONLY AIS:ON then contact your telco with this information (RAI:ON can also be a possibility in this case as well) Example call diagram of the scenario: Sangoma card <----------repeater <----------Telco
RAI	(Remote Alarm Indication): Indicates that the Far end (typically the Telco) is in RED alarm state and sending that message over the line. If ONLY RAI:ON then Telco is down, or TX wire in T1/E1 cable is damaged. You will also get this alarm, and only this alarm, if your framing is incorrect. This setting can be changed in the TDM Section.
Short Circuit	The wires in your cable connected to the port are crossed. If you see this alarm, check the pinouts for the cable you are using. You may also be plugging in the wrong form of cable (straight-through, or cross-over)
Open Circuit	No line plugged into the port. Make sure that your connector is plugged in and the wiring is making a good connection. If this alarm is on, you will also Rx Level='-36'->'-44'.
Loss of Signal	Cabling issue. Check the health of the cable plugged into the port, as well as its connection to the port it is plugged into.

		36 -> -44. It is typical to have this alarm triggers in combination with 'Open Circuit' if there is an issue with the physical connection
	YEL	When the equipment enters a Red-Alarm state, it returns a Yellow-Alarm back up the line of the received OOF. A typical scenario would be mis-configuration during the Sangoma card configuration (i.e selected CRC4 vs NCRC4). In this type of scenario also LOF and RED alarms will be triggered.
	Line Code Violation	This occurs upon a bipolar violation
	Far End Block Errors	is reported by the upstream end of the PHY (the wire between you and the switch) on the out-of-band management channel. This means the other end of the line received bad data from you. Possible reason are: line noise, corroded wires..etc. Also, check line Framing (E1: CRC4 vs NCRC4)
	CRC4 Errors	This occurs when the CRC polynomial calculation performed before transmission does not match the CRC calculation done upon reception.
	FAS Errors	(Frame alignment signal error). One or more incorrect bits in the alignment word
Check Clock	Note that NSG will not come out of Alarm state if there is NO clock on the T1/E1 line. If NSG configured for **NORMAL** (slave) clock • Re configure to MASTER clock • If E1 comes UP • Then there is NO clock on the line !!! • Contact the Telco	

19.2 TDM Signaling Link Debugging

If GUI TDM Status -> Data Link (MTP2) – is **DOWN**

Proceed to GUI -> Reporting -> Packet Capture

Check for Rx signaling packets.	Proceed to GUI -> Reporting -> Packet Capture Trace RX only packets on TDM T1/E1 port that contains a signaling link • eg: w1g1 – port 1 • eg: w2g1 – port 2 • Select RX Only Start Trace Wait a minute Stop Trace Download and open in Wireshark Check for RX FISSU and LSSU If NO RX packets at all • Then there is no signaling traffic on the T1/E1 timeslot There is probably only idle pattern • Telco needs to turn on the MTP2 Link OR • There is NO MTP2 link on this E1 timeslot
Check for Tx signaling packets	Proceed to GUI -> Reporting -> Packet Capture Trace RX only packets on TDM T1/E1 port that contains a signaling link • eg: w1g1 – port 1 • eg: w2g1 – port 2 • Select TX Only Start Trace Wait a minute Stop Trace Download and open in Wireshark Check for TX FISSU and LSSU If NO TX packets at all • Then MTP2 link might not be activated If in M2UA bridge mode, the M2UA must be active. Only when M2UA becomes active will the MTP2 link be activated. Caution: TX trace will only capture different FISU and LSSU due to hw optimization.
Capture all Signaling traffic and open in Wireshark	Proceed to GUI -> Reporting -> Packet Capture Trace Different only packets on TDM T1/E1 port that contains a signaling link • eg: w1g1 – port 1

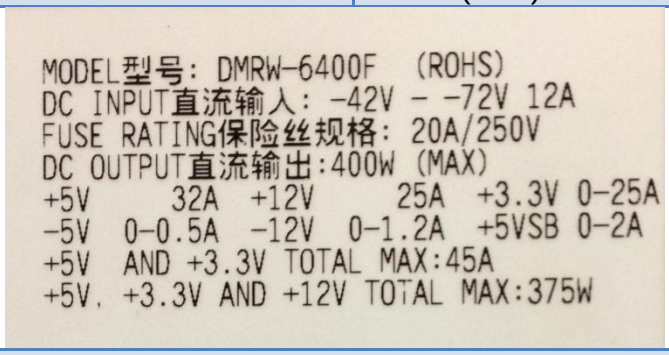
	• eg: w2g1 – port 2 • Select Different only packets Start Trace Wait a minute Stop Trace Download and open in Wireshark MTP2 • Check for LSSU size mismatch ISUP • Check for wrong OPC/SPC, APC, DPC

20 Appendix

20.1 Redundant DC PSU

Sangoma NSG appliances come with redundant DC power supply.

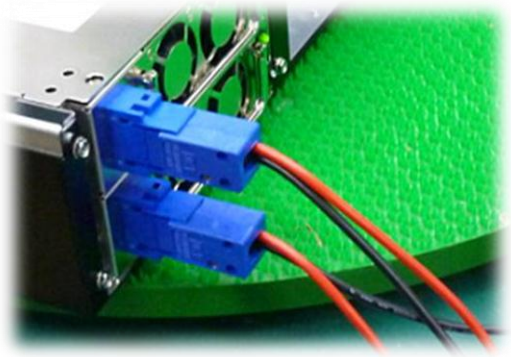
VOLTAGE	**DC -36V ~ -72V**
INPUT CURRENT:	12.0A (RMS). FOR -48 VDC
INRUSH CURRENT	20A (Max)
DC OUTPUT	400W (Max)



- **TEMPERATURE RANGE : OPERATING 100C --- 400C**
- **HUMIDITY:OPERATING:20%-95%, NON-OPERATING:10%-95%**
- **REMARKS:85% IS NORMAL CONDITION AND 95% IS WITH SPECIAL COATING PROCESS**
- **HOLD UP TIME: 1.6 ms MINIMUM AT FULL LOAD & NOMINAL INPUT VOLTAGE**
- **DIELECTRIC WITHSTAND: INPUT / OUTPUT 1500 VAC FOR 1 SECOND**
- **INPUT TO FRAME GROUND 1500 VAC FOR 1 SECOND**
- **EFFICIENCY: 65% TYPICAL, AT FULL LOAD**
- **POWER GOOD SIGNAL: ON DELAY 100 ms TO 500 ms, OFF DELAY 1 ms**
- **OVER LOAD PROTECTION: 130 ± 20%.**
- **OVER VOLTAGE PROTECTION: +5V → 5.5V ~ 7.0V, + 3.3V → 4.0V ~ 4.5V**
- **SHORT CIRCUIT:+5V,+12V,+3.3V**

- **EMI NOISE FILTER: FCC CLASS A, CISPR22 CLASS A**
- **SAFETY: UL 1950, CSA 22.2 NO/ 950, TÜV IEC 950**
- **REMOTE ON / OFF CONTROL**
- **THE UNIT SHALL ACCEPT A LOGIC OPEN COLLECTOR LEVEL WHICH WILL DISABLE / ENABLE ALL THE OUTPUT VOLTAGE (EXCLUDE +5V STANDBY), AS LOGIC LEVEL IS LOW, OUTPUTS VOLTAGE WERE ENABLE, AS LOGIC LEVEL IS HIGH, OUTPUTS VOLTAGE WERE DISABED**
- **COOLING : TWO 40 mm DC FANS (MODULE)**
  AC INLET IN EACH MODULE

20.1.1 DC PSU Cables



Connecting cables to a power supply depends on the remote power source.

Power Source Type	*Black Wire*	*Red Wire*
If power source -48V	-48V	0V (Ground)
If power source +48V	0V (Ground)	+48V

- The PSU **has** voltage reverse protection.
  If the red and black wires are connected the wrong way, the system will not power up. But there will be **no** damage to the PSU or the system.

20.1.2 Hot-swap procedures

Please refer to the following when either power module is defective.

- Locate the defective power module by examining the individual LED (if LED is distinguished, it indicates the power module is defective).

  *** WARNING
  please perform the following step carefully; otherwise, it may cause the whole system shutdown.
  *** WARNING

  Please do not remove the defective power module until you have worn gloves to keep from been burned. This is due to the cover of the power module is used as heat sink for cooling. Usually, its temperature is around 50-60 degree Celsius under full load condition.

- Loose the screws of power module bracket.
- Plug out the defective power module.

  *** WARNING
  please put aside the power module to wait for cooling down. Keep other people from toughing it until it is cooled.

- Replace a new / GOOD power module. Insert the power module into the power system till to the end.
- Check the LED of the power module, which should be in GREEN.
- Check the warning LED indicating the status of total power system, which should be in GREEN.
- Tighten the screws of the power module.
- If you want to test this new power module and simulate the defective situation, please refer to Section 1.7 Installation & Testing.

  Remarks: If the DC fan of the power module fails, you have to replace the power module. Please follow the Hot-Swap Procedures for replacement.

20.1.3 Trouble Shooting

If you have followed these instructions correctly, it should function normally.
Some common symptoms are, the system doesn't work, buzzer alarms, shutdown after running a very short period,… etc. If so, please check the following steps to verify and correct it.

- Check all connection (if pinouts is correct, if any connection loosed, if the direction is incorrect,… etc.).
- Check if any short-circuit or defective peripherals by plugging out the power connector from each peripheral, one at a time. Shall the system functions again, you have solved the problem.
- Once you hear the buzzer sound or see the warning LED in RED, please check,
- If the loading is under the minimum or over the maximum load of each channel.
- If the power source is well connected and supplied. Shall the above condition is happened, please disconnect the power source and wait for 2-3 minutes to release the protection status; then test it again.
- If buzzer keeps alarming or LED indicates the power module failure, please locate which power module is defective. Perform hot-swap procedures (ref. to Sec. 1.8 Hot-Swap Procedures). Return the defective power module back to your vendor for RMA procedure.
- If you cannot fix the problem, please contact your vendor for supporting.
  Note:
  * The description stated herein is subject to change without prior notice.
  * All brand names and trademarks are the property of their respective owners.

21 Theory

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Outline

- VoIP Network Introduction
- SS7 ISUP Overview
- M2UA Sigtran Overview
- Megaco Overview
- H323 Overview
- SIP Overview
- Sangoma NSG Overview

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VoIP Introduction

- Viable alternative / addition to traditional circuit-switched telephony.
- Large companies (which have their own private global IP networks) already realize the benefits of VoIP networks.
- VoIP is now penetrating the wider population of small offices and residential Internet users.

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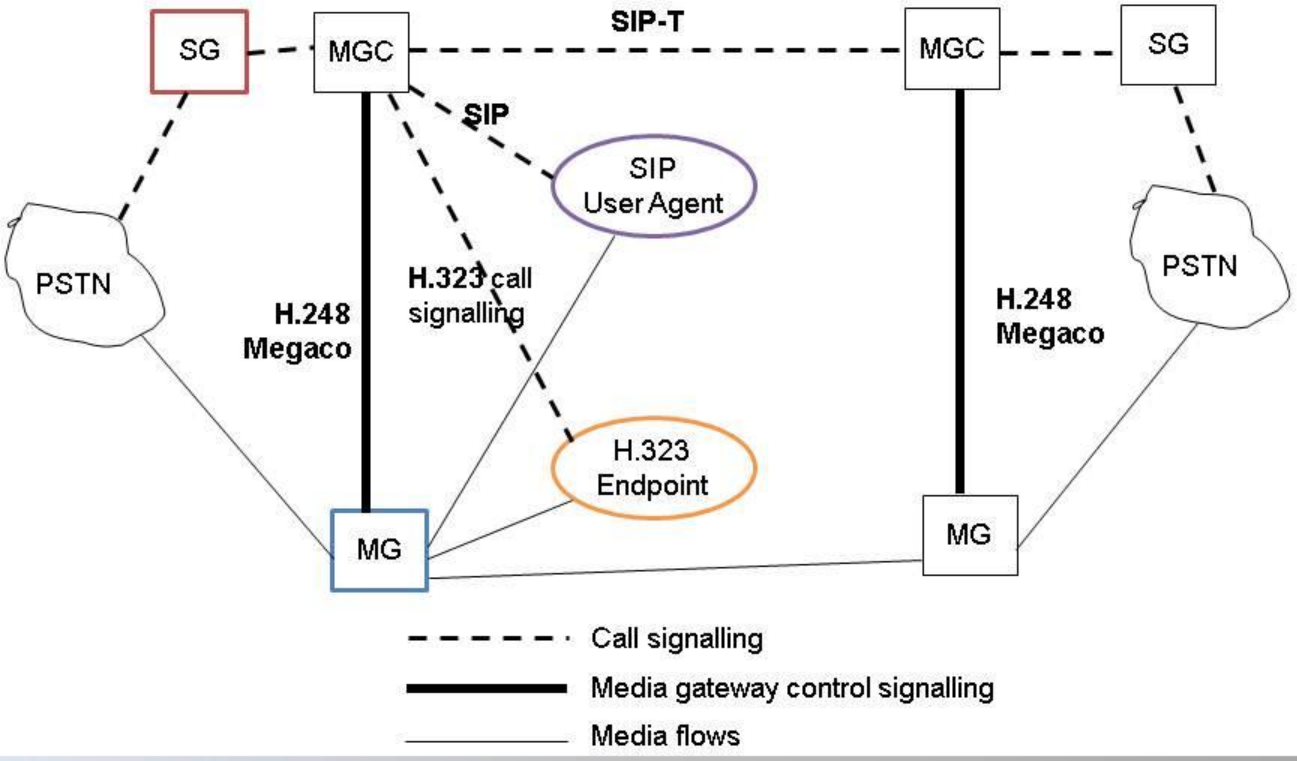
VoIP Introduction

- Today's VoIP quality of service (QoS) has improved tremendously from initial generations.
- However, the drawbacks are still apparent.
- The architecture of a carrier-grade VoIP network that provides telephony service for a wider customer base cannot be completely based on the peer-to-peer architecture of current VoIP call-processing standards.

VoIP Introduction

- Central management and call-routing functions are needed
- Interoperable, easy-to-design, cost effective client is imperative
- Megaco addresses the relationship between the Media Gateway (MG) and the Media Gateway Controller (MGC)
- A Master/Slave protocol that removes intelligence from MGs

Megaco/VoIP Network



VoIP Benefits

- Removing the signaling to a fast server is more practical than trying to integrate it into the MG.
- New services can be introduced without requiring any customer premises equipment (CPE) upgrades.
- Handled by simply upgrading the centralized software that contains the intelligence for implementing services.

SS7 Overview
10

SS7 Introduction

- What is SS7?
- SS7 Network Elements
  - SCP, STP, and SSP
- SS7 Link Types
  - A and F links
- Basic Call Messages
  - IAM, ACM, ANM, REL, and RLC
- SS7 Signaling Stack
  - MTP1, MTP2, MTP3, ISUP, and others
- SS7 Signaling Messages
  - FISU, LSSU, and MSUs
- SS7 Addressing
  - SPC, DPC, and APC

What is SS7?

- SS7 is a signaling control protocol
  - Primary signaling protocol in Carrier PSTN networks
  - Control messages include VOICE channel control such as "Place Call", "Hangup" ...
- Common Channel Signaling (CCS)
  - Signaling channel – carry telephony control messages (call,hangup...),
  - CICs – b-channels that carry media
  - One signaling channel can control many CICs
- Signaling System 7
  - CCSS7, C7, number 7, CCIS7
  - Official Standard by ITU in 1980

- Variants:
  ITU - Outside North America
  BT – United Kingdom
  Russia – Russia
  ANSI – North America
- Universal signaling via a suite of sub-protocols
  ISUP, TUP, DUP, etc.
- Each time you place and release a telephone call that extends beyond the local exchange, SS7/C7 signaling takes place

What is SS7?

- The SS7 network and protocol are used for:
  - Basic call setup, management and tear down
  - Wireless services, such as personal communications services (PCS), wireless roaming and mobile subscriber authentication
  - Local Number Portability (LNP)
  - Toll-free (800/888) and toll (900) wireline services
  - Enhanced call features, such as call forwarding, calling party name/number display and three-way calling
  - Efficient and secure worldwide telecommunications

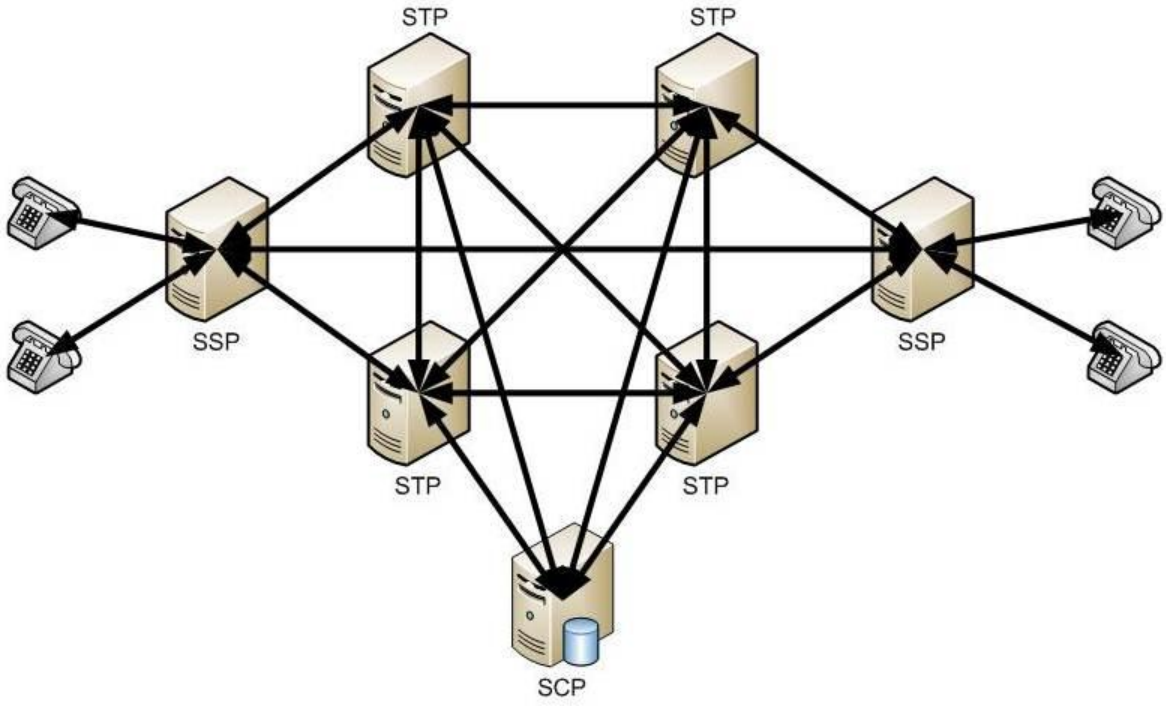
Signaling Links

- SS7 messages are transported over 56 or 64 kbps channels called signaling links
  - Note: 56kbps or 64kbps is SLOW ☺
- Signaling occurs out of band on dedicated T1/E1 ds0 timeslots rather than in-band on voice channels
- Example (F-Link):
  Timeslot 1 or 16 would be dedicated to carry SS7 traffic. While the rest of the channels would carry Voice.

SS7 Signaling Points

- There are 3 types of signaling points
  - Signal Switching Points (SSP)
    - Terminate signaling links
    - Start, end, and switch calls
  - Signal Transfer Points (STP)
    - Main routing switches
  - Signal Control Points (SCP)
    - Switches attached to databases
- Each signaling point in the SS7 network is uniquely identified by a numeric point code

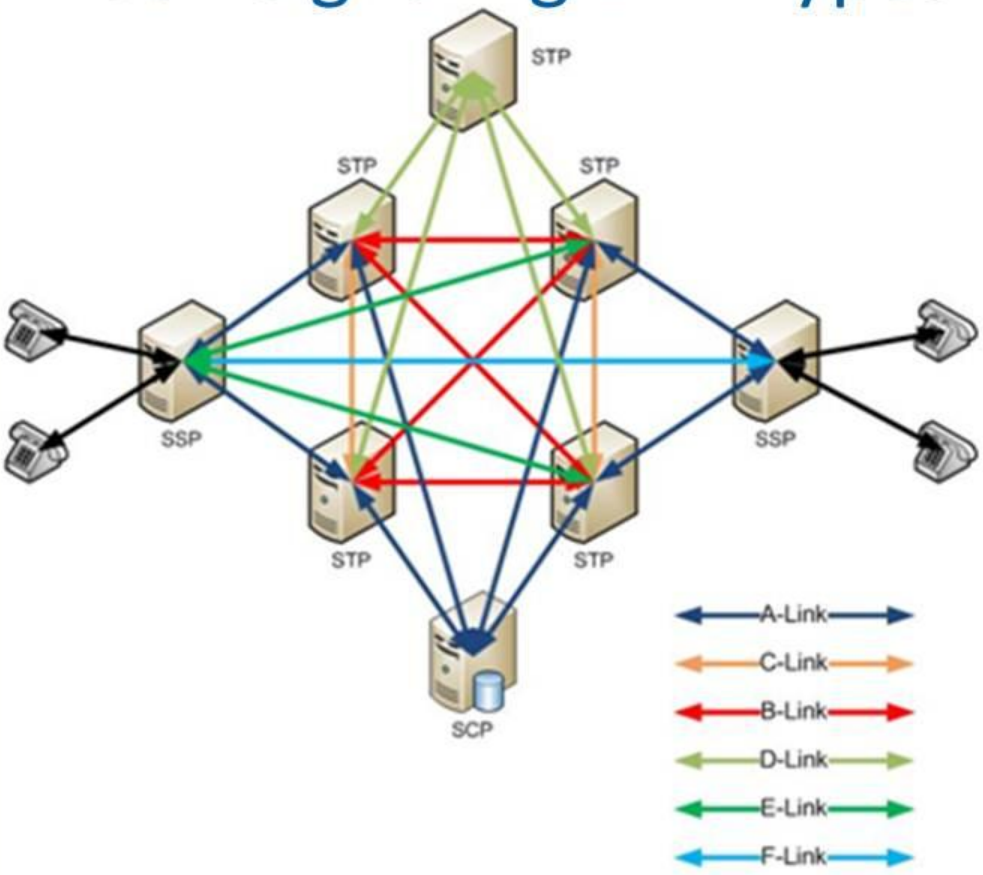
SS7 Signaling Points



SS7 Signaling Link Types

Link Type	Description
A Link (access)	Link between an SSP or SCP to an STP Its purpose is to deliver signaling messages
B Link (brigde)	Link between 2 mated STP pairs
C Link (cross)	Link between 2 STPs making them a mated pair
D Link (diagonal)	Link between 2 mated STPs pairs (different hierarchical levels)
E Link (external)	Link between an SSP and a secondary mated STP
F Link (fully associated)	Link between 2 SSPs

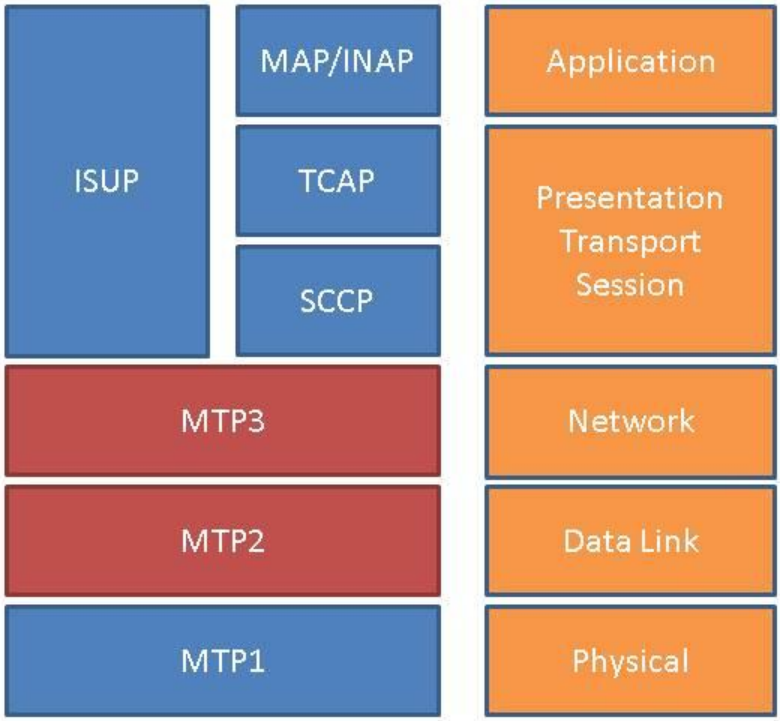
SS7 Signaling Link Types



SS7 Link Types: Most Common

- At the SS7 Edge one usually deals with only 2 link types
  - A Link & F Link
- F Link
  - Direct connection between two SSP's
  - Equivalent of Back to Back connection between two IP Servers using a back to back Ethernet cable. ie. No router in the middle
  - Adjacent Point Code is EQUAL to Destination Point Code.
- A Link
  - SSP to STP connection
  - Equivalent to an IP Server connected to an IP Router which provides end to end routing service.

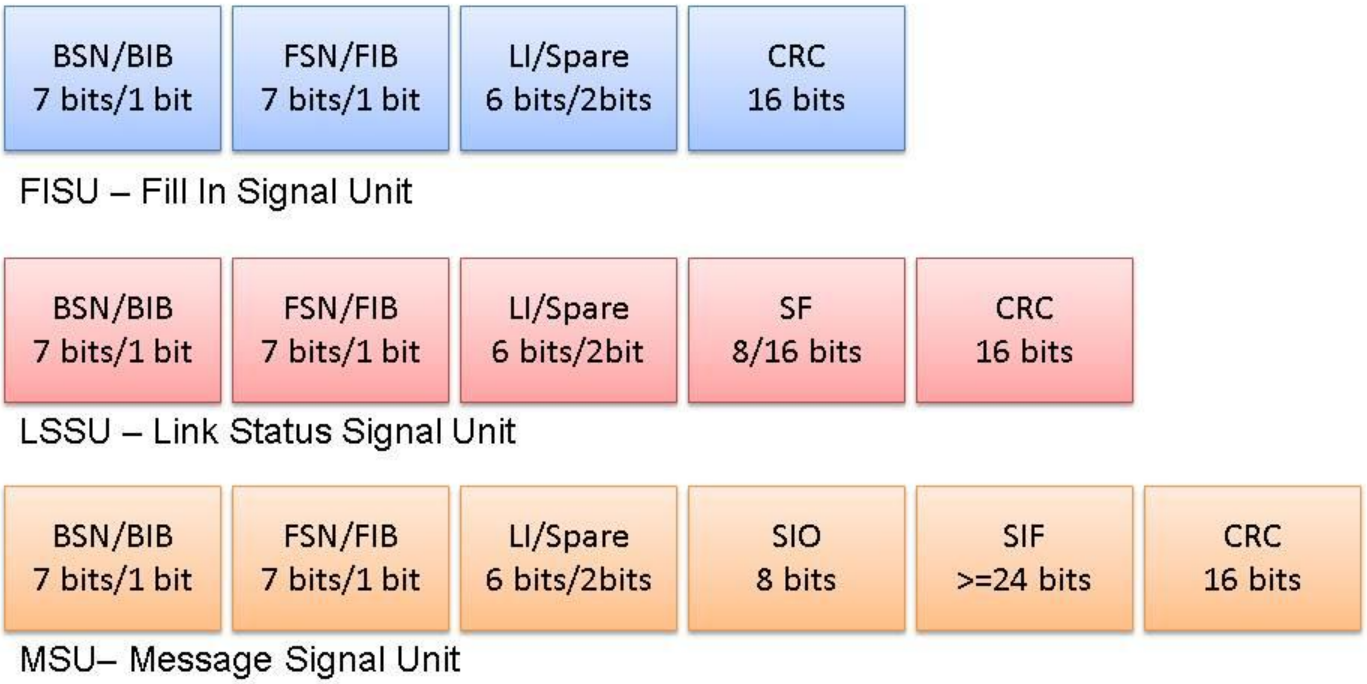
SS7 Stack Layers



MTP 2 Layer

- Data Link Layer Protocol
- Layer 2 in OSI Model
- Ensures reliable communications on a signaling link
- Error checking, flow control, sequence checking and data retransmission
- Line is filled with small packets which allows fast error detection.
- Guarantees error free and reliable data delivery to layers above.
- Defined in ITU Q.702, Q.703. ANSI equivalents exist.

MTP2 Messages



FISU Frames

- Used to idle the signaling link and keep track of transmitted and received frames
- Even though HDLC framing is used on signaling linking frames; MTP does not allow for more then 2-3 idle flags to be transmitted, instead FISU frames are transmitted repeatedly
- This allows very fast error detection on the network.
- Even though the SS7 network links are slow:
  - Usually 56kbps or 64kbps

LSSU Frames

- Used to sync the state machines on both sides of the link
- Frame length can be 1 or 2 (normally 1 and normally switches are supposed to be compatible with both) ***
- 6 types of frames identified by the value of the status field (Status Indication X)
- Service Information Octet (ITU Q.703, 11.1.3, page 23):

Val	Status Indication	Acronym	Meaning
0	O: Out of Alignment	SIO	Link not aligned; attempting alignment
1	N: Normal Alignment	SIN	Link is aligned
2	E: Emergency Alignment	SIE	Link is aligned
3	OS: Out of Service	SIOS	Link out of service; alignment failure
4	PO: Processor Outage	SIPO	MTP2 cannot reach MTP3
5	B: Busy	SIB	MTP2 congestion

*** Caution: source of configuration errors.
Same value must be configured on both Telco and User side.

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MSU Frames

- Used to carry higher layer traffic (MTP3 mng/test, ISUP, SCCP frames)
- Service Information Field divides into 2 fields:
  - Sub-Service Field, type of layer 4 traffic
  - Service Indicator, network indicator (aka NI)
- Service Indicator(ITU Q.704, 14.2.1, page 70):
  - 0, network management messages
  - 1, network test messages (For SLMT/A tests msgs)
  - 3, SCCP
  - 5, ISUP
- Sub-Service Field (ITU Q.704, 14.2.2, page 71):
  - 0, International Network
  - 1, Spare (international use only)
  - 2, National Network
  - 4, Reserved for national use

  TODO: Show the MSU diagram

MTP3 Layer

- Network Layer Protocol
- Layer 3 in OSI Model

- Provides logical end to end delivery of data in an SS7 network

- Addressing, routing, and congestion control

- Defined in ITU Q.703, Q.704. ANSI equivalents exist.

MTP3 Messages

- The primary purpose of this protocol level is to route messages between SS7 network nodes in a reliable manner. This is achieved by using
  - Signaling Message Handling (SMH): Handles the routing of messages based on the destination node address.
  - Signaling Network Management (SNM): Handles network failures to assure messages reach their destination.

MTP3 – Node Addressing

- ITU
  - 14 bits, 3-8-3 format or decimal
- ANSI
  - 24 bits, 8-8-8 format
- SPC
  - Self Point Code
- APC
  - Adjacent Point Code
- DPC
  - Destination Point Code
- OPC
  - Originating Point Code
- SLS/SLC
  - Signaling Link Selector/Code, identifiers used for load sharing across linksets and links

MTP3 Layer

- A comparison between the MTP3 and the IP protocol layers

IP	MTP3
Source IP Address	Originating Point Code
Destination IP Address	Destination Point Code
Protocol	Service Indicator
Precedence (part of TOS field)	Priority
Data	User Data

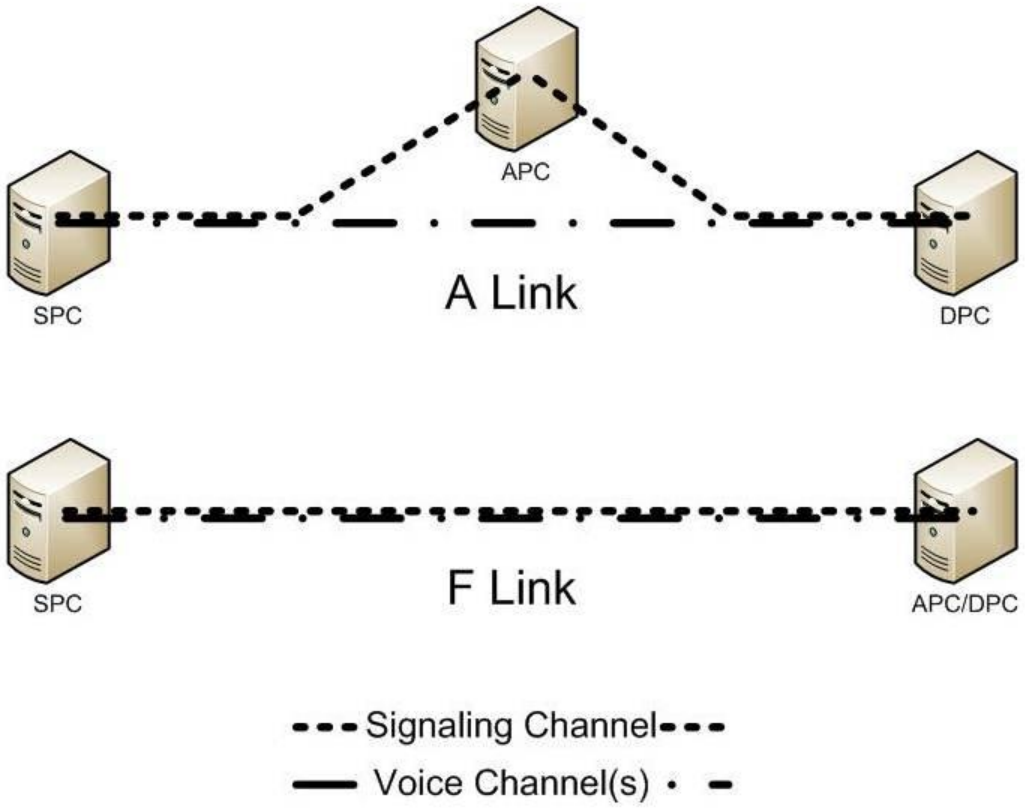
MTP3 Signaling Network Management

- Signaling Network Management is divided into three processes:
  - Traffic management keeps ISUP and SCCP messages moving towards destination
  - Route management exchanges info about routing status between nodes
  - Link management activates, deactivates and restores signaling links. After the MTP2 layer aligns, the MTP3 will send Signaling Link Test Control (SLTC) messages to bring up the MTP3 layer. The SLTC message consists of a STLM (Message) and SLTA (Acknowledge)

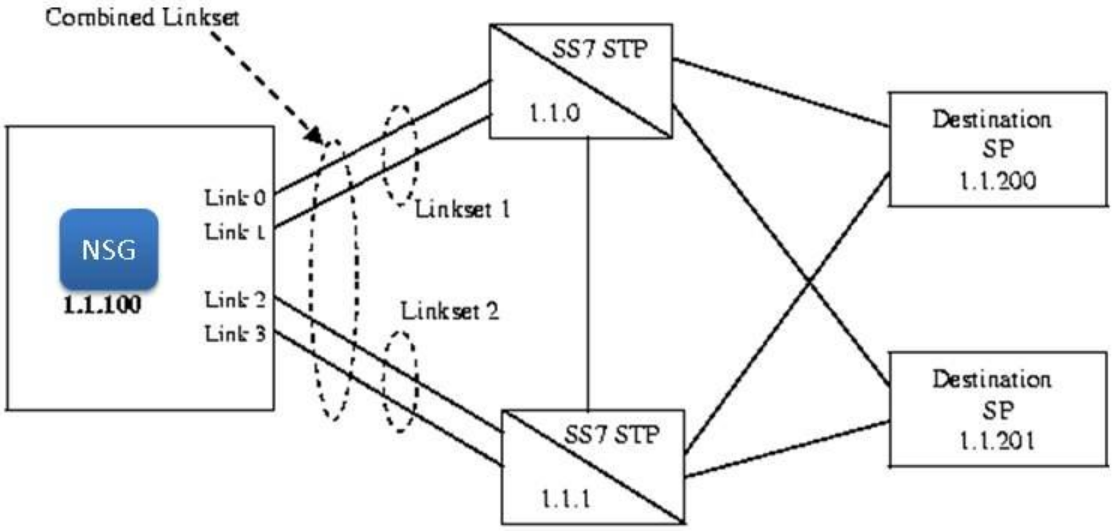
SS7 Links, Linksets and Routes

- A **Link** defines the channel where the signaling data is passed on the network
- A **Linkset** is constituted of multiple links that pass through the same APC
- A **Route** is determined by the path from the OPC to the DPC
- NSG 5.0 supports
  - 32 signalling links
  - 12 Originating Point Codes OPC
  - 12 Destination Point Codes DPC
  - 16 Linksets
  - A, F link support

Sample of Using SPC, APC and DPC



SS7 Network Diagram



Route 1,
DPC 1.1.0
Route 2,
DPC 1.1.1
Route 3,
DPC 1.1.200
Route4,
DPC 1.1.201

Linkset 1
Adj DPC 1.1.0
Route 1,0
Route 2,1
Route 3
Route 4
End

Linkset 2
Adj DPC 1.1.1
Route 1,1
Route 2,0
Route 3
Route 4
End

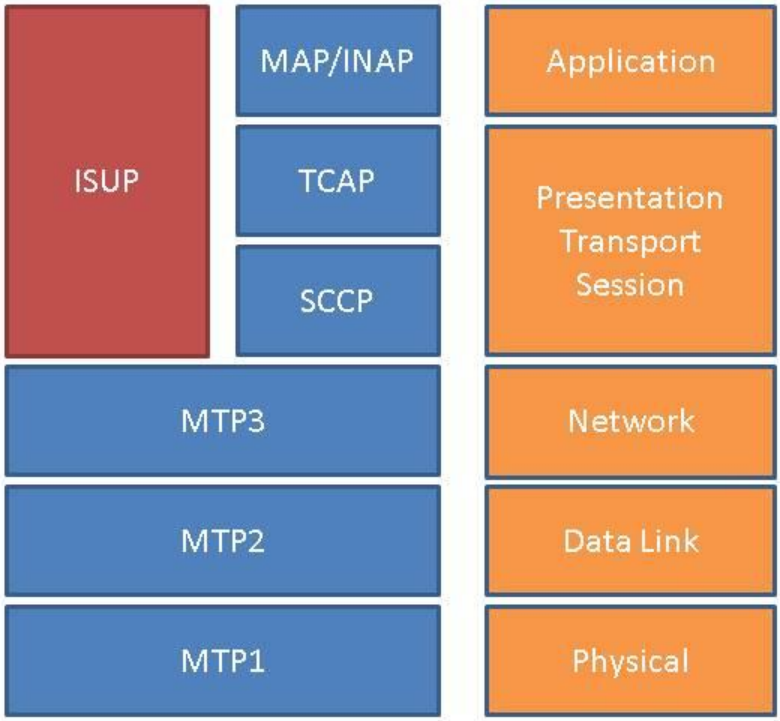
SS7 MTP2-MTP3 Quiz

- What are the 3 types of SS7 Signaling Points?
  1. Signal Switching Points (SSP)
  2. Signal Transfer Points (STP)
  3. Signal Control Points (SCP)
- What are the 2 signaling link types that link to an SSP?
  1. A link: SSP to STP link (access link)
  2. F link: SSP to SSP link (fully associated link)
- What is the role of the MTP2 and MTP3 layers?
  1. MTP2: Error checking, flow control, and sequence checking
  2. MTP3: Addressing, routing, and congestion control

SS7 MTP2-MTP3 Quiz (cont.)

- What are the 3 types of MTP2 messages?
    1. FISU – Fill in Signal Unit (Idle frame)
    2. LSSU – Link Status Signal Unit (Sync link)
    3. MSU – Message Signal Unit (Higher layer traffic)
- What are the 3 types of point codes?
    1. OPC: Originating point code
    2. DPC: Destination point code
    3. APC: Adjacent point code

SS7 Stack Layers



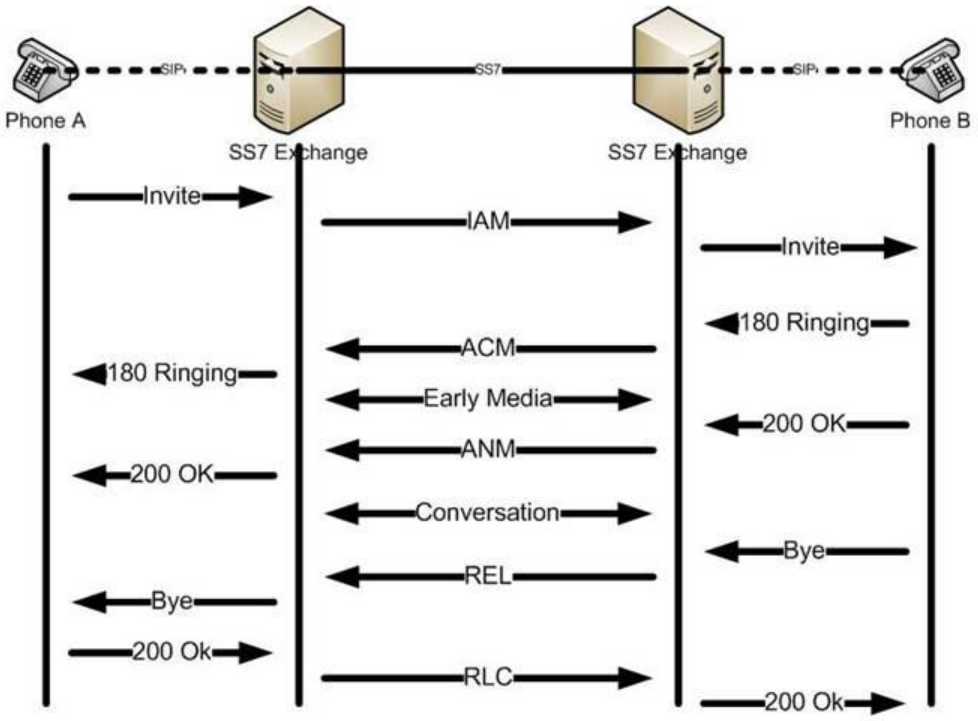
ISUP

- Call Signaling protocol of SS7
- Controls the status of CICs
  - Resets, blocks, etc.
- Controls the call setup
  - Setup, acknowledge, hang-up, etc.
- CICs
  - Circuit Identification Code
  - Integer number that uniquely identifies a time slot on a span
  - Used by all messages to ID the specific channel
- ITU Q.762, Q.763. and Q.764
- ANSI (T1.113-YEAR)

Basic Call Messages

- IAM
  - Initial Address Message
  - Call request
- ACM
  - Address Complete Message
  - Acknowledges IAM
  - Starts early media
- ANM
  - Answer Message
  - Answers the call
- REL
  - Release
  - Request to hang-up a call
- RLC
  - Release Complete
  - Acknowledges a REL

Basic Call Flow



Comparison of ISUP and SIP

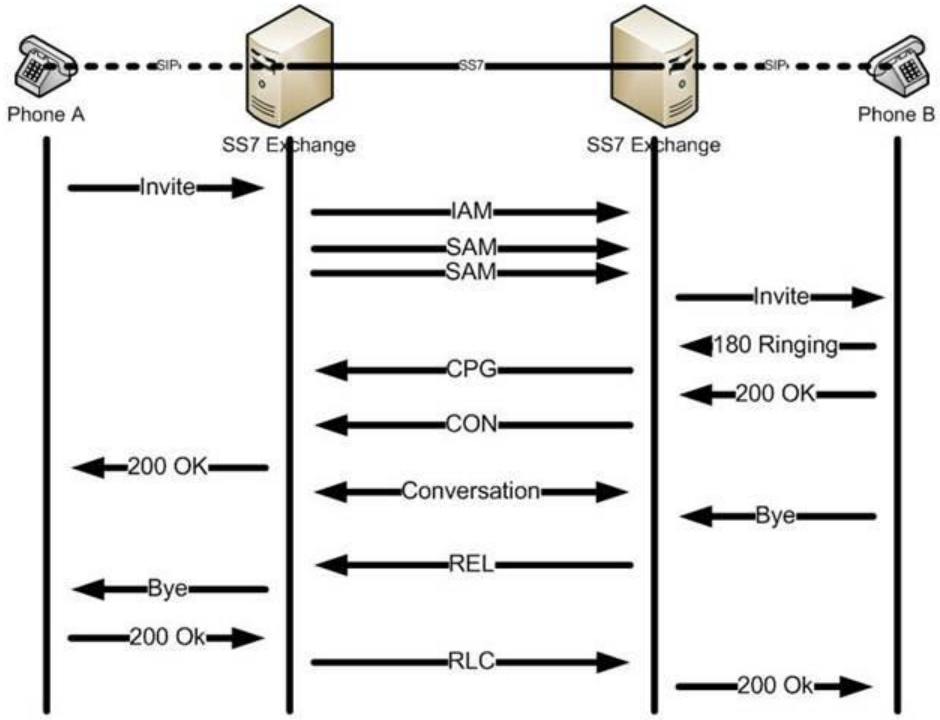
By comparing the ISUP and SIP call flow, we can establish that there are equivalent messages to both protocols.

ISUP Message	SIP Message
IAM	INVITE
ACM	183 Early Media
ANM	200 OK – Connect
REL	BYE
RLC	200 OK – ACK BYE

Advanced Call Messages

- SAM
  - Subsequent Address Message
  - Sends additional call setup info. (overlap)
- CPG
  - Call Progress Message
  - Used to signal additional info. during call setup
  - Really only used for Supplementary Services
- CON
  - Connect Message
  - Can replace ACM+ANM

Advanced Call Flow



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M2UA Overview
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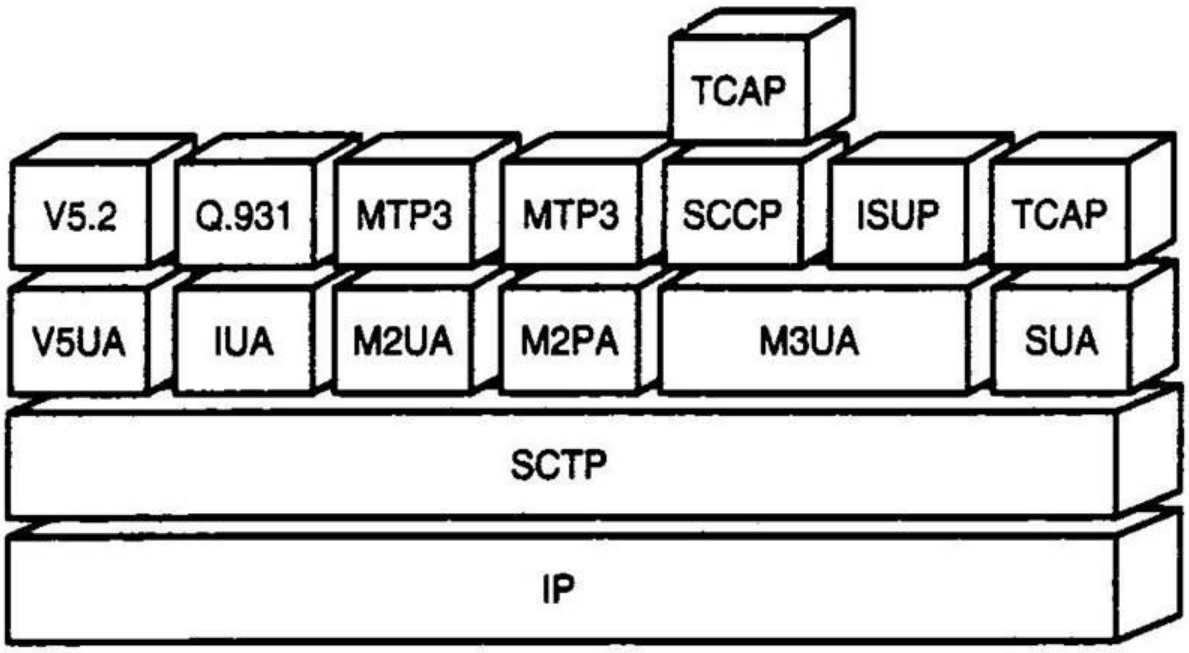
What is Sigtran?

- SS7 protocol transported over IP
- Two operation modes
  - SG: Signalling Gateway
  - ASP: ISUP Termination (softswitch)
- SG: Sigtran Signaling Gateway
  - Bridges T1/E1 SS7 packets to IP SS7 packets
  - Termination is done at a remote Softswitch (ASP)
- ASP: ISUP Termination
  - Termination of SS7 IP link
  - IP SS7 packets are terminated into ISUP protocol.
  - Usually implemented at the SoftSwitch
  - NSG supports both mode of operation.

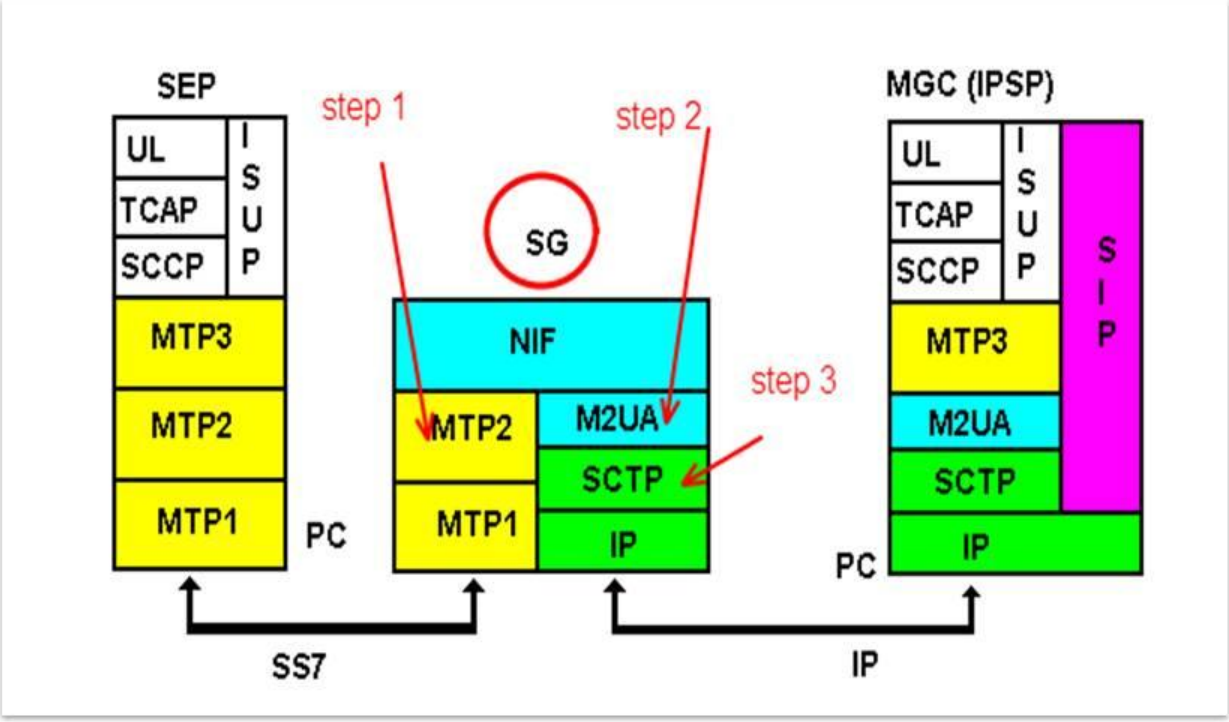
Sigtran Protocol Stack



Sigtran Signaling Gateway: SG

- Signaling Gateway: SG – is a **BRIDGE**
- M2UA
  - IP equivalent to MTP2
  - M2UA SG
    - **Bridges** T1/E1 MTP2 to M2UA IP
  - Benefit
    - SG is a dumb data pass through device.
    - All logic is in the SoftSwitch
- M3UA
  - IP equivalent of MTP3
  - M3UA SG
    - Bridges T1/E1 MTP3 to M3UA IP
  - SG is responsible for routing of SS7 data as its controlling MTP3 protocol. More complexity in SG.

M2UA SG



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M2UA Signaling Gateway

- Signaling Gateway: PSTN Side
  - SS7 MTP2 signaling over PSTN SS7 network interface.
- Signaling Gateway: IP Side
  - Transfer the SS7 MTP2 messages to and from MGC
  - IP Transport using SCTP Protocol

M2UA Signaling Gateway

- SCTP
  - explicit packet-oriented delivery (not stream-oriented)
  - sequenced delivery of user messages within multiple streams, with an option for order-of-arrival delivery of individual user messages,
  - optional multiplexing of user messages into SCTP datagrams,

M2UA Messages

M-ERROR
- indicate an error with a received M2UA message (e.g., an interface identifier value is not known to the SG).

M-SCTP_ESTABLISH
- establishment of a SCTP association to a peer M2UA node.

M-SCTP_RELEASE
- release of a SCTP association to a peer M2UA node.

M-SCTP_STATUS
- status of underlying SCTP association(s).

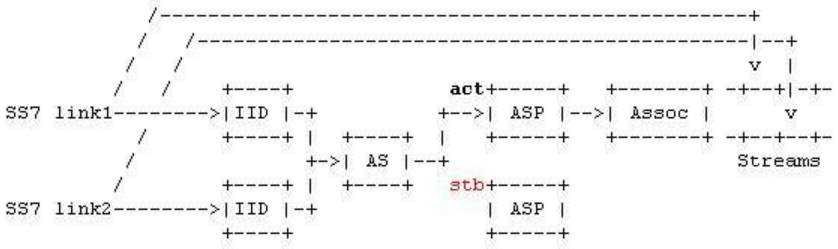
M-ASP_STATUS
- request the status of the Application Server Process from the M2UA layer.

M-ASP_MODIFY
- Modify the status of the Application Server Process. In other words, the Layer Management on the ASP side uses this primitive to initiate the ASPM procedures.

M-AS_STATUS
- Request or indicate the status of the Application Server.

M2UA Relationships

- An example of the logical view of the relationship between an SS7 link, Interface Identifier, AS and ASP in an SGP is shown below:

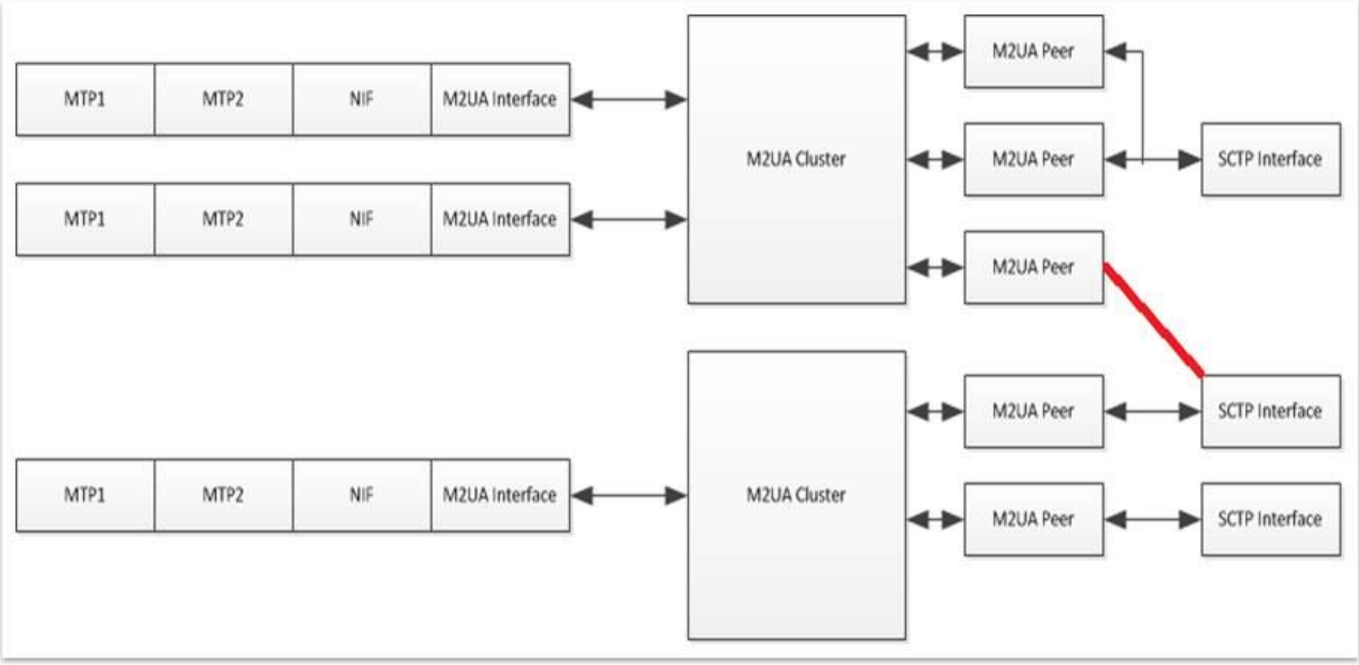


- M2UA keeps a map of Interface IDs
  - Interface ID's are mapped one-to-one with Physical MTP2 Links
- M2UA also keeps mapping between Interface ID's and STCP Streams.
  - Interface ID's do not have to be mapped one-to-one to streams
- AS and ASP manage the mapping between ID and Streams based on remote ASP state.

M2UA Bindings

- SCTP interfaces are standalone objects on which a ASP Peer bind to (regardless of its cluster).
  - 1 SCTP binds to 1 or more ASP Peers
  - 1 ASP Peer binds to 1 SCTP
    - Thus SCTP are shared across all peers
- ASP Peers are bound to AS Cluster.
  - 1 ASP peer binds to 1 AS cluster
  - 1 AS cluster binds/controls to 1 or more ASP Peers
- AS Cluster are bound to MTP2 through M2UA binding and NIF interface
  - 1 AS Cluster binds to 1 or many MTP2
    - (through M2UA->NIF relationship)
  - 1 MTP2 binds to 1 AS Cluster through NIF interface binding

M2UA Breakdown



M2UA Debugging

- MTP2 SS7 link will be set administratively down by the NIF function until M2UA link comes up.
  - E1 signaling trace will display IDLE tx packets.
  - E1 signaling trace should display MTP2 LSSUs
- Once M2UA link comes up, the MPT2 SS7 Link will become activated.
  - MTP2 link should become activated
- If MGC disables the M2UA link the MTP2 SS7 link will be set to administratively down.

Megaco Overview
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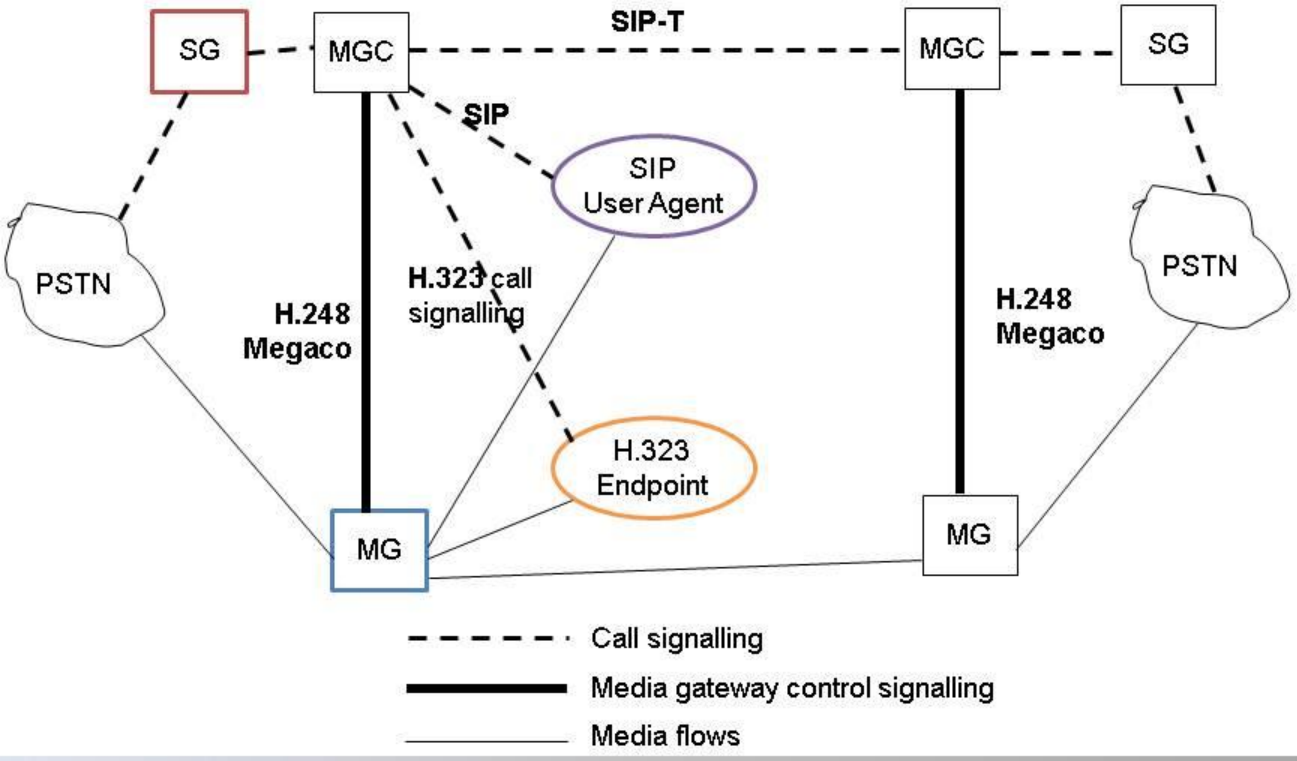
Megaco MG + M2UA SG

Third party Softswitch/MGC controls Netborder SS7 Media Gateway via Megaco/H.248 protocol.

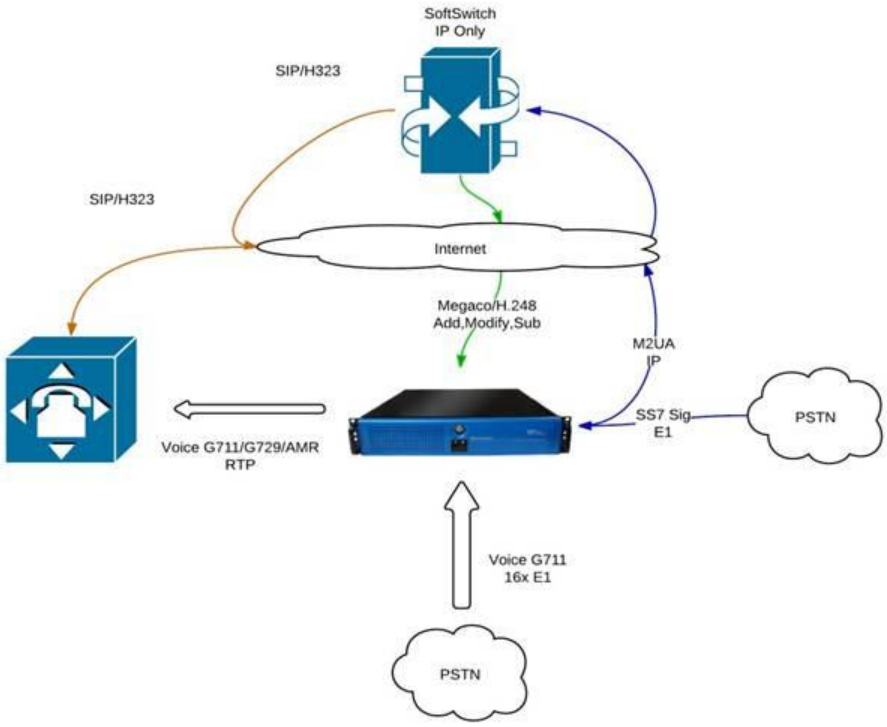
- NSG Megaco MG
  - Bridge RTP media to TDM Voice 64kb G.711 channels
  - Bridge TDM Voice 64kb G.711 channels to RTP media ports

- Media specific functions
  - Transcoding
  - DTMF
  - T.38 Faxing

Megaco/VoIP Network



Megaco MG + M2UA SG



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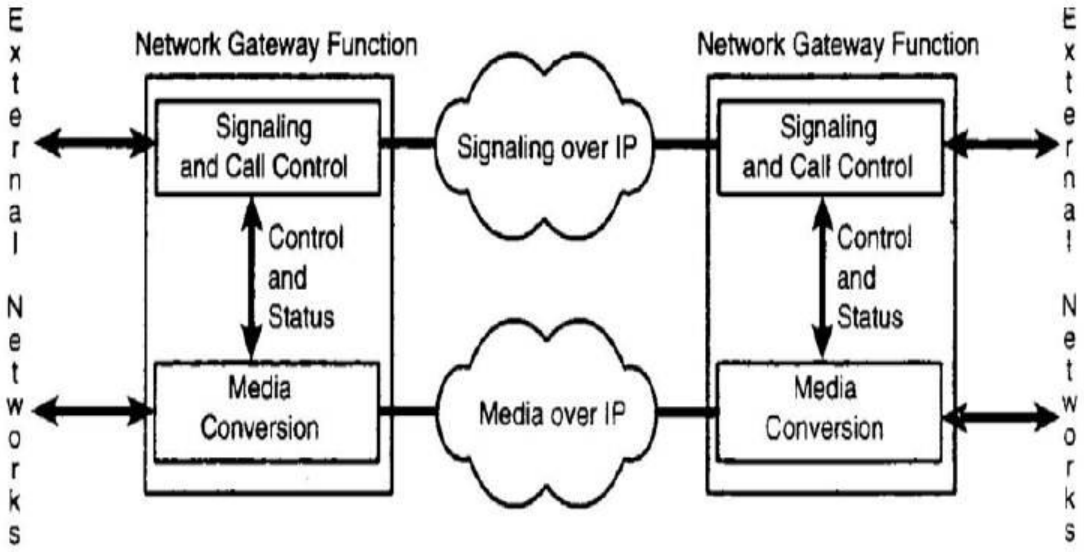
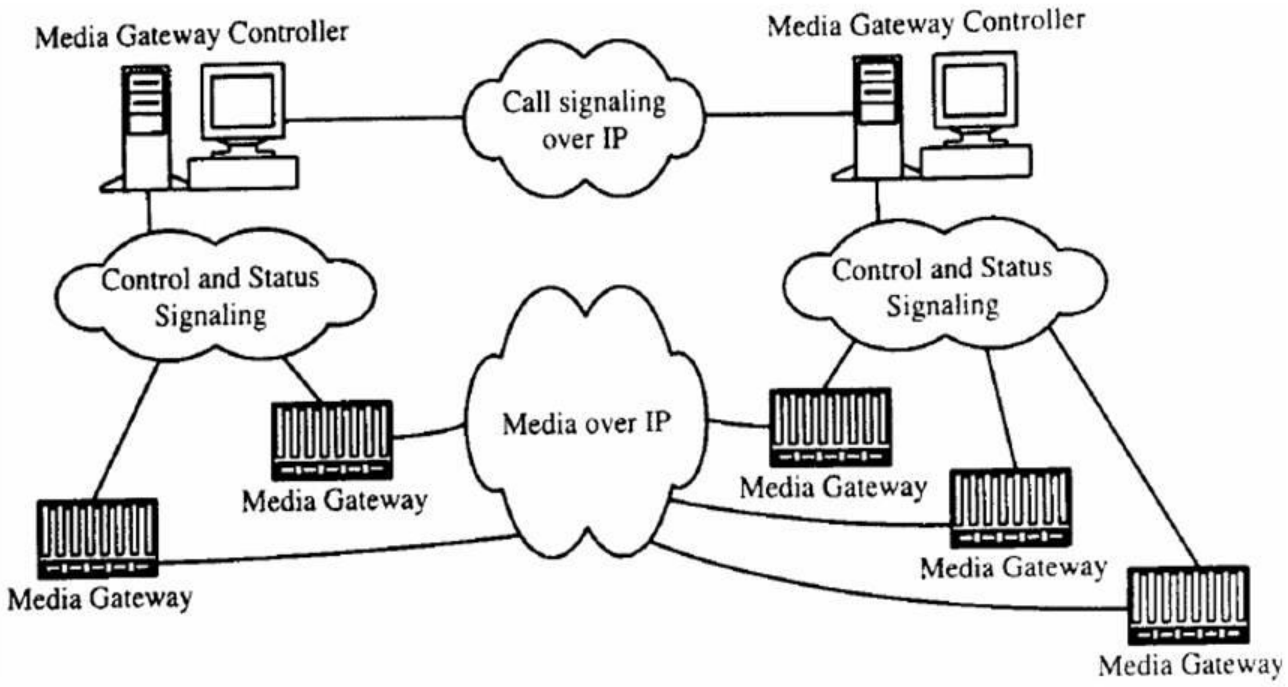
Megaco Architecture



Separation of Media and Call Control

- A network gateway has two related but separate functions.
- Signaling conversion
  - The call-control entities use signaling to communicate.
- Media conversion
  - A slave function (mastered by call-control entities)

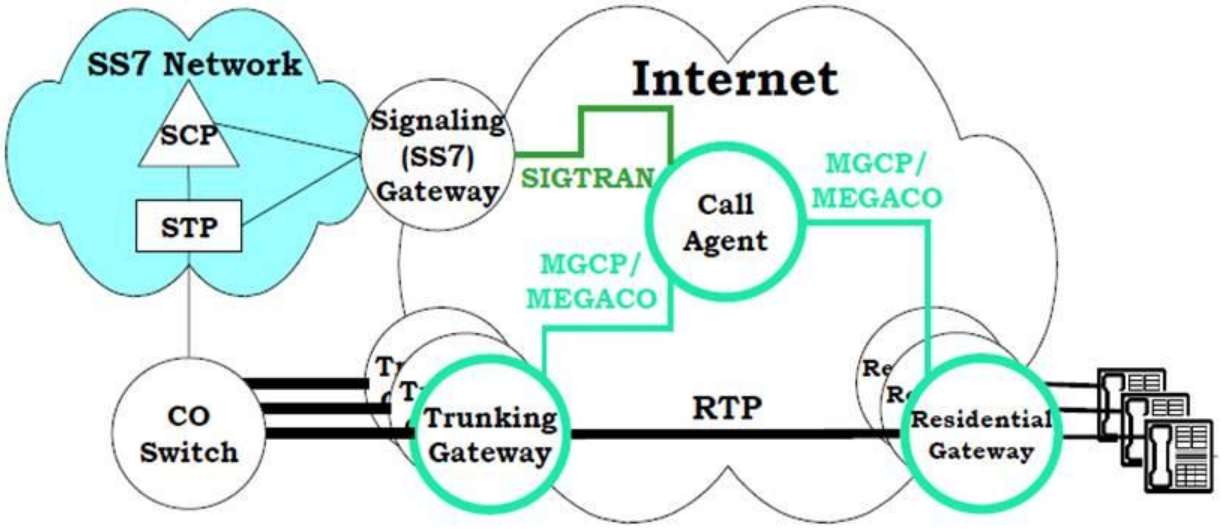
Softswitch Network Architecture [1/2]



MEGACO

- MEGACO is the protocol used to communicate between the Soft Switch and the Media Gateways.
- MEGACO can use any of TCP/UDP/SCTP transport layer protocols.
- MEGACO uses TEXT and BINARY (ASN) encoding modes.
- As of now we are supporting TEXT encoding with UDP/TCP transport protocol.

Softswitch Network Architecture [2/2]



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Megaco Connection Model

Based on 3 concepts:

Termination

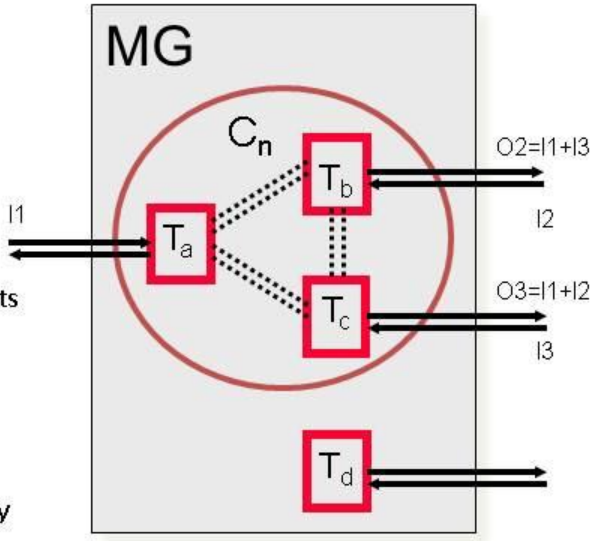
- Identifies an end point for media flows
- Implements Signals, and generates Events
- Can appear in at most one context.
- Permanent (provisioned) terminations can exist outside a context

Context

- Defines communication between Terminations, acts as a mixing bridge
- Contains 1 or more Terminations
- Supports multiple streams

Stream

- A context can have multiple streams, each typically for a medium, e.g. audio, video, etc
- The MGC specifies which streams a given termination supports



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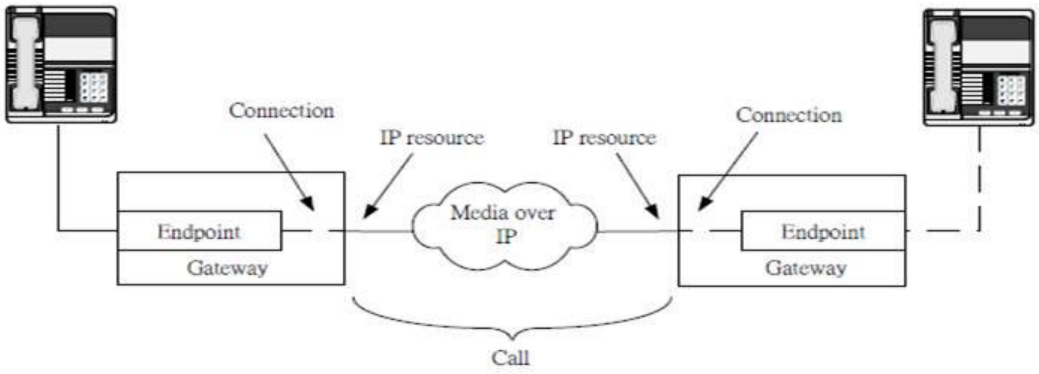
MEGACO Calls and Connection

- A **connection**
  - ❑ Relationship established between a given endpoint and an RTP/IP session
- A **call**
  - ❑ A group of connections
- The primary function of MEGACO is to enable
  - ❑ The connections to be created
  - ❑ The session descriptions to be exchanged between the connections



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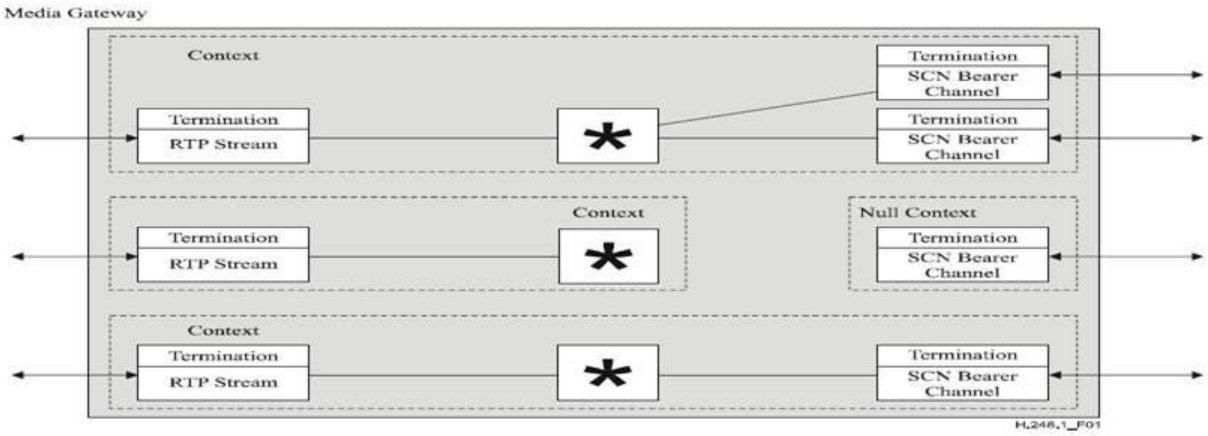
MEGACO Termination

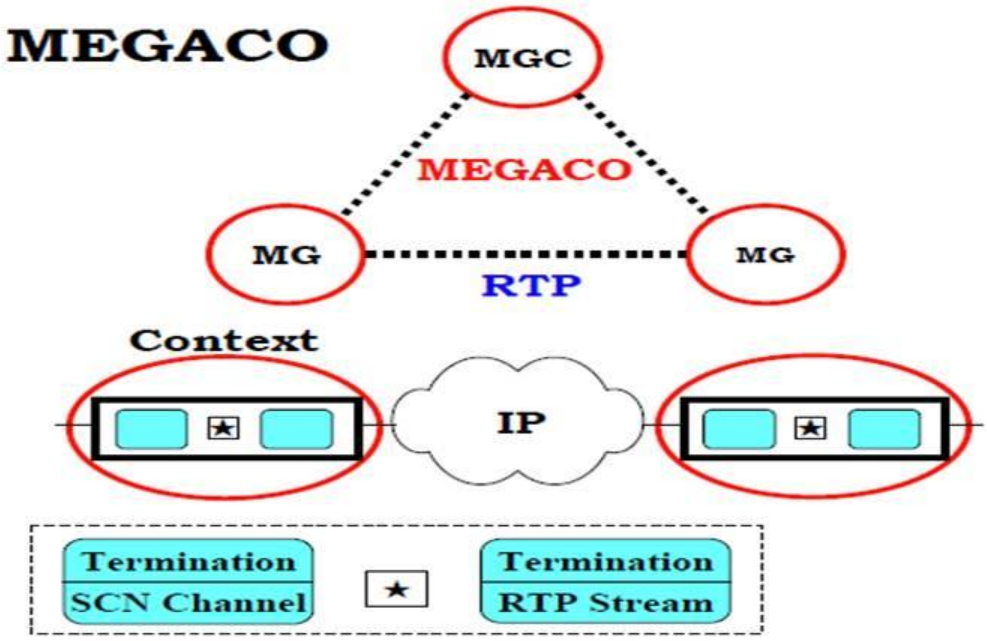
- A logical entity on a MG
  - It sources/sinks media/control streams.
- Termination ID
  - An arbitrary string assigned by the MG.
  - It uniquely maps to a real physical channel in MG and MGC.
- Two kinds of Termination
  - Physical Interface
  - RTP stream
- ROOT Termination
  - The entire MG itself

MEGACO Context

- A context is an association between a collection of terminations.
- Null Context
  - It contains terminations that are not present in any other context and therefore not associated to any other termination.
- Context ID
  - A 32 bit integer chosen by MG.
  - * = ALL ; - = NULL ; $ = choose



Concept of Context



Add 2 Terminations into 1 context

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MEGACO Commands

Add:

- The Add Command adds a termination to a context. The Add Command on the first termination in a context is used to create a context.

Modify:

- The Modify Command modifies the properties, events and signals of a termination.

Subtract:

- The Subtract Command disconnects a termination from its context and returns statistics on the termination's participation in the context.

Move:

- The Move Command atomically moves a termination to another context.

AuditValue:

- The AuditValue Command returns the current state of properties, events, signals and statistics of terminations.

AuditCapabilities:

- The AuditCapabilities Command returns all the possible values for termination properties, events and signals allowed by the Media Gateway.

Notify:

- The Notify Command allows the Media Gateway to inform the Media Gateway Controller of the occurrence of events in the Media Gateway.

ServiceChange:

- This command allows the MG to notify the MGC that a termination or group of terminations is about to be taken out of service or has just been returned to service. ServiceChange is also used by the MG to announce its availability to a MGC (registration).

Megaco Commands

Command	Initiator	Description
Add	MGC	Adds a termination to a context.
Modify	MGC	Modifies a termination's properties, events, and signals.
Move	MGC	Moves a termination from one context to another.
Subtract	MGC	Removes a termination from its context.
AuditValue	MGC	Returns current state of properties, events, signals, and statistics.
AuditCapabilities	MGC	Returns all possible values for termination properties, events, and signals allowed by an MG.
Notify	MG	Informs MGC of event occurrence(s).
ServiceChange	MGC	Takes or places a termination(s) out of or in service.
	MG	For registration and restart; notifies MGC termination(s) will be taken out of or returned to service.

MEGACO Transaction and Message

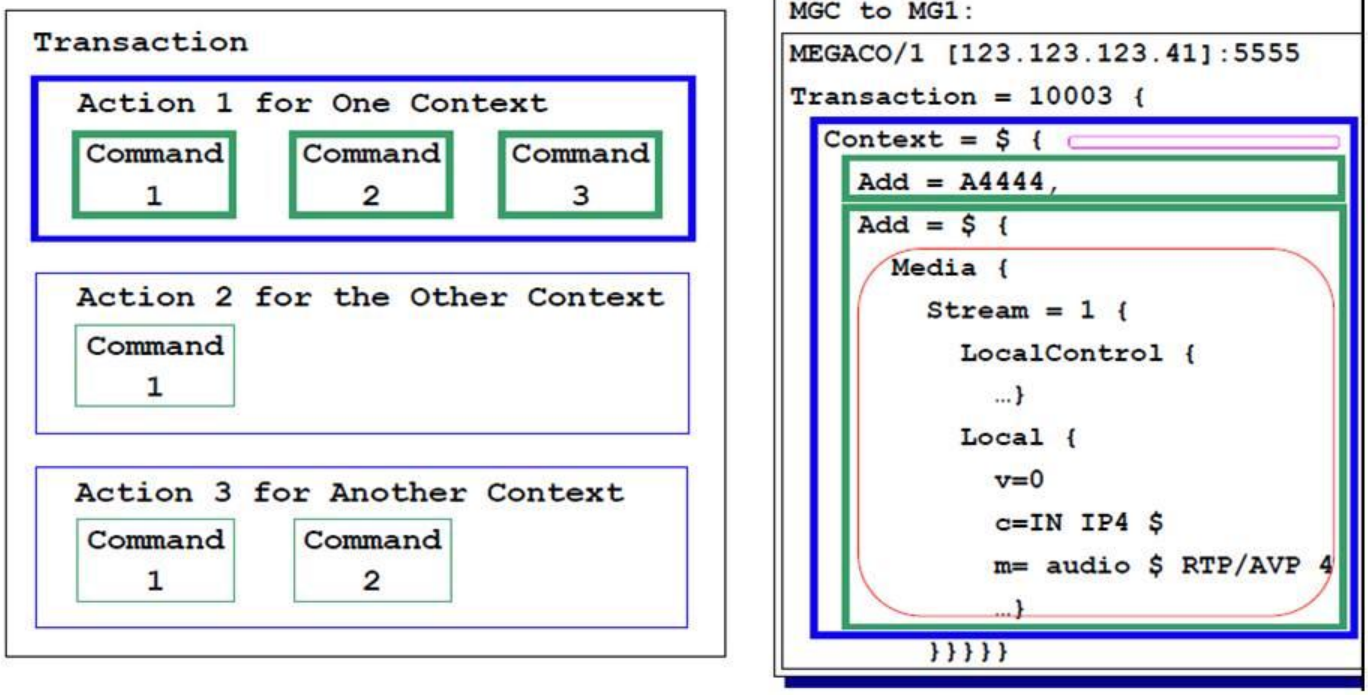
Transaction

Multiple commands can be grouped.
Commands are executed in sequence
If a command fails, the subsequent commands are not processed.
Not the case for optional commands
O-"command-name"

Messages

Concatenate multiple transactions
The transactions are treated independently

MEGACO Transactions



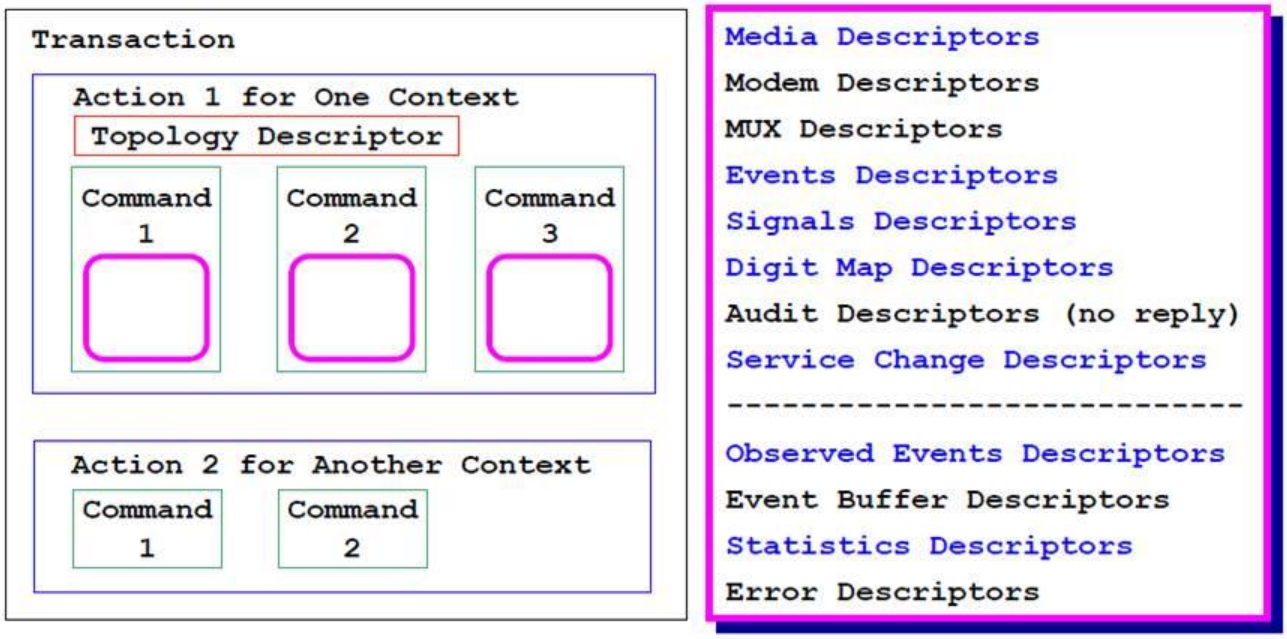
MEGACO Descriptors

To form the parameters of the commands/responses

To provide additional information to qualify a given command/response.

Termination Descriptors

MEGACO Termination Descriptors



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MEGACO Media Descriptor

Describe the various media streams

A hierarchical descriptor

Media descriptor
- **Termination state descriptor**
  - **Stream descriptor**
    - **Local control descriptor**
    - **Local descriptor**
    - **Remote descriptor**

MEGACO Termination State Descriptor

Service States

To indicate whether the termination is available for use.
"test", "out of service", "in service"

Event Buffer Control

To specify whether events detected by the termination are to be buffered following detection or processed immediately.

MEGACO Stream Descriptor

Stream ID

Local Control Descriptor

- Mode: sendonly, receiveonly, sendreceive, inactive, and loopback
- MGC specifies a set of choices for the session
- ReserveGroup and ReserveValue indicate the resources should be reserved

Local Descriptor and Remote Descriptor

- Usage of SDP

MEGACO Event & Signal Descriptor

Event Descriptor
- Request Identifier
- A list of events that the MG should detect and report

Signal Descriptor
- On/Off
- Timeout
- Brief

MEGACO Service Change Descriptor

Used only in association with the ServiceChange Command

Service Change Method (The type of service change)

- Graceful, the removal of existing terminations w/o interrupting existing connections
- Forced, an abrupt removal
- Restart, after a specified delay
- Disconnected, applied to the entire MG
- Handoff, from the old MGC; a new MGC is taking over
- Failover, from MG to MGC

Service Change Delay, a number of seconds.

Service Change Reason

MEGACO Digit Map Descriptor

- A Dialing plan
- A start timer, to start
- A short timer, when more digits are needed
- A long timer , to differentiate different routing
- Note: as of now we are not supporting this.

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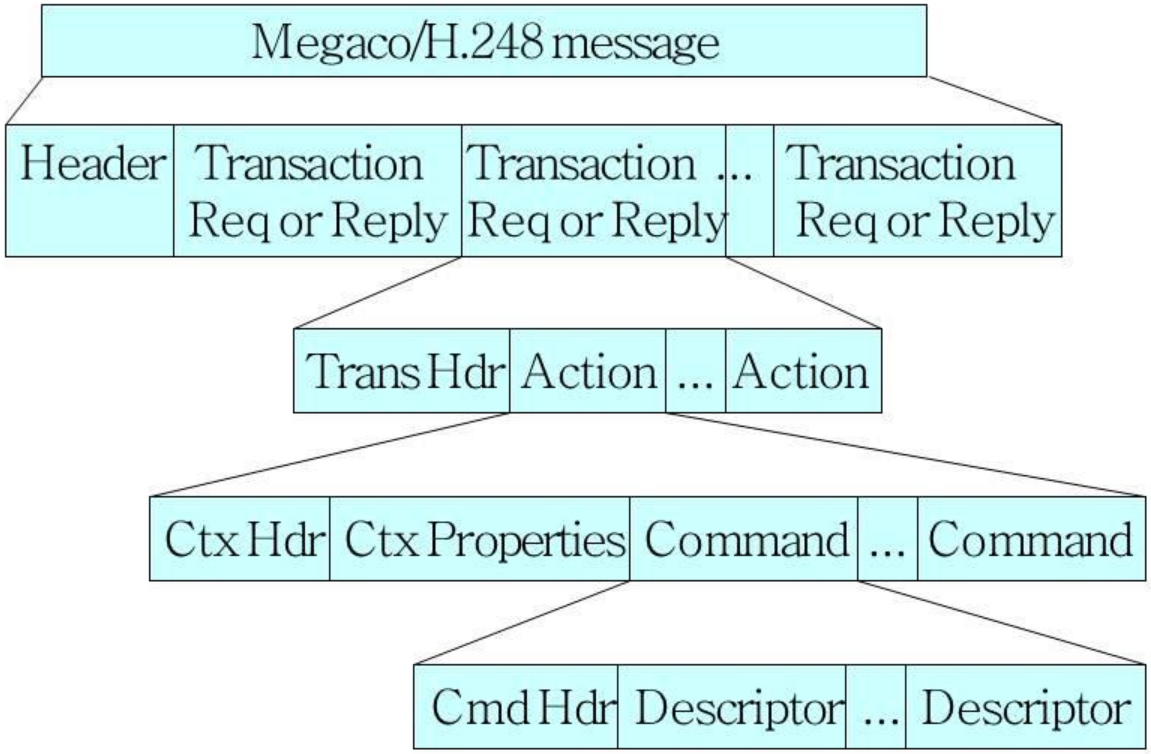
MEGACO Observed Events Descriptor

Observed Events is supplied with the Notify Command to inform the MGC of which event(s) were Detected at MG.

Mandatory in the Notify command. It contains

- Request Identifier
- Observed event identifier
- Optional time stamp for each observed event.

Megaco Messages



Megaco Commands

- Megaco uses some commands in order to manipulate terminations, contexts, signals and events.
- For termination manipulation: Add, Subtract, Move, Modify
- For event reporting: Notify
- For management: AuditCapability, AuditValue, ServiceChange

Megaco Commands

- From MGC to MC:
- Add: adds a termination to a context
- Subtract: removes a termination from a context
- Move: moves a termination from a context to another
- Modify: changes the characteristics of an existing termination , which can be in the null context
- AuditValue & AuditCapabilities: return information about terminations, contexts and general state and capabilities of MG

Megaco Commands

- From MG to MGC:
  - **Notify**: MG sends it to inform MGC that an event has occurred.
- Either from MG to MGC or from MGC to MG:
  - **ServiceChange**: creates a connection between MG and MGC.
- Descriptors are parameters for all these commands & return values of some of them.

Megaco Events

- Events are detected at MG and reported to MGC.
- (example: inband signaling)
- MGC controls what events it wants to learn about at any given time
  - sets the termination Events descriptor
- Events can have side effects
  - stop playout of signals
  - start new signals
  - automatically update the set of events of interest

Megaco Signals

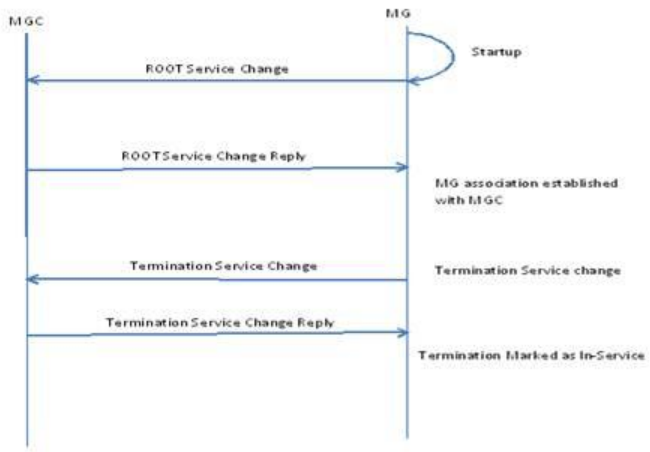
- Signals cause things to happen on terminations
  - play a tone, display text, ...
- Specified in the Signals descriptor for a termination
- MGC can specify duration of signal ahead of time or signal can play until explicitly stopped
- Signals stop playing when any event is detected unless MGC says otherwise.

Megaco Packages

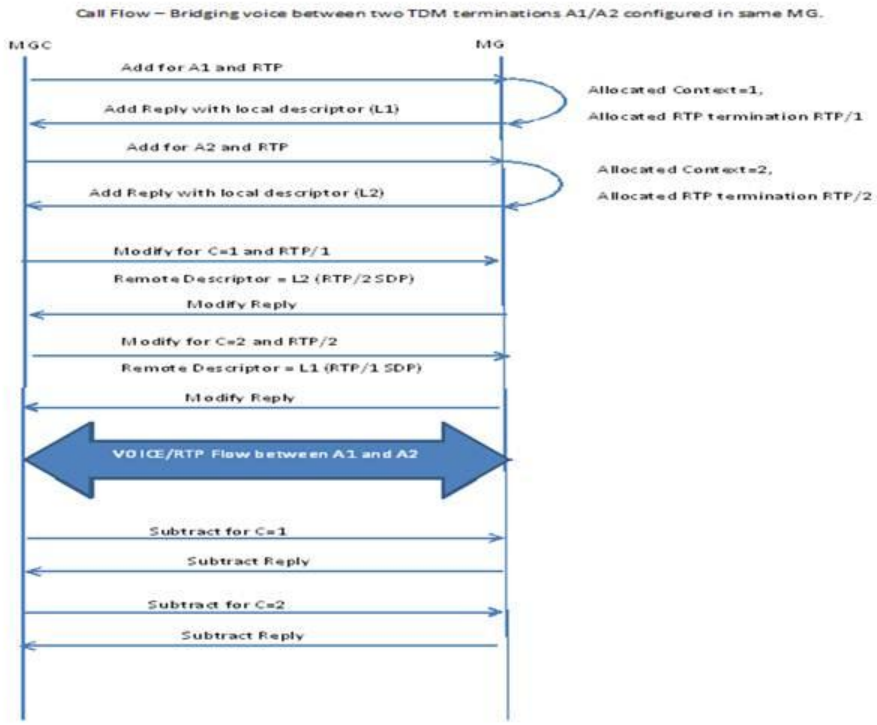
- Add detailed content to the protocol
  - all events, signals, and statistics are specified in packages
  - can also specify additional properties
- Package definition a continuing process
  - being created by multiple standards bodies
  - private packages also allowed
- Packages can inherit from and extend other packages.

MEGACO Call Flows

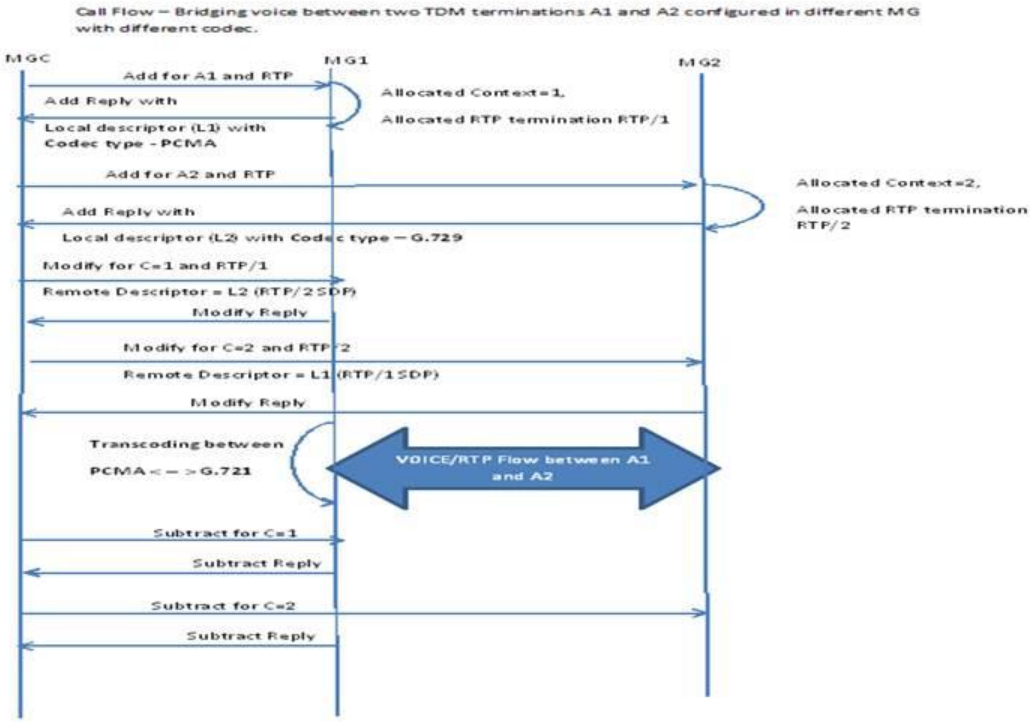
- Registration (association of MG with MGC)

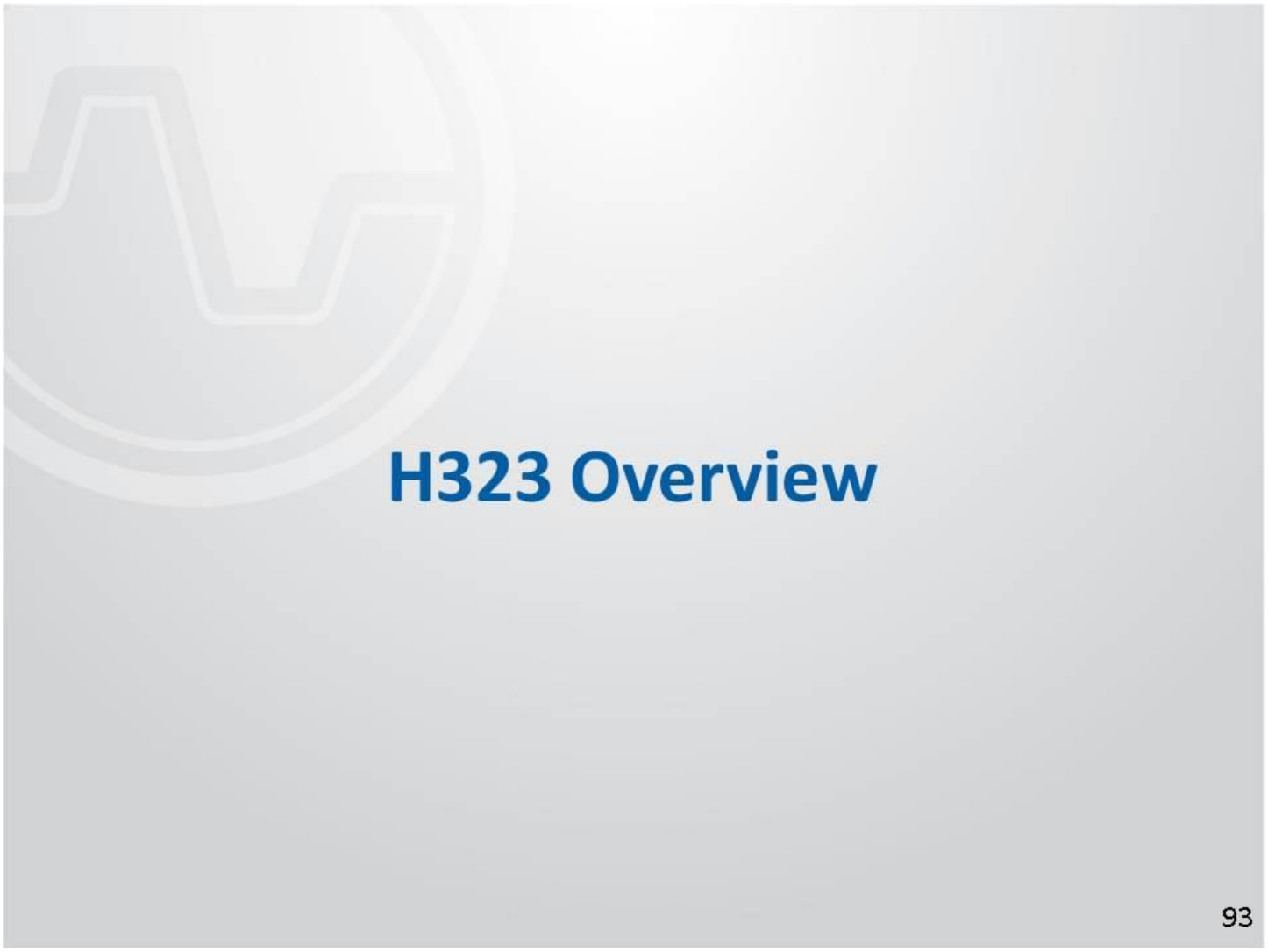


MEGACO Call Flows



MEGACO Call Flows




H323 Overview
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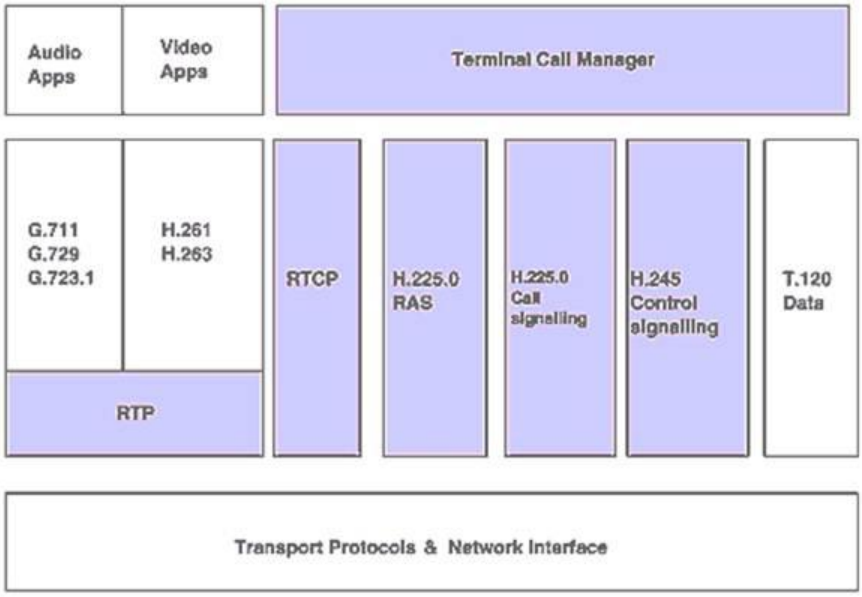
H323 protocol family



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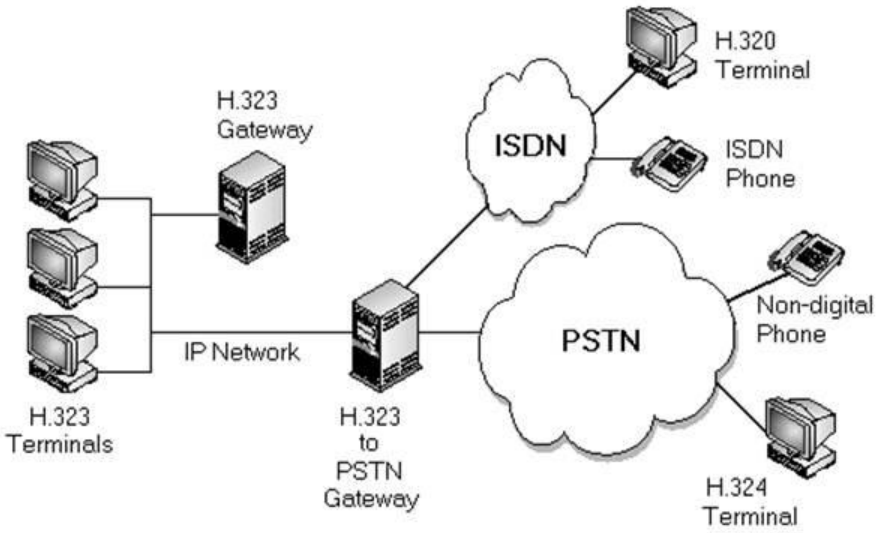
Network architecture



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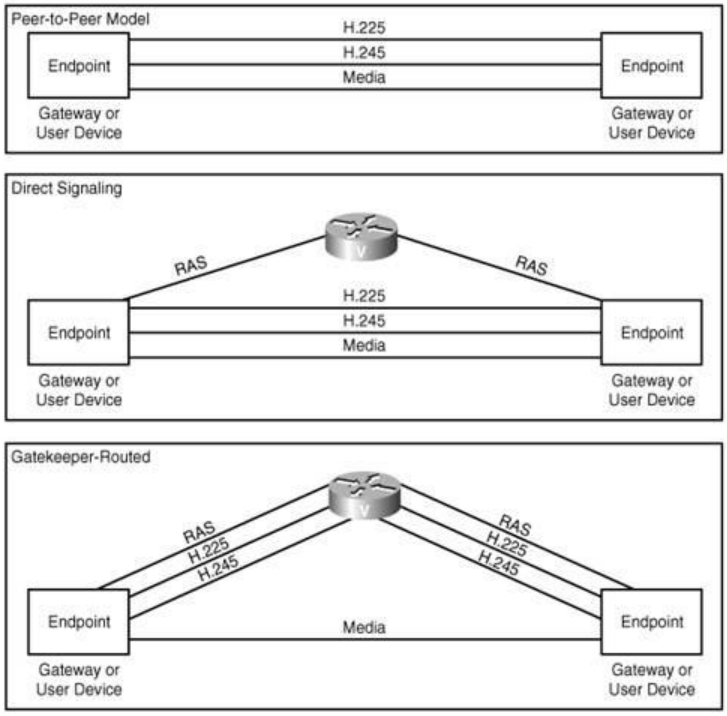
H323 call flow types



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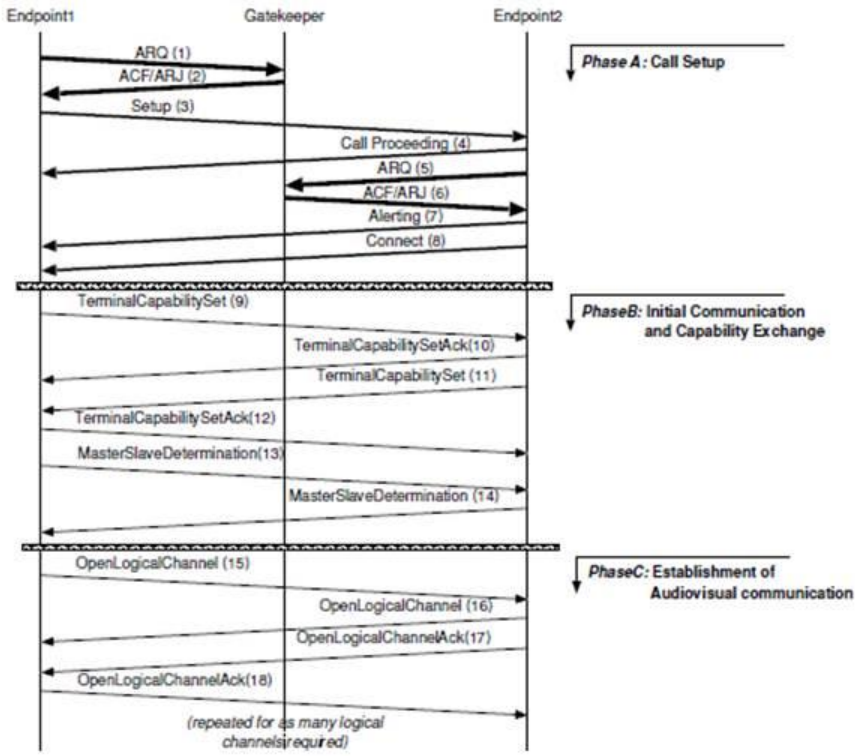
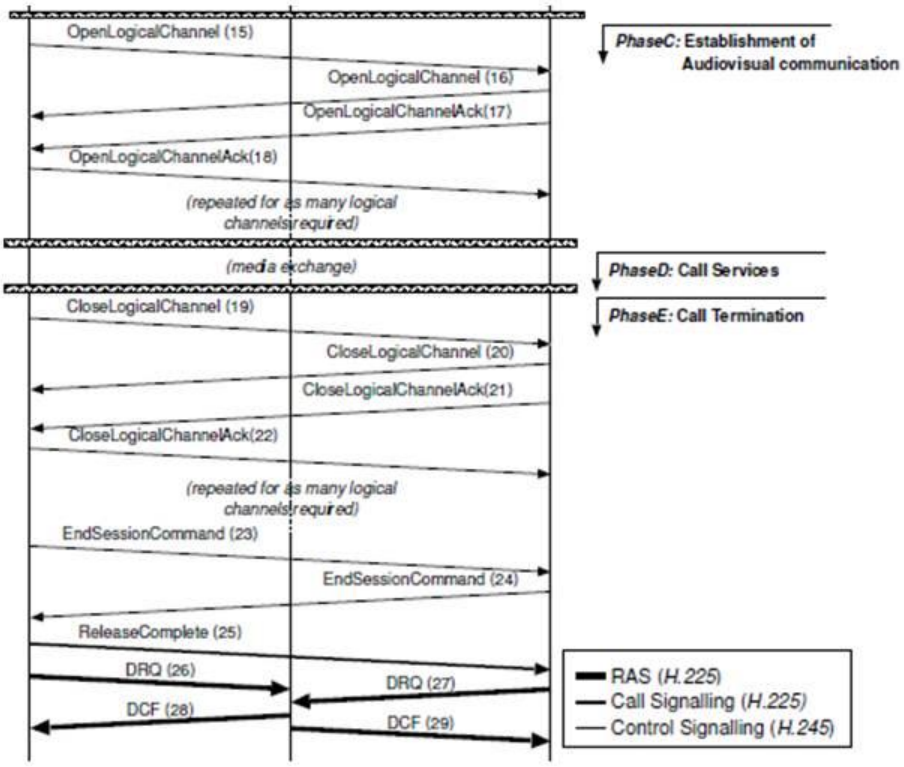
Direct signaling call flow



Direct signaling call flow - continued



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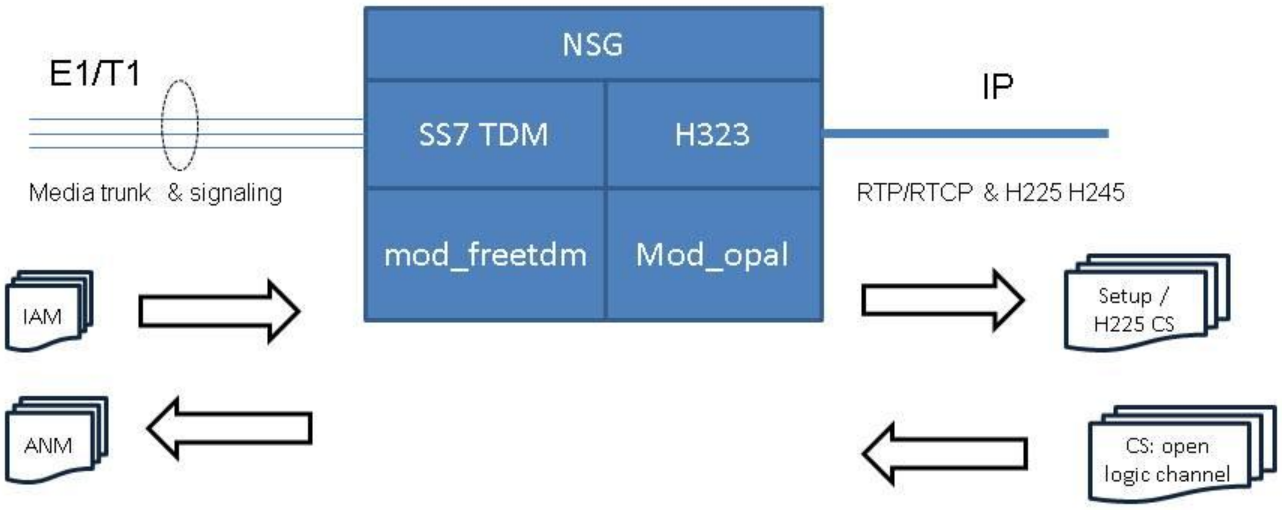
NSG implementation



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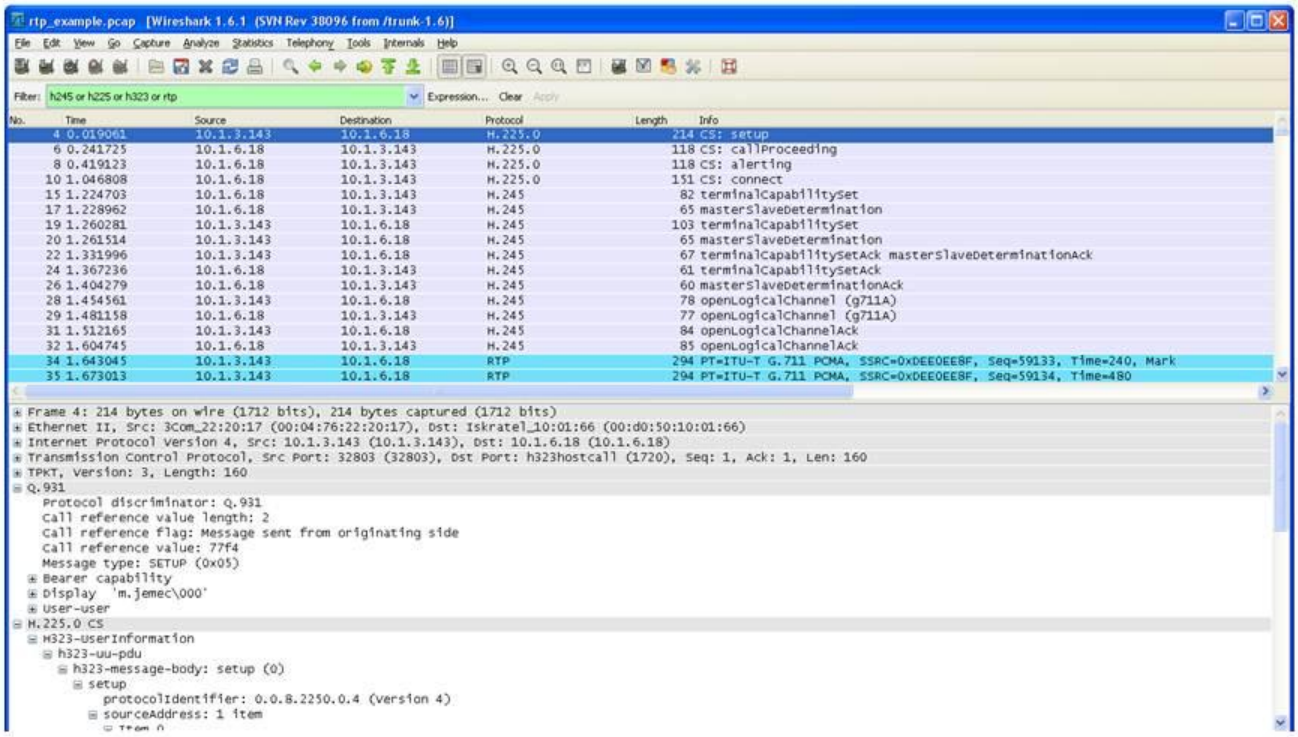
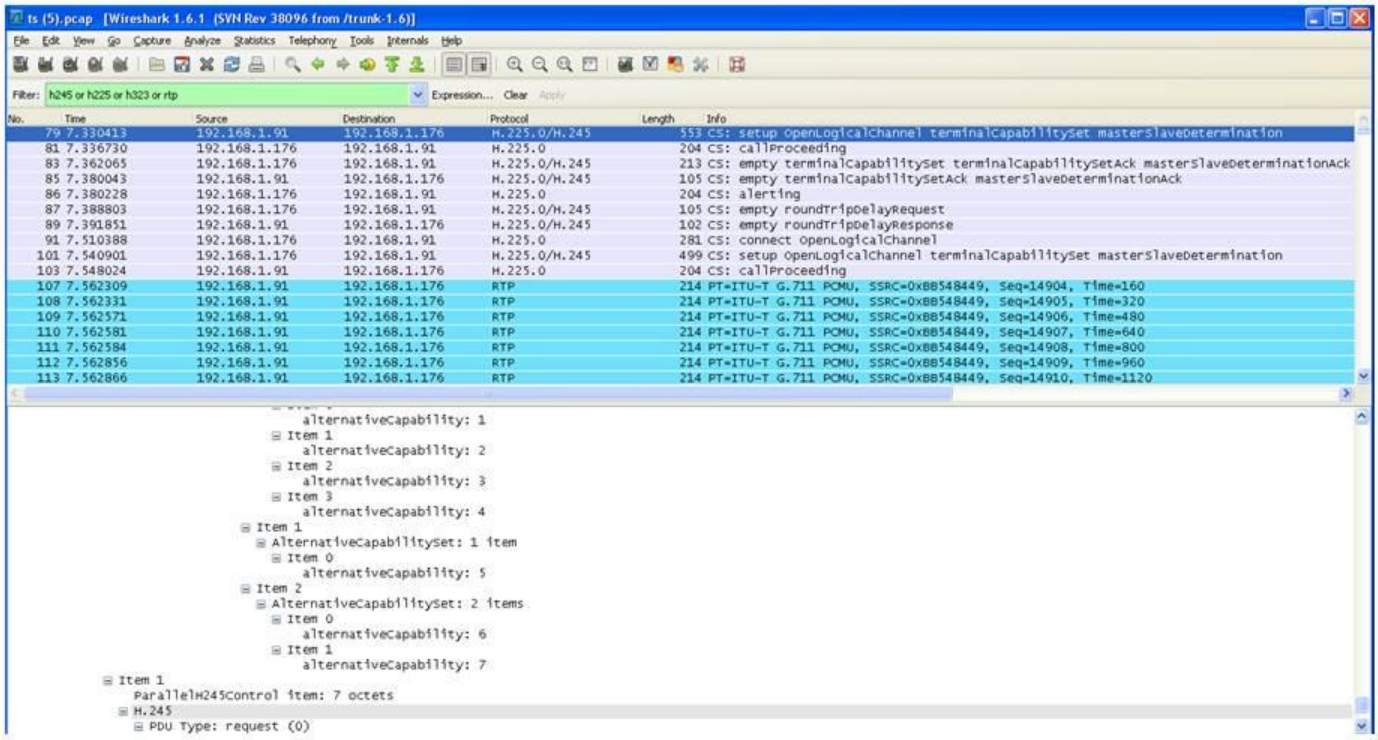
Fast start vs slow start

- Fast start
  - Push everything to the other end at one time
  - Inside the first message
  - Short call establishment time
  - More popular
- Slow start
  - Exchange capabilities in sequential packets
  - Multiple packets
  - Long call establishment time
  - Old standard

Slow start sample



Fast start



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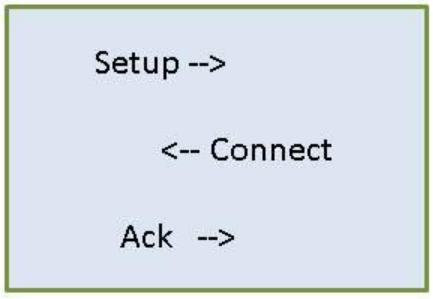
Messages

- Q.931
  - for call signaling. Used to setup connections between terminals
- H.225
  - for providing registration, admissions and status signaling functions.
  - Used between endpoints and gatekeepers.
- H.245
  - for call control including capability exchange, mode changes, flow control, commands, indications and others.
  - Provides the ability to open logical channels on the network.



H.245 important messages

Message	Function
Master-Slave Determination	Determines which terminal is the master and which is the slave. Possible replies: Acknowledge, Reject, Release (in case of a time out).
Terminal Capability Set	Contains information about a terminal's capability to transmit and receive multimedia treams. Possible replies: Acknowledge, Reject, Release.
Open Logical Channel	Opens a logical channel for transport of audiovisual and data information. Possible replies: Acknowledge, Reject, Confirm.
Close Logical Channel	Closes a logical channel between two endpoints. Possible replies: Acknowledge
Request Mode	Used by a receive terminal to request particular modes of transmission from a transmit terminal. General mode types include VideoMode, AudioMode, DataMode and Encryption Mode. Possible replies: Acknowledge, Reject, Release.
Send Terminal Capability Set	Commands the far-end terminal to indicate its transmit and receive capabilities by sending one or more Terminal Capability Sets.
End Session Command	Indicates the end of the H.245 session. After transmission, the terminal will not send any more H.245 messages.

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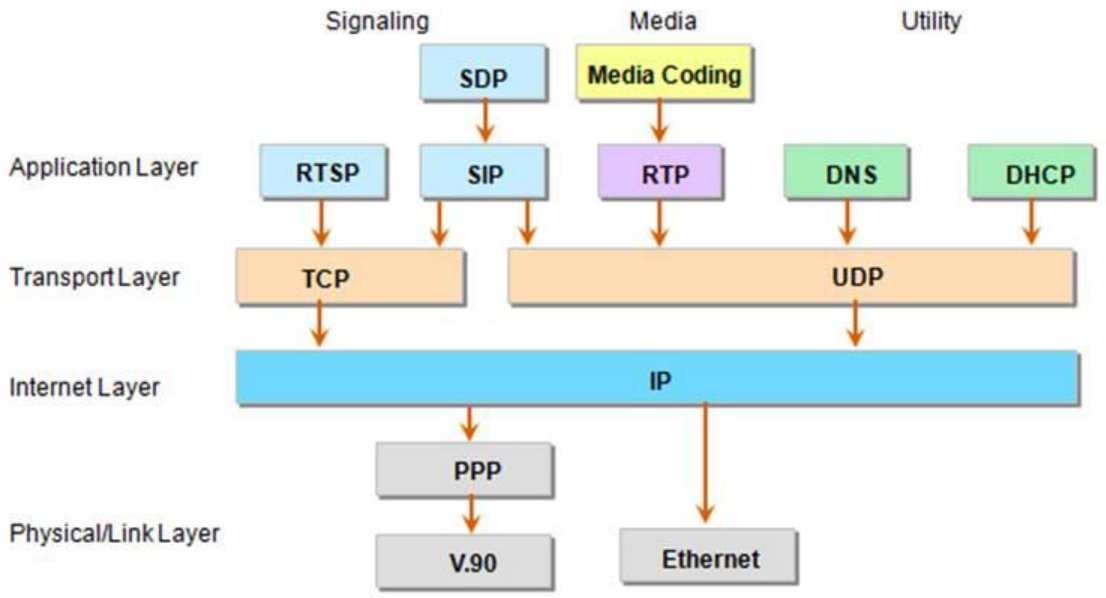
SIP Overview

- Light-weight generic signaling protocol
- Used to initiate sessions and invite members to a session
- Text-based protocol (good for prototyping)
- Syntax is textual and based on HTTP
- There have been several bake-offs with different vendors demonstrating interoperability of basic calls

SIP Architecture



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SIP Overview…

- SIP Transactions – SIP defined by RFC 3261
- User Agent Client (UAC):
  - Initiates the SIP transactions
- User Agent Server (UAS):
  - Sends the final response back to UAC
- SIP Messages structures
  - Start Line
  - SIP Headers
  - Empty line indicating the end of header fields
  - SIP Body (Session Description Protocol –SDP)
- There are 2 types of SIP Messages:
  - SIP Requests and SIP Responses

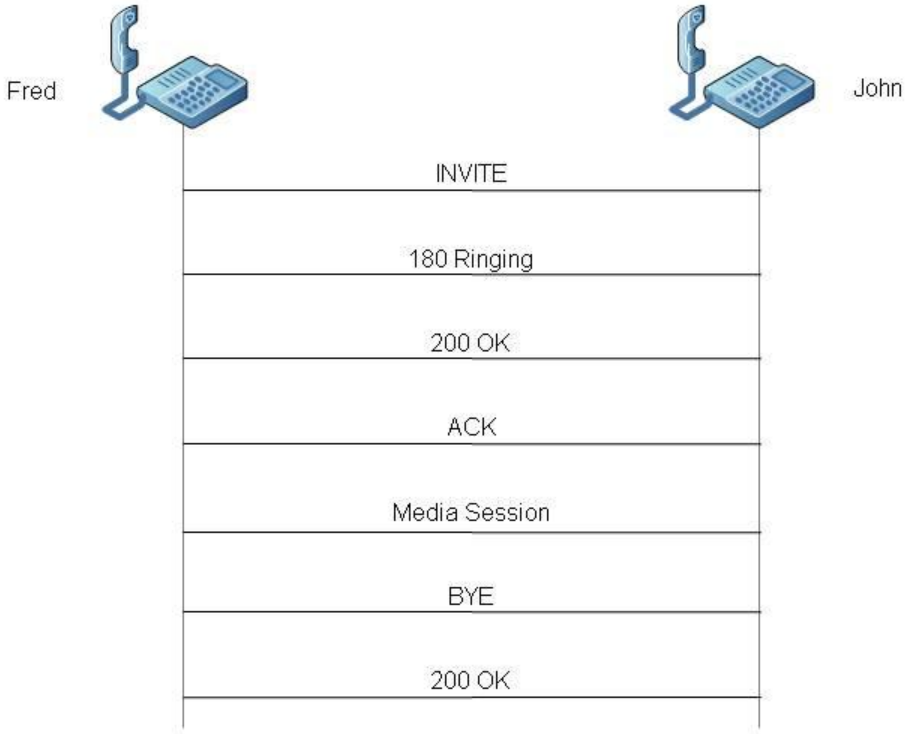
SIP Requests

- SIP requests are messages that are sent from client to server to invoke a SIP operation. RFC 3261 defines six requests or methods that enable a User Agent or SIP proxy to locate users and initiate, modify, and tear down sessions:
  - **INVITE**: An INVITE method indicates that the recipient user or service is invited to participate in a session. This method can also be used to modify the characteristics of a previously established session.
  - **ACK**: An ACK request confirms that the UAC has received the final response to an INVITE request. ACK is used only with INVITE requests.
  - **OPTIONS**: An OPTIONS request is used to query servers about their capabilities. If the UAS is capable of delivering a session to a user, it responds with its capability set.
  - **BYE**: A BYE request signifies the termination of a previously established session.
  - **CANCEL**: A CANCEL request allows UACs and network servers to cancel an in-progress request, such as an INVITE.
  - **REGISTER**: A REGISTER request is used to register the current contact information.
- In addition, RFC 3515 defines the **REFER** method. This SIP extension requests that the recipient REFER to a resource provided in the request. This method can be used to enable many applications, including call transfer.

SIP Responses

- A server sends a SIP response to a client to indicate the status of a SIP request that the client previously sent to the server. Specifically, the UAS or proxy server generates SIP responses in response to a SIP request that the UAC initiates.
- SIP responses are numbered from 100 to 600
  - **1xx**: Information/Provisionnal (100 Trying,180 Ringing, 183 Session Progress)
  - **2xx**: Successful (200 OK)
  - **3xx**: Redirection (302 Moved Temporarily)
  - **4xx**: Client Failure (404 User not found, 486 Busy Here)
  - **5xx**: Server Failure (500 Server Internal Error, 503 Service Unavailable)
  - **6xx:** Global Failure (603 Decline)

Basic SIP Call Flow



Sample SIP Request Message

INVITE sip:John@sangoma.com SIP/2.0	Request-Line
Via: SIP/2.0/UDP 192.168.11.156:5060;branch=z9hG4bK233E7363;rport=5060 Max-Forwards: 70 Contact: <sip:Username@192.168.11.156:5060;transport=udp> To: <sip:John@sangoma.com> From: Fred <sip:fred@sangoma.com>;tag= 660A2622E4A7F9B9A839A2B8B9381FD Call-ID: FC735409EBE21C394E69C4AD800FBF01@192.168.11.156:5062 CSeq: 1 INVITE Session-Expires: 1800;refresher=uac Content-Type: application/sdp Supported: timer, replaces User-Agent: Kapanga Softphone Desktop 1.00/2163e+1161886252_001372BDBCE2 Content-Length: 323	SIP message headers
	Blank Line
V=0 o=Username 1190134451 1190750689 IN IP4 192.168.11.156 s=Kapanga [1190134451] …	SIP body in SIP message

Sample SIP Response Message

SIP/2.0 200 OK	Status (Response) Line
Via: SIP/2.0/UDP 192.168.11.156:5060;branch=z9hG4bK233E73631D7A9A90B05D8C5A029 83766;rport=5060 Contact: "Sangoma NetBorder"<sip:NetBorder@127.0.0.1:5062> To: <sip:1024@192.168.11.103:5061>;tag=0f7fe217 From: "Your Long Name"<sip:Username@defaultproxy:5060>;tag=660A72622E4A7F9B9A839 A2B8B9381FD Call-ID: FC735409EBE21C394E69C4AD800FBF01@192.168.11.156:5062 CSeq: 1 INVITE Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, INFO Content-Type: application/sdp Content-Length: 236	SIP message headers
	Blank Line
v=0 o=Sangoma-Tech 1190750737 1190750739 IN IP4 192.168.11.103 …	SIP body in SIP 200 OK message

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Corporate Fact Sheet

- Founded in 1984
- Public company since 2000: TSXV:S technology fast 50™ | Deloitte.
  - Revenues of ~$14M
- Staff of 60 in offices in Toronto (corporate HQ), Montreal, New Jersey, London and Hong Kong.
- Sell via a network of distribution partners/resellers

Netborder Series

- Netboder for Carriers & Service Providers

– NetBorder SS7 to VoIP Gateway	SS7 to VoIP
– NetBorder Transcoding Gateway	SIP transcoding
– NetBorder Lync Express	MS Lync + PSTN
– NetBorder Session Controller (SBC)	SIP security/firewall

Vega Series

- Vega Series for Carrier CPE and Enterprise

– Vega 100,200,400	SIP to T1/E1 gateways
– Vega 50	SIP to Analog/BRI
– Vega 5000	SIP to 24/48 FXS gateways
– Vega 400 SBC	SIP to SIP gateways

Netborder SS7 to VoIP Appliance

NetBorder SS7 to VoIP Gateway: an elegant, self-contained, and cost-effective telco grade appliance.

Form Factor	Capacity	SKU	Appliance
1U	Up to 4 E1/T1	SS7-NSG-AP04	
	Up to 8 E1/T1	SS7-NSG-AP08	
2U	Up to 16 E1/T1	SS7-NSG-AP16	
	Up to 32 E1/T1	SS7-NSG-AP32	

NetBorder SS7 to VoIP Features

- SS7 ISUP Signalling with broad variant support
  - ITU, ANSI, Bellcore, UK, China, France, India and Russia
- Sigtran
  - M2UA SG and ASP mode
- VoIP
  - SIP, Megaco/H.248, H323
- Delivered as telco grade Appliance
- Up to 32 E1 per server
  - Echo cancelation 128ms
- Transcoding support on all channels
  - 32 E1 (G729, G722, iLBC, G723.X, AMR)
- Flexible XML based routing rules for call control
- All in features available on each appliance with single software image

NetBorder SS7 Advantages

- Easy to use!
- Wide range and support of SS7 PSTN protocols and variants
- Advanced VoIP configuration and protocol support.
- Scalable, and Flexible.
- Flexibility of software deployments – not stuck with monolithic hardware platforms
- Low cost installation
- Robust implementation with distribution, failover and redundancy
- No need to stock or provision different equipment and support multiple software images.