NORTEL

Secure Router 2330/4134 as Communication Server 1000 Survivable SIP Branch Solution

Quick Start Configuration Guide

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NN-SR-0001

Secure Router 2330/4134 as Communication Server 1000 Survivable SIP Branch Solution
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Contents

New in this release

The following section details what's new in *Secure Router 2330/4134 as Communication Server 1000 Survivable SIP Branch Solution (NN-SR-0001)* for Release 10.2.

Features

The following sections detail the Secure Router 2330/4134 based CS 1000 branch solution and its features.

Navigation

CS 1000 and Secure Router 2330/4134

In a centralized CS 1000 call server architecture, the remote branches make use of the call processing resources available at a central location, generally located at the corporate headquarters. The survivable branch solution based on Secure Router 4134 (SR 4134) and Secure Router 2330 (SR 2330) provides business continuity to the branch office in the event of a WAN connection outage to corporate headquarters. With this solution, employees at the branch office can continue to use SIP phones to place and receive intra-site calls and calls over the PSTN, including 911 calls.

Feature Background

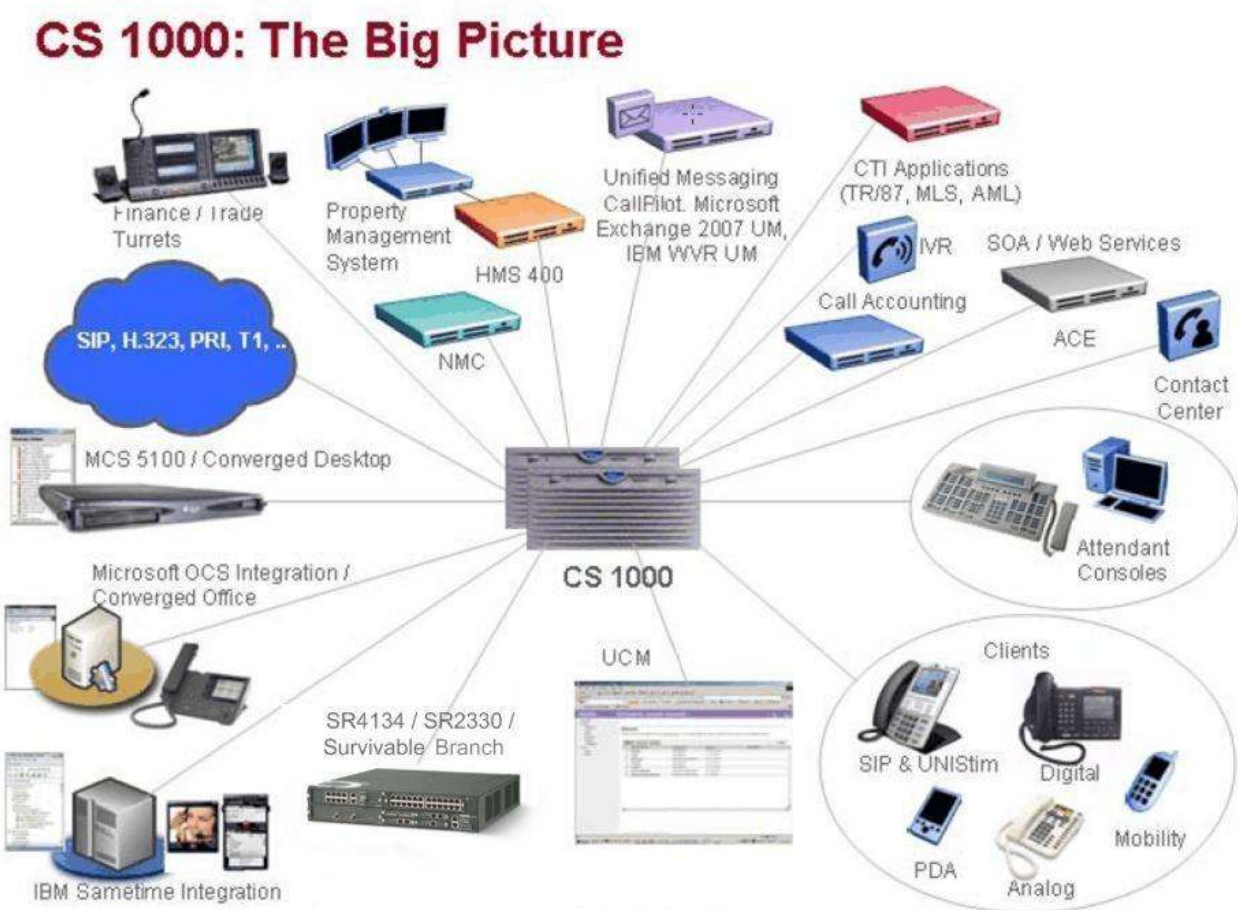
Communication Server 1000

Nortel Communication Server 1000 is a server-based, full-featured IP PBX and the cornerstone of Nortel Enterprise Unified Communication deployments. It provides the benefits of a converged network plus advanced applications and over 750 world-class telephony features. Fully distributed over IP LAN & WAN infrastructure with built-in reliability and survivability, Communication Server 1000 supports business-critical applications, including unified messaging, customer contact center, IVR, wireless VoIP and IP phones.

Key Features:

- Feature rich with over 750 call processing and telephony features
- Highly scalable with support for up to 22,500 IP users off of one Call Server, multiple Call Servers networked together can support unlimited scalability

- World class reliability and redundancy mechanisms - highly reliability architectural elements that maximize network uptime with extensive redundancy mechanisms to ensure network uptime including survivability options such as Campus and Geographic redundancy to support network failover
- Extensive desktop portfolio includes; Wireless, Soft-phones, IP, Digital and Analog set support, to meet diverse end-user requirements
- Supports business-critical applications, including IP Contact Center, CallPilot unified messaging, and integrated services such as conferencing, one-number-follow-me Personal Call Director, recorded announcement, network-wide attendant and messaging
- Telephony integration with desktop application providers such as Microsoft and IBM



For more details please refer Nortel CS 1000 Product Webpage

http://products.nortel.com/go/product_content.jsp?segId=0&catId=null&parId=0&prod_id=51121&locale=en-US

Secure Router 4134

The **Nortel Secure Router 4134** is a modular, multi-service platform that integrates multiple networking functions, including routing, WAN, Ethernet switching, security and Voice over IP (VoIP) into a single device. The platform's design ensures the consistently high throughput required by voice, data or unified communications applications. The first device of its kind to feature embedded Microsoft intelligence to simplify deployment of unified communications, the Secure Router 4134 can reduce the number of devices needed at the branch or regional site, generating substantial operational and capital cost savings for your business.

Key Features:

- Highly modular, high-performance platform - A wide range of LAN, WAN and multiservice options to support converged branch, regional or headquarters environments
- All-in-one voice, data and unified communications solution for enterprises – Nortel SCS Server hosted on 4134 provides complete unified communications and data networking solution for enterprise sites of up to 250 users by combining voice — call server, conferencing, collaboration applications and PSTN gateway — with data and security in an integrated, easy-to-manage platform.
- Only device of its kind to integrate Microsoft OCS services - Ideal for enterprises considering deploying Microsoft OCS services in their remote branch sites
- Voice media gateway services - Enables connection to the Public Switched Telephone Network (PSTN) or to traditional telephony devices
- Survivable voice services - Allows continued voice calling when the primary IP connection is lost.
- Robust routing services - Full IPv4 and IPv6, BGP-4 and multicast implementation for enterprise deployments
- Integrated Ethernet switching - High-density L2/L3 Gigabit, Fast Ethernet, as well as Power over Ethernet. Up to 58 Gigabit or 72 Fast Ethernet ports supported.
- Wide range of WAN connectivity - Low and high-speed WAN options include serial, T1/E1, DS3/T3, Channelized DS3/T3, HSSI and ISDN
- Integrated security - Stateful firewall and high-speed VPN encryption ensured the integrity of both voice and data traffic
- High-reliability / resiliency - Hot-swappable modules, redundant power and port/platform resiliency features deliver maximum uptime
- Unified Communications-ready platform - Superior small packet handling and low latency ensures the quality of multimedia applications. Integrated VoIP and Microsoft capabilities deliver on the promise of the unified communications branch.

For more details please refer Nortel SR 4134 Product Webpage

http://products.nortel.com/go/product_content.jsp?segId=0&catId=null&parId=0&prod_id=62360&locale=en-US

Secure Router 2330

The **Nortel Secure Router 2330** is a cost reduced 1RU version of 4134 with almost same feature set and lower capacity.

For more details please refer Nortel SR 2330 Product Webpage

http://products.nortel.com/go/product_content.jsp?segId=0&catId=null&parId=0&prod_id=69360&locale=en-US

Feature Description

The Secure Routers 4134 and 2330 combines high performance, robust routing, flexible WAN and voice media gateway connectivity and is targeted at enterprise branch and remote site environments. A rich suite of routing services and advanced WAN functionality makes these Secure Routers ideal for high-speed Internet access, private line WAN connectivity, IP Telephony and multimedia, IPSec VPN, stateful firewall and data applications. The SR 2330/4134 survivable branch solution for Nortel CS 1000 provides business continuity to the branch office in the event of a WAN connection outage to corporate headquarters.

Multiservice Branch Router

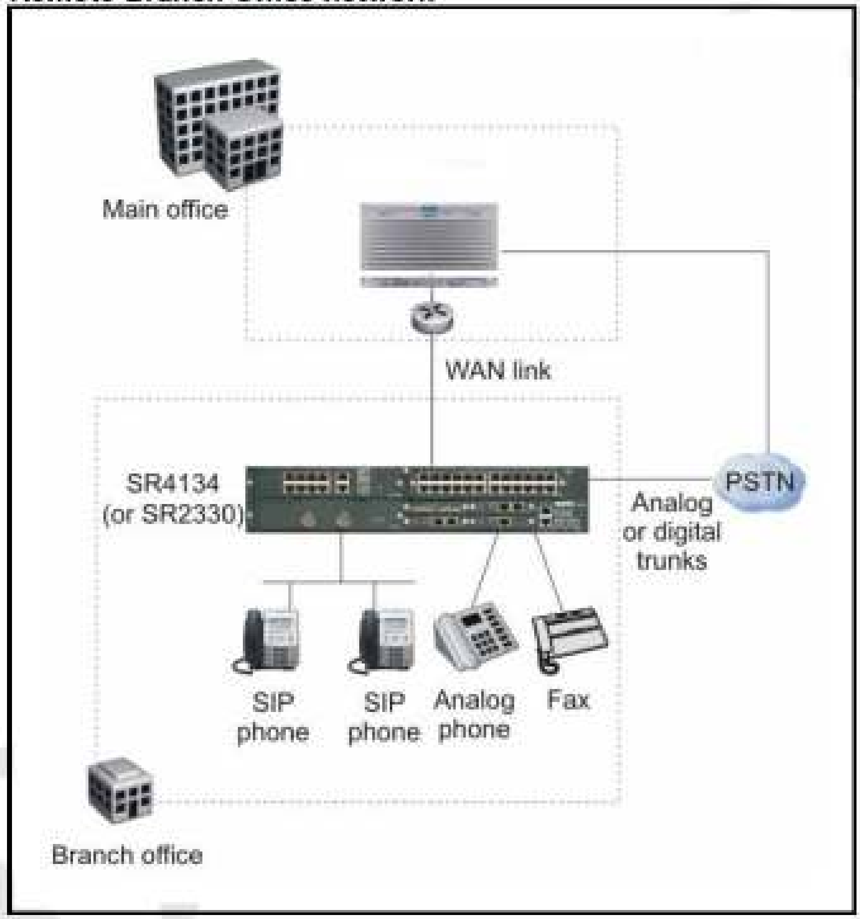
Figure 1 shows a survivable branch office deployment with CS 1000 Call Server located at the corporate main office or data center and Secure Router as branch office multi service router providing data routing, security and survivable SIP-PSTN gateway.

Data routing services include a full IPv4 and IPv6 protocol set, including BGP-4 and multicast capabilities. A full-function IPv6 implementation also enables deployment into environments that require extended IP addressing with the same routing services.

Powerful, fully-integrated security features include VPN and firewalls for increased reliability and user confidence. Capabilities include stateful packet firewall, detection and prevention of more than 60 Distributed Denial of Service (DDoS) attacks, VPN hardware acceleration for hub and spoke deployment over IPSec and VPN tunnels, and IPSec VPN data-encryption services with AES, 3DES, DES, SHA-1, MD-5 and Diffie-Hellman support.

The SR also offers a set of integrated voice interfaces that allow connection to the public switched telephone network (PSTN) as well as support of conventional TDM-based telephony devices. T1/E1, FXS and FXO interfaces are all available for flexible telephony connection with support for up to 128 simultaneous voice channels.

Figure 1
Remote Branch Office network



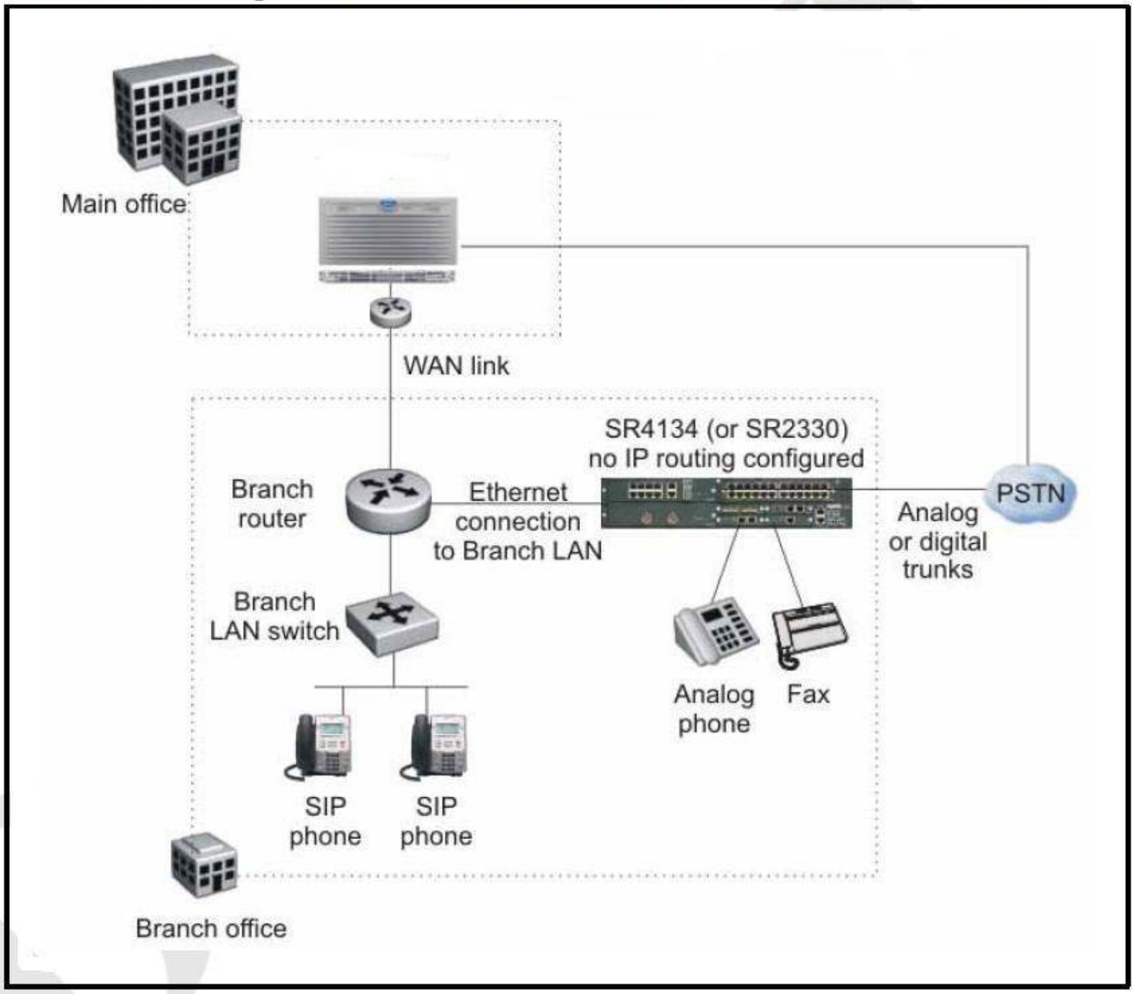
Survivable SIP PSTN Gateway

Figure 2 shows a survivable branch office deployment with CS 1000 Call Server located at the corporate main office or data center and Secure Router providing survivable SIP-PSTN gateway functionality complimenting the existing data infrastructure.

The SR 2330/4134 supports a variety of PSTN interfaces like T1/E1, BRI U, BRI S/T, FXS/DID and FXO/CAMA for connectivity to PSTN and legacy PBXs and telephony devices. Also supports a rich set of PSTN protocols including ISDN PRI, BRI, QSIG, T1 CAS, E1 R2 and analog signaling.

The Secure Router also includes a SIP Registrar and B2BUA based SIP Proxy which can function as a backup SIP Server supporting up-to 300 SIP end-points including Nortel and 3rd-party SIP phones Nortel 1120E/1140E, Nortel 1535 Video phone, LG Nortel 6800/8800, Polycom 330, SMC 3456, IP Dialog and Xlite. It can provide phone and call routing services to the branch office when main office call server connectivity is lost and is already tested with Nortel Call Servers and 3rd party Servers - CS 1000, CS 2100, CS 2000, A2E, SCS, Microsoft OCS and Broadsoft/Sylantro. Other main features include Call Admission Control, PSTN fallback and memory based load control.

Figure 2
SR with existing LAN infrastructure in Remote Branch Office network



Introduction

This document describes the quick start configuration of Nortel Secure Router 2330/4134 (Release 10.2) as survivable branch SIP-PSTN gateway for Nortel Communication Server 1000 (Release 6.0). For more information and detailed configuration guides on SR 2330, SR 4134 and CS 1000 go to the Nortel website:

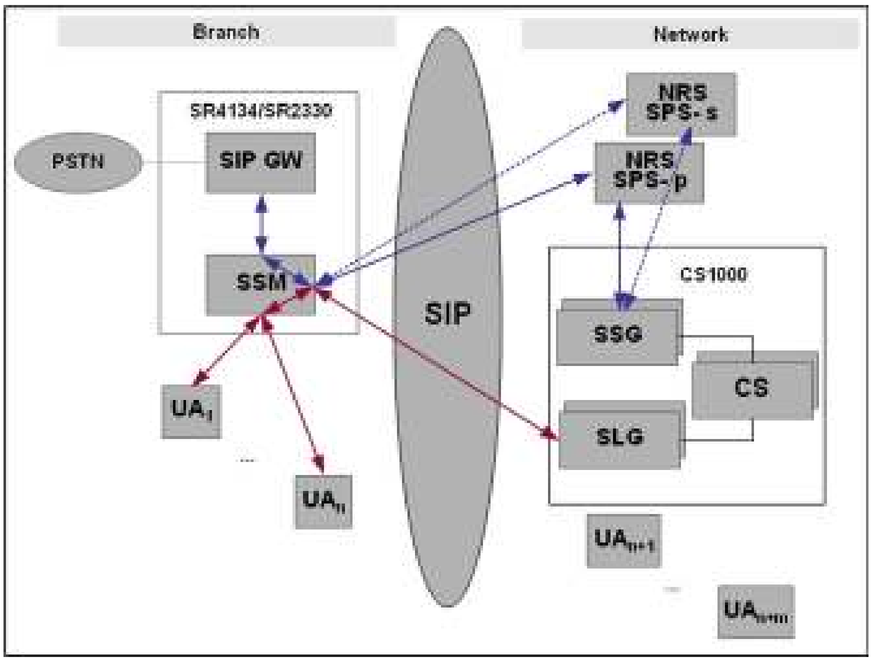
www.nortel.com/support

Navigation

SR 2330/4134 interoperability with CS 1000

SR 2330/4134, CS 1000 components

The following diagram shows the main components of Secure Router 2330/4134 and Communication Server 1000.



SR has two modules SIP Gateway (SIP GW) and SIP Survivability Module (SSM) that together interworks with CS 1000 to provide SIP survivable gateway functionality at the branch. SSM is a software-only subsystem on the Secure Router through which SIP calls are routed to the CS 1000. This module includes SIP B2BUA based proxy and SIP Registrar. SIP GW is software and hardware subsystem on the Secure Router that provides PSTN connectivity. The User Agents (UA) are SIP endpoints.

For detailed information about SSM operation please refer to Secure Router Release 10.2 guide NN47263-510 Configuration — SIP Survivability.

For detailed information on SIP GW please refer to Secure Router Release 10.2 guide NN47263-508 Configuration — SIP Media Gateway.

The main CS 1000 components are Call Server (CS), SIP Signaling Gateway (SSG), SIP Line Gateway (SLG), SIP Proxy Server (SPS) and Network Routing Service (NRS). SSG handles SIP trunking and SLG takes care of SIP endpoints or SIP Lines.

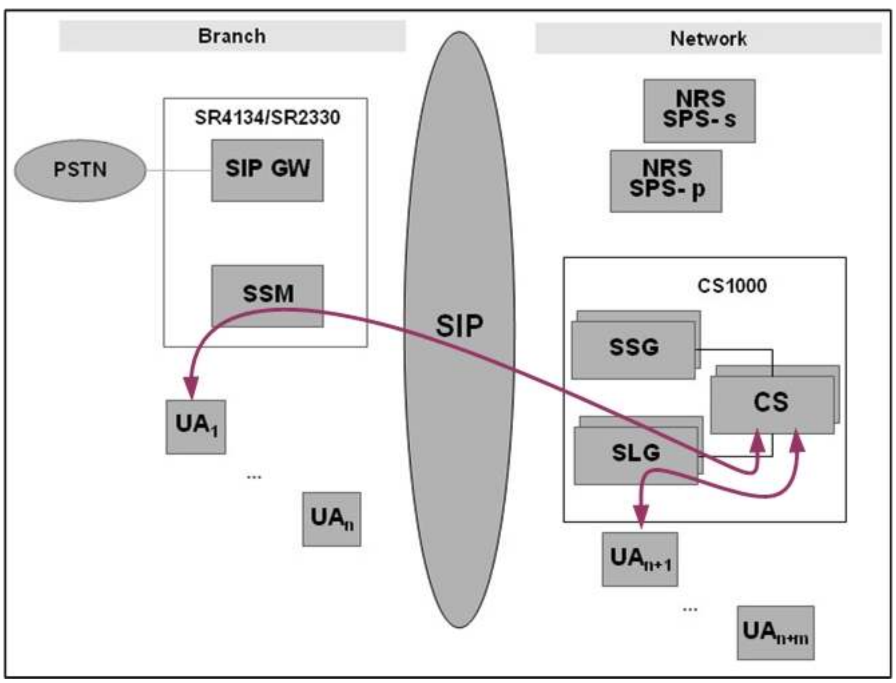
For detailed information on CS 1000 components and operation please refer to Communication Server 1000 Release 6.0 user guides.

SSM Operation

The SSM operates in two modes - Normal (Connected) and Survivable (Isolated). In normal mode, the SSM functions as an outbound proxy and proxies all SIP messages initiated from the SIP phones (UA) and the SIP GW to the SLG located in the head office. SSM acts as a B2BUA i.e. changes the Contact Header of SIP endpoint requests. Also the SIP endpoint registrations to the SLG are "cached" locally. In survivable mode, the SSM supports SIP server functionality to provide basic call features to the SIP endpoints at the branch, and also supports local registrar functionality to store registrations.

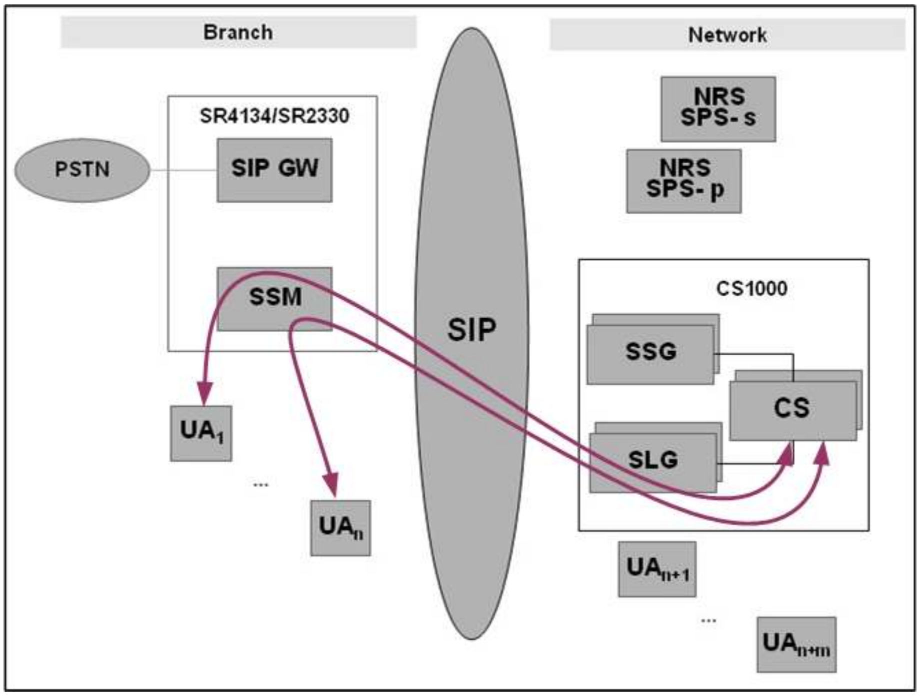
SSM monitors the reachability of SLG by sending OPTIONS messages. If SLG is not reachable or the link connected to SLG is down, SSM switches to the Survivable mode. The SSM will continue to monitor the reachability of SLG as long as the link is up. Once it is reachable, SSM will switch back to Normal mode.

SIP endpoints that have registered during Survivable mode will be registered with the SLG after the Normal mode is established and next registration is attempted. SSM forces SIP endpoints to register frequently (Default time 30 sec) in Survivable mode so that the endpoints are registered to SLG as soon as SSM switches to Normal mode.



The above diagram shows the call flow of a SIP endpoint in branch, calling a SIP endpoint connected to CS 1000 in Normal (Connected) mode. SSM proxies the calls to the SLG received from the SIP endpoint. SSM also modifies the contact header in the INVITE messages to point to the SSM bind IP address before forwarding the INVITE to the SLG to ensure that incoming calls are routed through the SSM. The same call flow holds good for the call originated by the SIP endpoint connected to CS 1000. When the call arrives at SSM, it will look for a mapped contact in its registration "cache" and routes the call to the UA. If SSM does not find the mapped contact then it forwards the call to the configured default gateway.

The following diagram shows the call flow between two SIP endpoints in branch in Normal (Connected) mode.

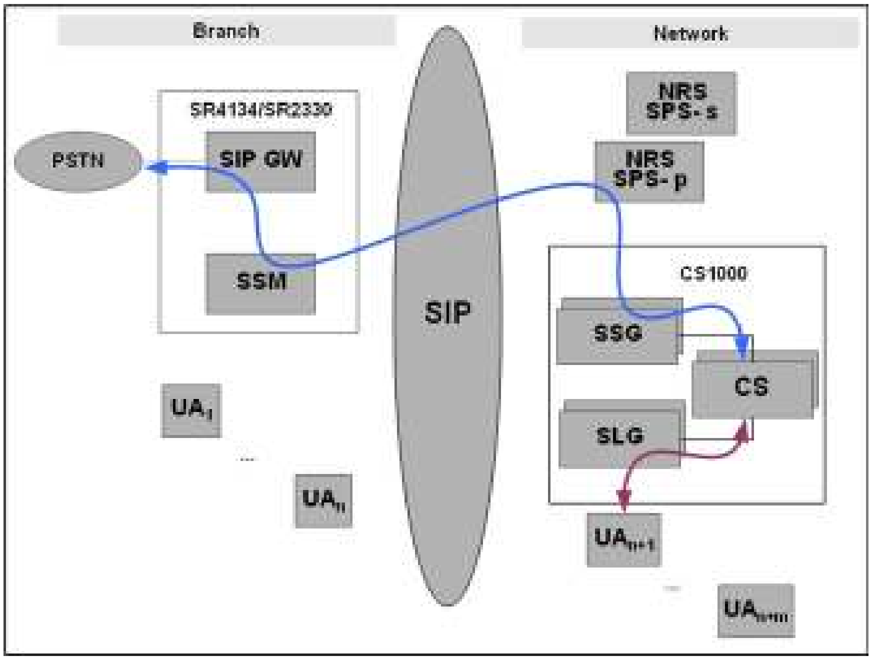


SIP Gateway Operation

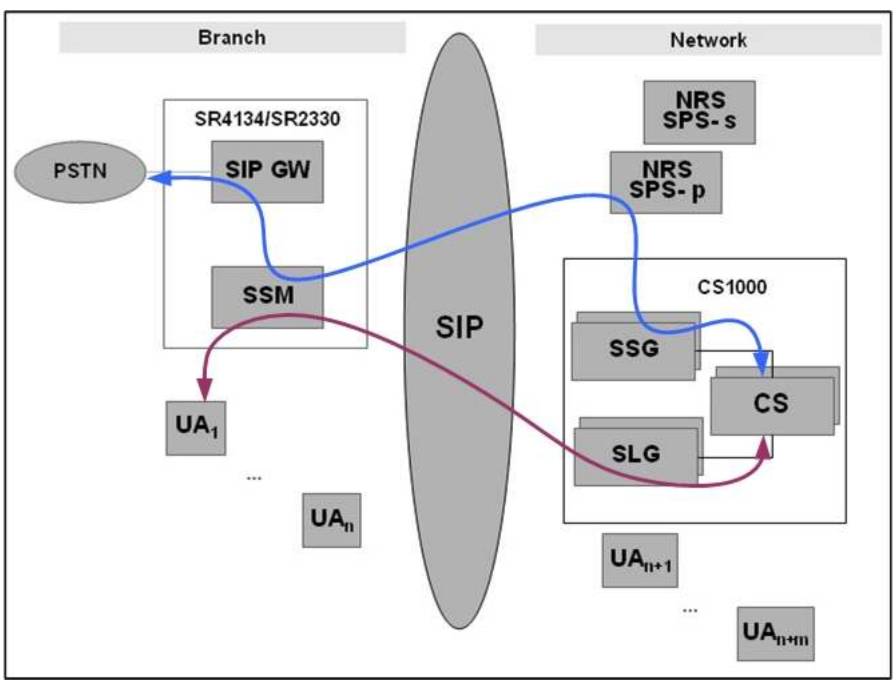
SIP GW interconnects SIP voice over IP networks with the PSTN. It also provides direct connections for analog phones, faxes and modems. In branch office deployment for CS 1000, SIP GW registers with the currently active NRS/SPS to enable newly active SPS to route calls to SIP GW. This is different than a user registration. SIP GW will monitor the reachability of NRS/SPS to know that current active NRS/SPS. SIP GW does this by sending OPTIONS messages to both primary and secondary NRS/SPS. If current active SPS is not reachable, then gateway will switch to the other SPS as the currently active SPS.

The following diagram shows the call flow between a branch PSTN interface/device and SIP endpoint connected to CS 1000. The SIP GW will route PSTN calls to an active SPS server with Req-Uri having IP address of the active NRS/SPS via SSM. The SSM will then replace the IP address with CS 1000 domain in 'From', 'To' and Req-Uri headers. If none of the NRS SPSs is available, then SIP GW will send calls to SSM with Req-Uri having host as domain configured for SLG in SSM e.g. nortel.com. This routing happens on the basis of entries defined in Normal mode NTML dial plan. Normal mode dial plan defines routes for both primary SPS and secondary SPS. SSM will choose the route from NTML depending upon destination set in SIP messages by SIP GW.

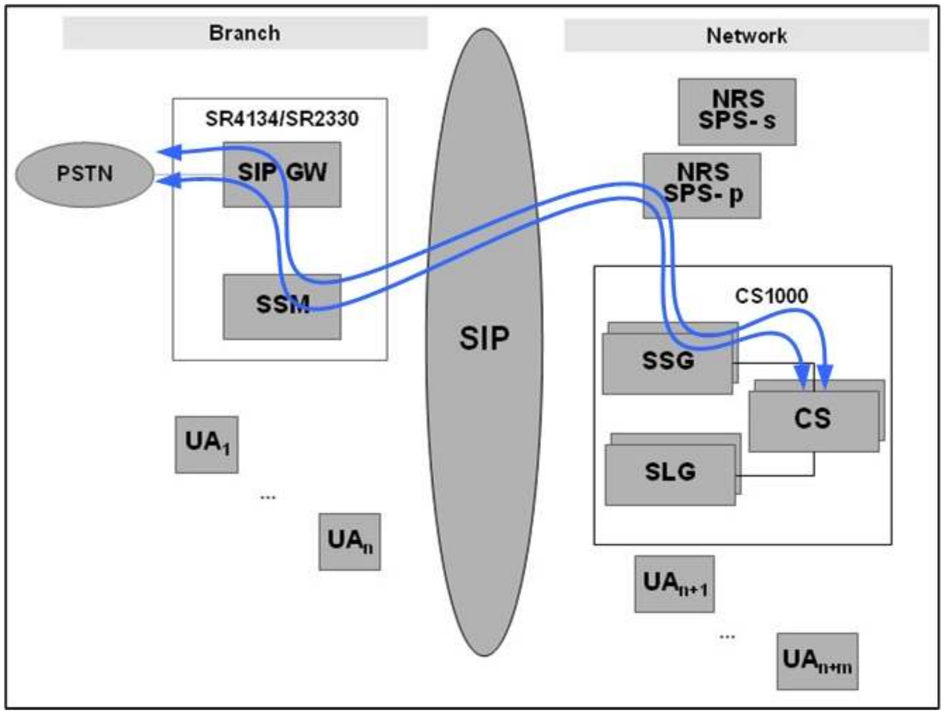
SSM will receive the incoming calls from SPS that are destined for PSTN. For such calls SSM will not be able to find a mapped contact in its registration cache and hence routes the call to configured default gateway which is SIP GW.



The following diagram shows the call flow between a branch PSTN interface/device and branch SIP endpoint.

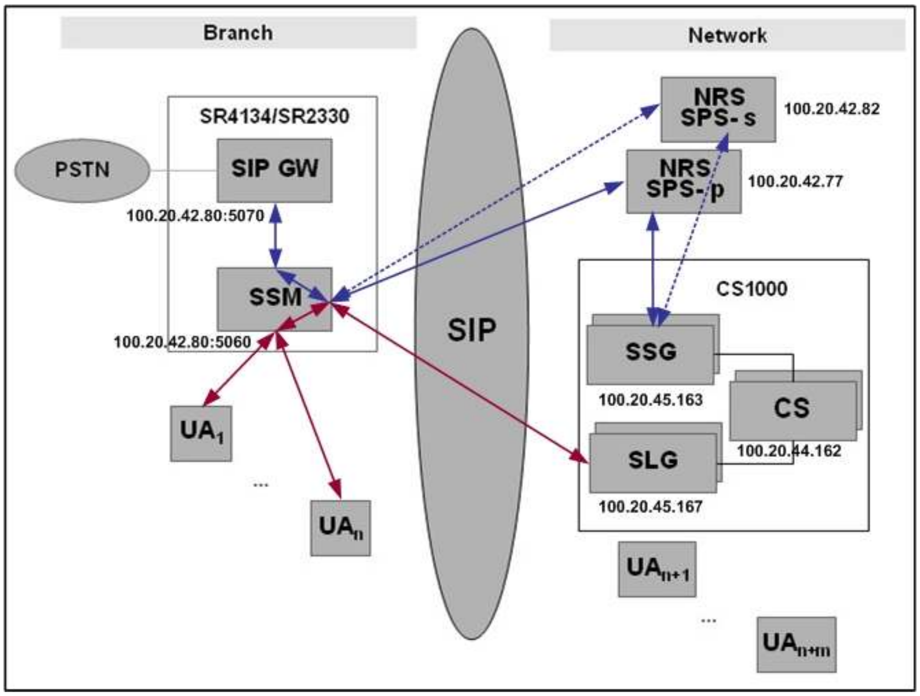


The following diagram shows the call flow between branch PSTN interfaces/devices.



CS 1000 Configuration

This section describes the configuration steps needed on CS 1000 components like SLG, SSG and SPS for SR 2330/4134 based branch solution. There is no SR 2330/4134 specific configuration required on CS 1000. Please refer to CS 1000 user guides for the detailed configuration steps. The following diagram complete with IP addresses will be used as reference for the configuration chapters.



SLG Configuration

Please refer to CS 1000 NN43001-508 SIP Line Fundamentals document for detailed information on SLG configuration. There is no Secure Router specific configuration need to done on SLG.

Steps

1. SIP Line (SIPL) feature depends on the following packages to be enabled in keycode.

Package Mnemonic	Package Number	Package Description	Package Type (New or Existing or Dependency)	Applicable Market
SLS_Package	417	SIP Line Service	New	Global
FFC	139	Flexible Feature Codes	Existing	Global
SIP_LINE_NT_PKG	415	Nortel SIP Line Package	Existing	
SIP_LINE_3P_PKG	416	3rdParty SIP Line Package	Existing	

2. Deploy SIP Line software application on the Linux server

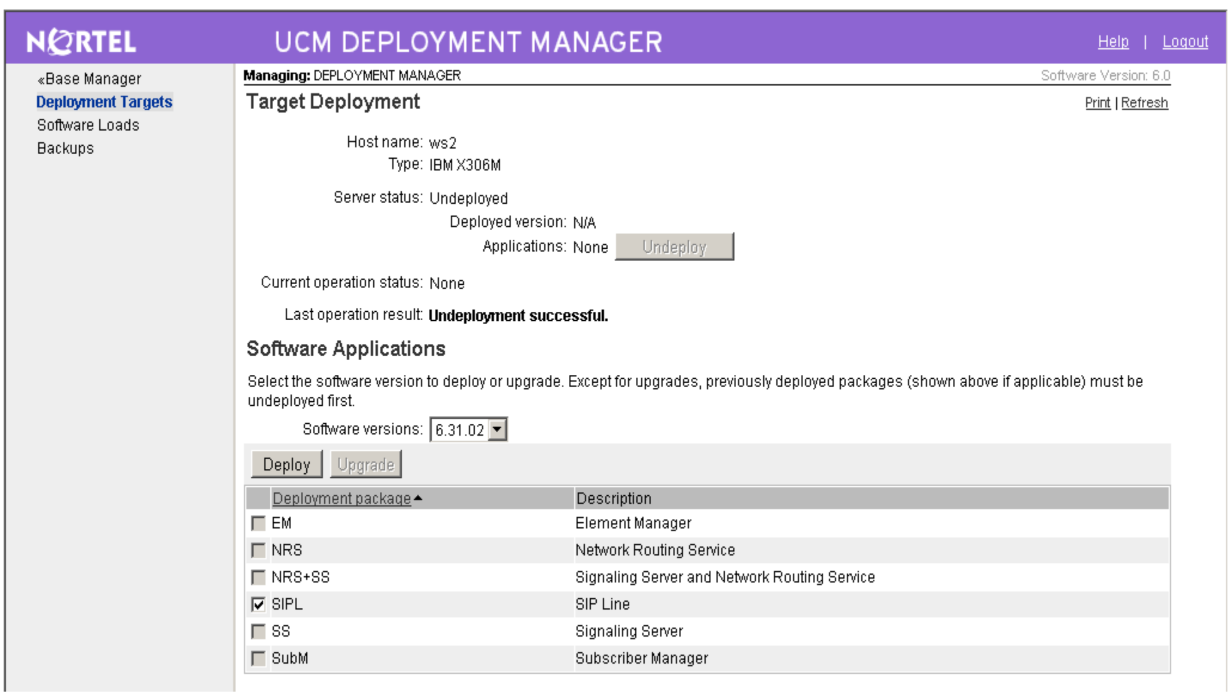


Figure SIPL 1 – SIP Line application deployment

3. SIPL node configuration

+ Nodes: Servers, Media Cards

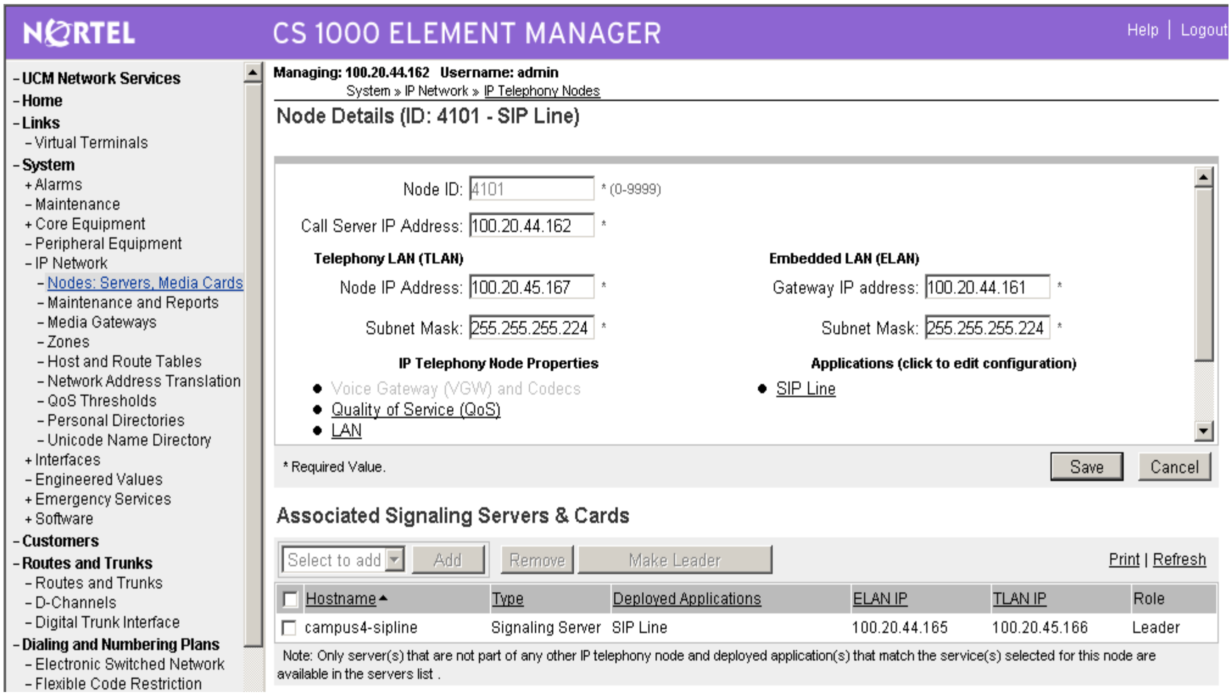


Figure SIPL 2 – SIPL node details

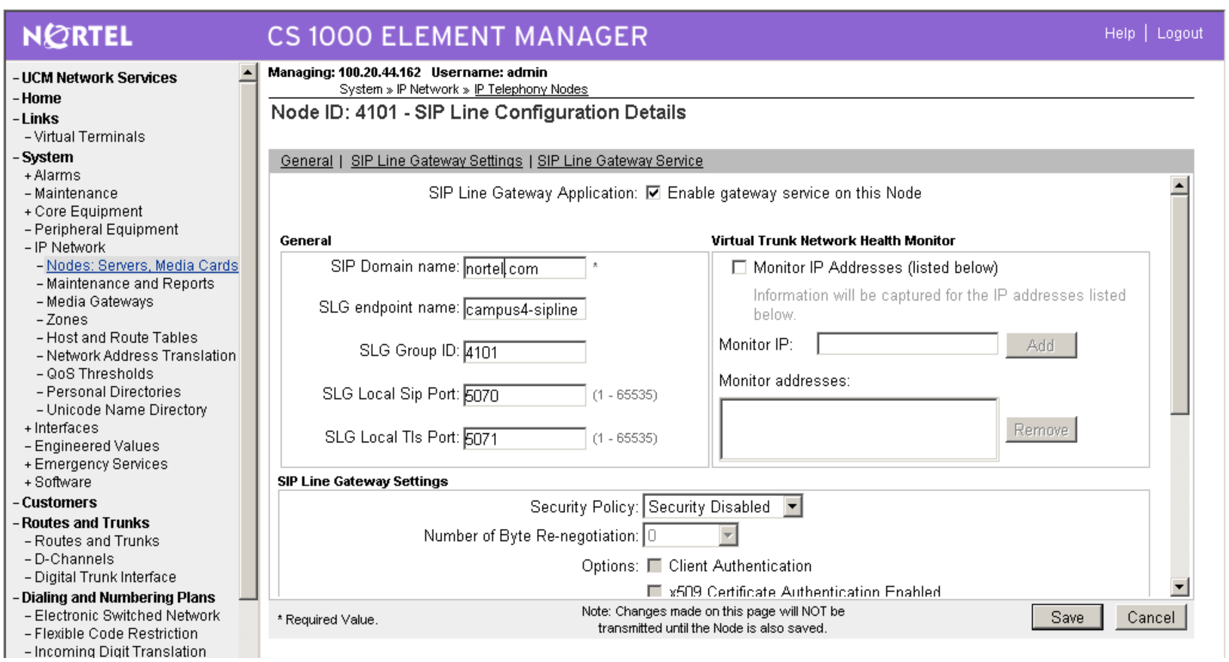


Figure SIPL 3 – General SIPL node configuration

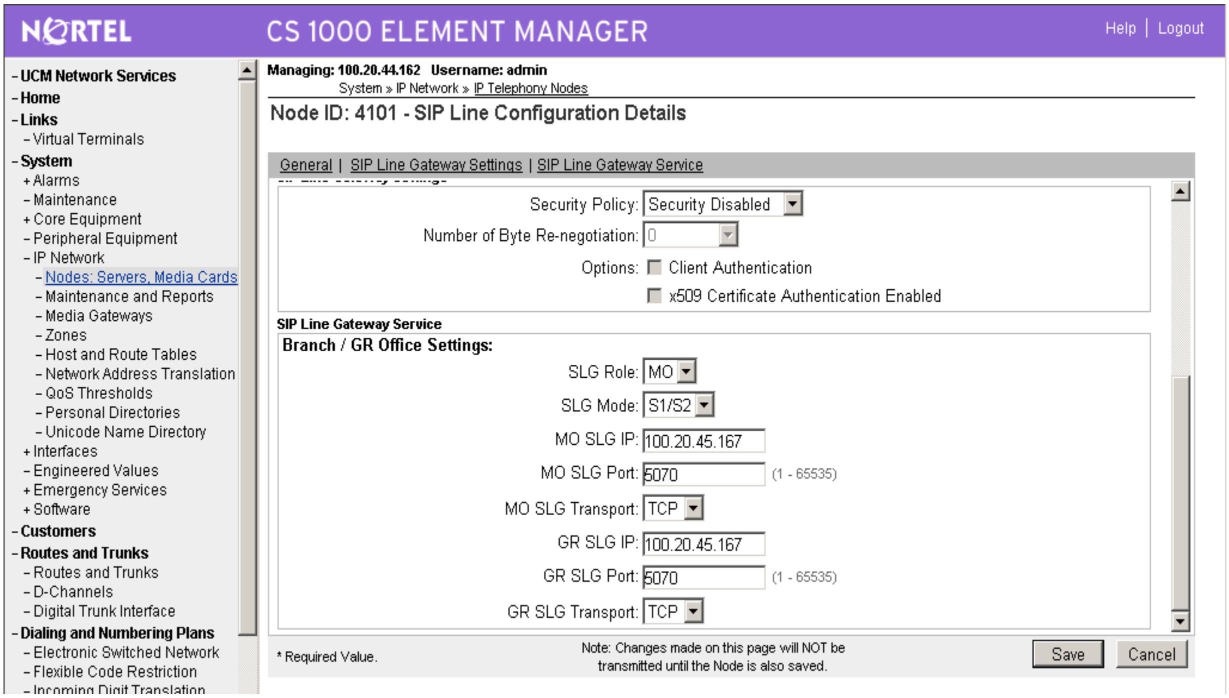


Figure SIPL 4 – SIP Line Gateway Service configuration

4. The SIP Line service must be enabled on a customer level

+ Customer --> Cus Number --> SIP Line Service

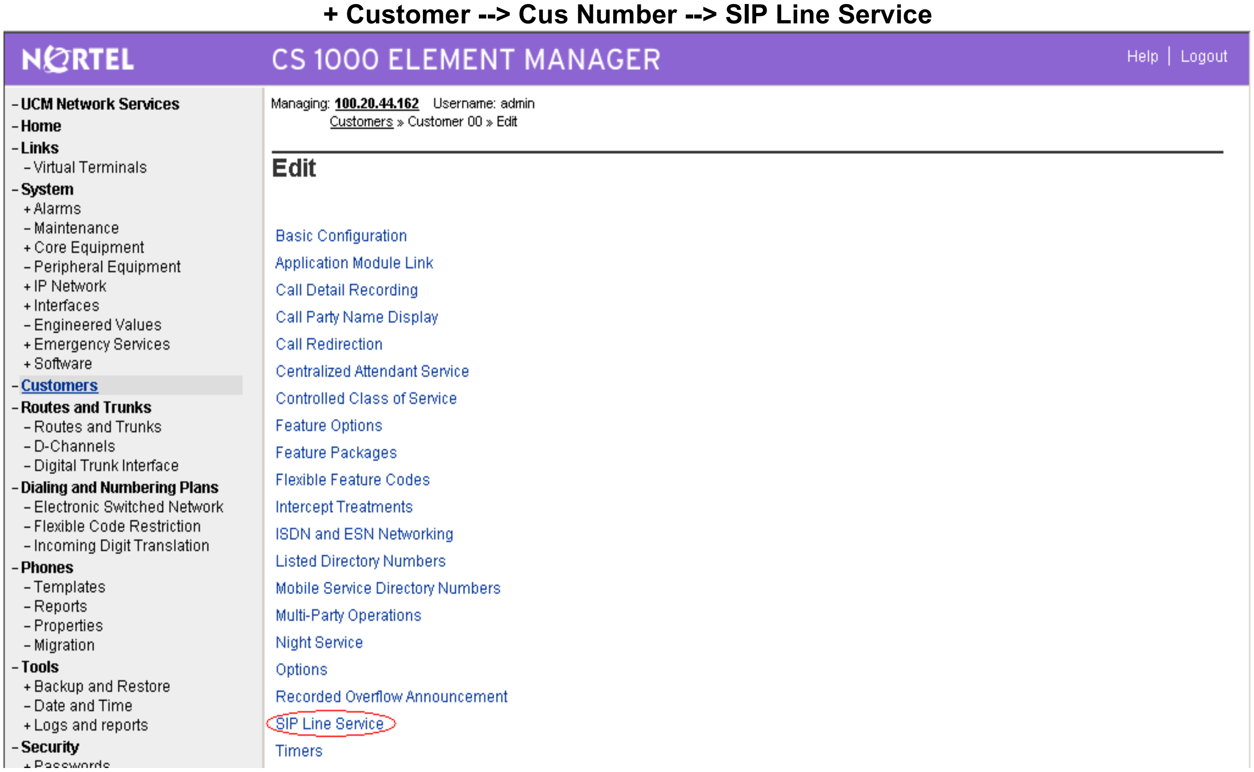


Figure SIPL 5 - SIP Line service inside Customer edit page

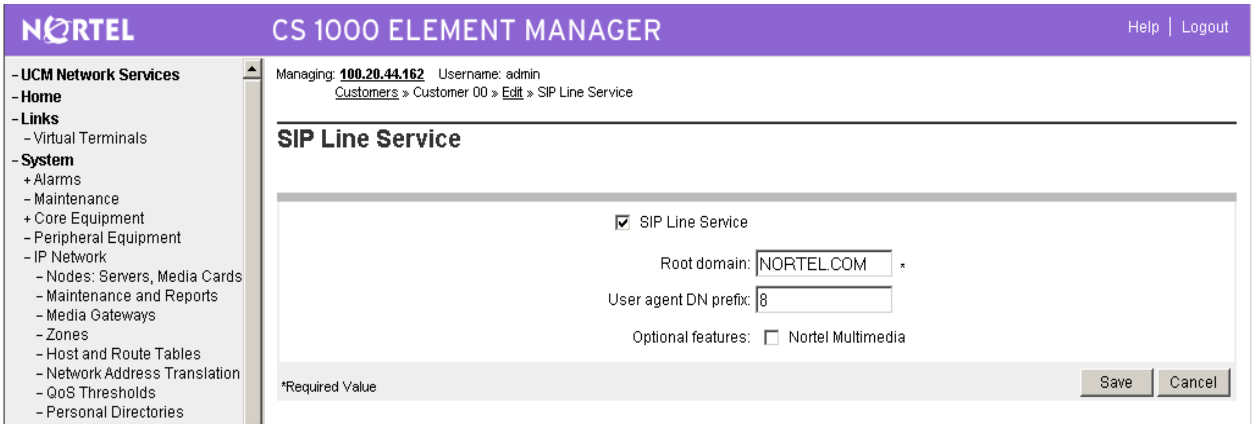


Figure SIPL 6 - SIP Line service enable page

5. Password length configuration for SIP clients

+ Customer -->CUS# -->Flexible Feature Codes

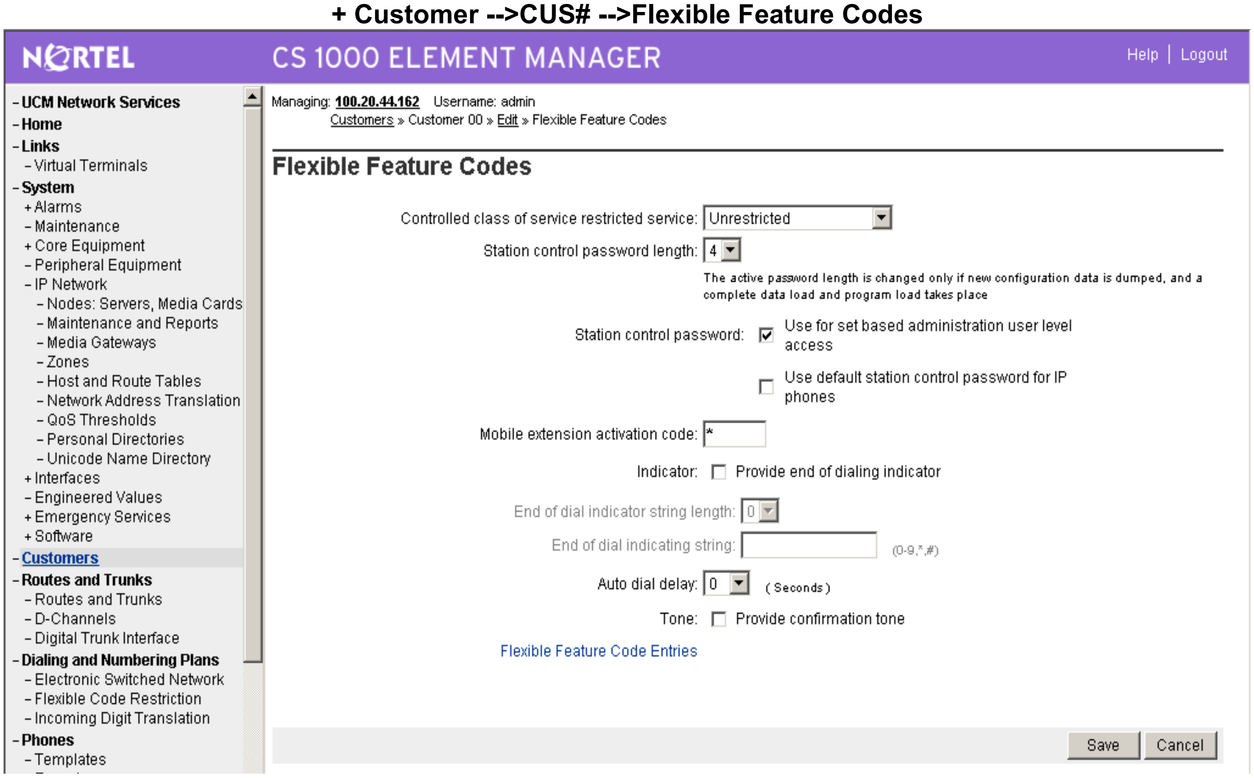


Figure SIPL 7 - Password length configuration

6. Enable ISDN for trunking

+ Customer --> CUS# --> Features Packages

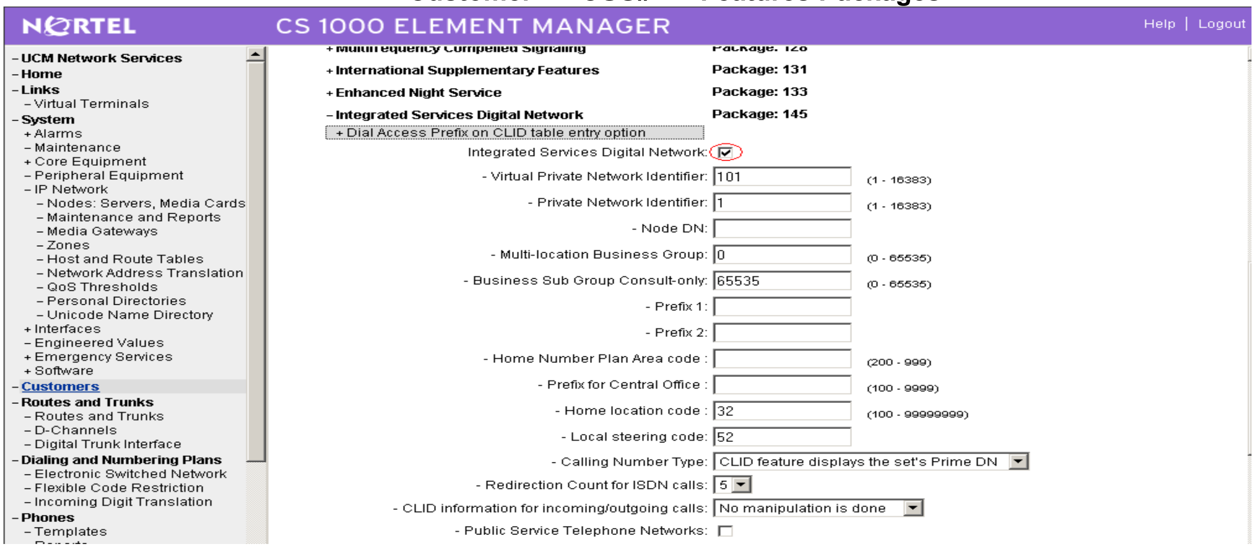


Figure SIPL 8 – Enable ISDN feature

7. AML and VAS configuration

+ System --> Interfaces --> Application Module Link (must over 32)

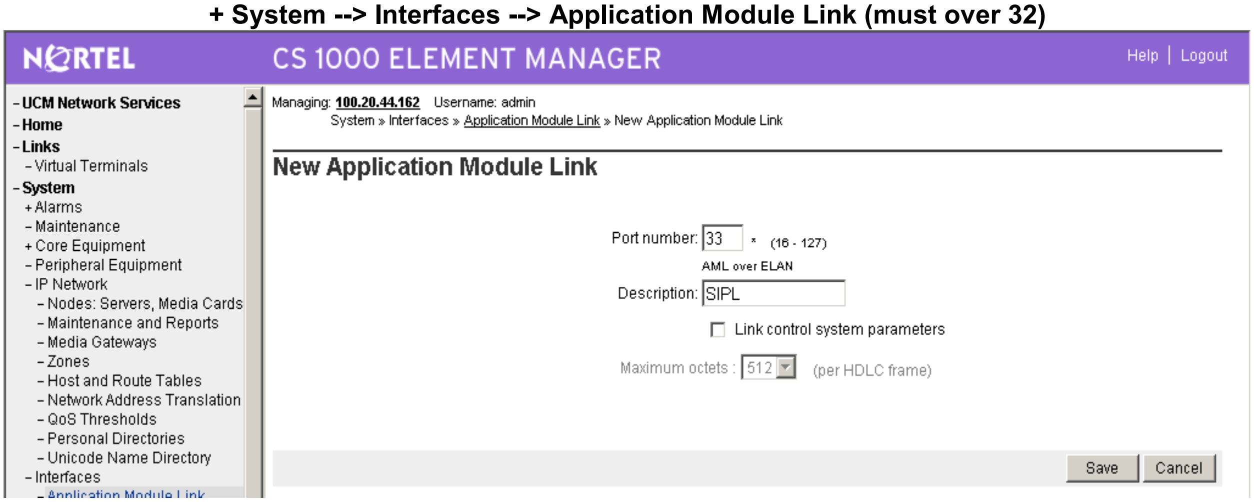


Figure SIPL 9 – AML configuration

+ System --> Interfaces --> Value Added Server --> Add -->Application Module Link

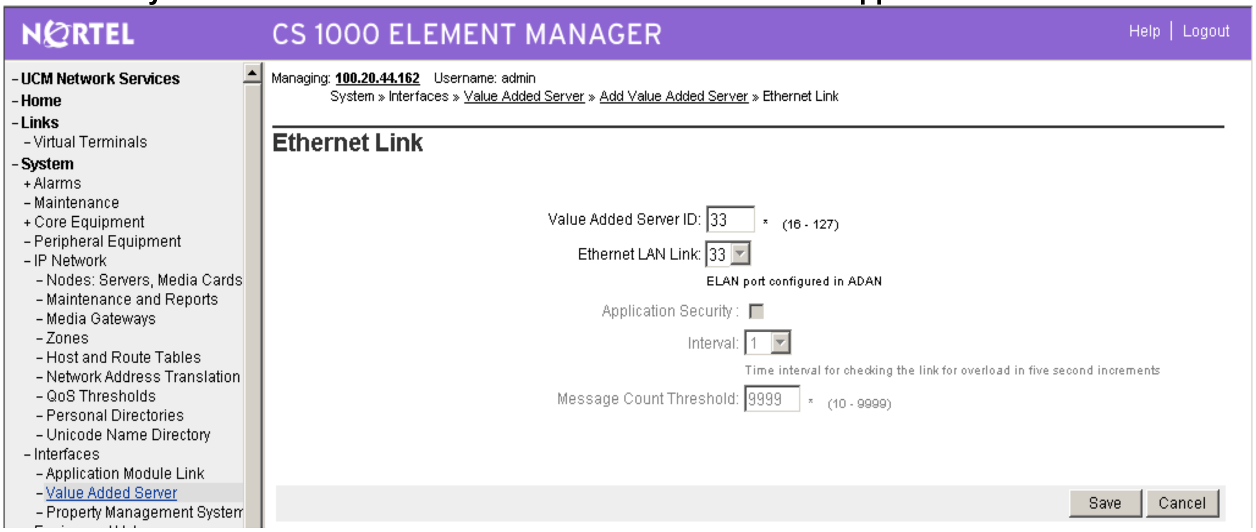


Figure SIPL 10 – AML configuration

8. D-channel/Route/Trunk for SIPL service

+ Routes and Trunks --> D-Channel

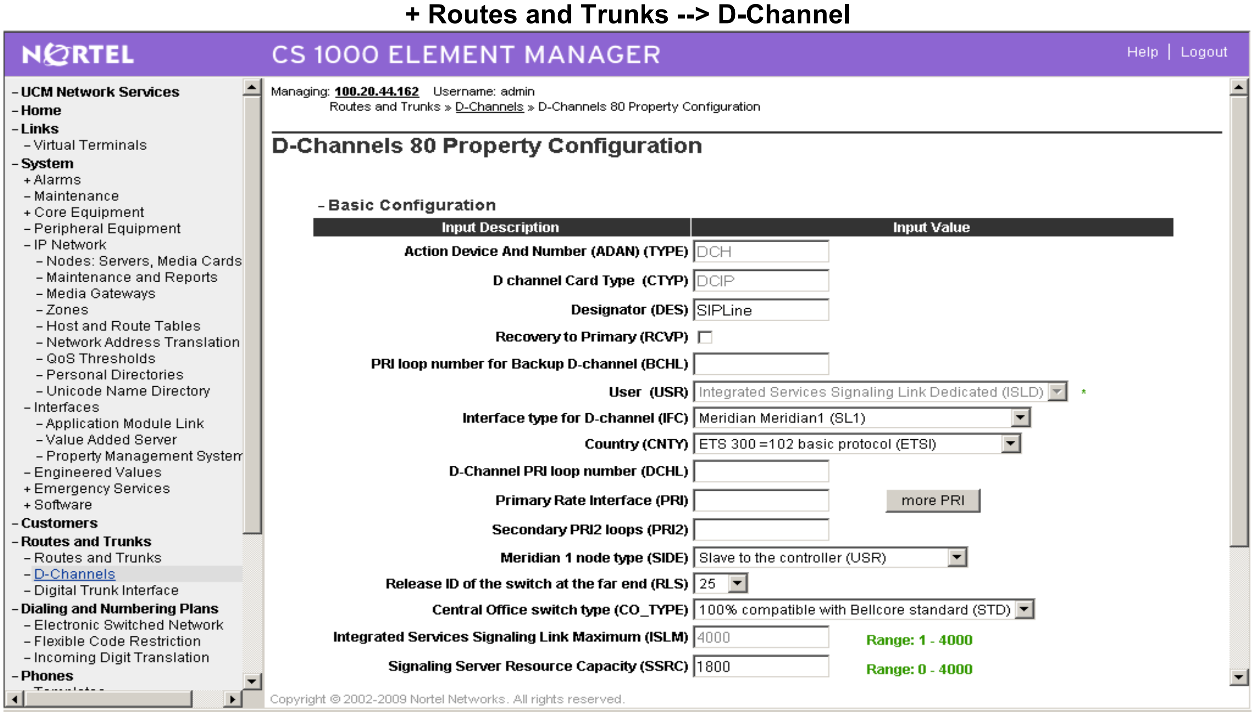


Figure SIPL 11 – D-Channel configuration for SIPL service

+ Routes and Trunks --> Routes and Trunks --> CUS# --> Add route

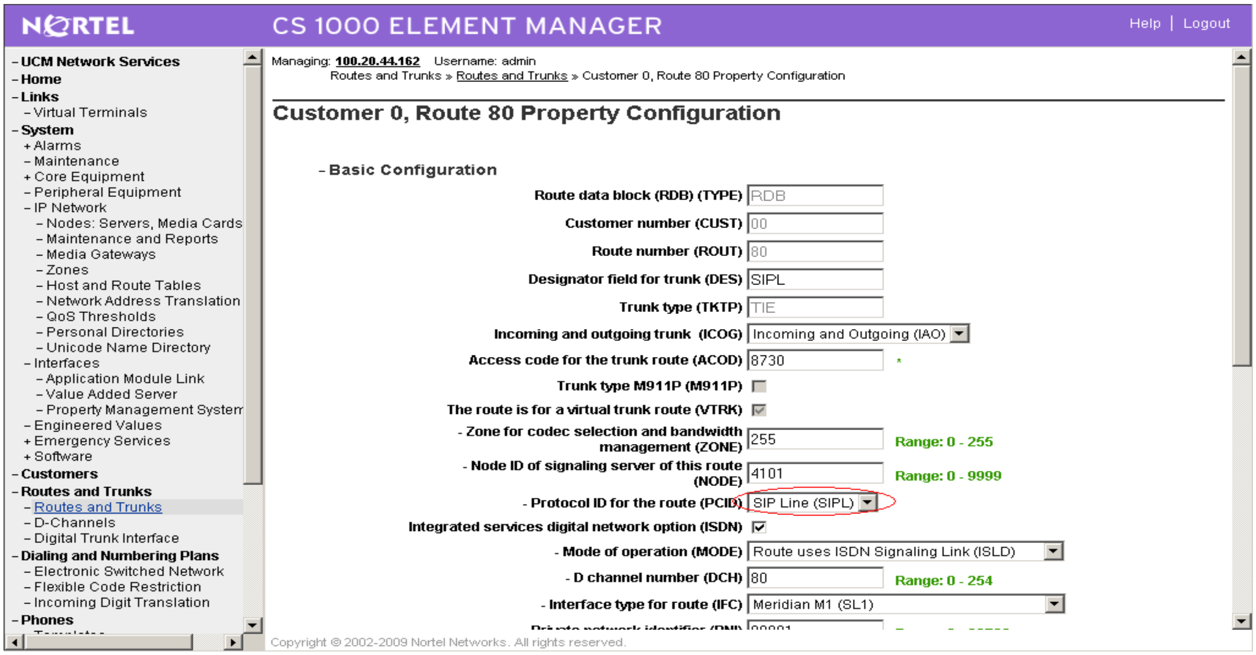


Figure SIPL 12a – Route configuration for SIPL service

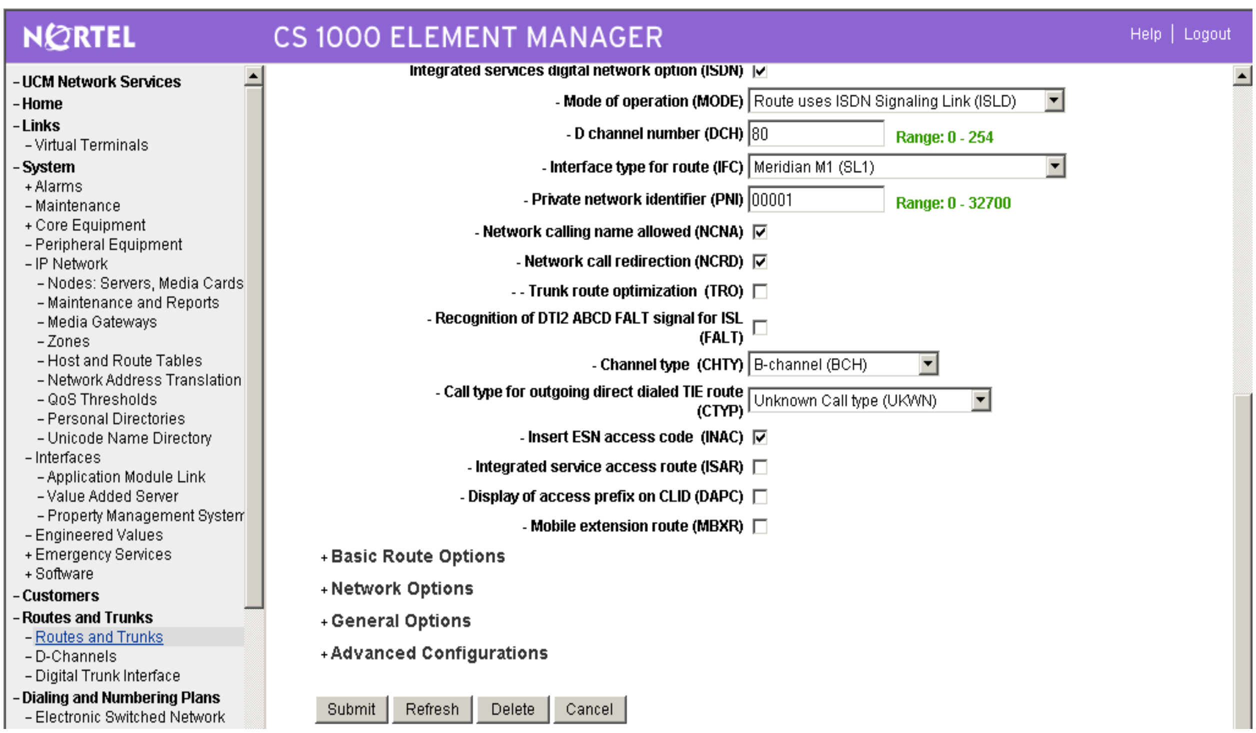


Figure SIPL 12b – Route configuration for SIPL service

+ Routes and Trunks --> Routes and Trunks --> CUS# --> Route# --> Add Trunks

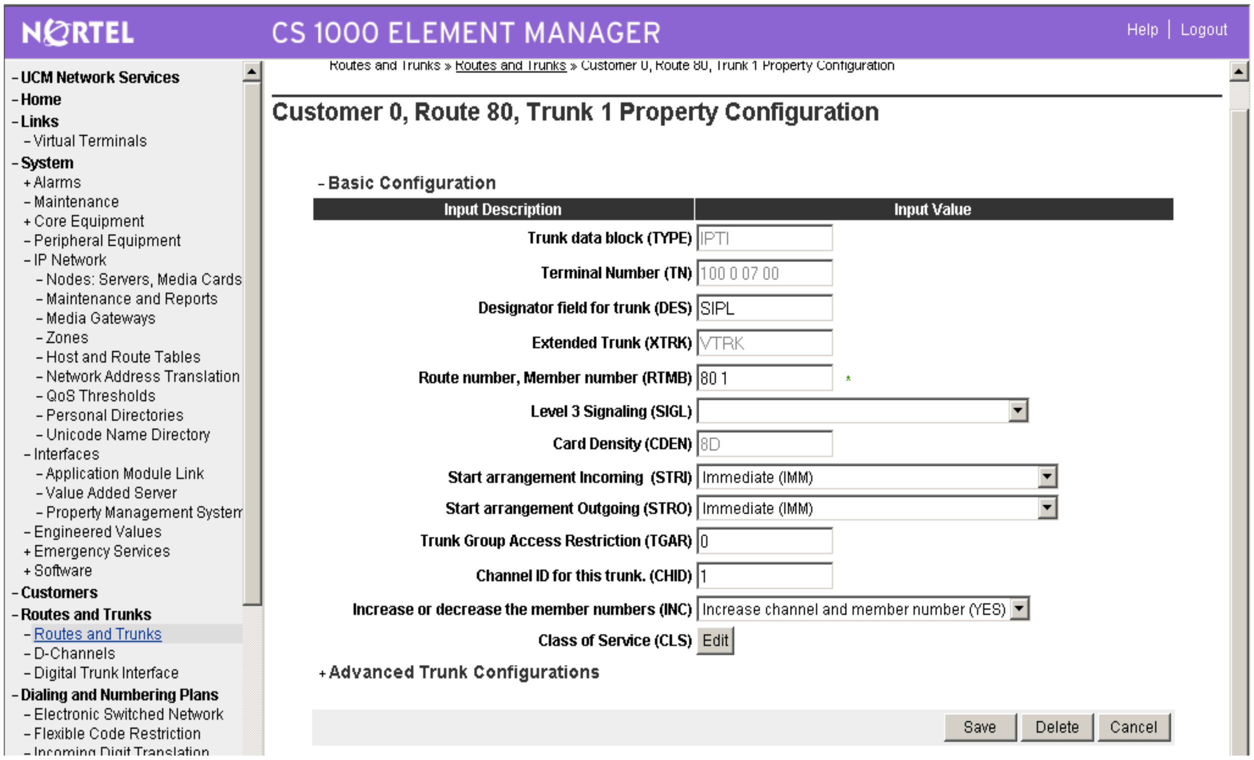


Figure SIPL 13 – Trunk configuration for SIPL service

9. SIPL phone configuration

+ Phones --> Add --> choice UEXT-SIPL phone --> preview

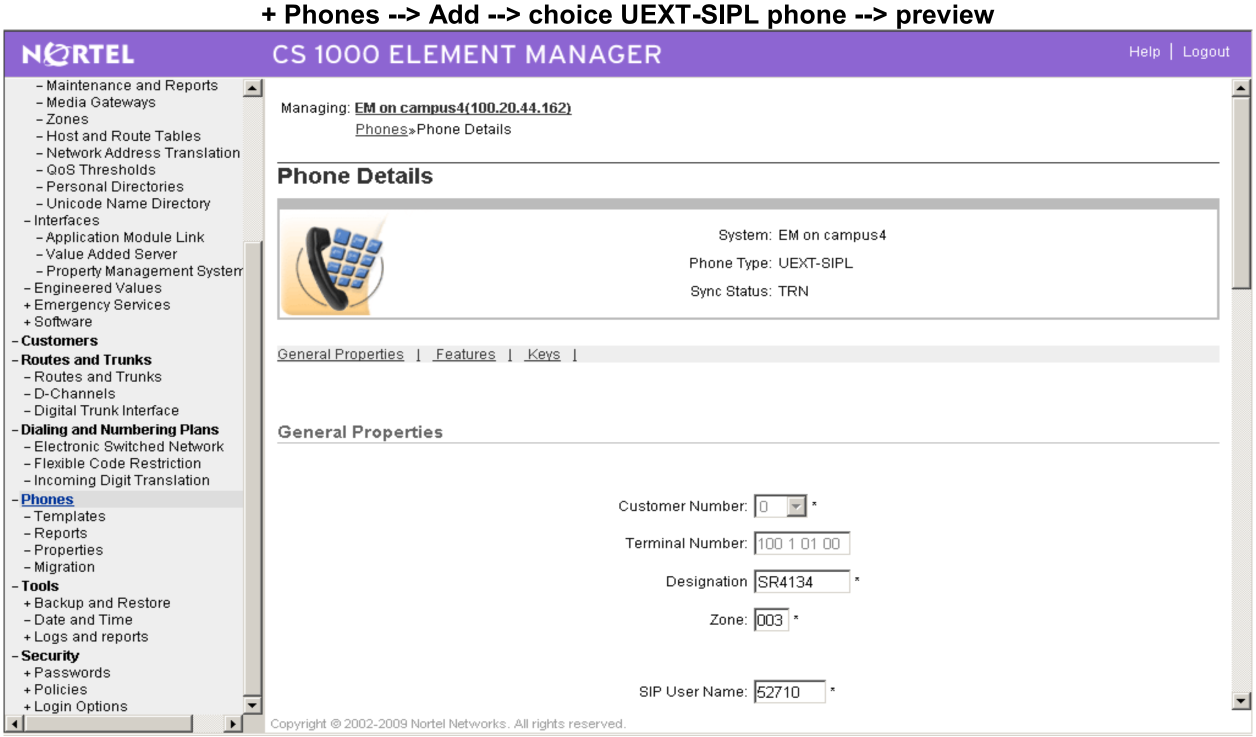


Figure SIPL 14a – SIPL phone configuration

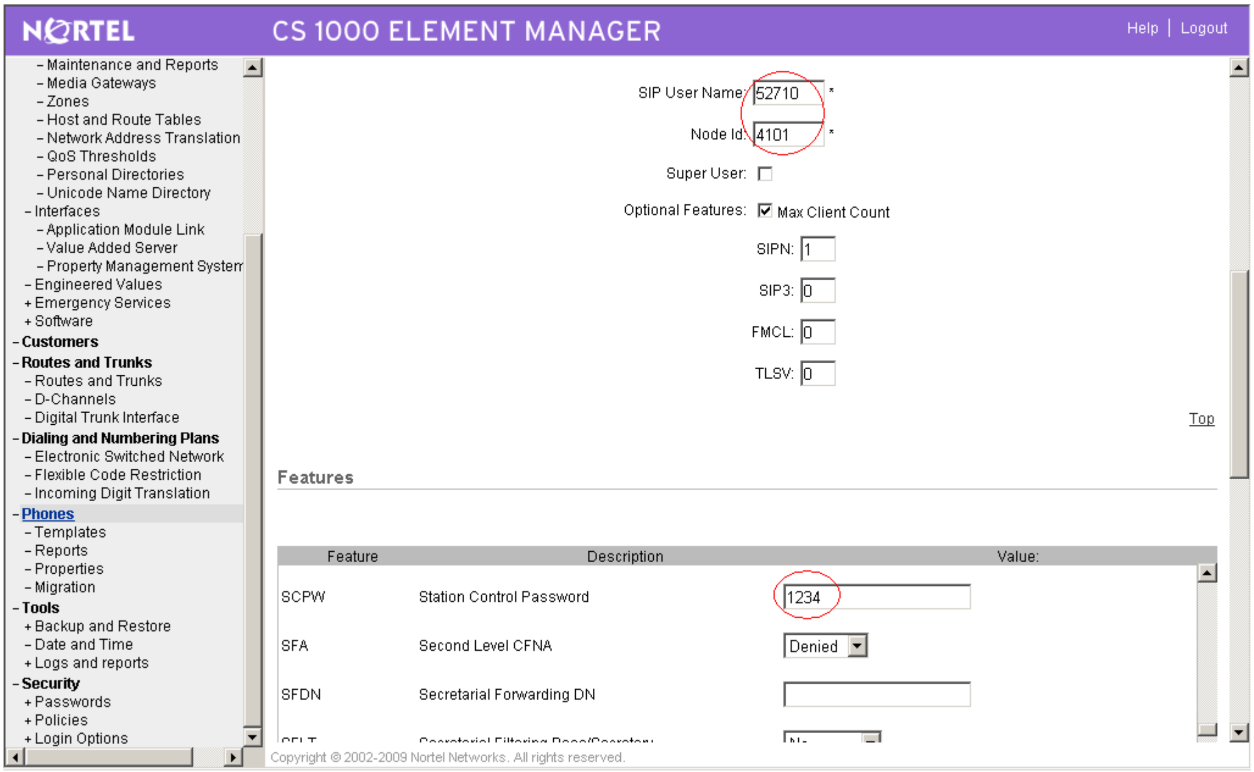


Figure SIPL 14b – SIPL phone configuration

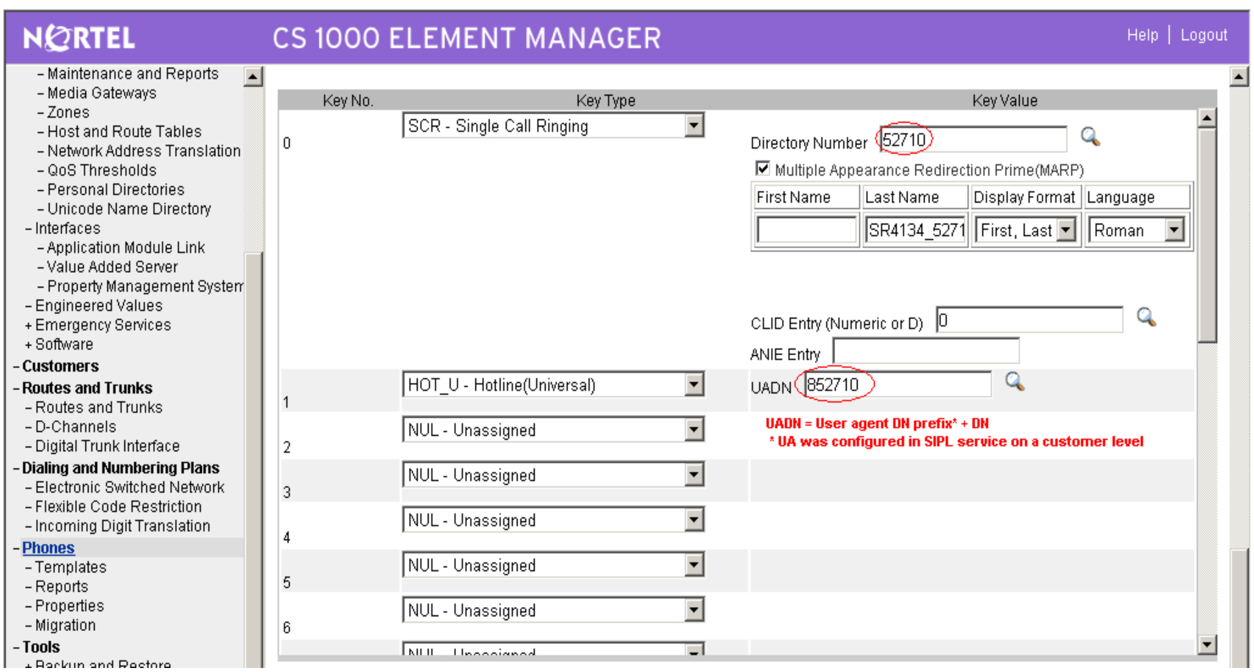


Figure SIPL 14c – SIPL phone configuration

SSG Configuration

Please refer to CS 1000 document for detailed information on SSG configuration. SIPL needs a SSG server to route an external call to NRS (Network Routing Service). SIPGW and H323GW endpoints will be configured in SSG to register on specific NRS server.

Steps

1) Deploy Signaling Server software application on the Linux server

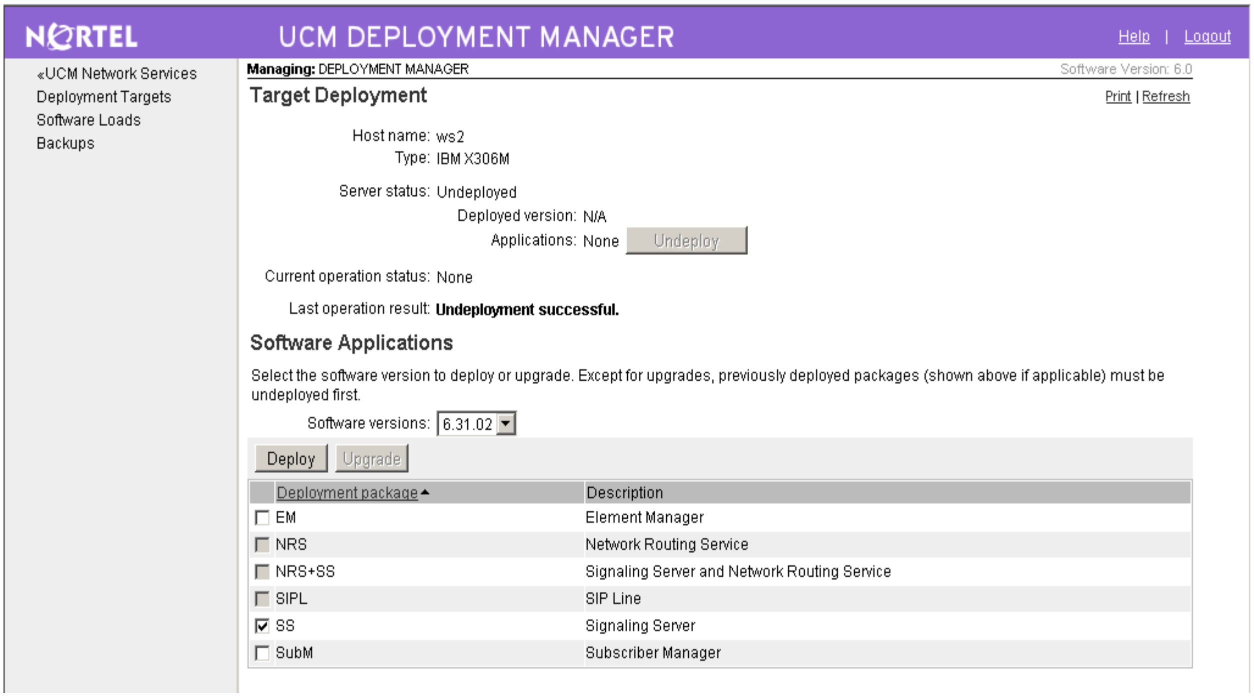


Figure SSG 1 – SSG application deployment

2) SSG node configuration:

+ Nodes: Servers, Media Cards

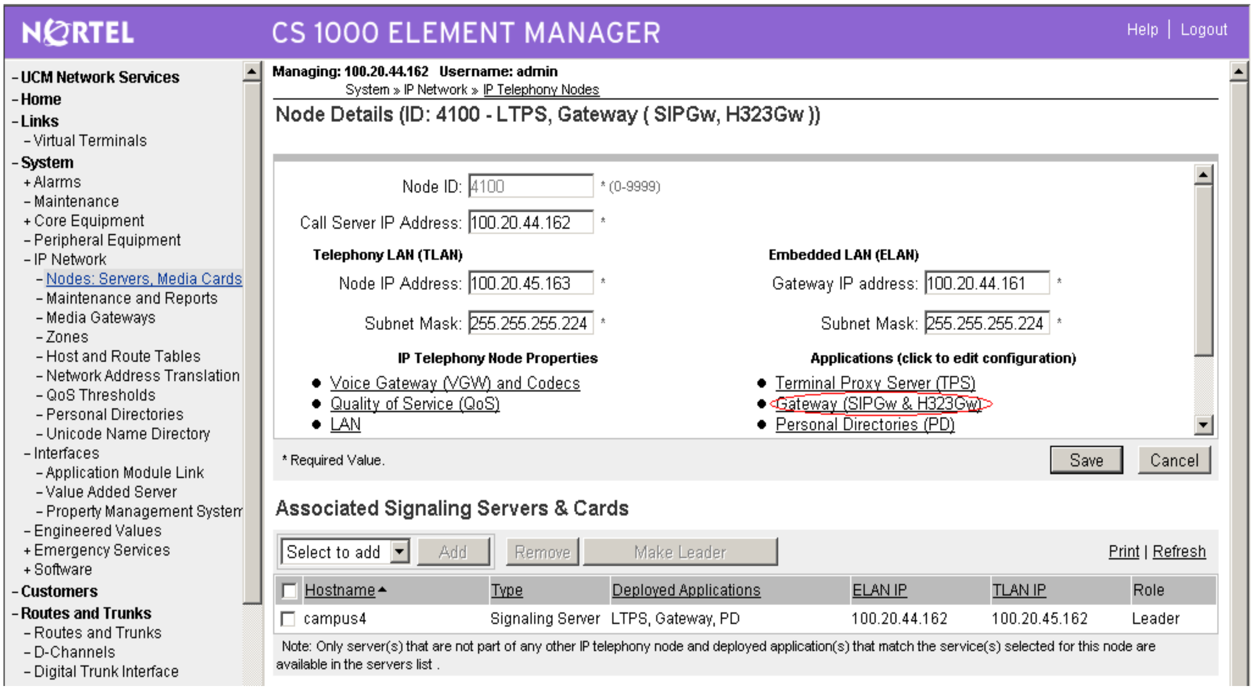


Figure SSG 2 – SSG node details

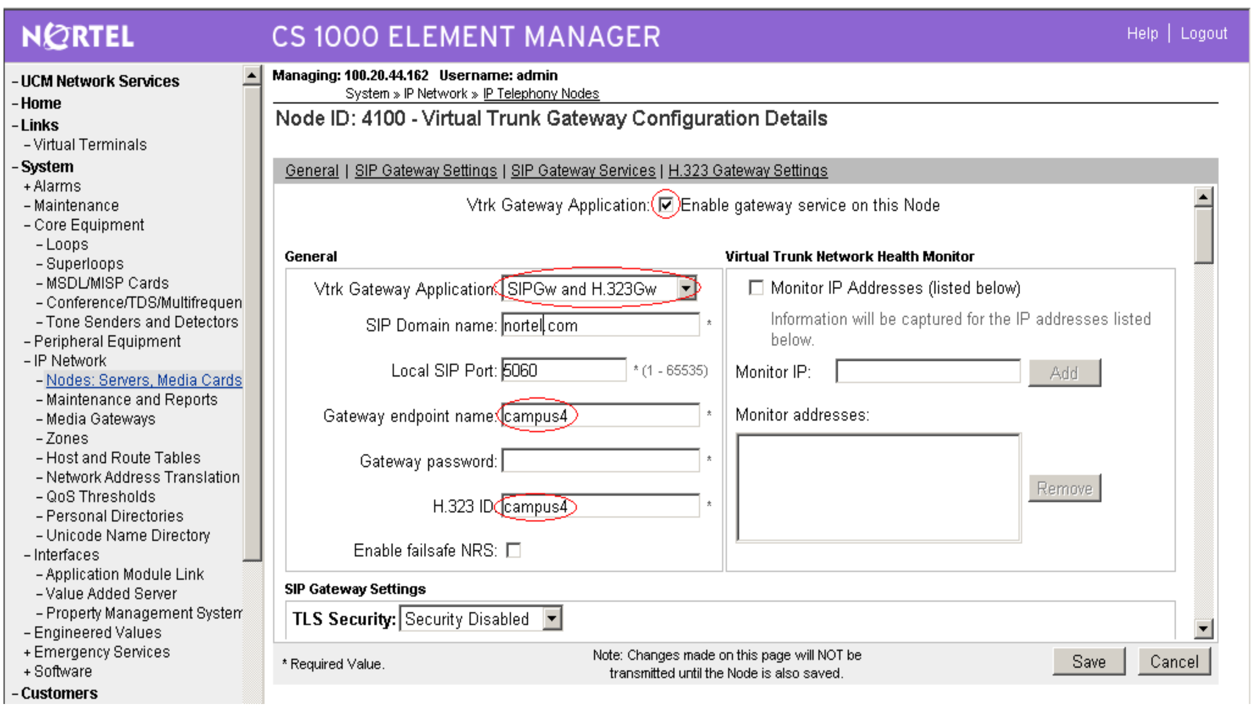


Figure SSG 3 – SIPGw and H323Gw endpoints configuration

3) Specify a NRS server for SIPGw endpoint:

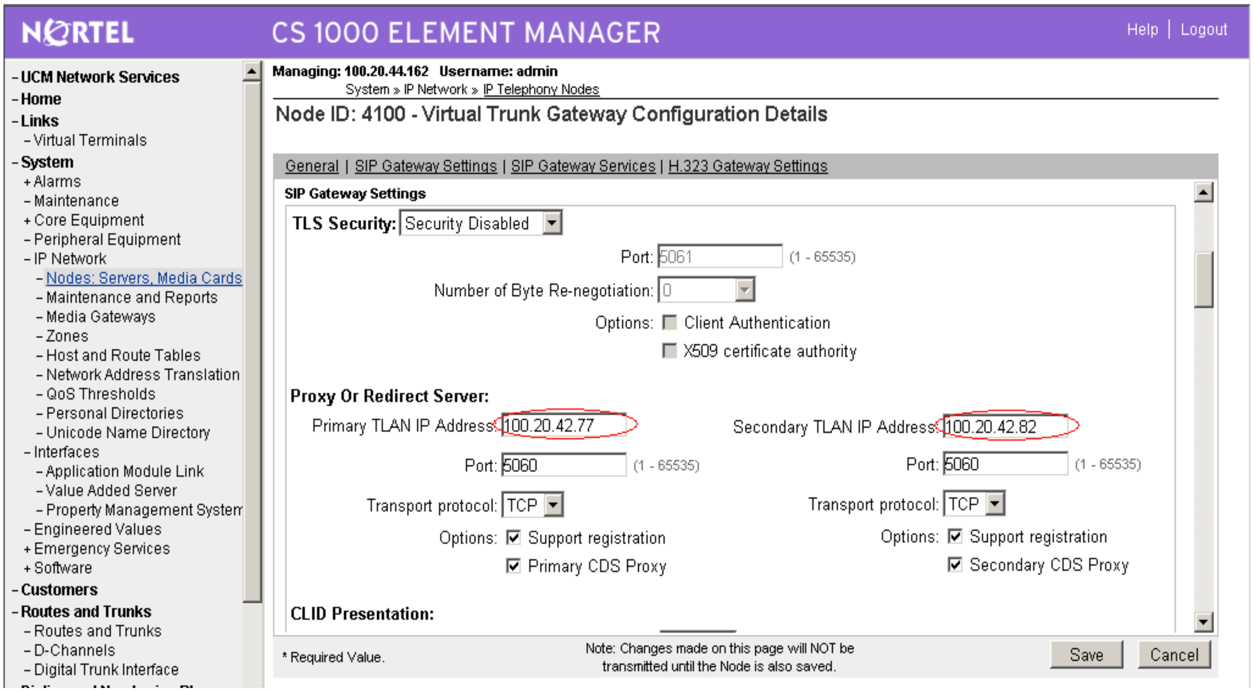


Figure SSG 4 – Specify a NRS server for SIPGw endpoint

4) Specify a GateKeeper server (NRS) for H323Gw endpoint:

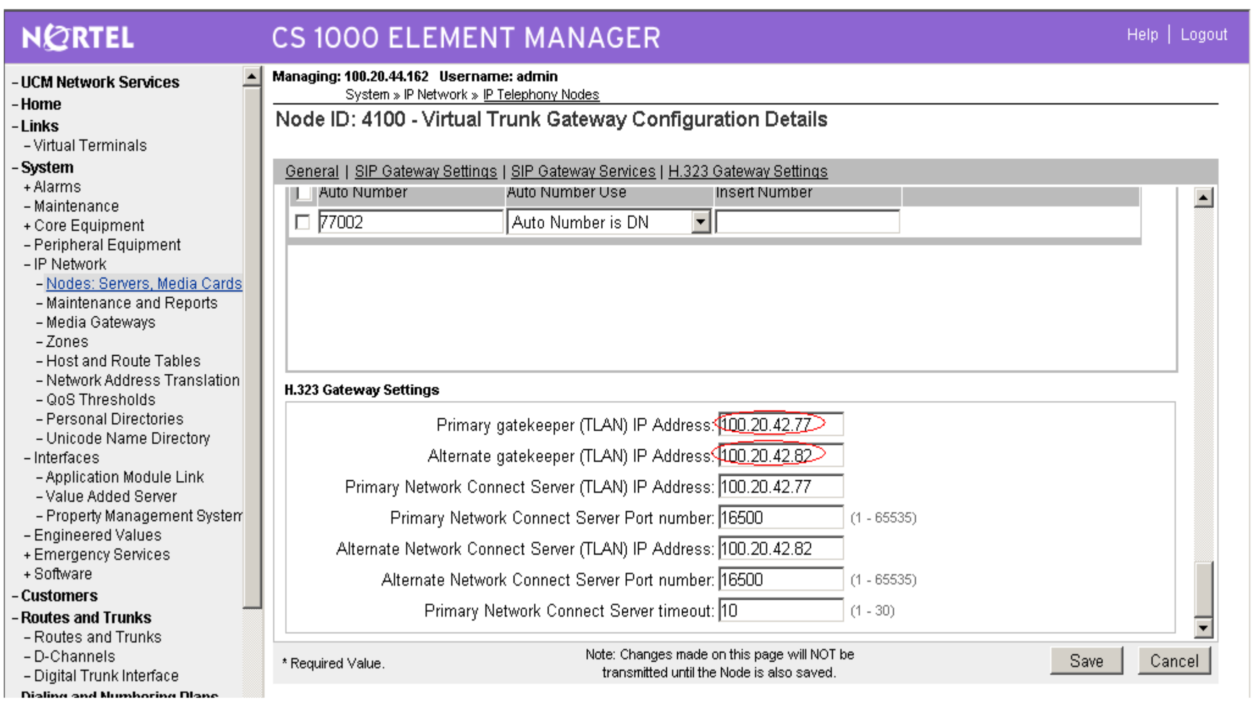


Figure SSG 5 – Specify a GateKeeper server(NRS) for H323Gw endpoint

NRS/SPS Configuration

Please refer to CS 1000 NN43001-130 Network Routing Service Fundamentals document for detailed information on NRS/SPS configuration. The Network Routing Service (NRS) provides routing services to both SIP and H.323-compliant devices. The NRS allows customers to manage a single network dialing plan for SIP, H.323, and mixed SIP/H.323 networks. Therefore SIPGw and H323Gw endpoints of SSG and Secure Router endpoint have to register on NRS server.

Steps

1) Deploy NRS software application on the Linux server

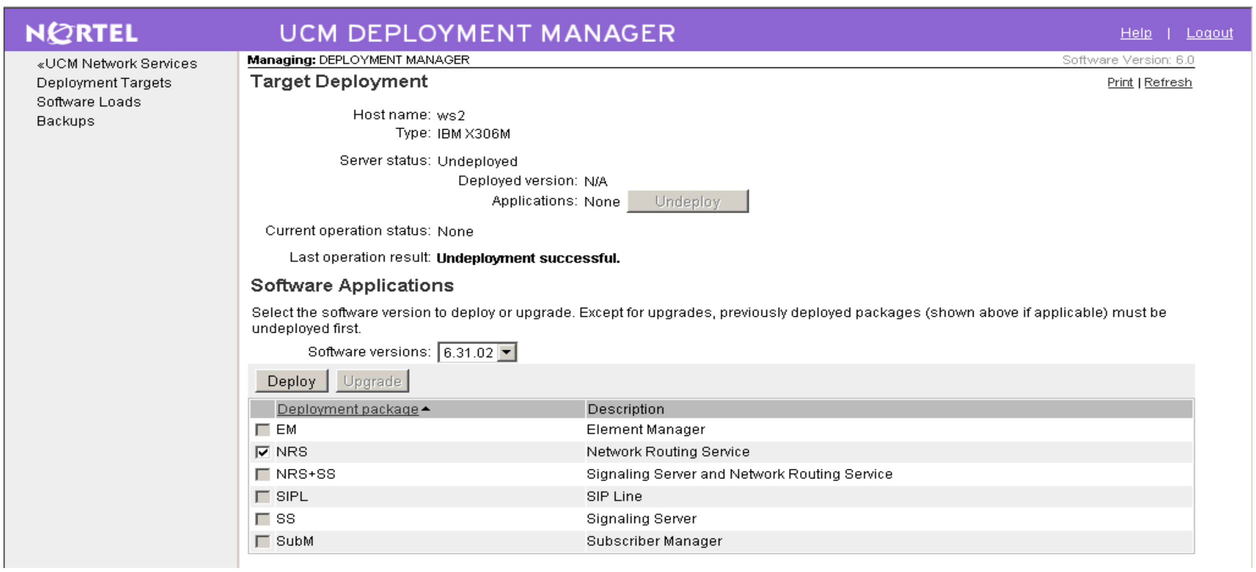


Figure NRS 1 – NRS application deployment

2) NRS server configuration

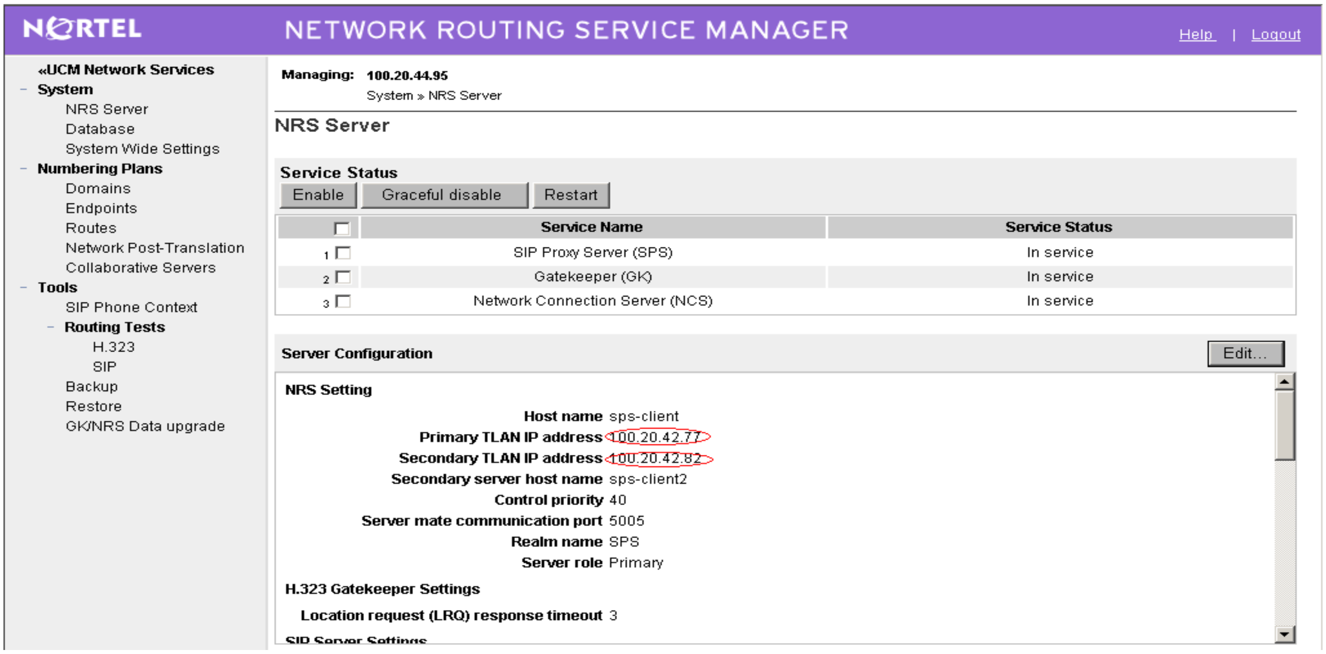


Figure NRS 2 – NRS server configuration

3) SSG endpoint configuration

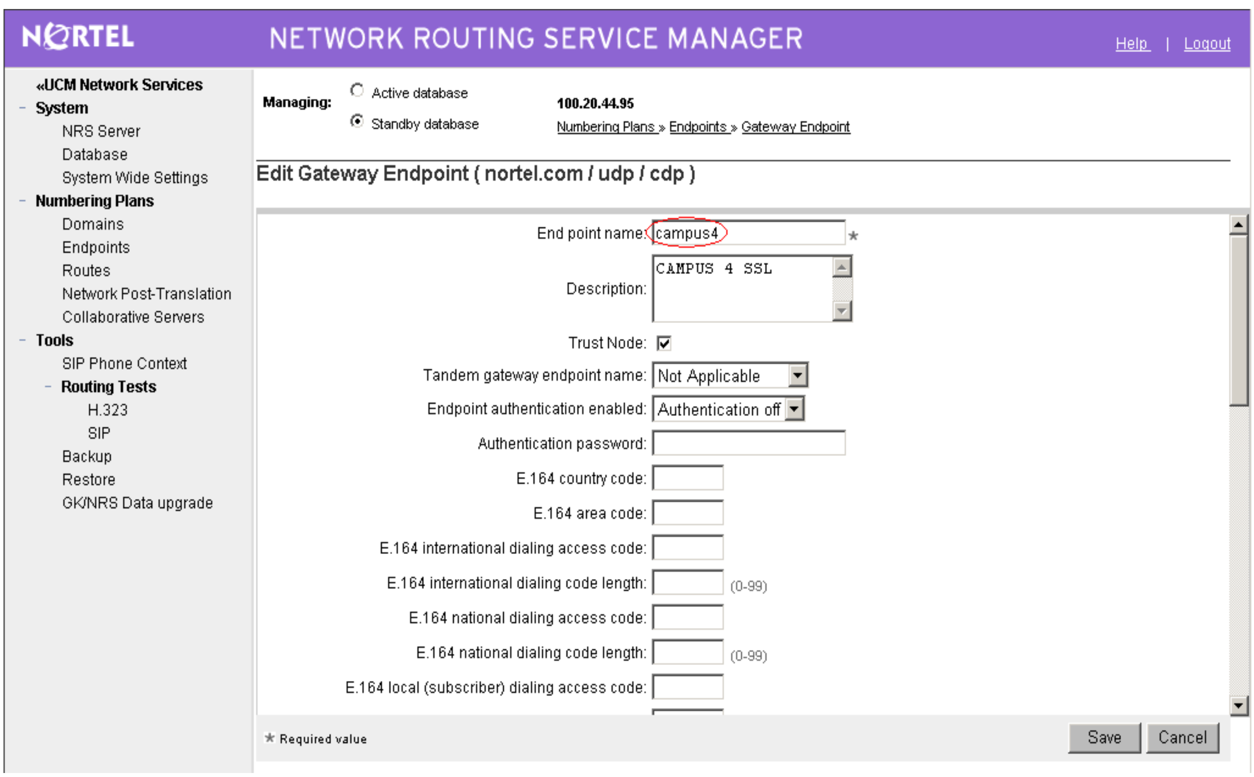


Figure NRS 2a – SSG endpoint configuration

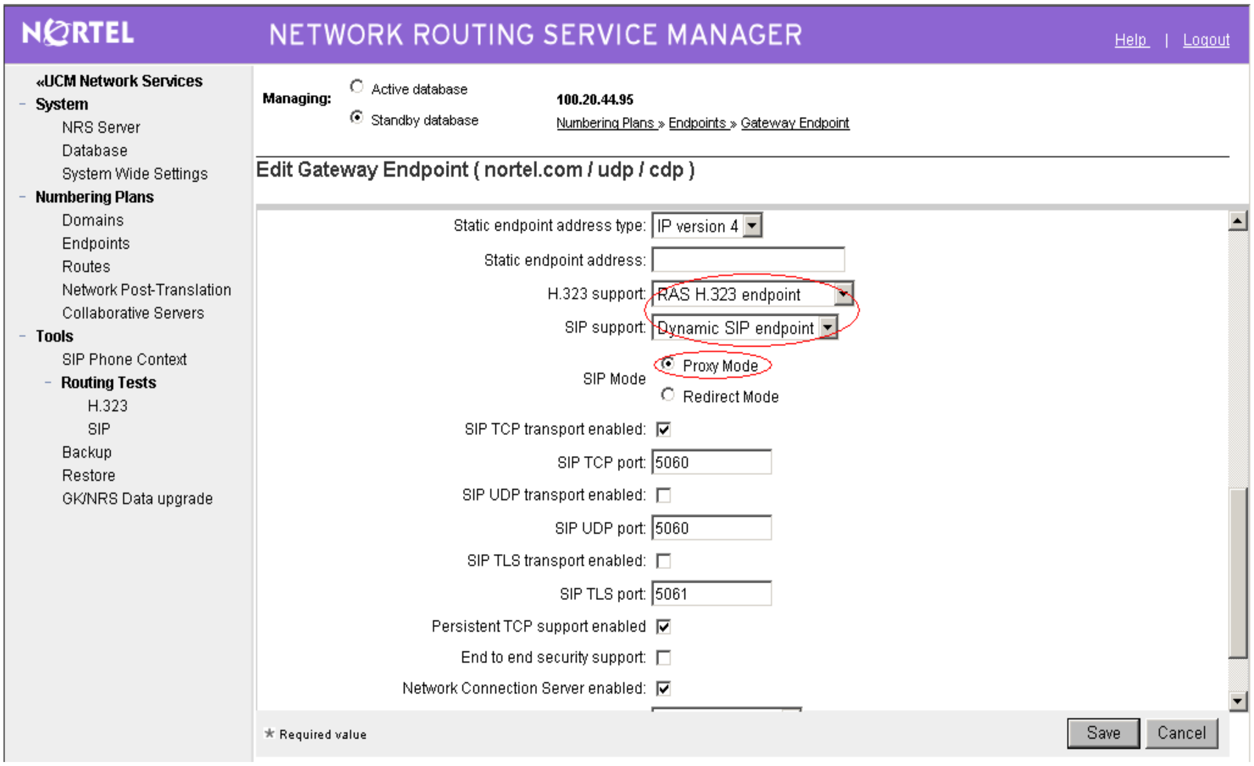


Figure NRS 2b – SSG endpoint configuration

4) Dialing plan routes for SSG endpoint configuration:

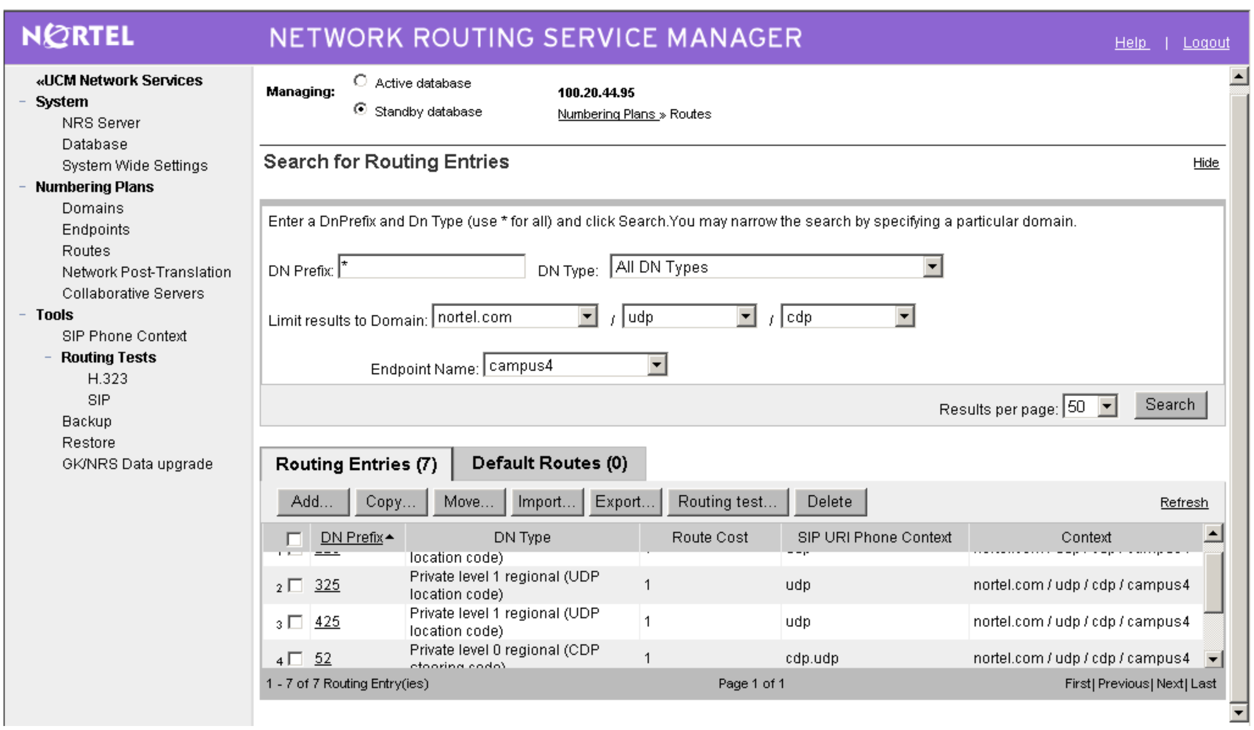


Figure NRS 3 – Dialing plan routes for SSG endpoint configuration

5) SSG endpoint registration status

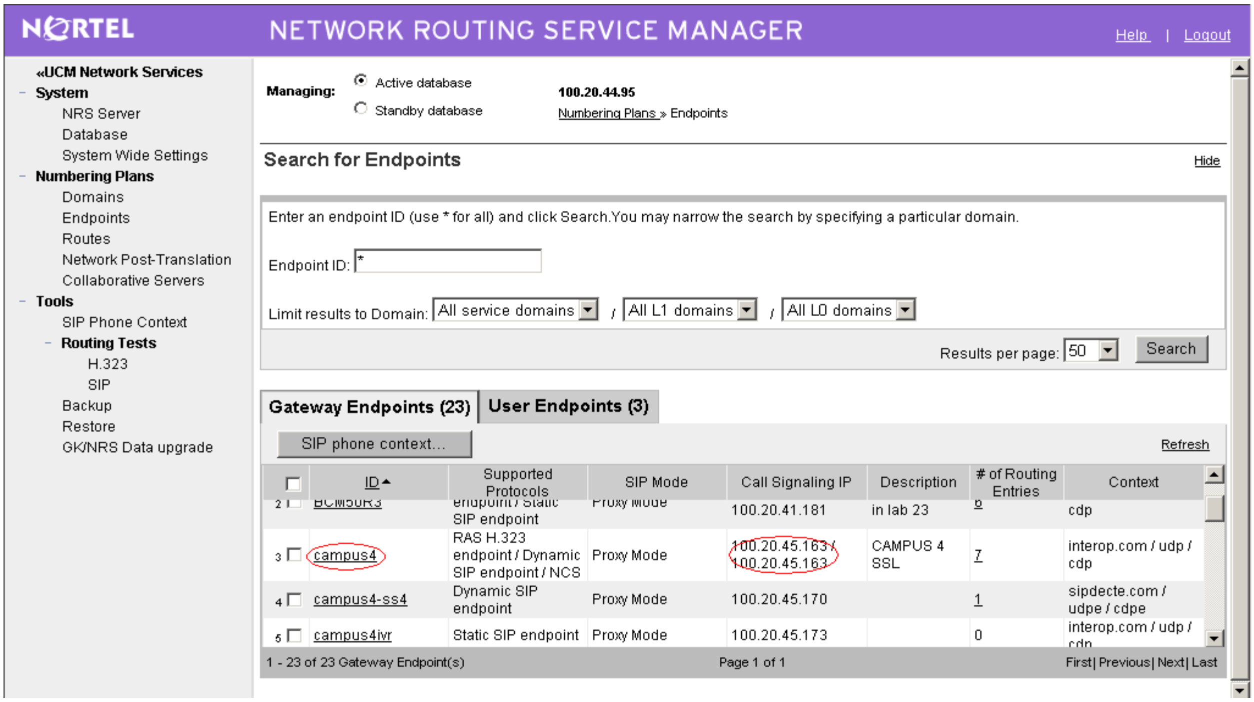


Figure NRS 4 – SSG endpoint registration status

6) SR 2330/4134 endpoint configuration

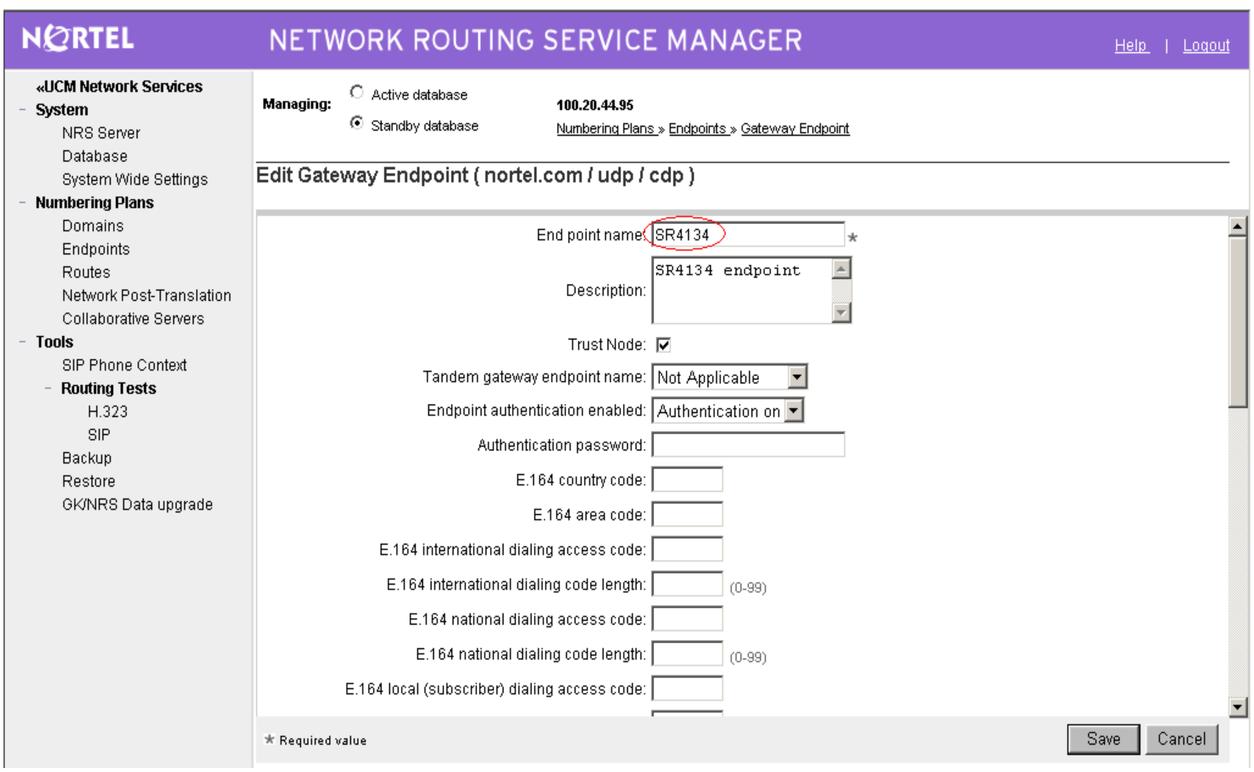


Figure NRS 5a – SR 2330/4134 endpoint configuration

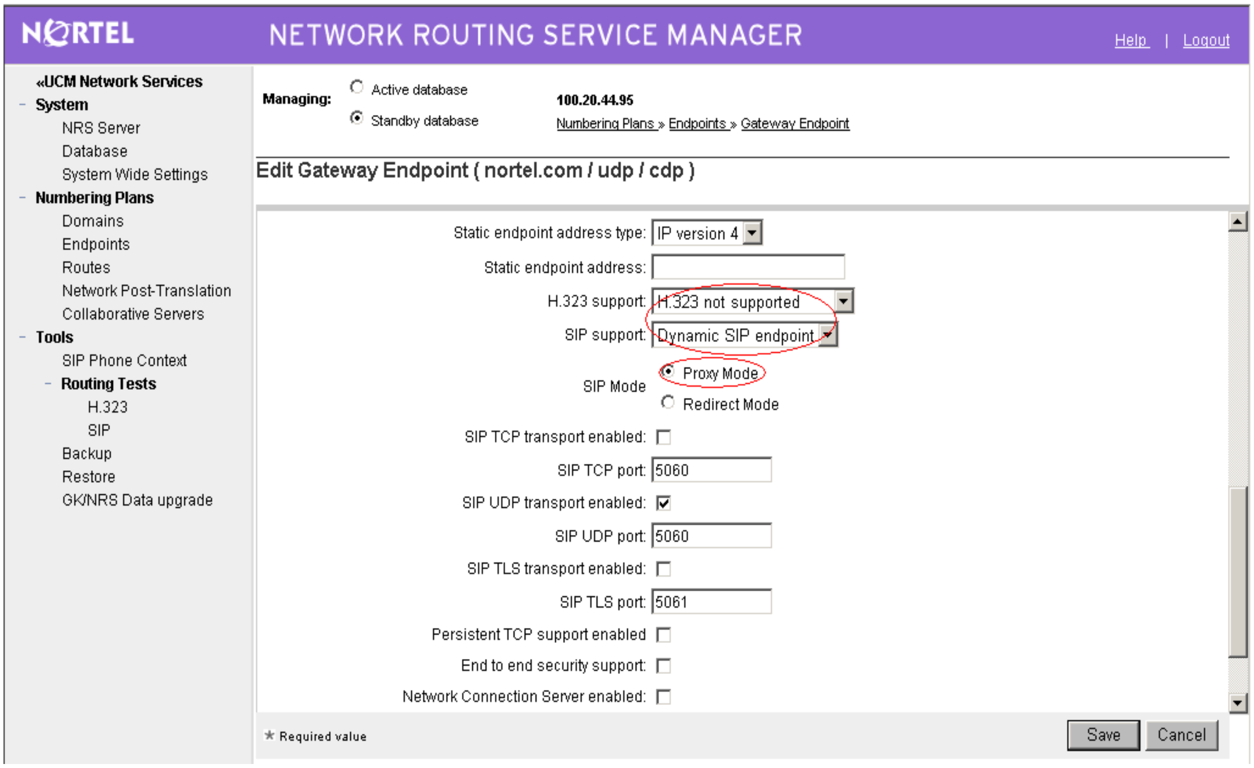


Figure NRS 5b – SR 2330/4134 endpoint configuration

7) Dialing plan routes for SR 2330/4134 endpoint configuration:

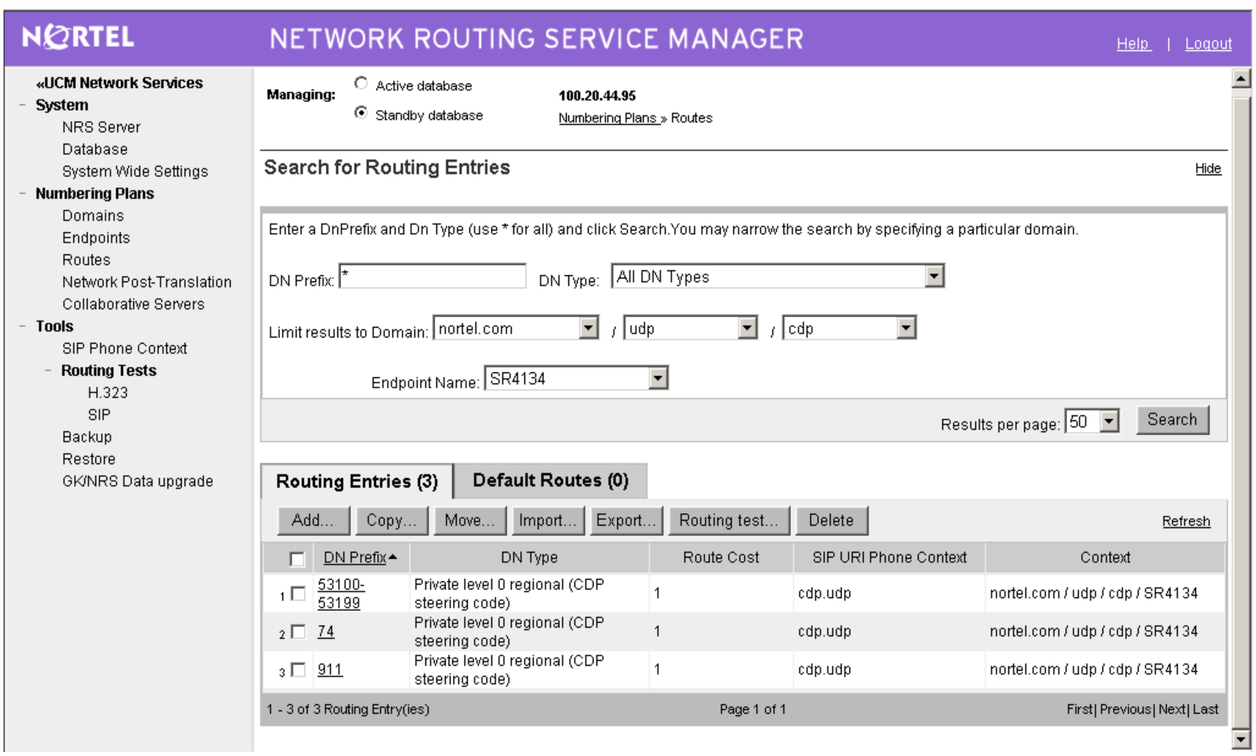


Figure NRS 6 – Dialing plan routes for SR 2330/4134 endpoint configuration

8) SR 2330/4134 endpoint registration status

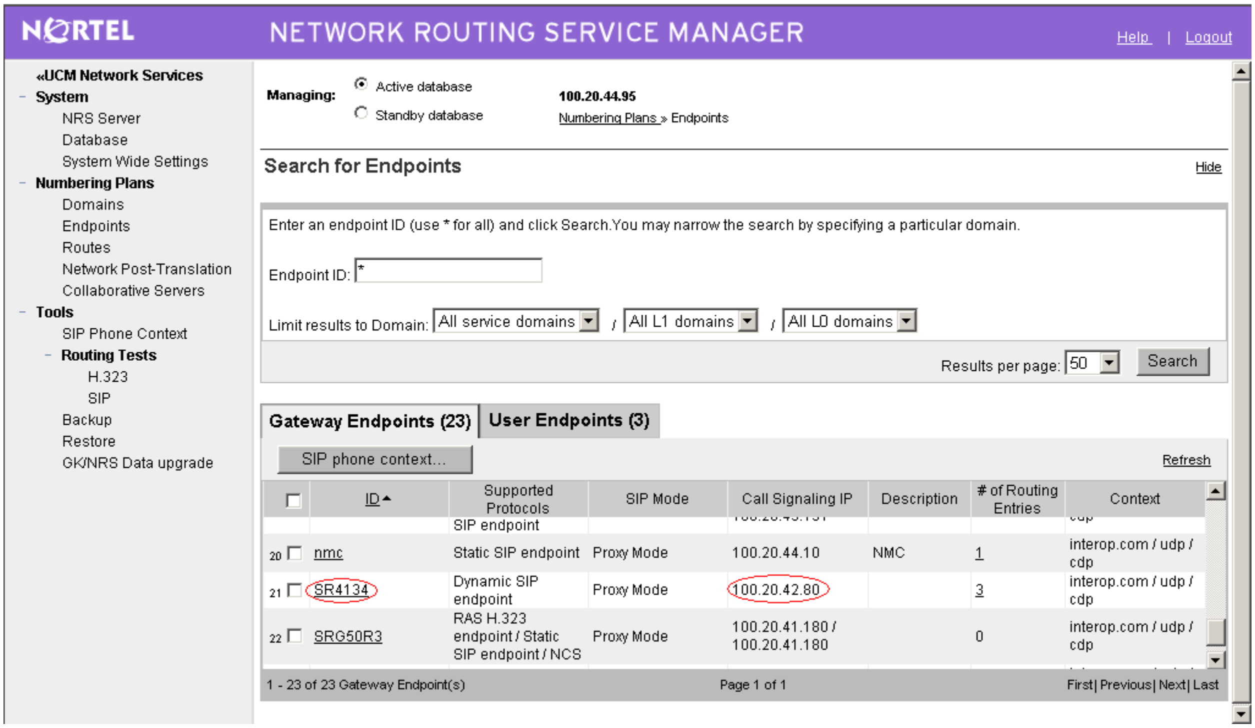


Figure NRS 7 – SR 2330/4134 endpoint registration status

SIP Clients Configuration

Configure the branch SIP endpoints to use SSM bind IP address as the Outbound Proxy ie.100.20.42.80:5060.

Please ensure that the SIP username and domain need to match the CS 1000 SIP Line settings. username@domain represents a globally unique identifier for a SIP user.

CS 1000 Patches

Below patches are needed for version **CS: 600R LB: 6.00.18 LA: 6.00.18** of the CS 1000 system:

- ✓ nortel-CS 1000-sps-6.00.18.17-01.i386.000
- ✓ nortel-CS 1000-vtrk-6.00.18.23-08.i386.000

For higher versions of CS 1000, please refer Meridian PEP Library at

http://qtcfs0n6.ca.nortel.com/mpl/core_menu_view.cfm

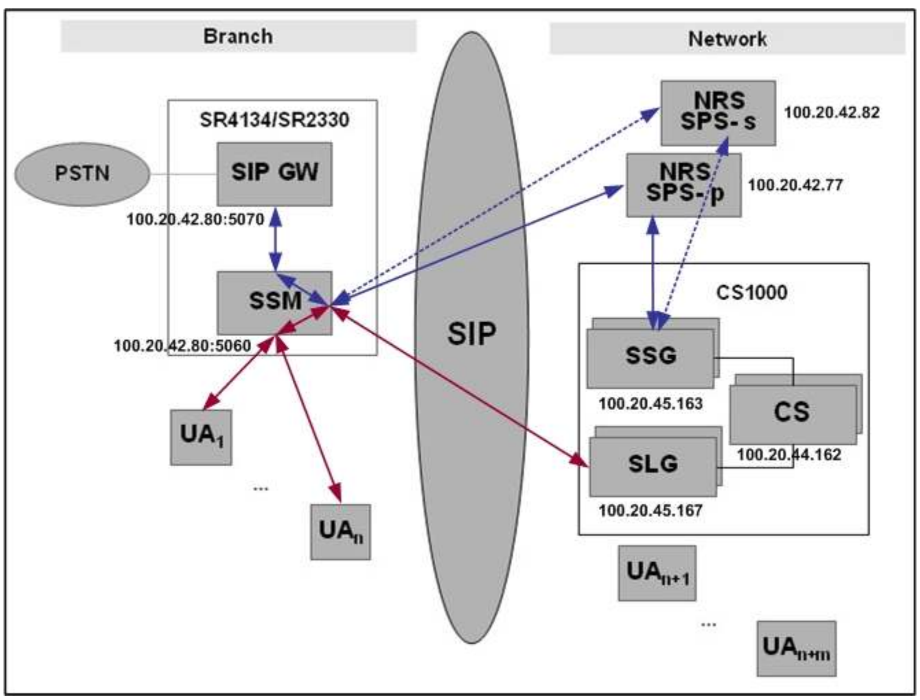
If you don't find the required versions in PEP Library please contact Nortel support.

SR 2330/4134 Configuration

This section describes the configuration steps needed for SR 2330/4134 for CS 1000 branch solution. The following diagram complete with IP addresses will be used as reference for this chapter.

For configuration details on SSM, please refer to Secure Router Release 10.2 guide NN47263-510 Configuration — SIP Survivability.

For configuration details on SIP Gateway (SIP GW) please refer to Secure Router Release 10.2 guide NN47263-508 Configuration — SIP Media Gateway.



Steps

1. Configure the Ethernet interface for connection to the SIP server and SIP phones:

configure terminal
interface ethernet 0/1
ip address 100.20.42.80 255.255.255.224
exit ethernet

2. Configure a default route to the branch router:

   ip route 0.0.0.0/0 100.20.42.65

3. Configure the SIP Media Gateway to listen on port 5070:

   voice service voip
   sip
   bind all ipv4:100.20.42.80:5070
   exit sip
   exit voip

4. To configure the SIP Survivability Module, bind the IP interface for SIP traffic using default port 5060:

   voice service voip
   ssm
   bind ip ipv4:100.20.42.80

5. Enable SSM:

   enable

6. Configure dialpan. Normal mode NTML is used to route gateway calls to SPSs and to replace IP address of SPS to domain name. Survivable dial plan is optional. Configure it only if number translation is required in survivable mode.

   dialplan
   load normal normal_cs1k.ntm
   load survivable backup_cs1k.ntm
   exit dialplan

7. Enable SSM keepalives to configure SLG ip-address and monitor connectivity with SLG:

   sip-server
   keepalive-server ipv4:100.20.45.167:5070 interval 60 retries 2 transport udp

8. Configure SSM domain to specify SLG domain:

   domain dns:interop.com
   exit sip-server

9. Configure SSM Call Admission Control on the WAN interface connecting SLG:

cac
max-calls ethernet0/1 256

10. Configure the CAC exclusion pool that identifies the IP address range of the SIP endpoints that use SSM:

    exclude-pool 100.20.47.0 255.255.255.0
    exit cac

11. Point the SSM to the SIP Media Gateway IP interface as the default gateway (specifying the non-default port), Port should be same as configured for gateway's listening port in step 3.:

    default-gateway ipv4:100.20.42.80:5070 transport udp
    exit ssm
    exit voip

12. Configure the outbound proxy on the SIP Media Gateway to point to the SSM:

    sip-ua
    outbound-proxy ipv4:100.20.42.80:5060

13. Configure the primary and secondary SPS for the SIP Media Gateway:
    (No need to configure secondary sip-server if there is only one SPS used):

    sip-server ipv4:100.20.42.77:5060
    sip-server ipv4:100.20.42.82:5060 secondary

14. Configure authentication parameters to be used for gateway to SPS calls and registration:

    authentication SR 4134 1234

15. Configure keepalive to monitor primary and secondary SPS connectivity:

    keepalive target sip-server
    keepalive target sip-server secondary

16. Configure dynamic registration from SR gateway to active SPS.

    register dynamic
    exit sip-ua

17. Configure voice ports for FXS phones
    (example for 2 FXS phones connected to port 2/1 and 2/2)

    voice-port 2/1
    signal loop-start
    station number 74001

no shutdown
exit voice-port

voice-port 2/2
signal loop-start
station number 74002
no shutdown
exit voice-port

18. Configure PRI interface bundle.
(example to configure PRI E1 bundle on port 3/1 with switch-type as qsig)

interface bundle E1PSTN
link pri_e1 3/1 voice
isdn
switch-type primary-qsig
activate
exit isdn
exit bundle

19. Optional translation profile configuration.
a. Configure translation profile for PSTN to SPS calls. (Example to translate 335.. number to 5.. numbers.) There can be more that 1 rules which can be called or calling or both numbers.

voice translation-rule 100
rule 1 /335/ /5/
exit translation-rule

voice translation-profile pstn2sps
translate calling 100
translate called 100
exit translation-profile

b. Configure translation profile for SPS to PSTN calls. (Example to translate 5.. numbers to 335..)

voice translation-rule 200
rule 1 /5/ /335/
exit translation-rule

voice translation-profile sps2pstn
translate calling 200
translate called 200
exit translation-profile

20. Configure Dial peer for FXS phones:

    dial-peer voice pots 1
    destination-pattern 74001
    port 2/1
    forward-digits all
    no shutdown
    exit pots

    dial-peer voice pots 2
    destination-pattern 74002
    port 2/2
    forward-digits all
    no shutdown
    exit pots

21. Configure Dial peer for PRI interface and (optionally) apply translation profile. (Following dial peer is for calls coming from SPS with a number starting with 53 to be translated to a number starting with 3353 before sending on PRI. Don't use translation profile if no translation required.)

    dial-peer voice pots 3
    destination-pattern 53.%
    port 3/1
    forward-digits all
    no shutdown
    translation-profile outgoing sps2pstn
    exit pots

    (Following dial peer is for calls coming from PSTN with a number starting with 3353 to be translated to a number starting with 53 before sending to SPS. Don't use translation profile if no translation required.)

    dial-peer voice pots 4
    destination-pattern 3353.%
    port 3/1
    forward-digits all
    no shutdown
    translation-profile incoming pstn2sr
    exit pots

➔ If no translation required than instead of dial-peer 3 and 4, only 1 dial peer is required with destination-pattern 53.% (assuming that all PRI numbers are starting with 53).

➔ Following is an example if a 911 call is to be routed to PRI interface.

dial-peer voice pots 5
destination-pattern 911
port 3/1
forward-digits all
no shutdown

NTML Examples

Example of Normal mode NTML (normal_cs1k.ntm)

This NTML is used by SSM to route gateway calls to active SPS (100.20.42.77 or 100.20.42.82) with domain nortel.com.

```
<translation>
 <address-switch field = "original-destination" subfield = "host">
  <address is = "100.20.42.77">
    <replace string=" nortel.com" field="destination" subfield="host">
     <replace string=" nortel.com" field="origin" subfield="host">
       <replace string=" nortel.com" field="original-destination" subfield="host">
        <route host="100.20.42.77" add-route="yes" replace-host ="no"/>
       </replace>
     </replace>
    </replace>
  </address>
  <address is = "100.20.42.82">
   <replace string=" nortel.com" field="destination" subfield="host">
    <replace string=" nortel.com" field="origin" subfield="host">
      <replace string=" nortel.com" field="original-destination" subfield="host">
        <route host="100.20.42.82" add-route="yes" replace-host ="no"/>
      </replace>
    </replace>
   </replace>
  </address>
 </address-switch>
</translation>
```

Example of Backup mode NTML (backup_cs1k.ntm)

This NTML is optional. Here is an example assuming that calls with certain prefix in backup mode needs to be converted to a 5 digit number extension.

```
<translation>
  <number-switch>
        <number is = "967?????">
       <drop literals = "3"/>
     </number>
     <number is = "613967?????">
       <drop literals = "6"/>
     </number>
       <number is = "1613967?????">
     <drop literals = "7"/>
   </number>
  </number-switch>
</translation>
```