Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Communication Server 1000E R7.0 with Trio Enterprise R3.0 (SIP Trunks) – Issue 1.0

Abstract

These Application Notes describe how to configure an Avaya Communication Server 1000E R7.0 to interface with Trio Enterprise R3.0, which is operating as an attendant answering position. Trio Enterprise is a software application installed on a Windows server that interfaces with Avaya Communication Server 1000E using SIP trunks and provides users with the call functions of an attendant console without having to install a hardware attendant position.

Information in these Application Notes has been obtained through DevConnect Compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the test configuration for Avaya Communication Server 1000E R7.0 with Trio Enterprise R3.0. Trio Enterprise is a client/server application running on Microsoft Windows operating systems. Trio Enterprise provides users with an attendant answering position for Avaya Communication Server 1000E that does not need attendant telephony hardware (e.g., 2250 attendant console). Trio Enterprise connects to the Avaya Communication Server 1000E using SIP trunks and calls are made over SIP trunks to PSTN destinations as well as internal Avaya Communication Server 1000E users. Trio Enterprise can perform the usual range of attendant call functions, i.e., centralized answering position, extending PSTN calls to users, placing PSTN calls on behalf of internal users, and performing internal telephone directory lookups. In addition, Trio Enterprise uses Intercept Computer Protocol (ICP) as a tool to manage user absences, by allowing Avaya Communication Server 1000E users to forward their telephones to the Trio Enterprise attendant position with an absence reason code which is displayed on the attendant screen.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise voice network using a Communication Server 1000E Network Routing Server (NRS) and a Communication Server 1000E system, connected via SIP trunks. The Trio Enterprise server connects to the NRS via SIP trunks. All SIP traffic was routed by the NRS. See **Figure 1** for a network diagram.

A basic Distance Steering Code configuration (DSC) was configured on the Communication Server 1000E to route all incoming PSTN calls to the Trio attendant position. An Avaya 1140E IP telephone was used as the Trio attendant telephony device.

During tests, calls are placed to a four digit number which is associated with the Trio attendant position. The Communication Server 1000E routes all calls destined for the Trio attendant to the NRS, which routes the calls to the Trio server over SIP trunks. The call is terminated by extending the call to the 1140E telephone associated with the Trio attendant and when answered, the Trio server bridges the two calls. It is possible to have multiple Trio attendant positions on a Communication Server 1000E system

A variety of Avaya telephones were installed and configured on the Communication Server 1000E. The NRS was configured to route calls between the Communication Server 1000E and the Trio server. The Trio attendant client was installed on the same server as the Trio Server, but can be installed on a separate platform if required.

2.1. Interoperability Compliance Testing

The compatibility tests included the following:

- NRS configuration
- SIP trunk configuration
- Attendant answers direct call
- Supervised and unsupervised transfer with answer
- Directing calls to busy extensions
- Call queuing and retrieval
- Loop detection for busy and unanswered extensions

2.2. Test Results

The following observation was made during Trio 3.0 testing.

- Trio Enterprise attendant calls that were placed to a Communication Server 1000E basic ACD telephone always had just one-way speech if the (answering) agent telephone was an IP telephone (Avaya 1140E or 2007 type telephone). If the answering ACD agent was using a digital telephone (e.g., 3905), the speech path was proper two-way audio. This was subsequently found to be caused by a setting of 10ms frame size for the G.711 codec. When this was changed to 20ms/frame, the problem could not be reproduced. It is recommended to use a G.711 frame size of 20ms for all Trio Enterprise installations.

2.3. Support

For technical support on Trio products, please use the following web link:
http://www.trio.com/web/Support.aspx

For support on Avaya products, please use the following web link:
http://support.avaya.com

3. Reference Configuration

The following **Figure 1** shows the network configuration used for all test cases in the test plan.

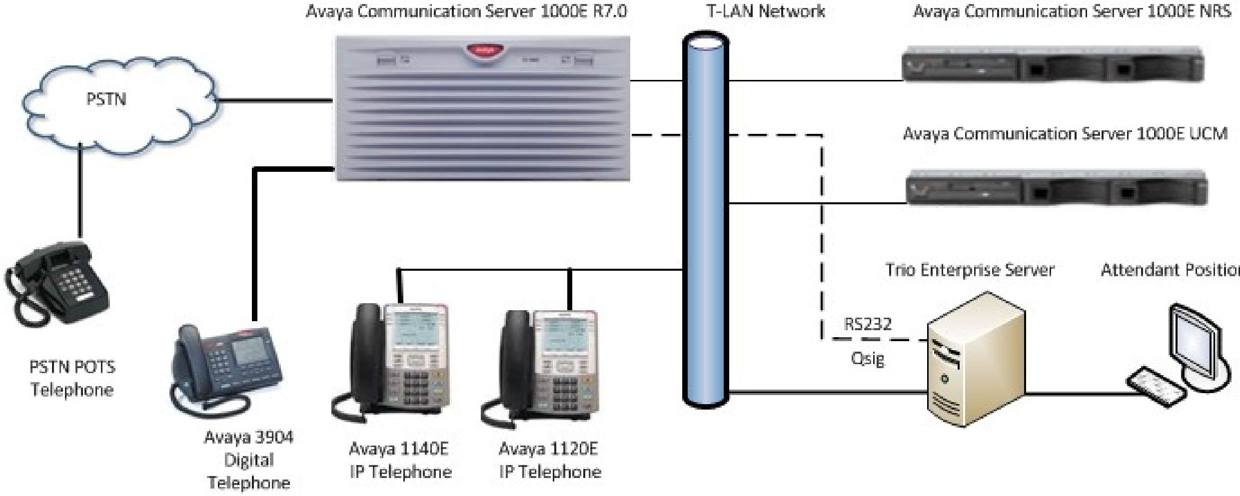


Figure 1: Test Configuration for Avaya Communication Server 1000E and Trio Enterprise V3.0

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Avaya Equipment	Software Name / Firmware Version
Avaya Communication Server 1000E	Avaya Communication Server 1000E 07.00Q / 7.00.20 (PSWV 100 with latest Patches and Deplist)
Avaya Communication Server 1000E Media Gateway	CSP VERSION: MGCC BD01 MSP VERSION: MGCM AB01 APP VERSION: MGCA BA07 FPGA VERSION: MGCF AA18 BOOT VERSION: MGCB BA07 DSP1 VERSION: DSP1 AB03 DSP2 VERSION: DSP2 AB03 DSP3 VERSION: DSP3 AB03 DSP4 VERSION: DSP4 AA01 DSP5 VERSION: DSP5 AA01
Avaya 1100 series IP Telephones • 1140e • 1120e Avaya M3900 series Telephones • M3904	 0625C8A (UniStim 5.0) 0624C8A (UniStim 5.0) Version: AA93
Trio Equipment	**Software Name / Firmware Version**
Trio Enterprise Server platform	Trio Enterprise® 3.0.37.105

5. Configure Avaya Communication Server 1000E NRS

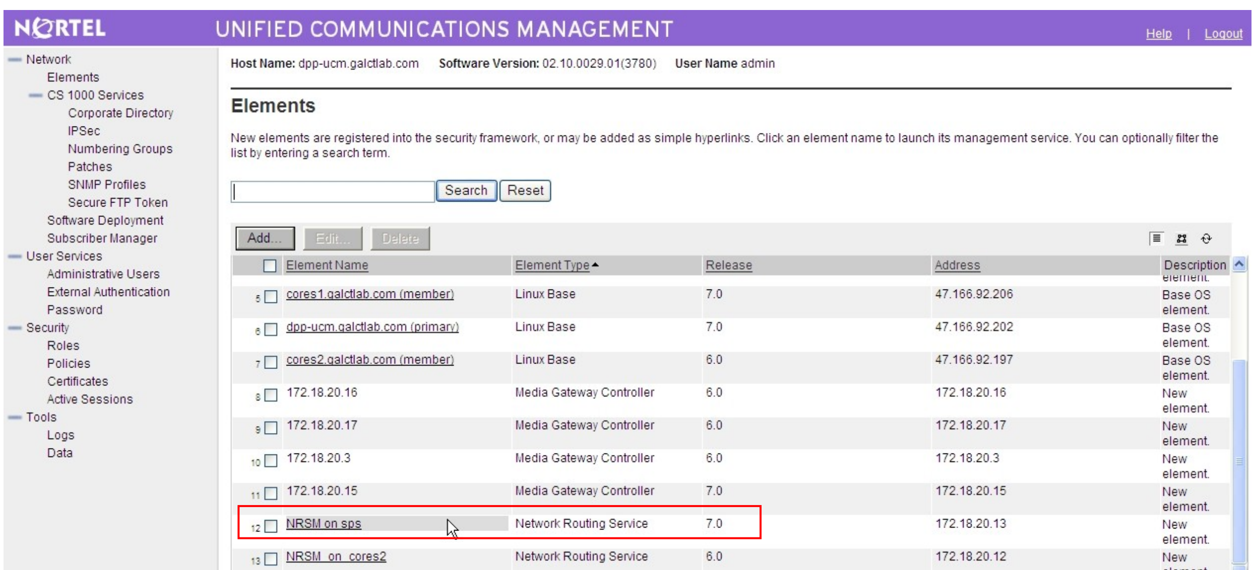
This section describes the steps required to configure Communication Server 1000E Network Routing Service (NRS) prior to testing with Trio Enterprise 3.0. The general installation and configuration of Communication Server 1000E NRS and UCM is presumed to have been previously completed and is not discussed here. The function of the NRS is to route SIP traffic between the Communication Server 1000E system and the Trio Enterprise server. NRS configuration requires the following actions:

- Log on to Avaya Unified Communications Management and Network Routing Services Manager and configure System Wide Settings
- Administer SIP,L1 and L0 domains
- Administer SIP Endpoints
- Administer Routing Entries

For detailed information on installing and configuring Communication Server 1000E NRS, see item **[3]** in **Section 10** of this document.

5.1. Unified Communications Management (UCM), Network Routing Services Manager (NRSM) and System Wide Settings

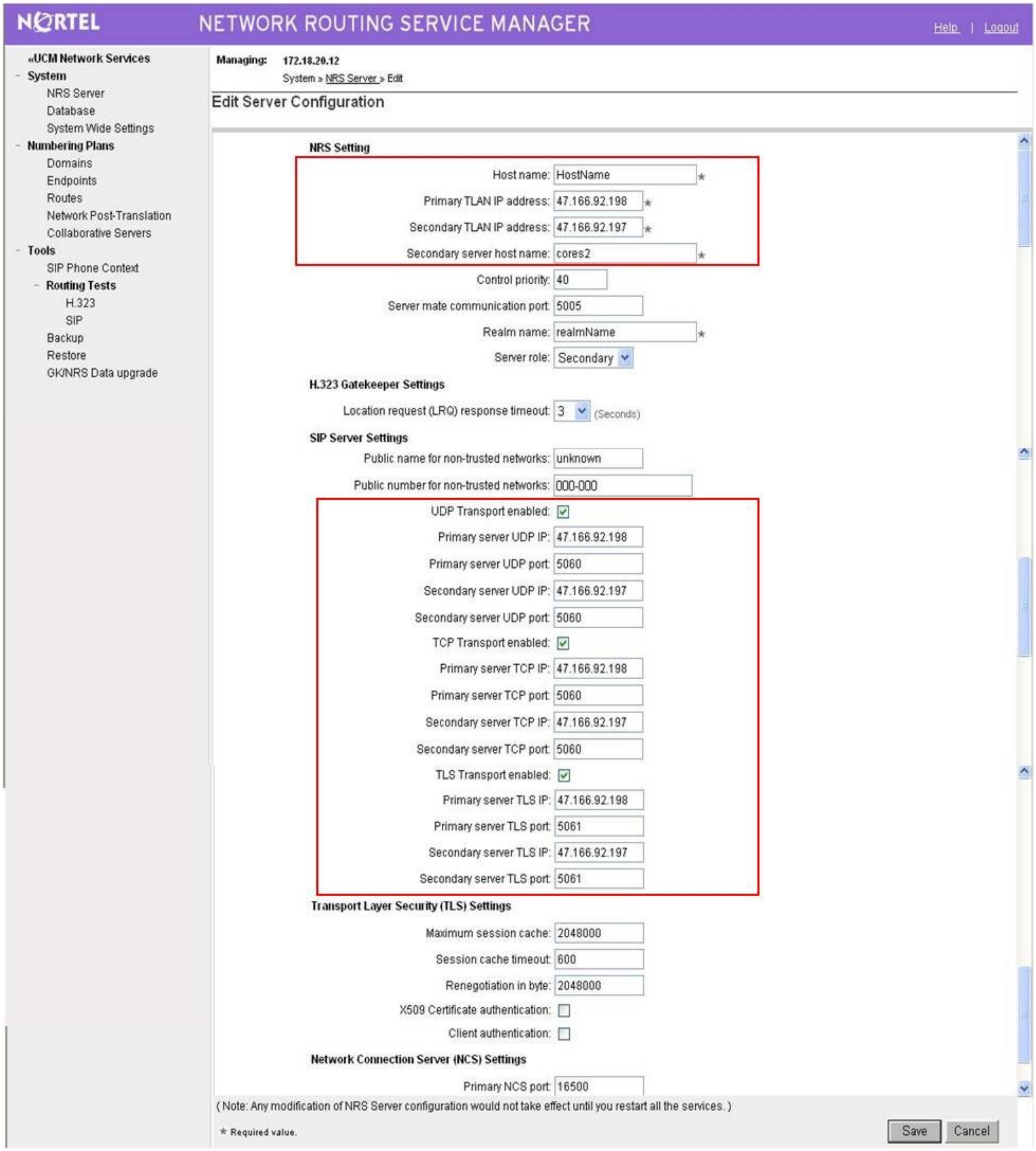
Access UCM using a Web Browser by entering **http://<FQDN >/**, where **<FQDN>** is the fully qualified domain name of the UCM server. Log in using appropriate credentials (not shown) and the Home page will be presented with menu options shown below. Scroll down and select **NRSM on sps** from the **Elements** list.



The **Network Routing Service Manager** (NRSM) page will open in a new window. Click on the **Edit** button (not shown), the **Edit Server Configuration** window opens (see following screenshot). The following settings are required to be configured:

- **Hostname**, this is the primary NRS network name
- **Primary TLAN IP address**, the primary NRS IP address
- **Secondary TLAN IP address**, IP address of a secondary NRS (if required)
- **Secondary server host name**, secondary NRS network name
- **UDP Transport enabled**, checkbox must be ticked
- **Primary server UDP IP**, same as Primary TLAN IP address
- **Primary server UDP port**, must be set to **5060**
- **Secondary server UDP IP**, same as Secondary TLAN IP address (if required)
- **Secondary server UDP port**, must be set to **5060** (if required)
- **TCP Transport enabled**, checkbox must be ticked
- **Primary server TCP IP**, same as Primary TLAN IP address
- **Primary server TCP port**, must be set to **5060**
- **Secondary server TCP IP**, same as Secondary TLAN IP address (if required)
- **Secondary server TCP port**, must be set to **5060** (if required)
- **TLS Transport enabled**, checkbox must be ticked
- **Primary server TLS IP**, same as Primary TLAN IP address
- **Primary server TLS port**, must be set to **5061**
- **Secondary server TLS IP**, same as Secondary TLAN IP address (if required)
- **Secondary server TLS port**, must be set to **5061** (if required)

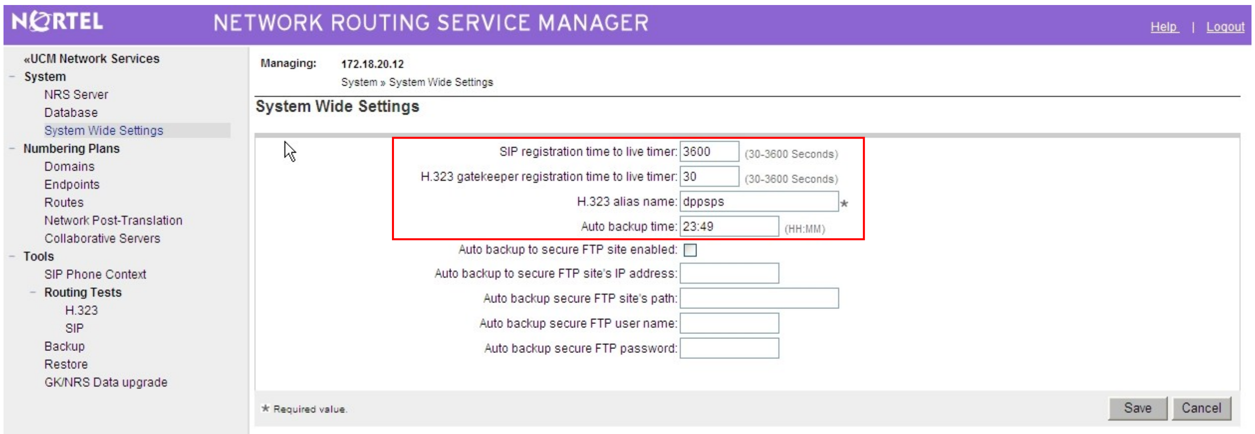
When finished, click on the **Save** button.


NORTEL
NETWORK ROUTING SERVICE MANAGER
Help | Logout
«UCM Network Services
System
NRS Server
Database
System Wide Settings
Numbering Plans
Domains
Endpoints
Routes
Network Post-Translation
Collaborative Servers
Tools
SIP Phone Context
Routing Tests
H.323
SIP
Backup
Restore
GK/NRS Data upgrade
Managing: 172.18.20.12
System » NRS Server » Edit
Edit Server Configuration
NRS Setting
Host name: HostName
Primary TLAN IP address: 47.166.92.198
Secondary TLAN IP address: 47.166.92.197
Secondary server host name: cores2
Control priority: 40
Server mate communication port: 5005
Realm name: realmName
Server role: Secondary
H.323 Gatekeeper Settings
Location request (LRQ) response timeout: 3 (Seconds)
SIP Server Settings
Public name for non-trusted networks: unknown
Public number for non-trusted networks: 000-000
UDP Transport enabled:
Primary server UDP IP: 47.166.92.198
Primary server UDP port: 5060
Secondary server UDP IP: 47.166.92.197
Secondary server UDP port: 5060
TCP Transport enabled:
Primary server TCP IP: 47.166.92.198
Primary server TCP port: 5060
Secondary server TCP IP: 47.166.92.197
Secondary server TCP port: 5060
TLS Transport enabled:
Primary server TLS IP: 47.166.92.198
Primary server TLS port: 5061
Secondary server TLS IP: 47.166.92.197
Secondary server TLS port: 5061
Transport Layer Security (TLS) Settings
Maximum session cache: 2048000
Session cache timeout: 600
Renegotiation in byte: 2048000
X509 Certificate authentication:
Client authentication:
Network Connection Server (NCS) Settings
Primary NCS port: 16500
(Note: Any modification of NRS Server configuration would not take effect until you restart all the services.)
* Required value.
Save
Cancel

Select **System Wide Settings** from the side menu, the **System Wide Settings** page will appear (see the following screenshot). Configure the following values:

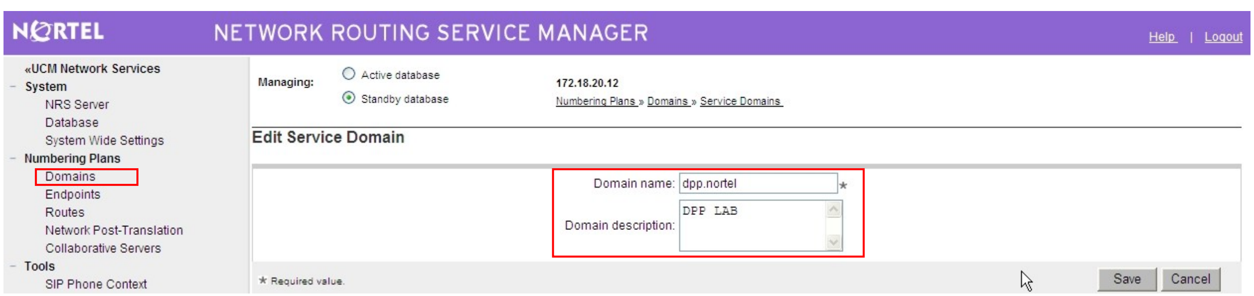
- **Sip registration time to live timer**, set to **3600** seconds
- **H.323 gatekeeper time to live timer**, set to **30** seconds
- **H.323 alias name,** set to **dppsps** in this example
- **Auto backup time**, the NRS automatic backup script runs at this time

Click on the Save button when finished.

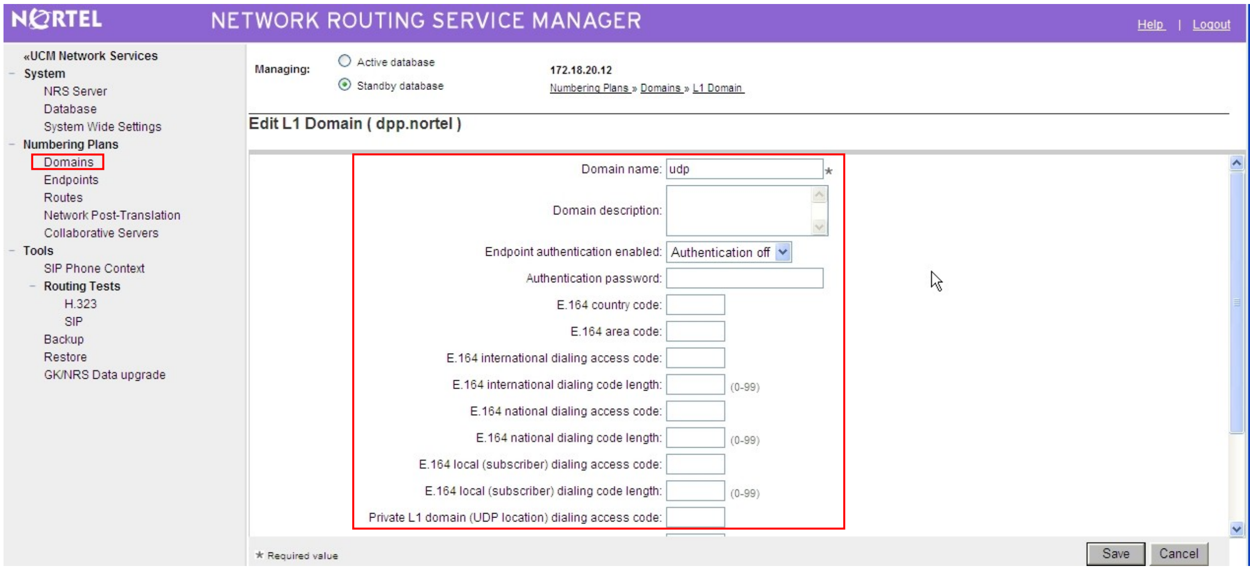


5.2. Administer SIP Service Domain, L1 and L0 domains

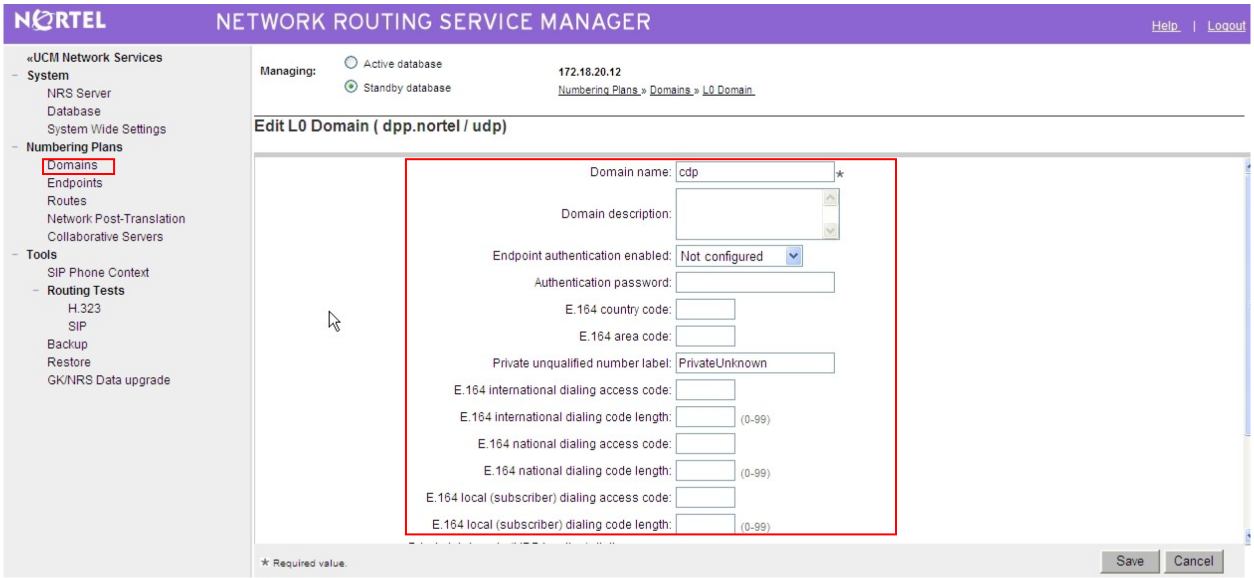
Ensure the **Managing Standby database** radio button is checked. Click on **Domains** from the left hand side menu, the **Edit Service Domain** page appears. Enter the required **Domain name** and an (optional) **Domain description**. Click on the **Save** button when finished.



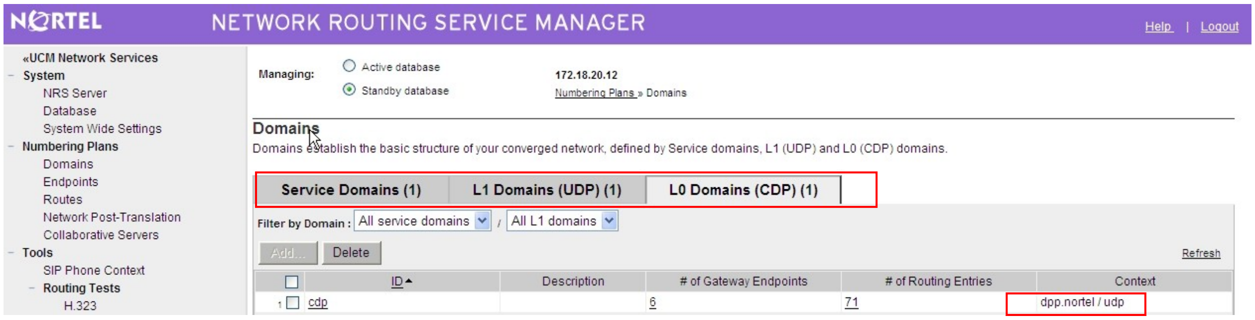
The Service Domain page now reappears with three tabs. Ensuring the **Managing Standby database** radio button is checked, click on the **L1** tab (not shown) and select the service domain previously configured from the **Filter by Domain** drop-down box (not shown). Enter the **Domain name** (**UDP** in the example) in the **Edit L1 Domain** page (see the following screenshot). Ensure **Endpoint authentication enabled** is set to **Authentication off**. The remaining parameters can be set to values appropriate for the installation. Click on the **Save** button when completed.



Ensure the **Managing Standby database** radio button is checked. Click on the **L0** tab (not shown), the **Edit L0 Domain** page appears. Select the configured service domain from the **Filter by Domain** drop-down box (not shown) and then the previously configured L1 domain from the **Filter by L1 Domain** drop-down box (not shown). Enter the **Domain name** (**CDP** in the example shown**)**, ensure **Endpoint authentication enabled** is set to **Not configured** (see following screenshot). The remaining parameters can be set to values appropriate for the installation. Click on the **Save** button when completed.



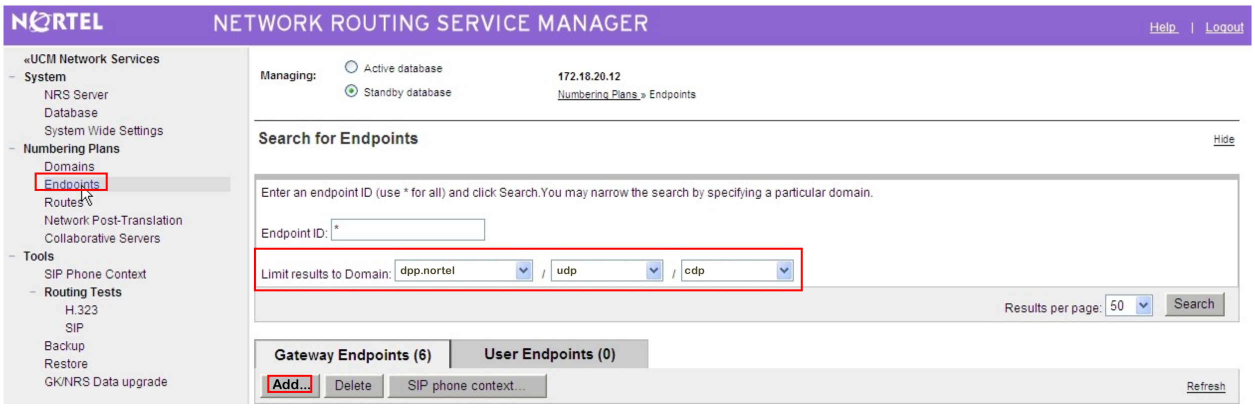
This completes the NRS Domain configuration. The next screenshot is an example of the fully configured **SIP Service Domain**, **L1Domains (UDP)** and **L0 Domains (CDP)**.



5.3. Administer SIP Endpoints

SIP endpoints must register with the NRS before sending or receiving SIP traffic. Endpoints are typically Communication Server 1000E systems, but may also be SIP telephones or third party SIP proxies.

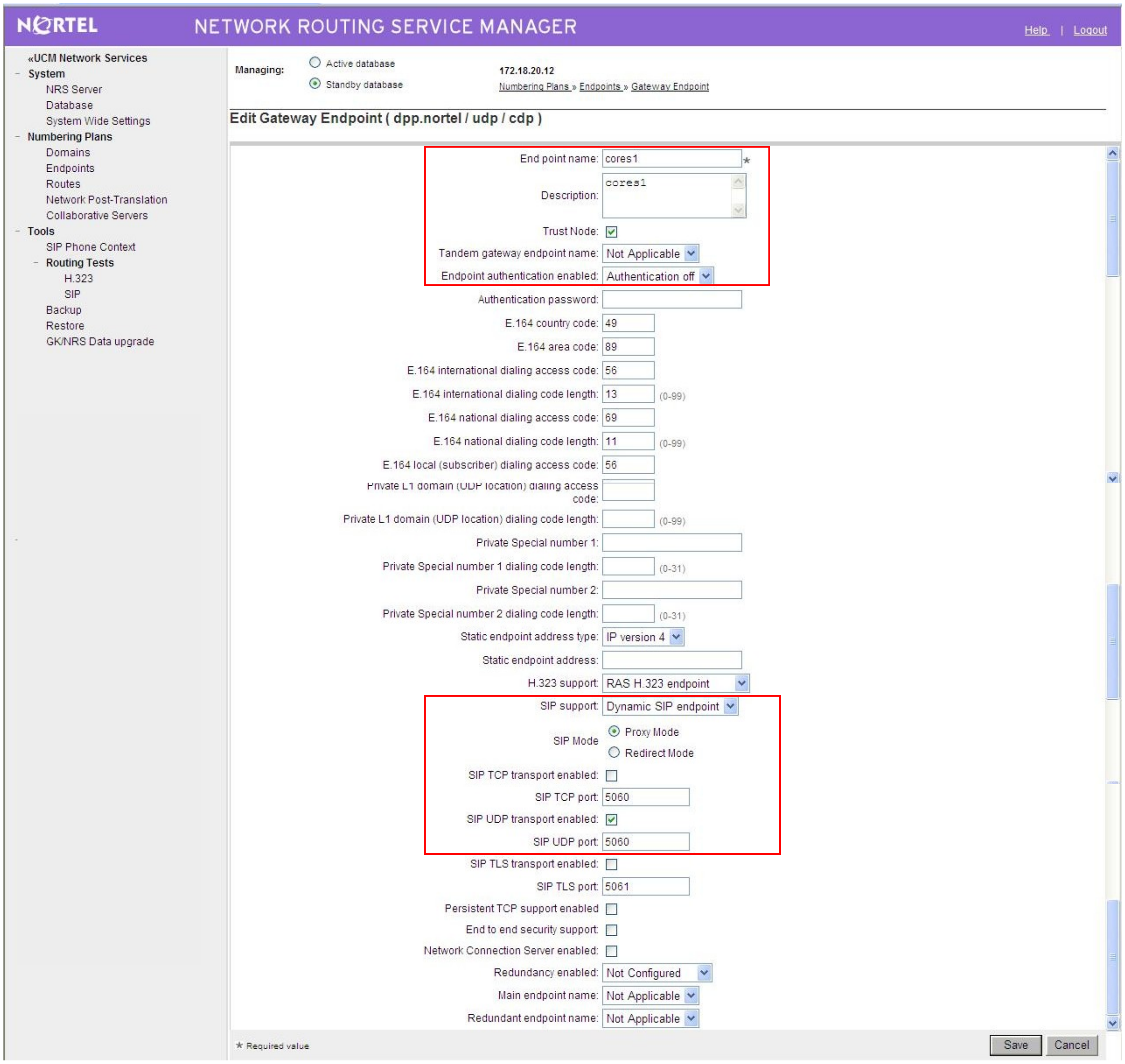
Ensure the **Managing Standby database** radio button is checked. Click on the **Endpoint** entry in the left side menu and in the resulting page select the previously configured SIP, L1 and L0 domains from the **Limit results to Domain** suite of drop-down lists (not shown). The following screenshot shows the previously entered SIP, L1 and L0 domains highlighted.



When drop-down lists have been populated with the correct values, the **Add** button is then activated and a new endpoint can be added. Click on the **Add** button and enter the endpoint data. See the highlighted area in the following screenshot for information.

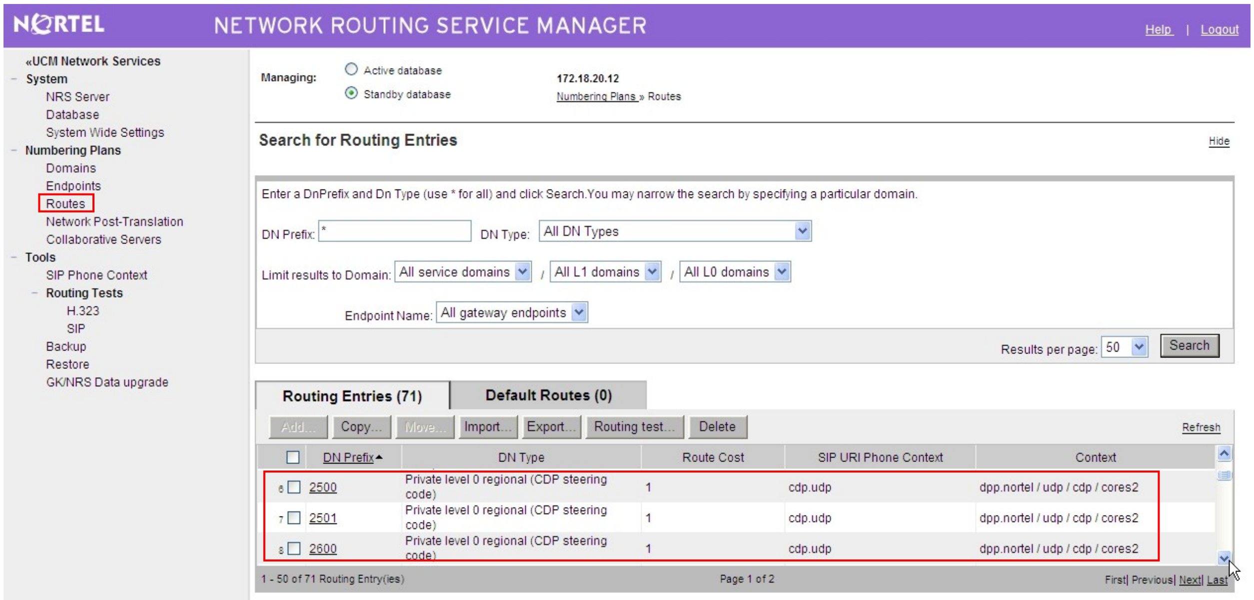
- **End point name** (**cores1** in this example) must match that configured later in **Section 6.4**
- **Description** is typically some text to describe the endpoint
- **Trusted Node** must be checked
- **Tandem gateway endpoint name** is set to **Not Applicable**
- **Endpoint Authentication enabled** is set to **Authentication off**
- **SIP Support** must be set to **Dynamic SIP endpoint**
- **SIP Mode** must be set to **Proxy**
- **SIP UDP transport enabled** checkbox must be ticked
- **SIP UDP port** must be **5060**

The remaining values will be specific to the particular location and endpoint being configured, examples of typical values are shown, and the correct values must be entered before the endpoint is brought into service. Click the **Save** button to confirm the settings.



5.4. Administer Routing Entries

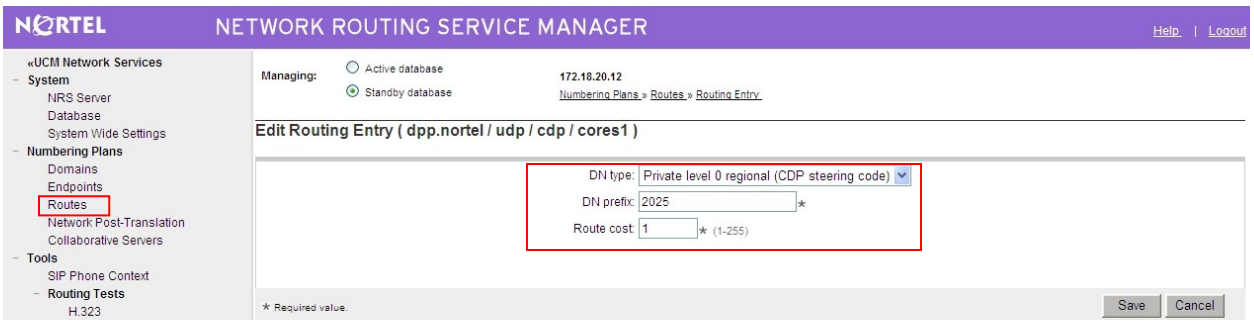
Routing entries are telephone numbers associated with an endpoint. When a telephone number is dialed, the NRS searches the endpoint database to find a match and then directs the call to the endpoint with the first returned match. Endpoints can be entered as a range of telephone numbers (e.g., 756*, which matches all numbers beginning with 756) or as a list of unique numbers. Unique listings reduce unnecessary SIP messaging, but require significantly more effort to setup and maintain. The following screenshot shows the routing entries initially setup on the Communication Server 1000E.



To add a new routing entry, ensure the **Managing Standby database** radio button is checked. Click on the **Routes** entry in the left side menu and in the resulting page select the previously configured **SIP, L1 and L0** domains from the **Limit results to Domain** suite of drop-down lists. When drop-down lists have been populated with the correct values, the **Add** button is activated and a new route can be added. Click on the **Add** button and enter the route data.

- Select **Private level 0 regional CDP steering code** from the **DN type** drop-down list
- **DN prefix** is a four digit telephone number
- **Route cost** is set to **1**

Click on the **Save** button when finished. The following screenshot shows an example routing entry.



To add more endpoints, repeat **Sections 5.3** and **5.4.**

This completes the Communication Server 1000E NRS setup.

6. Configure the Communication Server 1000E

This section describes the steps required to configure Communication Server 1000E SIP trunks and the necessary configuration for terminals (digital, analog, attendant and IP phones). SIP trunks are established between Communication Server 1000E and the NRS and are used for all off-switch calls. To reach telephone numbers on other Communication Server 1000E systems, calls are placed via the NRS, which proxies SIP messages. The general installation of the Communication Server 1000E, NRS and UCM is presumed to have been previously completed and is not discussed further here.

6.1. Confirm System Features

The keycode installed on the Call Server controls the maximum values for system attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional feature/capacity. Use the Communication Server 1000E system terminal and manually load **overlay 22** to print the **System Limits** (the required command is SLT), and verify there are sufficient Traditional Telephones, IP Users, Basic IP Users and SIP Access Ports to meet requirements.

Overlay 22 – system Limits (SLT) Printout

```
System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz
IPMGs Registered:              2IPMGs Unregistered:             0IPMGs
Configured/unregistered:  0

TRADITIONAL TELEPHONES 32767    LEFT 32764    USED     9
DECT USERS             32767    LEFT 32767    USED     0
IP USERS               32767    LEFT 32744    USED    12
BASIC IP USERS         32767    LEFT 32761    USED     7
TEMPORARY IP USERS     32767    LEFT 32767    USED     0
DECT VISITOR USER      10000    LEFT 10000    USED     0
ACD AGENTS             32767    LEFT 32752    USED     2
MOBILE EXTENSIONS      32767    LEFT 32767    USED     0
TELEPHONY SERVICES     32767    LEFT 32767    USED     0
CONVERGED MOBILE USERS 32767    LEFT 32767    USED     0
NORTEL SIP LINES       32767    LEFT 32765    USED     2
THIRD PARTY SIP LINES  32767    LEFT 32761    USED     6
SIP CONVERGED DESKTOPS 32767    LEFT 32767    USED     0
SIP CTI TR87           32767    LEFT 32767    USED     0
SIP ACCESS PORTS       32767    LEFT 32752    USED    30
```

Load **overlay 21**, and confirm the customer is setup to use ISDN trunks (see below).

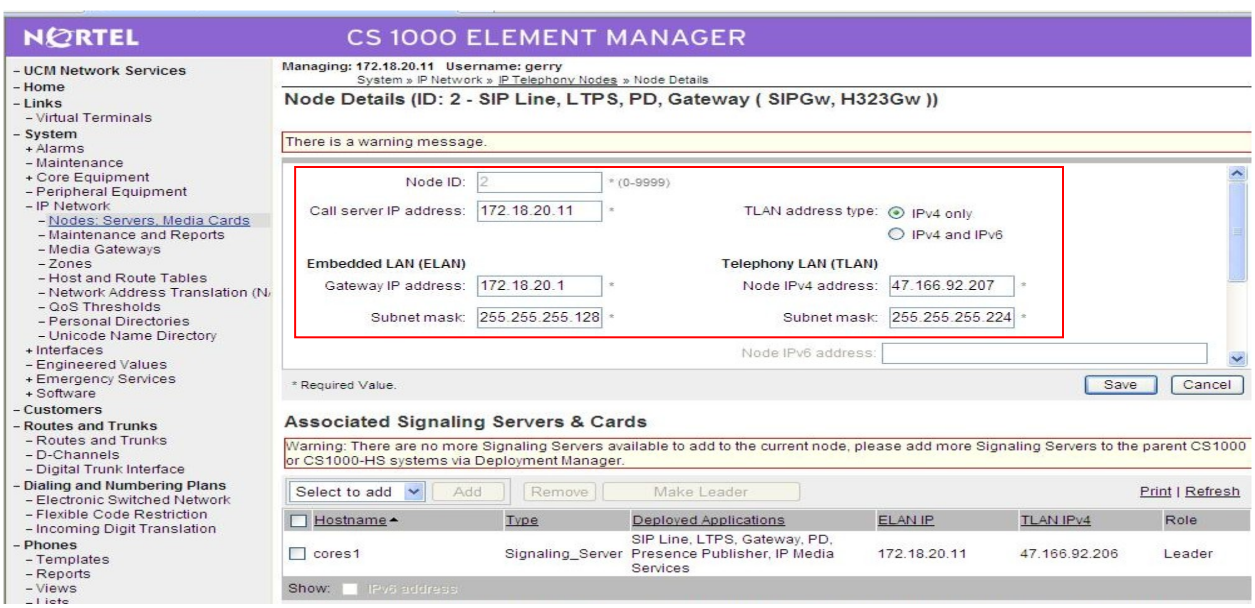
```
                              Overlay 21 Customer Network Data
REQ: prt
TYPE: net
TYPE NET_DATA
CUST 0

TYPE NET_DATA
CUST 00
OPT RTD
AC1 INTL NPA SPN NXX LOC
AC2
FNP YES
ISDN YES
```

6.2. Configure System Node Information

Use Communication Server 1000E Element Manager to configure the system node properties. Navigate to the **System → IP Networks → IP Telephony Nodes → Node Details** and click on the **Add** button (not shown), the node details page appears (see following screenshot). Fill in the following settings:

- **Node ID**, a unique numerical value to identify the node
- **Call Server IP Address**, IP address of the Communication Server 1000E call processor
- **Embedded LAN (ELAN) Gateway IP Address**, the node ELAN gateway IP address
- **Embedded LAN (ELAN) Subnet Mask**, the ELAN network Subnet Mask
- **TLAN address type,** click on the IPv4 only radio button
- **Telephony LAN (TLAN) Node IP Address**, a unique IP address for the node
- **Telephony LAN (TLAN) Subnet Mask**, the TLAN network Subnet Mask

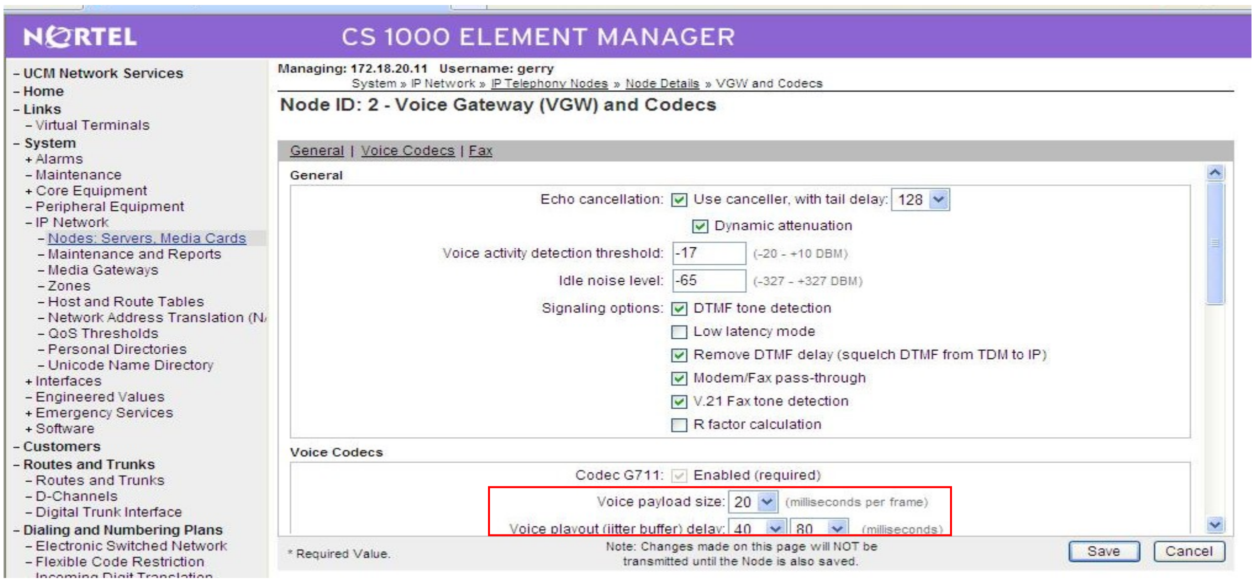
Click on the **Save** button when finished.



6.3. Configure System Codecs

Communication Server 1000E uses codecs to convert digital and analog telephone speech into a format suitable for SIP trunks. Before SIP trunks can be utilized, system codecs must be selected. Using the Communication Server 1000E element manager sidebar, click on **Nodes: Servers, Media Cards** and navigate to the **IP Network → IP Telephony Nodes → Node Details → VGW and Codecs** property page and configure the Communication Server 1000E General codec settings as in the next screenshot (see highlighted area).

- **Voice payload size** must be set to **20**
- **Voice playout (jitter buffer) delay**, set to **40** and **80**

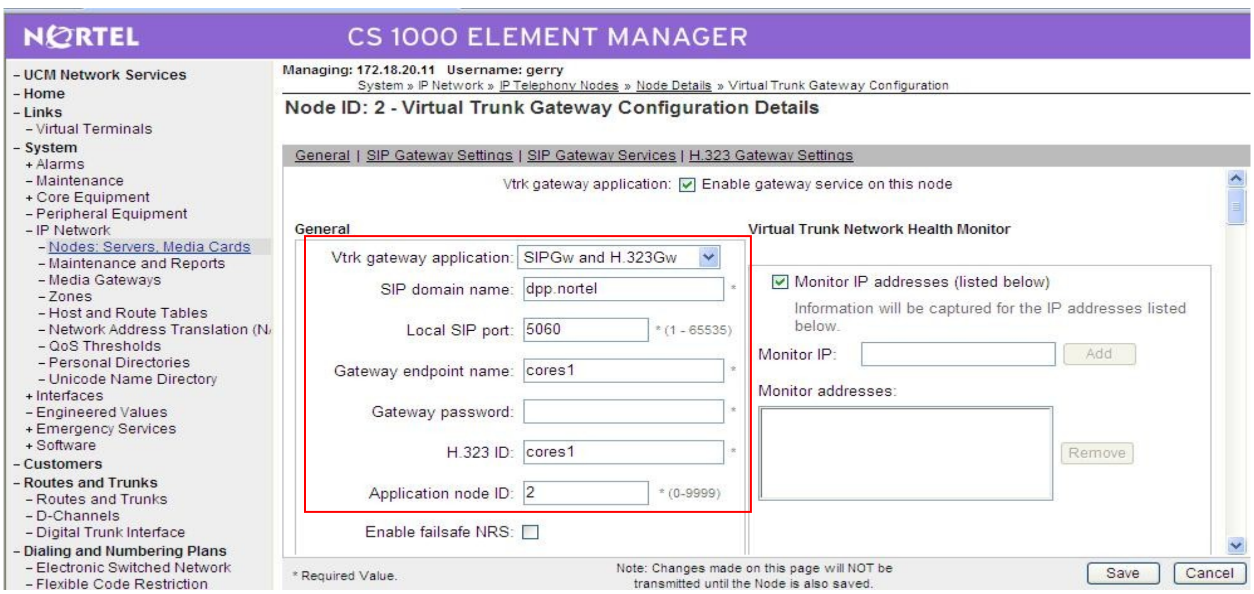


Scroll down to the bottom of the page and click on the **Save** button.

6.4. Virtual Trunk Gateway Configuration

The next screenshot shows the SIP Virtual Trunk Gateway configuration. To successfully set up Virtual trunks, the Virtual Trunk Gateway (i.e., the application that registers with the NRS to route call traffic) settings must be configured correctly. The majority of settings on this page will match those previously used when setting up the NRS properties because the gateway needs to be in the same SIP domain as the NRS and use known aliases and ports. Navigate to **System → IP Network → IP Telephony Nodes → Node Details → Virtual Trunk Gateway Configuration** and fill in the highlighted areas with the following settings

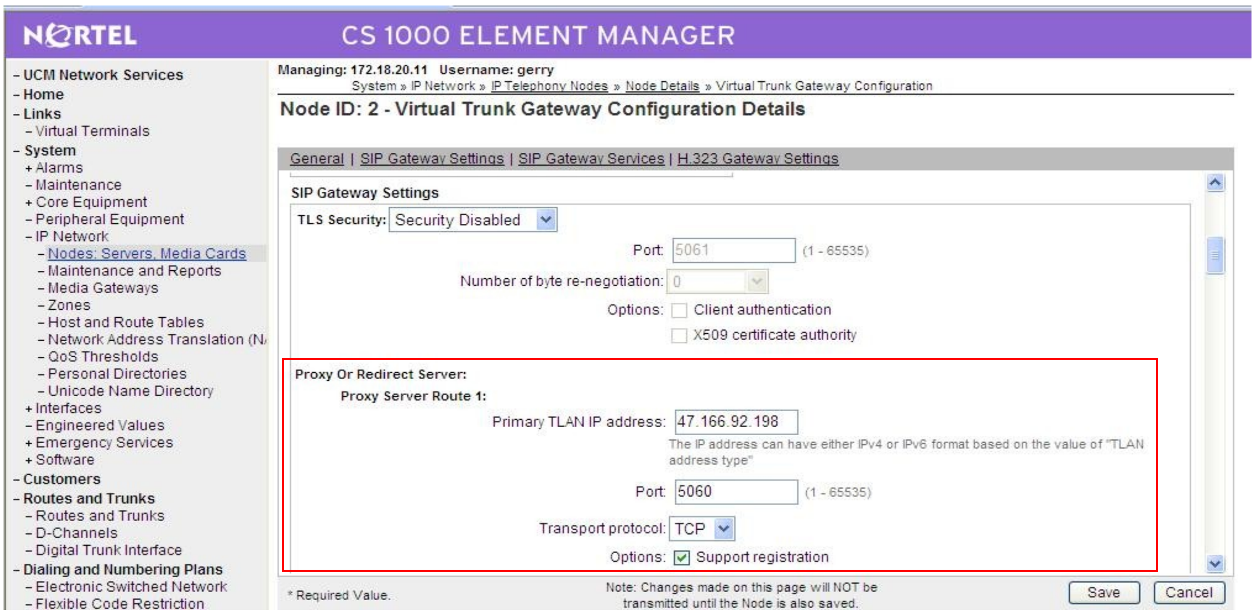
- **Vtrk gateway application**, select **SIPGw and H.323Gw**
- **SIP domain name**, must be same value as configured in **Section 5.2**
- **Local SIP port,** must be set to **5060** as in **Section 5.2**
- **Gateway endpoint name,** must be endpoint name in **Section 5.3**
- **H.323 ID,** must be as in **Section 5.3**
- **Application node ID,** must be the same as the Node ID configured in **Section 6.2**



Scroll down the page and enter the **Proxy or Redirect Server** (i.e., the NRS) settings. In the following screenshot:

- **Primary TLAN IP Address,** set to the value used in **Section 5.1**
- **Port** setting should match the setting in **Section 5.1**
- **Transport protocol** was set to **TCP**
- **Options Support registration** checkbox must be ticked

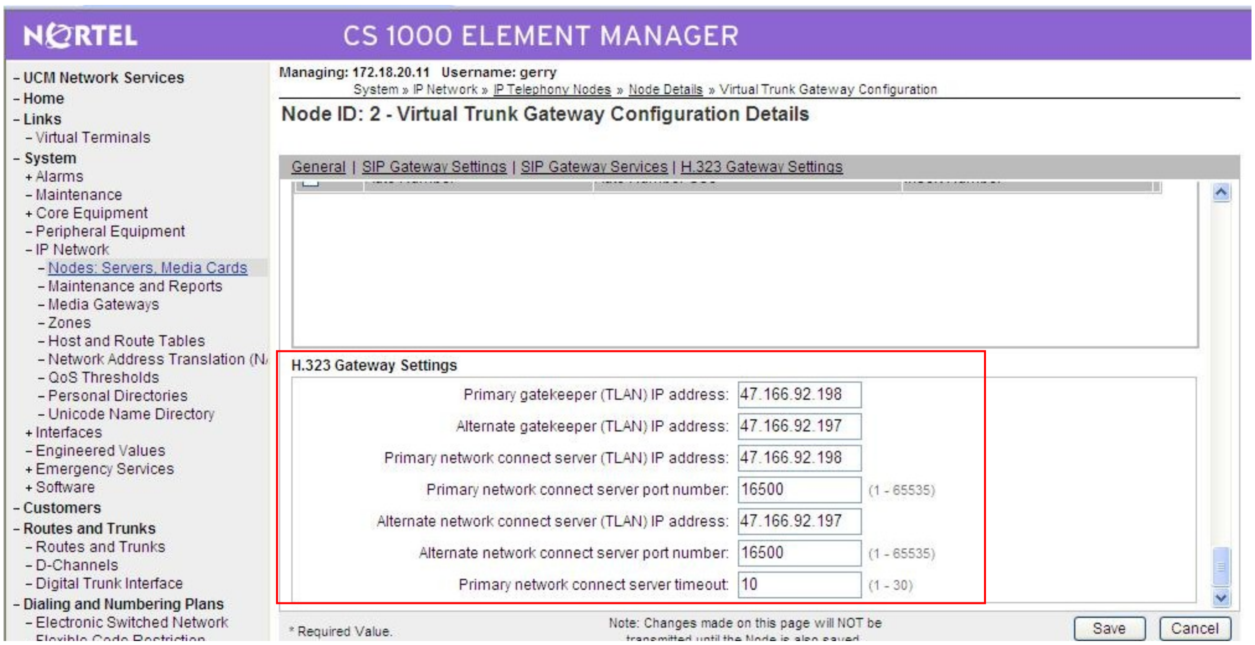
These settings configure the Virtual Trunk Gateway to allow successful registration with the NRS.



Scroll down the page to the **H.323 Gatekeeper Settings**. Use the following settings:

- **Primary gatekeeper (TLAN) IP address,** same as used in **Section 5.1**
- **Alternate gatekeeper (TLAN) IP address,** same as used in **Section 5.1**
- **Primary network connect server (TLAN) IP address,** same as **Section 5.1**
- **Primary network connect server port number** is set to **16500**
- **Alternate network connect server (TLAN) IP address,** same as **Section 5.1**
- **Alternate network connect server port number** is also **16500**
- **Primary network connect server timeout** is set to **10** seconds

Click on the **Save** button when finished.

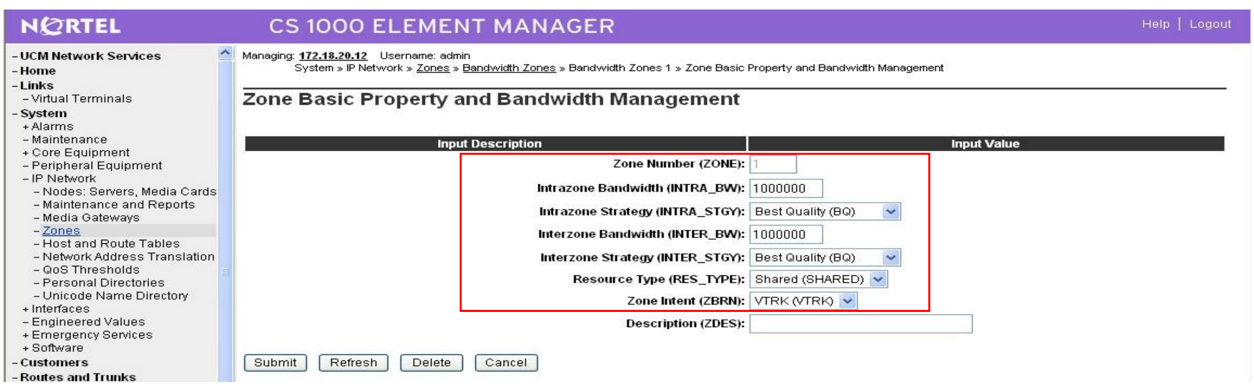


6.5. Configure Bandwidth Zones

Bandwidth Zones are used for alternate call routing between IP telephones and for call Bandwidth Management. SIP trunks require a unique zone, and best practice dictates that IP Trunks, IP telephones and Media Gateways are placed in separate zones. Use Element Manager to define bandwidth zones as in the following highlighted example. Select **Zones** from the left side menu and navigate to **Zones → Bandwidth Zones** and add new zones as required. The following screenshot shows an example Virtual Trunk zone configuration.

- **Zone Number** must be a unique non zero value.
- **Intrazone Bandwidth** is usually set to the network speed (10, 100 or 1000 M/bS)
- **Interzone Bandwidth** is usually set to the network speed (10, 100 or 1000 M/bS)
- **Intrazone Strategy** sets the preferred codec quality for in zone calls
- **Interzone Strategy** sets the preferred codec quality for zone to zone calls
- **Resource Type** can be set to **Shared**
- **Zone Intent** defines the function; in this case it is used for **VTRK** (Virtual Trunks)

Click on the Submit button when completed.



6.6. Configure SIP Trunks

Communication Server 1000E virtual trunks will be used for all inbound and outbound calls. Four separate steps are required to configure Communication Server 1000E virtual trunks:

- Configure a D-Channel Handler (DCH); configure using the Communication Server 1000E system terminal and overlay 17.
- Configure a SIP trunk Route Data Block (RDB); configure using the Communication Server 1000E system terminal and overlay 16.
- Configure SIP trunk members; configure using the Communication Server 1000E system terminal and overlay 14.
- Configure a Route List Block (RLB); configure using the Communication Server 1000E system terminal and overlay 86.

The following is an example DCH configuration for SIP trunks. Load **overlay 17** at the Communication Server 1000E system terminal and enter the following values. The highlighted entries are required for correct SIP trunk operation. Exit overlay 17 when completed.

Overlay 17 D-Channel Handler Configuration

```
ADAN     DCH 79
  CTYP DCIP
  DES  VIR_TRK
  USR  ISLD
  ISLM 4000
  SSRC 1800
  OTBF 32
  NASA YES
  IFC  SL1
  CNEG 1
  RLS  ID  5
  RCAP ND2
  MBGA NO
  H323
    OVLR NO
    OVLS NO
```

Next, configure the SIP trunk Route Data Block (RDB) using the Communication Server 1000E system terminal and overlay 16. Load **overlay 16**, enter **RDB** at the prompt, press return and commence configuration. The value for **DCH** is the same as previously entered in overlay 17. The value for **NODE** should match the node value in **Section 6.2**. The value for **ZONE** should match that used in **Section 6.5**. The remaining highlighted values are important for correct SIP trunk operation.

```
       Overlay 16 RDB
TYPE: rdbCUST 00
ROUT 100
TYPE RDB
CUST 00
ROUT 66
DES  VIR_TRK
TKTP TIE
NPID_TBL_NUM   0
ESN  NO
RPA  NO
CNVT NO
SAT  NO
RCLS EXT
VTRK YES
ZONE 1
PCID SIP
CRID NO
NODE 2
DTRK NO
ISDN YES
    MODE ISLD
    DCH  79
    IFC  SL1
    PNI  00001
    NCNA YES
    NCRD YES
    TRO  NO
    FALT NO
    CTYP UKWN
    INAC NO
    ISAR NO
    DAPC NO
MBXR NO
MBXOT NPA
MBXT 0
PTYP ATT
CNDP UKWN
AUTO NO
DNIS NO
DCDR NO
ICOG IAO
SRCH LIN
TRMB YES
STEP
```

```
ACOD 130
TCPP NO
PII NO
AUXP NO
TARG
CLEN 1
BILN NO
OABS
INST
IDC  NO
DCNO 10
NDNO 10 *
DEXT NO
DNAM NO
SIGO STD
STYP SDAT
MFC  NO
ICIS YES
OGIS YES
TIMR ICF  1920
     OGF  1920
     EOD  13952
     LCT  256
     DSI  34944
     NRD  10112
     DDL  70
     ODT  4096
     RGV  640
     GTO  896
     GTI  896
     SFB  3
     PRPS  800
     NBS  2048
     NBL  4096
     IENB  5
     TFD  0
     VSS  0
     VGD  6
     EESD  1024
SST  5 0
DTD  NO
SCDT NO
2 DT NO
NEDC ORG
FEDC ORG
```

```
CPDC NO
DLTN NO
HOLD 02 02 40
SEIZ 02 02
SVFL 02 02
DRNG NO
CDR  NO
NATL YES
SSL
CFWR NO
IDOP NO
VRAT NO
MUS  YES
MRT  21
PANS YES
RACD NO
MANO NO
FRL  0 0
FRL  1 0
FRL  2 0
FRL  3 0
FRL  4 0
FRL  5 0
FRL  6 0
FRL  7 0
OHQ  NO
OHQT 00
CBQ  NO
AUTH NO
TTBL 0
ATAN NO
OHTD NO
PLEV 2
OPR  NO
ALRM NO
ART  0
PECL NO
DCTI 0
TIDY 1600 100
ATRR NO
TRRL NO
SGRP 0
ARDN NO
CTBL 0
AACR NO
```

Next, configure virtual trunk members using the Communication Server 1000E system terminal and overlay 14. Configure sufficient trunk members to carry both incoming and outgoing PSTN calls. The following example shows a single SIP trunk member configuration. Load **overlay 14** at the system terminal and type **new X**, where X is the required number of trunks. Continue entering data until the overlay exits. The **RTMB** value is a combination of the **ROUT** value entered in the previous step and the first trunk member (usually 1). The remaining highlighted values are important for correct SIP trunk operation.

Overlay 14 Trunk Member Configuration

```
TN   120 0 0 0 DATE PAGE DES  VIR_TRK
TN   120 0 00 00  VIRTUAL
TYPE IPTI
CDEN 8D
CUST 0
XTRK VTRK
ZONE 00253
TIMP 600
BIMP 600
AUTO_BIMP NO
NMUS NO
TRK  ANLG
NCOS 0
RTMB 66 1
CHID 1
TGAR 0
STRI/STRO WNK WNK
SUPN YES
AST  NO
IAPG 0
CLS  TLD DTN CND ECD WTA LPR APN THFD XREP SPCD MSBT
     P10 NTC
TKID
AACR NO
```

Configure a Route List Block (RLB) in overlay 86. Load **overlay 86** at the system terminal and type **new**. The following example shows the values used. The value for **ROUT** is the same as previously entered in overlay 16. The **RLI** value is unique to each RLB.

```
Overlay 86
CUST 0
FEAT rlb
RLI  24
ELC  NO
ENTR 0
LTER NO
ROUT 66
TOD  0 ON  1 ON  2 ON  3 ON
     4 ON  5 ON  6 ON  7 ON
VNS  NO
SCNV NO
CNV  NO
EXP  NO
FRL  0
DMI  0
CTBL 0
ISDM 0
```

```
FCI  0
FSNI 0
BNE  NO
DORG NO
SBOC NRR
PROU 1
IDBB DBD
IOHQ NO
OHQ  NO
CBQ  NO

ISET 0
NALT 5
MFRL 0
OVLL 0
```

Next, configure Special Prefix Number(s) (**SPN**) which users will dial to reach PSTN numbers. Use the Communication Server 1000E system terminal and overlay 90. The following are some example SPN entries used. The highlighted **RLI** value previously configured in overlay 86 is used as the Route List Index (RLI); this is the default route to the NRS.

SPN 999 FLEN 3 ITOH NO CLTP NONE **RLI 24** SDRR NONE ITEI NONE	SPN 90 FLEN 7 ITOH NO CLTP NONE **RLI 24** SDRR NONE ITEI NONE	SPN 2 FLEN 7 ITOH NO CLTP NONE **RLI 24** SDRR NONE ITEI NONE	SPN 15 FLEN 3 ITOH NO CLTP NONE **RLI 24** SDRR NONE ITEI NONE

6.7. Configure Analog, Digital and IP Telephones

A variety of telephone types were used during the testing, the following is the configuration for the Avaya 1140e Unistim IP telephone. Load **overlay 20** at the system terminal and enter the following values. A unique four digit number is entered for the **KEY 00** and **KEY 01** value.

Overlay 20 IP Telephone configuration

```
DES  1140
TN   096 0 01 16  VIRTUAL
TYPE 1140
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 2
CUR_ZONE 2
ERL  0
ECL  0
FDN  0
TGAR 0
LDN  NO
NCOS 0
SGRP 0
RNPG 1
SCI  0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS  UNR FBA WTA LPR PUA MTD FNA HTA TDD HFA CRPD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LNA CNDA
     CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDA CDMD LLCN MCTD CLBD AUTR
     GPUD DPUD DNDA CFXA ARHD FITD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
     UDI RCC HBTA AHD IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
     FDSD NOVD VOLA VOUD CDMR PRED RECA MCDD T87D SBMD KEM3 MSNV FRA  PKCH MUTA MWTD

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```

```
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DVLD CROD CROD
CPND_LANG ENG
RCO  0
HUNT 0
LHK  0
PLEV 02
PUID
DANI NO
AST  00
IAPG 1
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 MCR 5200 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME IP1140
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     01 MCR 5200 0
        CPND
          CPND_LANG ROMAN
            NAME IP1140
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     02
     03 BSY
     04 DSP
     05
     06
     07
     08
     09
     10
     11
     12
     13
     14
     15
     16
     17 TRN
     18 AO6
     19 CFW 16
     20 RGA
     21 PRK
     22 RNP
     23
     24 PRS
     25 CHG
     26 CPN
```

Digital telephones are configured using the overlay 20; the following is a sample 3904 digital set configuration. Again, a unique number is entered for the **KEY 00** and **KEY 01** value.

Overlay 20 – Digital Set configuration

```
TYPE: 3904
DES  3904
TN   000 0 09 08  VIRTUAL
TYPE 3904
CDEN 8D
CTYP XDLC
CUST 0
MRT
ERL  0
FDN  0
TGAR 0
LDN  NO
NCOS 0
SGRP 0
RNPG 1
SCI  0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS  UNR FBD WTA LPR PUA MTD FND HTD TDD HFA GRLD CRPA STSD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LNA CNDA
     CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDA CDMA LLCN MCTD CLBD AUTU
     GPUD DPUD DNDA CFXA ARHD FITD CNTD CLTD ASCD
     CPFA CPTA ABDA CFHD FICD NAID BUZZ AGRD MOAD
     UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
     FDSD NOVD CDMR PRED RECA MCDD T87D SBMD PKCH CROD CROD
CPND_LANG ENG
RCO  0
HUNT
PLEV 02
PUID
DANI NO
SPID NONE
AST
IAPG 1
AACS
ACQ
ASID
SFNB
SFRB
USFB
CALB
FCTB
ITNA NO
DGRP
PRI  01
MLWU_LANG 0

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```

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```
MLNG ENG
DNDR 0
KEY  00 MCR 5201 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME Digital Set
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     01 MCR 5201 0
        CPND
          CPND_LANG ROMAN
            NAME Digital Set
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     02 DSP
     03 MSB
     04
     05
     06
     07
     08
     09
     10
     11
     12
     13
     14
     15
     16
     17 TRN
     18 AO6
     19 CFW 16
     20 RGA
     21 PRK
     22 RNP
     23
     24 PRS
     25 CHG
     26 CPN
     27 CLT
     28 RLT
     29
     30
     31
```

Analog telephones are also configured using overlay 20; the following example shows an analog port configured for Plain Old Telephone Service (POTS) and also configured to allow T.38 Fax transmission. A unique value is entered for **DN**, this is the extension number. **DTN** is required if the telephone uses DTMF dialing.

Overlay 20 – Analog Telephone Configuration

```
DES  500
TN   100 0 00 03
TYPE 500
CDEN 4D
CUST 0
MRT

ERL 00000
WRLS NO
DN   5202
AST  NO
IAPG 0
HUNT
TGAR 0
LDN  NO
NCOS 0
SGRP 0
RNPG 0
XLST
SCI  0
SCPW
SFLT NO
CAC_MFC 0
CLS  UNR DTN FBD XFD WTA THFD FND HTD ONS
     LPR XRD AGRD CWD SWD MWD RMMD SMWD LPD XHD SLKD CCSD LND TVD
     CFTD SFD MRD C6D CNID CLBD AUTU
     ICDD CDMD LLCN EHTD MCTD
     GPUD DPUD CFXD ARHD OVDD AGTD CLTD LDTD ASCD SDND
     MBXD CPFA CPTA UDI RCC HBTD IRGD  DDGA NAMA MIND
     NRWD NRCD NROD SPKD CRD PRSD MCRD
     EXR0 SHL SMSD ABDD CFHD DNDY DNO3
     CWND USMD USRD CCBD BNRD OCBD RTDD RBDD RBHD FAXA CNUD CNAD PGND FTTC
     FDSD NOVD CDMR PRED MCDD T87D SBMD PKCH MPTD
PLEV 02
PUID
AACS NO
MLWU_LANG 0
FTR  DCFW 4
```

6.8. Configure Intercept Computer Protocol (ICP) serial port

A serial port (TTY) must be configured on the Communication Server 1000E which Trio Enterprise uses to retrieve ICP data. Use **overlay 17** to configure a new TTY port as in the following example. **User** must be **ICP.**

Overlay 17 – ICP TTY Configuration

```
ADAN     TTY 9
  CTYP MGC
  IPMG 4 0
  DNUM 6
  PORT 1
  DES  ICP_DATA
  BPS  9600
  BITL 8
  STOP 1
  PARY NONE
  FLOW NO
  USER ICP
  TTYLOG       0
  BANR YES
```

This completes the configuration required for the Communication Server 1000E.

7. Configure Trio Enterprise Server

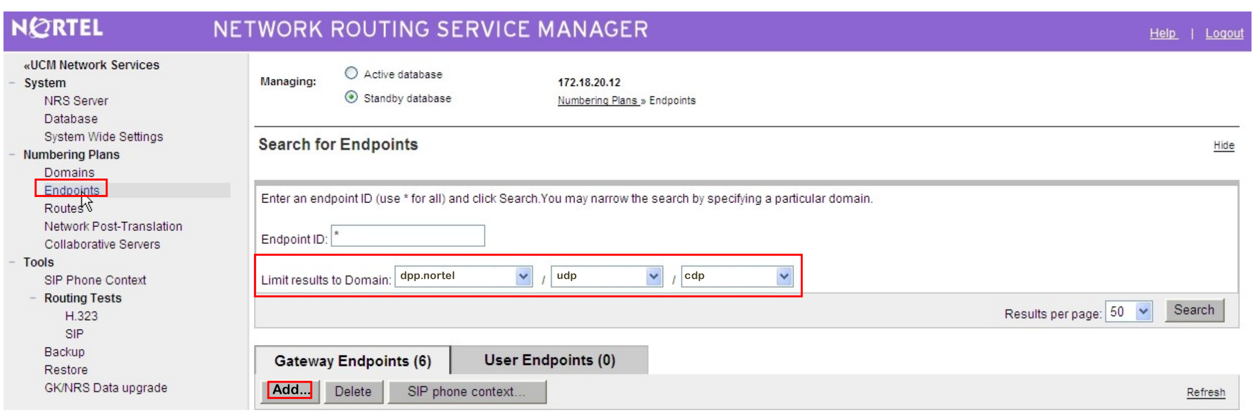
The primary purpose of Trio Enterprise is to provide an attendant position to Communication Server 1000E systems. Trio Enterprise overcomes the installation limitation of 1000 feet from the call server by using a SIP trunk connection to the NRS. This allows the Trio attendant to be located anywhere in the building or offsite if required. The Trio server consists of a Windows PC running Microsoft XP or Server 2003/2008 with the Trio Enterprise R3.0 software installed. An Ethernet connection to the Communication Server 1000E T-LAN is required, as is a serial connection to a call server serial port, configured to support Intercept Computer Protocol.

This section describes the steps necessary to configure Trio Enterprise 3.0 to operate with Communication Server 1000E. The following procedures are discussed:

- Configure Trio Enterprise as a dynamic SIP endpoint on the NRS
- Configure routing entries for Trio Enterprise
- Configure Trio Enterprise to interface with the NRS
- Configure Trio Attendant

7.1. Configure Trio Enterprise as a dynamic SIP endpoint on the NRS

Access UCM as in **Section 5.1**. Scroll down and select **NRSM on sps** from the **Elements** list (see the highlighted section in **Section 5.1** screenshot). The Network Routing Service Manager (NRSM) page will open in a new window. Ensure the **Managing Standby database** radio button is checked. Click on the **Endpoint** entry in the left hand side menu and in the resulting page select the previously configured SIP, L1 and L0 domains from the **Limit results to Domain** suite of drop-down lists. The following screenshot shows the previously entered SIP, L1 and L0 domains highlighted.

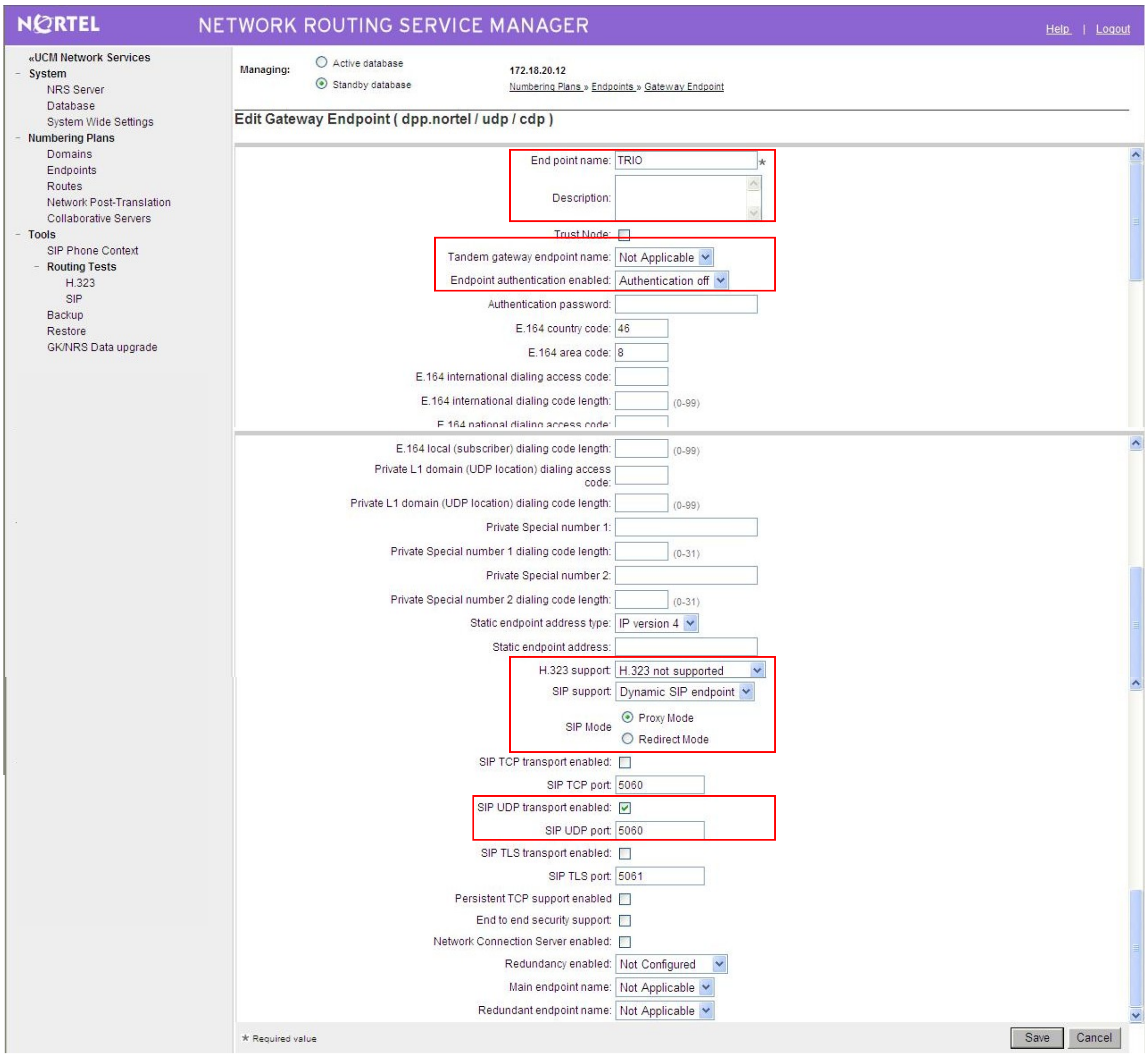


When drop-down lists have been populated with the correct values, the **Add** button is then activated and a new endpoint can be added. Click on the **Add** button and enter the endpoint data. See the highlighted area in the following screenshot for information.

- **End point name** (**TRIO** in this example)
- **Description** is typically some text to describe the endpoint
- **Tandem gateway endpoint name** is set to **Not Applicable**
- **Endpoint Authentication enabled** is set to **Authentication off**

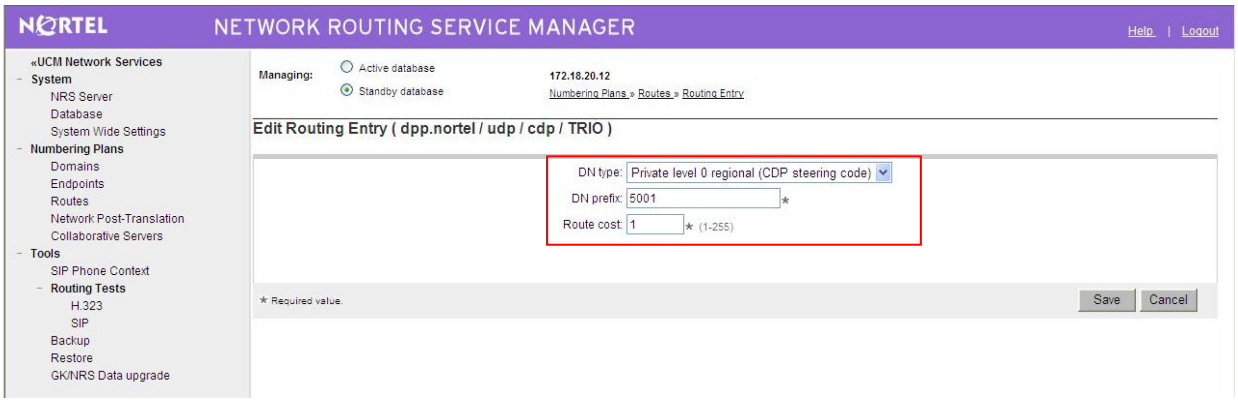
- **H.323 support** must be set to **H.323 not supported**
- **SIP support** must be set to **Dynamic SIP endpoint**
- **SIP Mode** must be set to **Proxy Mode**
- **SIP UDP transport enabled** checkbox must be ticked
- **SIP UDP port** must be **5060**

The remaining values will be specific to the particular location and endpoint being configured. Examples of typical values are shown, and the correct values must be entered before the endpoint is brought into service. Click on the **Save** button to confirm the settings.

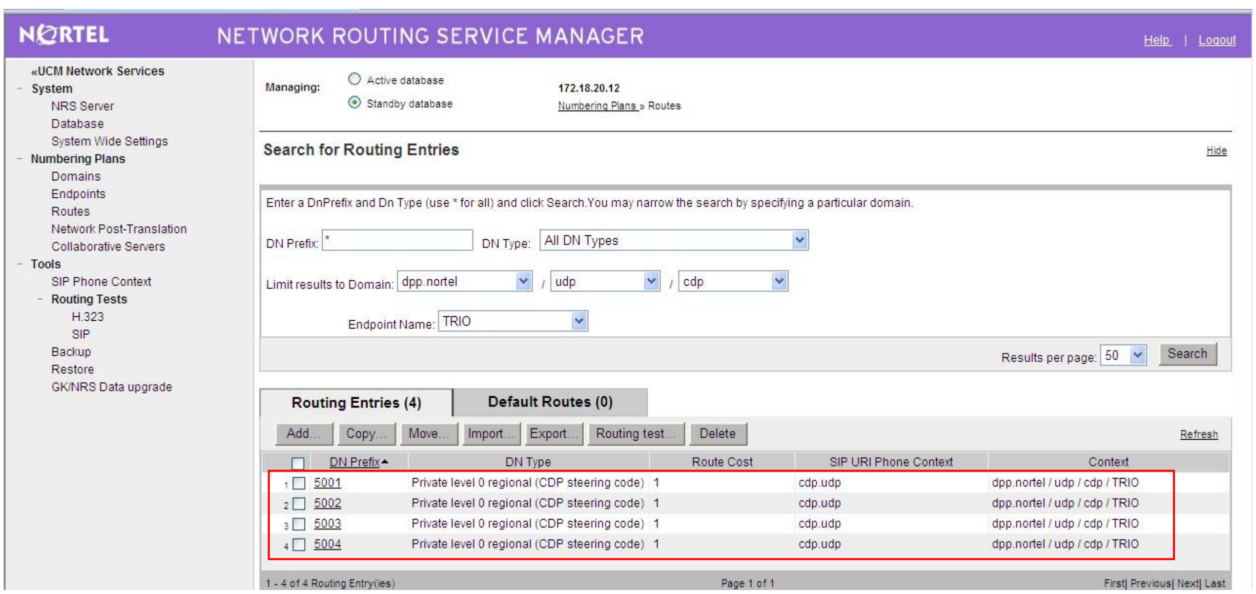


7.2. Configure Routing Entries for Trio Enterprise

Trio Enterprise needs NRS routing entries to carry out call control operations. To add a new routing entry, ensure the **Managing Standby database** radio button is checked. Click on the **Routes** entry in the left side menu (see screenshot in **Section 5.4**) and in the resulting page select the previously configured **SIP, L1 and L0** domains from the **Limit results to Domain** suite of drop-down lists. When drop-down lists have been populated with the correct values, select the endpoint from the **Endpoint Name** drop down list (endpoint = TRIO). The **Add** button is activated and a new route can be added. Click on the **Add** button and enter the route data. The following screenshot shows an example routing entry for the TRIO endpoint.

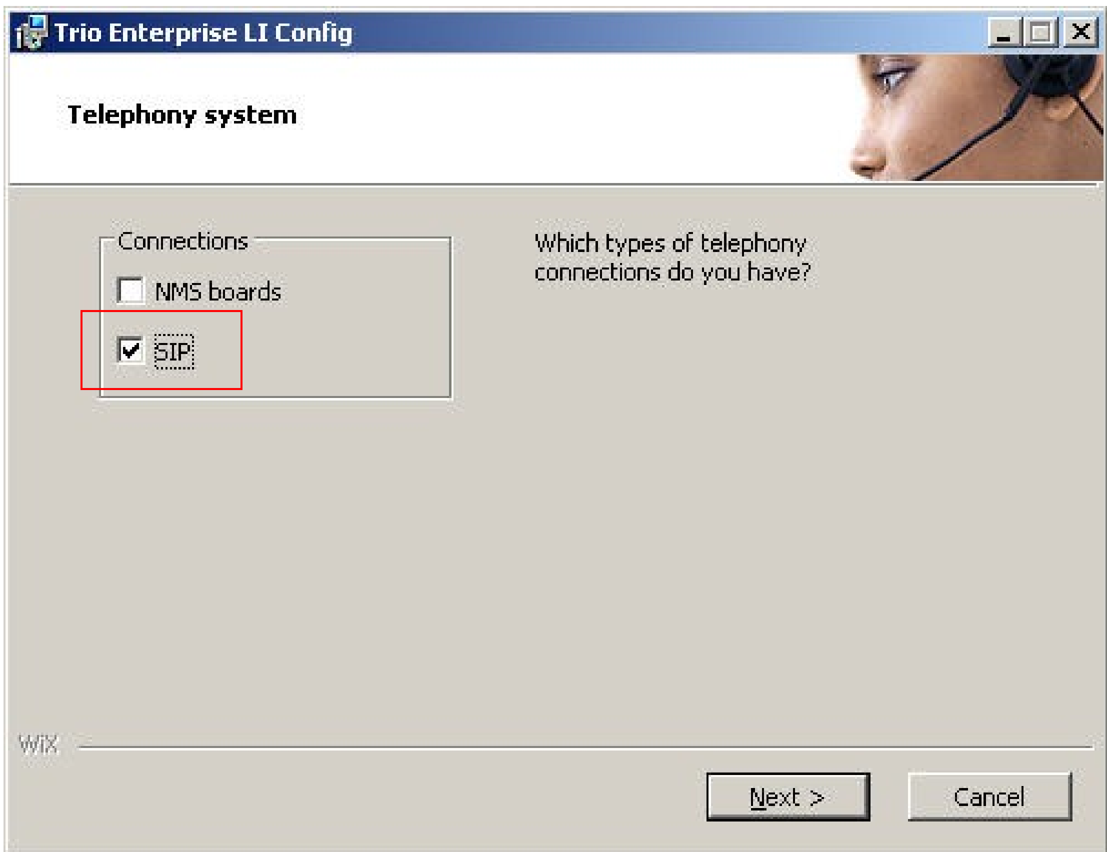


Click on the **Save** button when finished. The following screenshots shows the complete list of routing entries configured for the TRIO endpoint.

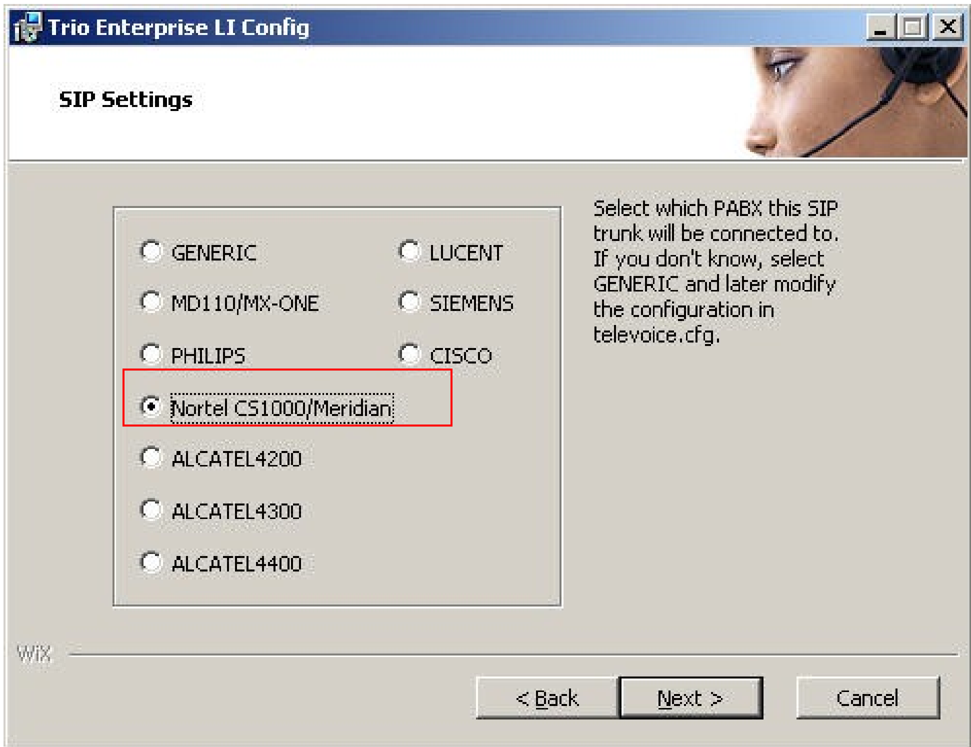


7.3. Configure Trio Enterprise to interface with the NRS

Trio Enterprise needs to register with the NRS before it can make and receive calls. The Trio endpoint details have already been configured on the NRS, this section shows how to configure Trio Enterprise to register with the NRS for SIP trunk operation. The installation of the Trio Enterprise software is assumed to be completed and the Trio services are up and running. Stop the Trio Televoice service by selecting **Start → Run**, then type **services.msc** into the command line and press return. When the standard services window opens, locate the Trio TeleVoice service and stop the service running. Launch the Trio configuration application by selecting it from **Start → Programs → Trio Enterprise → Line Interface** and clicking on the **Config** entry (not shown). The configuration application starts up and presents the following screen. Ensure the **SIP** entry in the **Connections** area is checked. Click on **Next** when ready.



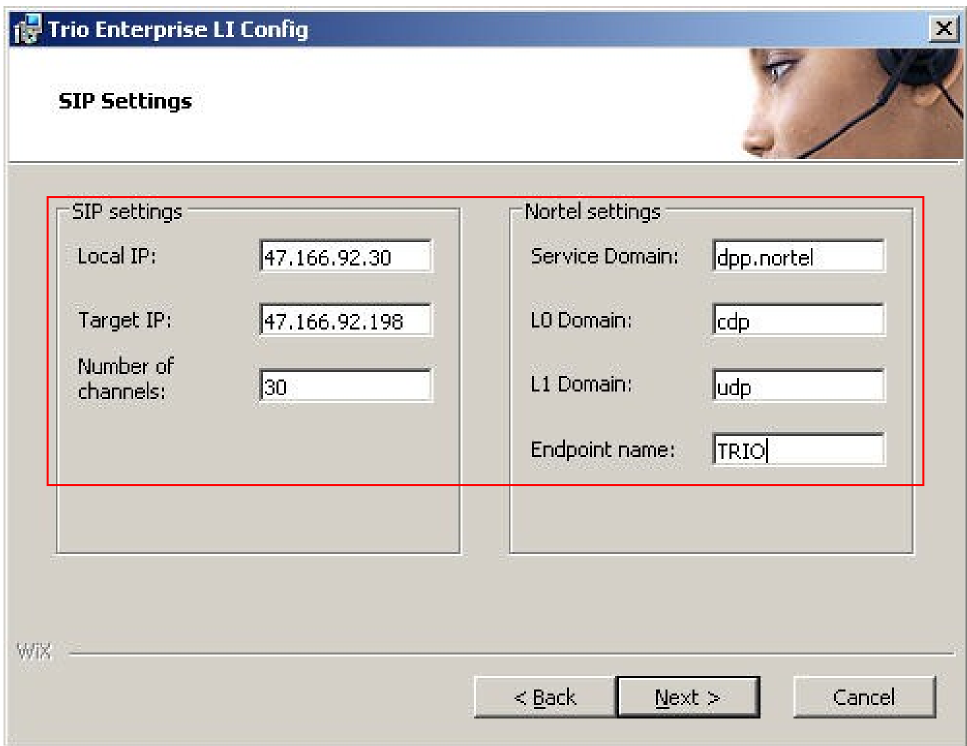
In the next screenshot (first page of SIP Settings), ensure the **Nortel CS1000/Meridian** radio button entry is selected. Click on **Next** when ready.



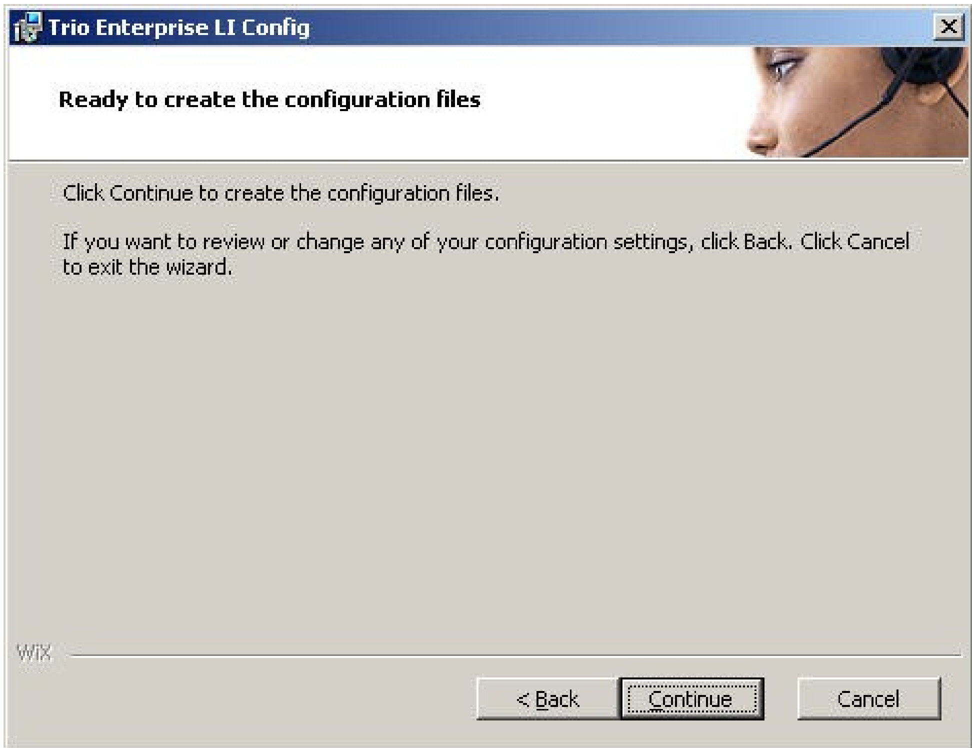
The second page of SIP Settings opens up. The following needs to be configured:

- For **Local IP:** configure the Trio server TLAN IP address
- For **Target IP:** configure the NRS TLAN IP address
- **Number of Channels:** refers to the number of SIP trunks in the Trio license file.
- **Service Domain:** is the SIP domain configured in **Section 5.2**
- **L0 Domain:** is the L0 Domain configured in **Section 5.2**
- **L1 Domain:** is the L1 Domain configured in **Section 5.2**
- **Endpoint name:** is the endpoint name configured in **Section 7.1** for Trio Enterprise

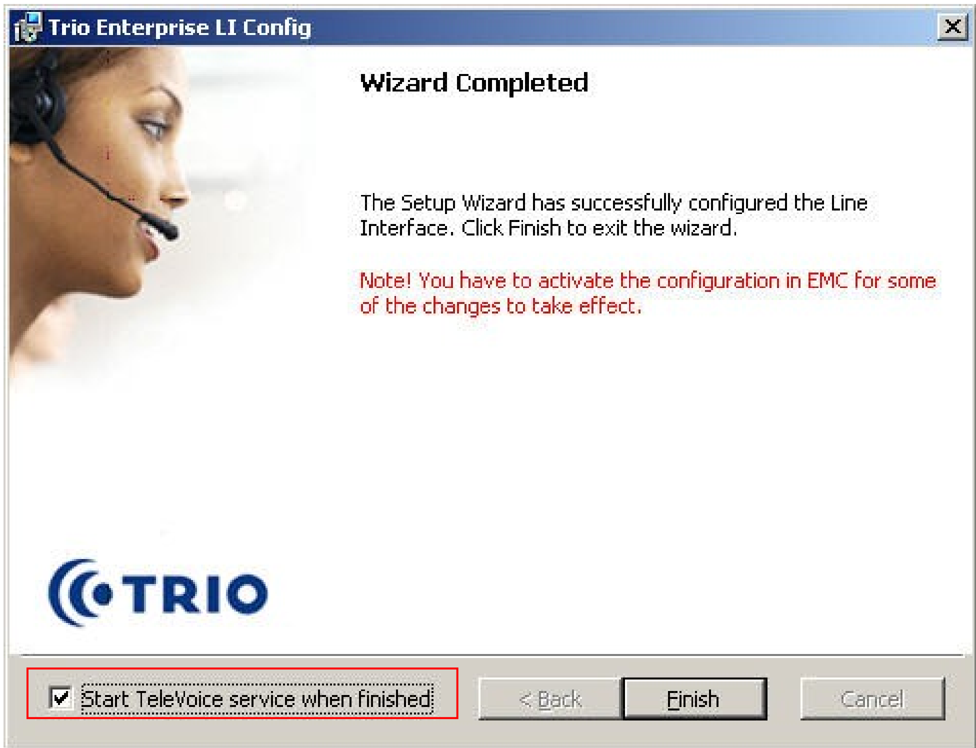
Click on **Next** when ready.



If all configuration steps have been completed, click on the **Continue** button.



If the configuration wizard completes successfully, the following screen appears. Ensure the **Start TeleVoice service when finished** option is checked. Click on the **Finish** button to complete the configuration procedure. This completes the Trio Enterprise server side configuration steps. The Trio Enterprise services are now restarted with the new configuration.

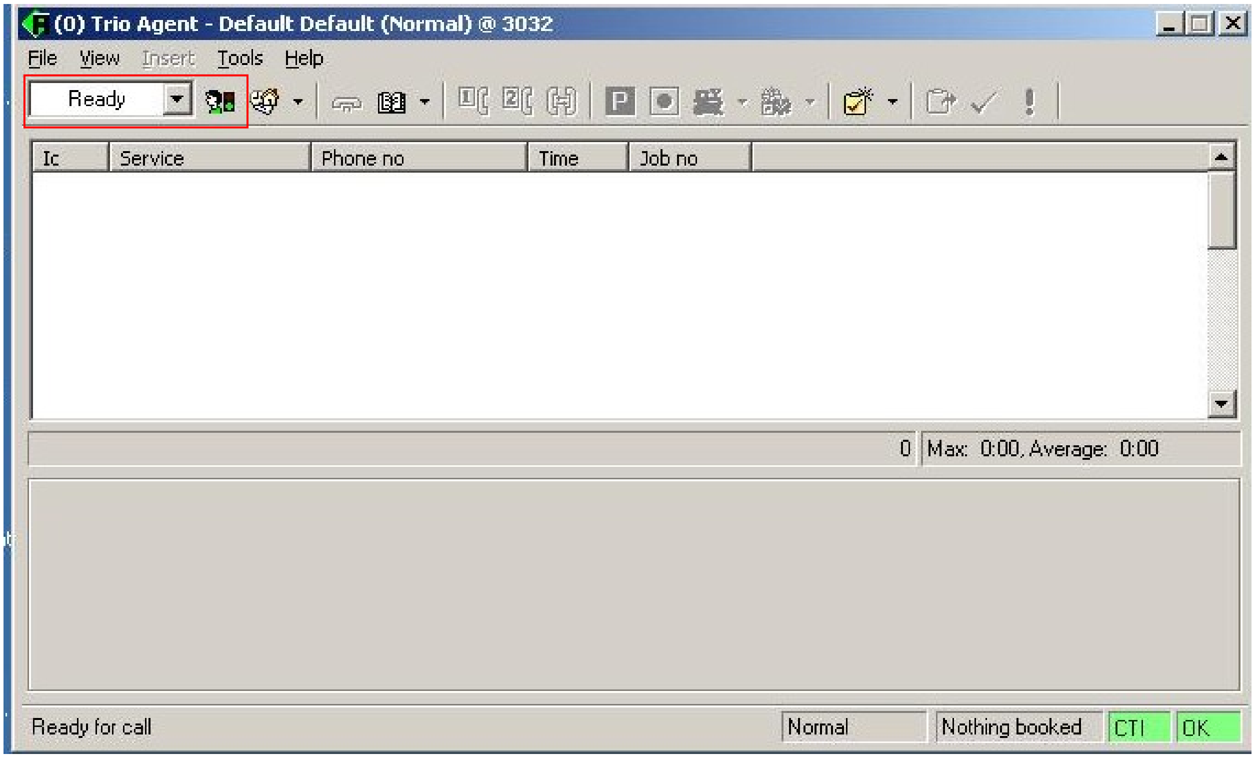


7.4. Configure Trio Attendant

Trio attendant is a separate application that can run concurrently on the same server platform. The attendant uses a regular Communication Server 1000E telephone to make and receive calls, which are directed to the phone by Trio Enterprise server using SIP messaging.

Launch Trio attendant by clicking on **Start → Programs → Trio Enterprise → Contact Centre → Agent Client**. The following window opens. Enter a valid **User ID** and **Password**. For **Extension,** select the Communication Server 1000E telephone number that will be used as the agent's audio device (number **3032** in this example). Ensure the correct Trio Enterprise server is selected if there is more than one on the network (default is the current Trio server). Confirm **Phone type** is set to **Standard phone**. Click on the **OK** button when finished.

The Trio Agent window appears. Select **Ready** from the drop down box (confirm the traffic light goes green in the small icon to the right of the drop-down box).

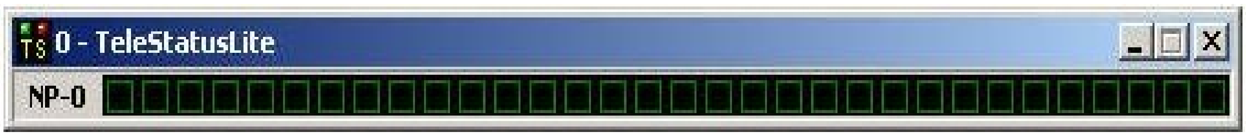


This completes configuration of the Trio attendant.

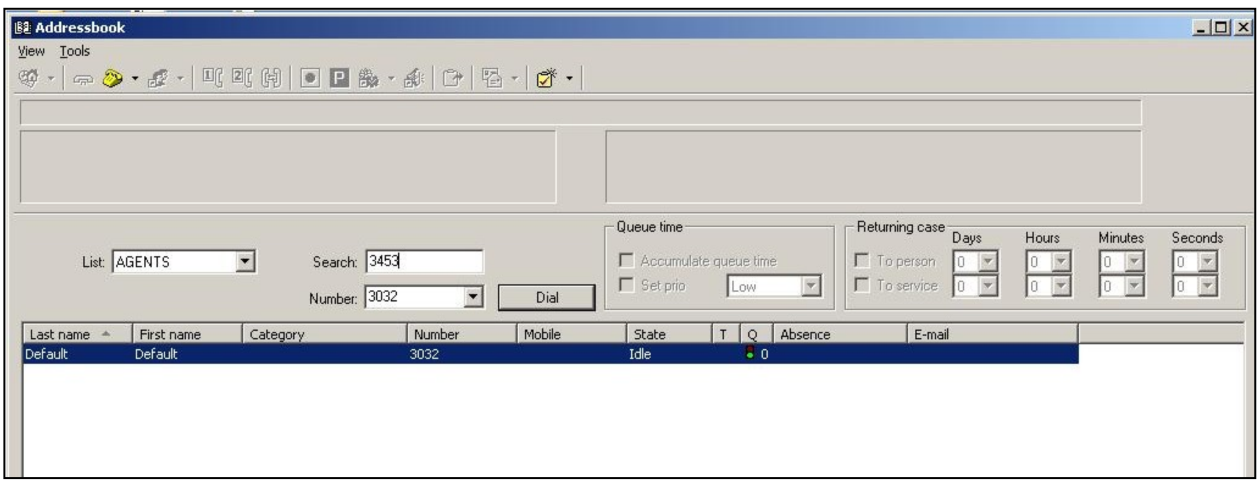
8. Verification Steps

To confirm successful Trio Enterprise configuration with the CS 1000E perform the following two actions:

- Click on **Start → Programs → Trio Enterprise → Line Interface** and then select the **Telestatus** entry. A new window opens, showing the SIP trunk channel status as a series of green squares. Confirm the SIP trunks are all in the idle state (unfilled green squares).



- Click on **Start → Programs → Trio Enterprise → Contact Centre → Agent Client**. The Trio attendant window opens, log in with a valid user ID and password (as in **Section 7.4**) and click on the **Tools → Dial** entry (not shown). A new **Addressbook** window opens (see the following screenshot). Make a call to a known telephone number and confirm the call completes normally. Place a call to extension 5001and confirm the Trio client displays an incoming call and the telephone alerts.



9. Conclusion

As illustrated in these application notes Avaya Communication Server 1000E and Trio Enterprise R3.0 can be configured to interoperate successfully. This solution provides Avaya Communication Server 1000E users with the capability to have an attendant answering position located remotely from the PBX installation if required.

10. Additional References

The following documents and external references may be helpful in understanding operation of particular CS1000 features and may provide more detailed information:

1 Information on how to install and configure Linux and Telephony applications - Avaya Communication Server 1000E – Documentation Library – Document NN43001-315 (*Linux Platform Base and Applications Installation and Commissioning)*

2 Installation procedures and guidelines for CS1000E system installers - Avaya Communication Server 1000E – Documentation Library – Document NN43041-310 *(Communication Server 1000E Installation and Commissioning)*.

3 System Management platform (UCM) provides security, software deployment and other services to CS1000E elements - Avaya Communication Server 1000E – Documentation Library – Document NN43001-116 *(Unified Communications Management Common Services Fundamentals)*.

4 For information on configuring and operating SIP Line services on the CS1000E, see the following document:- Avaya Communication Server 1000E – Documentation Library – Document NN43001-508 (*Configuration SIP Line Fundamentals)*

5 A complete and detailed account of all CS1000E telephony features and services can be found in the following document:- Avaya Communication Server 1000E – Documentation Library – Document NN43001-106-B1 through NN43001-106-B6 (*Communication Server 1000 Features and Services Fundamentals - Book 1 through Book 6)*.

6 An installation guide for Trio Enterprise product version 3.0:- *Standard configuration of Trio Enterprise 3.0 with Avaya CS1K 7.0*, published February 2011 – (public version).

Appendix A: Linux Base software and installed Applications

```
Product Release: 7.00.20.00
Base Applications
   base                          7.00.20     [patched]
   NTAFS                         7.00.20
   sm                            7.00.20
   nortel-Auth                   7.00.20
   Jboss-Quantum                 n/a
   lhmonitor                     7.00.20
   baseAppUtils                  7.00.20
   dfoTools                      7.00.20
   nnnm                          7.00.20
   cppmUtil                      n/a         [patched]
   oam-logging                   7.00.20
   dmWeb                         n/a
   baseWeb                       7.00.20
   ipsec                         7.00.20
   Snmp-Daemon-TrapLib           7.00.20
   ISECSH                        7.00.20
   patchWeb                      n/a         [patched]
   EmCentralLogic                7.00.20
Application configuration: CS+SS+EM
Packages:
CS+SS+EM
Configuration version:     7.00.20-00
   cs                            7.00.20
   dbcom                         7.00.20
   cslogin                       7.00.20
   sigServerShare                7.00.20     [patched]
   csv                           7.00.20.01  [patched]
   tps                           7.00.20     [patched]
   vtrk                          7.00.20
   pd                            7.00.20
   sps                           7.00.20
   ncs                           7.00.20
   gk                            7.00.20
   EmConfig                      7.00.20
   emWeb_6-0                     7.00.20
   emWebLocal_6-0                7.00.20
   csmWeb                        7.00.20
   bcc                           7.00.20
   ftrpkg                        7.00.20
   cs1000WebService_6-0          7.00.20
   managedElementWebService      7.00.20
   mscAnnc                       7.00.20     [patched]
   mscAttn                       7.00.20
   mscConf                       7.00.20     [patched]
   mscMusc                       7.00.20
   mscTone                       7.00.20     [patched]
```

Appendix B: Installed Linux Base and Application Patches and Service Updates

```
Product Release: 7.00.20.00
In system patches: 2
PATCH#  NAME       RPM
22      p30179_1   nortel-cs1000-OS-1.00.00.00-00.noarch
23      p30181_1   nortel-cs1000-OS-1.00.00.00-00.noarch

In System service updates: 22
PATCH#  NAME
0       nortel-cs1000-linuxbase-7.00.20.09-00.i386.000
1       nortel-cs1000-patchWeb-7.00.20.04-00.i386.000
2       nortel-cs1000-csv-7.00.20.01-00.i386.000
3       nortel-cs1000-tps-7.00.20-03.i386.000
4       nortel-cs1000-shared-tpselect-7.00.20.01-00.i386.000
5       nortel-cs1000-cnd-3.2.22-00.i386.000
6       nortel-cs1000-mscAnnc-7.00.20-01.i386.000
7       nortel-cs1000-mscTone-7.00.20-01.i386.000
8       nortel-cs1000-mscConf-7.00.20-01.i386.000
9       nortel-cs1000-cppmUtil-7.00.20.01-00.i686.000
10      nortel-cs1000-mscMusc-7.00.20-01.i386.000
11      nortel-cs1000-dbcom-7.00.20-01.i386.000
12      nortel-cs1000-mscAttn-7.00.20-02.i386.000
13      nortel-cs1000-dmWeb-7.00.20.01-00.i386.001
14      nortel-cs1000-csmWeb-7.00.20.03-00.i386.000
15      nortel-cs1000-ftrpkg-7.00.20.01-00.i386.000
16      nortel-cs1000-cs1000WebService_6-0-7.00.20.03-00.i386.000
17      nortel-cs1000-Jboss-Quantum-7.00.20.04-00.i386.001
18      nortel-cs1000-emWeb_6-0-7.00.20.04-00.i386.000
19      nortel-cs1000-bcc-7.00.20.06-00.i386.000
20      nortel-cs1000-vtrk-7.00.20-08.i386.000
21      nortel-cs1000-sps-7.00.20-07.i386.000

The following SP is in service: Service_Pack_Linux_7.00_20_20101116.ntl
```

Appendix C: Installed call server dependency lists

```
VERSION 4121
RELEASE 7
ISSUE 00 Q +
DepList 1: core Issue: 01 (created: 2010-09-14 13:43:30 (est))

IN-SERVICE PEPS
PAT# CR #            PATCH REF #     NAME      DATE        FILENAME
SPECINS
000  Q02162391       ISS1:1OF1       p30272_1  30/05/2011  p30272_1.cpl   NO
001  Q02151971-01    ISS1:1OF1       p30183_1  30/05/2011  p30183_1.cpl   NO
002  Q02152936-01    ISS1:1OF1       p30249_1  30/05/2011  p30249_1.cpl   NO
003  Q02162037       ISS1:1OF1       p30266_1  30/05/2011  p30266_1.cpl   YES
004  Q02149076-01    ISS1:1OF1       p30206_1  30/05/2011  p30206_1.cpl   NO
005  Q02158718-01    ISS1:1OF1       p30311_1  30/05/2011  p30311_1.cpl   NO
006  Q02143641-01    ISS1:1OF1       p30159_1  30/05/2011  p30159_1.cpl   NO
007  Q02159250-01    ISS1:1OF1       p30280_1  30/05/2011  p30280_1.cpl   NO
008  Q02156594       ISS1:1OF1       p30276_1  30/05/2011  p30276_1.cpl   YES
009  Q02143605-02    ISS1:1OF1       p30089_1  30/05/2011  p30089_1.cpl   NO
010  Q02152254       ISS1:1OF1       p30271_1  30/05/2011  p30271_1.cpl   NO
011  Q02159545       ISS1:1OF1       p30277_1  30/05/2011  p30277_1.cpl   YES
012  Q02145107-02    ISS1:1OF1       p30126_1  30/05/2011  p30126_1.cpl   NO
013  Q02161860       ISS2:1OF1       p30263_2  30/05/2011  p30263_2.cpl   NO
014  Q02152968-01    ISS1:1OF1       p30168_1  30/05/2011  p30168_1.cpl   NO
015  Q02157114       ISS1:1OF1       p30251_1  30/05/2011  p30251_1.cpl   NO
016  Q02154023       ISS1:1OF1       p30157_1  30/05/2011  p30157_1.cpl   NO
017  Q02154408       ISS1:1OF1       p30162_1  30/05/2011  p30162_1.cpl   NO
018  Q02165164       ISS1:1OF1       p30304_1  30/05/2011  p30304_1.cpl   NO
019  Q02156744       ISS2:1OF1       p30248_2  30/05/2011  p30248_2.cpl   NO
020  Q02150582-02    ISS2:1OF1       p30144_2  30/05/2011  p30144_2.cpl   NO
MDP>LAST SUCCESSFUL MDP REFRESH :2010-10-12 14:18:19(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2010-10-12 09:11:33(est)
```

Appendix D: Installed call server patches and plug-ins

```
14/02/11 13:58:36
TID: 46379
VERSION 4121

System type is - Communication Server 1000E/CPPM LinuxCPPM - Pentium M 1.4
GHz
IPMGs Registered:               2IPMGs Unregistered:             0IPMGs
Configured/unregistered:  0

RELEASE 7
ISSUE 00 Q  +
IDLE_SET_DISPLAY Rls 7.0 CoRes1
DepList 1: core Issue: 01(created: 2010-09-14 13:43:30 (est))

MDP>LAST SUCCESSFUL MDP REFRESH :2010-10-12 14:18:19(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2010-10-12 09:11:33(est)
SYSTEM HAS NO USER SELECTED PEPS IN-SERVICE


LOADWARE VERSION: PSWV 100

INSTALLED LOADWARE PEPS : 0

ENABLED PLUGINS : 7

PLUGIN STATUS  DESCRIPTION
---------------------------------------------------
 34 ENABLED MPLR12678 Intermittent Orion lockups
 37 ENABLED MPLR10894 ERR5413 When disc an unanswered euro call.
 48 ENABLED MPLR12169 ORION: NO SPEECHPATH WHEN TRANSFER CALL TO ORION
 55 ENABLED MPLR05511 EUROISDN: UIPE BRIT status mess. on EST CONFIRM Layer 2
 59 ENABLED MPLR10160 TO DO BLIND TRANSFER ACROSS MCDN NETWORK
 68 ENABLED MPLR15764 DASS CALL TO BE ROUTED TO ACD QUEUE FOR AODN.
 74 ENABLED MPLR16079 Time of day displayed on MDECT handsets
```

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.