



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.

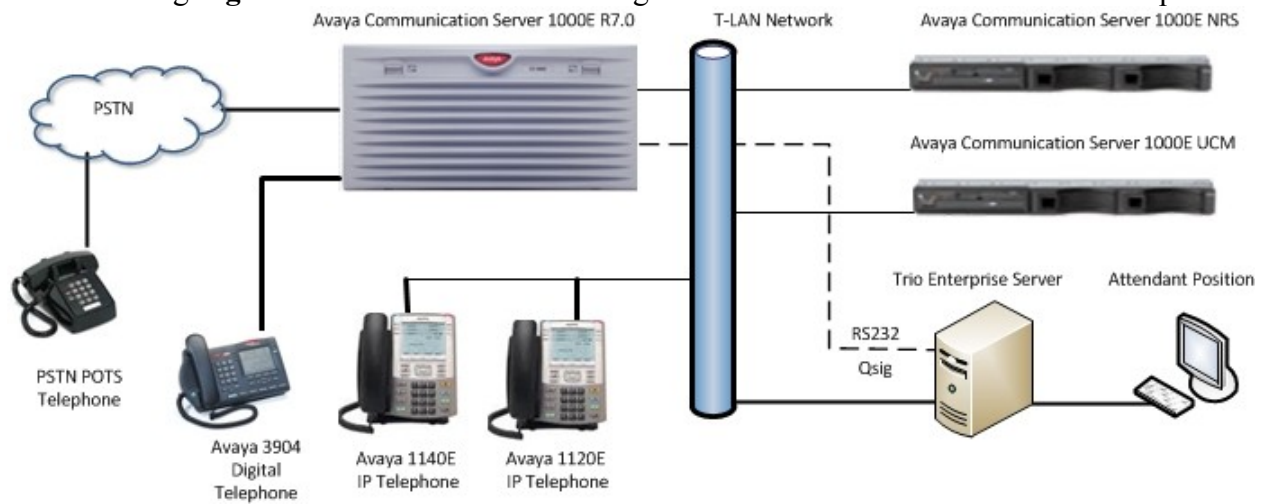


Figure 1: Test Configuration for Avaya Communication Server 1000E and Trio Enterprise V3.0

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 <ul style="list-style-type: none">• 1140e• 1120e Avaya M3900 series Telephones <ul style="list-style-type: none">• 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 screenshot shows the Avaya Unified Communications Management (UCM) web interface. The top header bar is purple with the Avaya logo and the text 'UNIFIED COMMUNICATIONS MANAGEMENT'. Below the header, there is a navigation sidebar on the left with categories like Network, Elements, CS 1000 Services, User Services, Security, and Tools. The main content area is titled 'Elements' and contains a table of registered elements. The table has columns for Element Name, Element Type, Release, Address, and Description. The element 'NRSM on sps' is highlighted with a red box.

Element Name	Element Type	Release	Address	Description
cores1.galctlab.com (member)	Linux Base	7.0	47.166.92.206	Base OS element.
dpp-ucm.galctlab.com (primary)	Linux Base	7.0	47.166.92.202	Base OS element.
cores2.galctlab.com (member)	Linux Base	6.0	47.166.92.197	Base OS element.
172.18.20.16	Media Gateway Controller	6.0	172.18.20.16	New element.
172.18.20.17	Media Gateway Controller	6.0	172.18.20.17	New element.
172.18.20.3	Media Gateway Controller	6.0	172.18.20.3	New element.
172.18.20.15	Media Gateway Controller	7.0	172.18.20.15	New element.
NRSM on sps	Network Routing Service	7.0	172.18.20.13	New element.
NRSM on cores2	Network Routing Service	6.0	172.18.20.12	New element.

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.

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: *

Primary TLAN IP address: *

Secondary TLAN IP address: *

Secondary server host name: *

Control priority:

Server mate communication port:

Realm name: *

Server role:

H.323 Gatekeeper Settings

Location request (LRQ) response timeout: (Seconds)

SIP Server Settings

Public name for non-trusted networks:

Public number for non-trusted networks:

UDP Transport enabled: ☒

Primary server UDP IP:

Primary server UDP port:

Secondary server UDP IP:

Secondary server UDP port:

TCP Transport enabled: ☒

Primary server TCP IP:

Primary server TCP port:

Secondary server TCP IP:

Secondary server TCP port:

TLS Transport enabled: ☒

Primary server TLS IP:

Primary server TLS port:

Secondary server TLS IP:

Secondary server TLS port:

Transport Layer Security (TLS) Settings

Maximum session cache:

Session cache timeout:

Renegotiation in byte:

X509 Certificate authentication: ☐

Client authentication: ☐

Network Connection Server (NCS) Settings

Primary NCS port:

(Note: Any modification of NRS Server configuration would not take effect until you restart all the services.)

★ Required value.

GOR; Reviewed:
SPOC 7/20/2011

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8 of 48
te30_sip_CS1KR7

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.

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 » System Wide Settings

System Wide Settings

SIP registration time to live timer: 3600 (30-3600 Seconds)

H.323 gatekeeper registration time to live timer: 30 (30-3600 Seconds)

H.323 alias name: dppsps *

Auto backup time: 23:49 (HH:MM)

Auto backup to secure FTP site enabled: ☐

Auto backup to secure FTP site's IP address:

Auto backup secure FTP site's path:

Auto backup secure FTP user name:

Auto backup secure FTP password:

* Required value.

Save Cancel

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.

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

Managing: ☐ Active database ☒ Standby database 172.18.20.12
[Numbering Plans » Domains » Service Domains](#)

Edit Service Domain

Domain name: dpp.nortel *

Domain description: DPP LAB

* Required value.

Save Cancel

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.

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: ☐ Active database 172.18.20.12
☒ Standby database [Numbering Plans » Domains » L1 Domain](#)

Edit L1 Domain (dpp.nortel)

Domain name: *

Domain description:

Endpoint authentication enabled:

Authentication password:

E.164 country code:

E.164 area code:

E.164 international dialing access code:

E.164 international dialing code length: (0-99)

E.164 national dialing access code:

E.164 national dialing code length: (0-99)

E.164 local (subscriber) dialing access code:

E.164 local (subscriber) dialing code length: (0-99)

Private L1 domain (UDP location) dialing access code:

* Required value

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.

Domain name: cdp

Domain description:

Endpoint authentication enabled: Not configured

Authentication password:

E.164 country code:

E.164 area code:

Private unqualified number label: PrivateUnknown

E.164 international dialing access code:

E.164 international dialing code length: (0-99)

E.164 national dialing access code:

E.164 national dialing code length: (0-99)

E.164 local (subscriber) dialing access code:

E.164 local (subscriber) dialing code length: (0-99)

Save Cancel

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)**.

Domains

Domains establish the basic structure of your converged network, defined by Service domains, L1 (UDP) and L0 (CDP) domains.

Service Domains (1) L1 Domains (UDP) (1) L0 Domains (CDP) (1)

Filter by Domain: All service domains / All L1 domains

Add Delete Refresh

ID	Description	# of Gateway Endpoints	# of Routing Entries	Context
cdp	dpp.nortel / udp	6	71	dpp.nortel / udp

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.

The screenshot displays the Nortel Network Routing Service Manager (NRS) interface. The left sidebar shows the navigation menu with 'Endpoints' highlighted. The main content area is titled 'Search for Endpoints'. It includes a search bar for 'Endpoint ID' and a section for 'Limit results to Domain' with three dropdown menus: 'dpp.nortel', 'udp', and 'cdp'. These dropdowns are highlighted with a red box. Below the search bar, there are tabs for 'Gateway Endpoints (6)' and 'User Endpoints (0)'. The 'Add...' button is highlighted with a red box. The interface also shows a 'Results per page' dropdown set to '50' and a 'Search' button.

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.

NORTEL NETWORK ROUTING SERVICE MANAGER Help | Logout

Managing: ☐ Active database 172.18.20.12
☒ Standby database [Numbering Plans > Endpoints > Gateway Endpoint](#)

Edit Gateway Endpoint (dpp.nortel / udp / cdp)

End point name: *

Description:

Trust Node: ☒

Tandem gateway endpoint name:

Endpoint authentication enabled:

Authentication password:

E.164 country code:

E.164 area code:

E.164 international dialing access code:

E.164 international dialing code length: (0-99)

E.164 national dialing access code:

E.164 national dialing code length: (0-99)

E.164 local (subscriber) dialing access code:

Private L1 domain (UDP location) dialing access code:

Private L1 domain (UDP location) dialing code length: (0-99)

Private Special number 1:

Private Special number 1 dialing code length: (0-31)

Private Special number 2:

Private Special number 2 dialing code length: (0-31)

Static endpoint address type:

Static endpoint address:

H.323 support:

SIP support:

SIP Mode: ☒ Proxy Mode ☐ Redirect Mode

SIP TCP transport enabled: ☐

SIP TCP port:

SIP UDP transport enabled: ☒

SIP UDP port:

SIP TLS transport enabled: ☐

SIP TLS port:

Persistent TCP support enabled: ☐

End to end security support: ☐

Network Connection Server enabled: ☐

Redundancy enabled:

Main endpoint name:

Redundant endpoint name:

★ Required value

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.

The screenshot displays the Nortel Network Routing Service Manager (NRS) interface. The left sidebar contains a navigation menu with categories like «UCM Network Services», System, Numbering Plans, Tools, and Routing Tests. The 'Routes' option under Numbering Plans is highlighted. The main content area is titled 'Search for Routing Entries' and includes a search form with fields for DN Prefix, DN Type, and Limit results to Domain. Below the search form, there are two tabs: 'Routing Entries (71)' and 'Default Routes (0)'. The 'Routing Entries' tab is active, showing a table of routing entries. The table has columns for DN Prefix, DN Type, Route Cost, SIP URI Phone Context, and Context. Three entries are visible, all with a Route Cost of 1 and a Context of 'dpp.nortel / udp / cdp / cores2'. The first entry has a DN Prefix of 2500, the second 2501, and the third 2600. The table is paginated, showing 1 of 50 of 71 Routing Entry(ies) on Page 1 of 2.

DN Prefix	DN Type	Route Cost	SIP URI Phone Context	Context
2500	Private level 0 regional (CDP steering code)	1	cdp.udp	dpp.nortel / udp / cdp / cores2
2501	Private level 0 regional (CDP steering code)	1	cdp.udp	dpp.nortel / udp / cdp / cores2
2600	Private level 0 regional (CDP steering code)	1	cdp.udp	dpp.nortel / udp / cdp / cores2

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.

The screenshot displays the 'Nortel NETWORK ROUTING SERVICE MANAGER' interface. On the left is a navigation menu with categories like 'System', 'Numbering Plans', and 'Tools'. The 'Routes' option under 'Numbering Plans' is highlighted with a red box. The main content area shows the 'Edit Routing Entry (dpp.nortel / udp / cdp / cores1)' form. A red box highlights the input fields: 'DN type' is set to 'Private level 0 regional (CDP steering code)', 'DN prefix' is '2025', and 'Route cost' is '1'. The 'Route cost' field has a '(1-255)' range indicator. At the bottom right are 'Save' and 'Cancel' buttons. A small note at the bottom left of the form area says '* Required value.'

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.

Node Details (ID: 2 - SIP Line, LTPS, PD, Gateway (SIPGw, H323Gw))

There is a warning message.

Node ID: 2 * (0-9999)

Call server IP address: 172.18.20.11 *

TLAN address type: ☒ IPv4 only
☐ IPv4 and IPv6

Embedded LAN (ELAN)

Gateway IP address: 172.18.20.1 *

Subnet mask: 255.255.255.128 *

Telephony LAN (TLAN)

Node IPv4 address: 47.166.92.207 *

Subnet mask: 255.255.255.224 *

Node IPv6 address: *

* Required Value.

Associated Signaling Servers & Cards

Warning: There are no more Signaling Servers available to add to the current node, please add more Signaling Servers to the parent CS1000 or CS1000-HS systems via Deployment Manager.

Select to add	Hostname	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
<input type="checkbox"/>	cores1	Signaling_Server	SIP Line, LTPS, Gateway, PD, Presence Publisher, IP Media Services	172.18.20.11	47.166.92.206	Leader

Show: ☐ IPv6 address

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**

The screenshot shows the Nortel CS 1000 Element Manager interface. The left sidebar contains a tree view with categories like UCM Network Services, Links, System, Interfaces, Customers, Routes and Trunks, and Dialing and Numbering Plans. The main content area is titled 'Node ID: 2 - Voice Gateway (VGW) and Codecs'. It has tabs for 'General', 'Voice Codecs', and 'Fax'. The 'General' tab is active, showing settings for Echo cancellation, Voice activity detection threshold, Idle noise level, and Signaling options. The 'Voice Codecs' section is highlighted with a red box, showing 'Codec G711' as 'Enabled (required)'. Below this, 'Voice payload size' is set to '20' (milliseconds per frame), and 'Voice playout (jitter buffer) delay' is set to '40' and '80' (milliseconds). At the bottom right, there are 'Save' and 'Cancel' buttons.

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**

NORTEL CS 1000 ELEMENT MANAGER

Managing: 172.18.20.11 Username: gerry
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

Node ID: 2 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services | H.323 Gateway Settings

Vtrk gateway application: ☒ Enable gateway service on this node

General

Vtrk gateway application: SIPGw and H.323Gw
SIP domain name: dpp.nortel
Local SIP port: 5060 * (1 - 65535)
Gateway endpoint name: cores1
Gateway password:
H.323 ID: cores1
Application node ID: 2 * (0-9999)
Enable failsafe NRS: ☐

Virtual Trunk Network Health Monitor

☒ Monitor IP addresses (listed below)
Information will be captured for the IP addresses listed below.
Monitor IP: Add
Monitor addresses: Remove

* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved.

Save Cancel

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.

The screenshot displays the Nortel CS 1000 Element Manager interface. The left sidebar shows a navigation tree with categories like UCM Network Services, Home, Links, System, Interfaces, Customers, Routes and Trunks, and Dialing and Numbering Plans. The main content area is titled 'Node ID: 2 - Virtual Trunk Gateway Configuration Details'. It includes a breadcrumb trail: 'System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration'. Below this, there are tabs for 'General', 'SIP Gateway Settings', 'SIP Gateway Services', and 'H.323 Gateway Settings'. The 'SIP Gateway Settings' tab is active, showing fields for 'TLS Security' (set to 'Security Disabled'), 'Port' (5061), 'Number of byte re-negotiation' (0), and 'Options' (Client authentication and X509 certificate authority). A red box highlights the 'Proxy Or Redirect Server' section, which contains 'Proxy Server Route 1' with 'Primary TLAN IP address' (47.166.92.198), 'Port' (5060), 'Transport protocol' (TCP), and 'Options' (Support registration checked). A note at the bottom states: 'Note: Changes made on this page will NOT be transmitted until the Node is also saved.' Buttons for 'Save' and 'Cancel' are at the bottom right.

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.

NORTEL CS 1000 ELEMENT MANAGER

Managing: 172.18.20.11 Username: gerry
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

Node ID: 2 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services | **H.323 Gateway Settings**

H.323 Gateway Settings

Primary gatekeeper (TLAN) IP address:	<input type="text" value="47.166.92.198"/>
Alternate gatekeeper (TLAN) IP address:	<input type="text" value="47.166.92.197"/>
Primary network connect server (TLAN) IP address:	<input type="text" value="47.166.92.198"/>
Primary network connect server port number:	<input type="text" value="16500"/> (1 - 65535)
Alternate network connect server (TLAN) IP address:	<input type="text" value="47.166.92.197"/>
Alternate network connect server port number:	<input type="text" value="16500"/> (1 - 65535)
Primary network connect server timeout:	<input type="text" value="10"/> (1 - 30)

* Required Value.

Note: Changes made on this page will NOT be transmitted until the Node is also saved.

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.

Input Description	Input Value
Zone Number (ZONE):	1
Intrazone Bandwidth (INTRA_BW):	1000000
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	1000000
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	VTRK (VTRK)
Description (ZDES):	

Submit Refresh Delete Cancel

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.

```
ADAN      DCH 79      Overlay 17 D-Channel Handler Configuration
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	ACOD 130	
TYPE: rdbCUST 00	TCPP NO	CPDC NO
ROUT 100	PII NO	DLTN NO
TYPE RDB	AUXP NO	HOLD 02 02 40
CUST 00	TARG	SEIZ 02 02
ROUT 66	CLEN 1	SVFL 02 02
DES VIR_TRK	BILN NO	DRNG NO
TKTP TIE	OABS	CDR NO
NPID_TBL_NUM 0	INST	NATL YES
ESN NO	IDC NO	SSL
RPA NO	DCNO 10	CFWR NO
CNVT NO	NDNO 10 *	IDOP NO
SAT NO	DEXT NO	VRAT NO
RCLS EXT	DNAM NO	MUS YES
VTRK YES	SIGO STD	MRT 21
ZONE 1	STYP SDAT	PANS YES
PCID SIP	MFC NO	RACD NO
CRID NO	ICIS YES	MANO NO
NODE 2	OGIS YES	FRL 0 0
DTRK NO	TIMR ICF 1920	FRL 1 0
ISDN YES	OGF 1920	FRL 2 0
MODE ISLD	EOD 13952	FRL 3 0
DCH 79	LCT 256	FRL 4 0
IFC SL1	DSI 34944	FRL 5 0
PNI 00001	NRD 10112	FRL 6 0
NCNA YES	DDL 70	FRL 7 0
NCRD YES	ODT 4096	OHQ NO
TRO NO	RGV 640	OHQT 00
FALT NO	GTO 896	CBQ NO
CTYP UKWN	GTI 896	AUTH NO
INAC NO	SFB 3	TTBL 0
ISAR NO	PRPS 800	ATAN NO
DAPC NO	NBS 2048	OHTD NO
MBXR NO	NBL 4096	PLEV 2
MBXOT NPA	IENB 5	OPR NO
MBXT 0	TFD 0	ALRM NO
PTYP ATT	VSS 0	ART 0
CNDP UKWN	VGD 6	PECL NO
AUTO NO	EESD 1024	DCTI 0
DNIS NO	SST 5 0	TIDY 1600 100
DCDR NO	DTD NO	ATRR NO
ICOG IAO	SCDT NO	TRRL NO
SRCH LIN	2 DT NO	SGRP 0
TRMB YES	NEDC ORG	ARDN NO
STEP	FEDC ORG	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.

<pre> 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 </pre>	<pre> 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 </pre>
--	--

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	SPN 90	SPN 2	SPN 15
FLEN 3	FLEN 7	FLEN 7	FLEN 3
ITOH NO	ITOH NO	ITOH NO	ITOH NO
CLTP NONE	CLTP NONE	CLTP NONE	CLTP NONE
RLI 24	RLI 24	RLI 24	RLI 24
SDRR NONE	SDRR NONE	SDRR NONE	SDRR NONE
ITEI NONE	ITEI NONE	ITEI 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 RCBF
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 OCBF FLXD FTTC DNDY DNO3 MCBN
FDSD NOVD VOLA VOUD CDMR PRED RECA MCDD T87D SBMD KEM3 MSNV FRA PKCH MUTA MWTD

---continued on next page---
```

---continued from previous page----

```
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 RCBF
    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 CCBF 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
---continued on next page---
```

---continued from previous page----

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 CCBF BNRD OCBF 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.

The screenshot displays the Network Routing Service Manager (NRSM) web interface. On the left, a navigation menu lists various system components, with 'Endpoints' highlighted. The main panel shows the 'Managing' status as 'Standby database' and a search area for endpoints. A search box is present, and below it, the 'Limit results to Domain' section features three dropdown menus configured with 'dpp.nortel', 'udp', and 'cdp'. At the bottom of this section, the 'Add...' button is highlighted with a red rectangle, indicating the next step in the configuration process.

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.

NORTEL NETWORK ROUTING SERVICE MANAGER Help | Logout

Managing: ☐ Active database 172.18.20.12
☒ Standby database [Numbering Plans > Endpoints > Gateway Endpoint](#)

Edit Gateway Endpoint (dpp.nortel / udp / cdp)

End point name: *

Description:

Trust Mode: ☐

Tandem gateway endpoint name:

Endpoint authentication enabled:

Authentication password:

E.164 country code:

E.164 area code:

E.164 international dialing access code:

E.164 international dialing code length:

E.164 national dialing access code:

E.164 local (subscriber) dialing code length:

Private L1 domain (UDP location) dialing access code:

Private L1 domain (UDP location) dialing code length:

Private Special number 1:

Private Special number 1 dialing code length:

Private Special number 2:

Private Special number 2 dialing code length:

Static endpoint address type:

Static endpoint address:

H.323 support:

SIP support:

SIP Mode: ☒ Proxy Mode ☐ Redirect Mode

SIP TCP transport enabled: ☐

SIP TCP port:

SIP UDP transport enabled: ☒

SIP UDP port:

SIP TLS transport enabled: ☐

SIP TLS port:

Persistent TCP support enabled: ☐

End to end security support: ☐

Network Connection Server enabled: ☐

Redundancy enabled:

Main endpoint name:

Redundant endpoint name:

* Required value

Save Cancel

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.

«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: ☐ Active database 172.18.20.12
☒ Standby database [Numbering Plans » Routes » Routing Entry](#)

Edit Routing Entry (dpp.nortel / udp / cdp / TRIO)

DN type: Private level 0 regional (CDP steering code) ▼

DN prefix: 5001 *

Route cost: 1 * (1-255)

* Required value.

Save Cancel

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

«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: ☐ Active database 172.18.20.12
☒ Standby database [Numbering Plans » Routes](#)

Search for Routing Entries [Hide](#)

Enter a DnPrefix and Dn Type (use * for all) and click Search. You may narrow the search by specifying a particular domain.

DN Prefix: * DN Type: All DN Types ▼

Limit results to Domain: dpp.nortel / udp / cdp ▼

Endpoint Name: TRIO ▼

Results per page: 50 Search

Routing Entries (4) **Default Routes (0)**

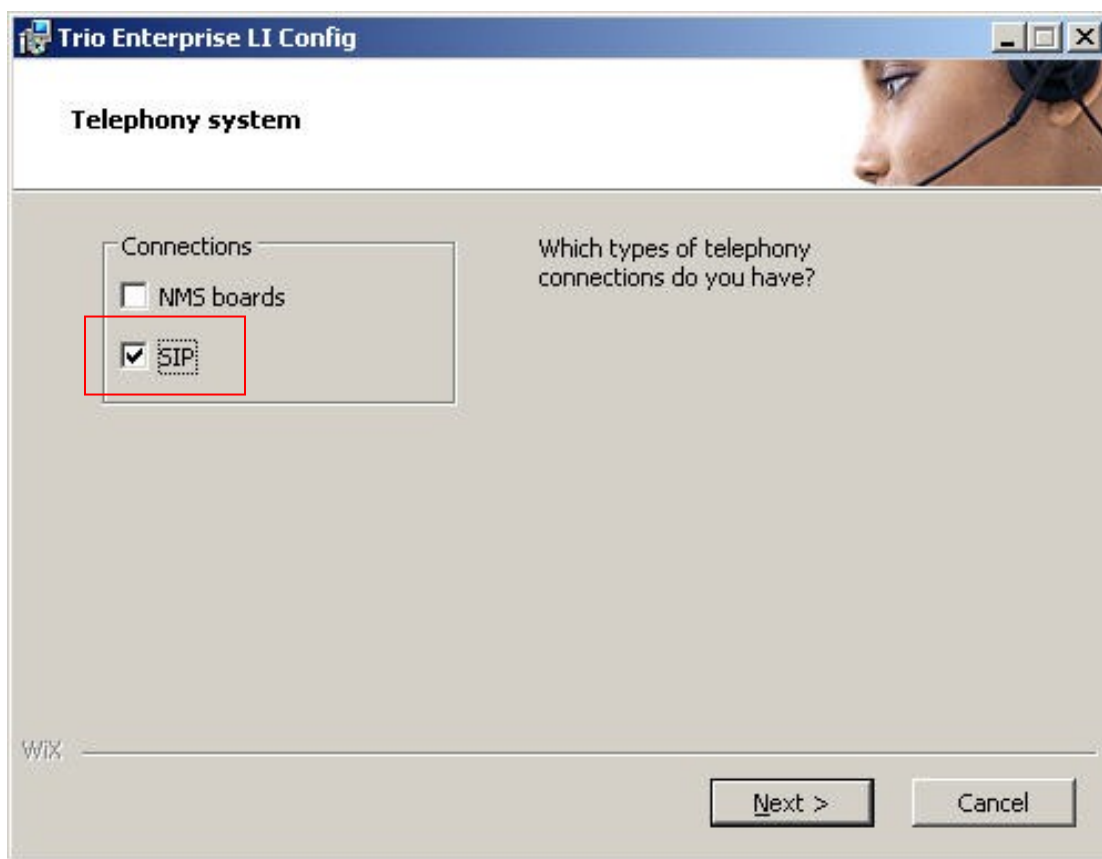
Add... Copy... Move... Import... Export... Routing test... Delete Refresh

	DN Prefix	DN Type	Route Cost	SIP URI Phone Context	Context
1	5001	Private level 0 regional (CDP steering code)	1	cdp.udp	dpp.nortel / udp / cdp / TRIO
2	5002	Private level 0 regional (CDP steering code)	1	cdp.udp	dpp.nortel / udp / cdp / TRIO
3	5003	Private level 0 regional (CDP steering code)	1	cdp.udp	dpp.nortel / udp / cdp / TRIO
4	5004	Private level 0 regional (CDP steering code)	1	cdp.udp	dpp.nortel / udp / cdp / TRIO

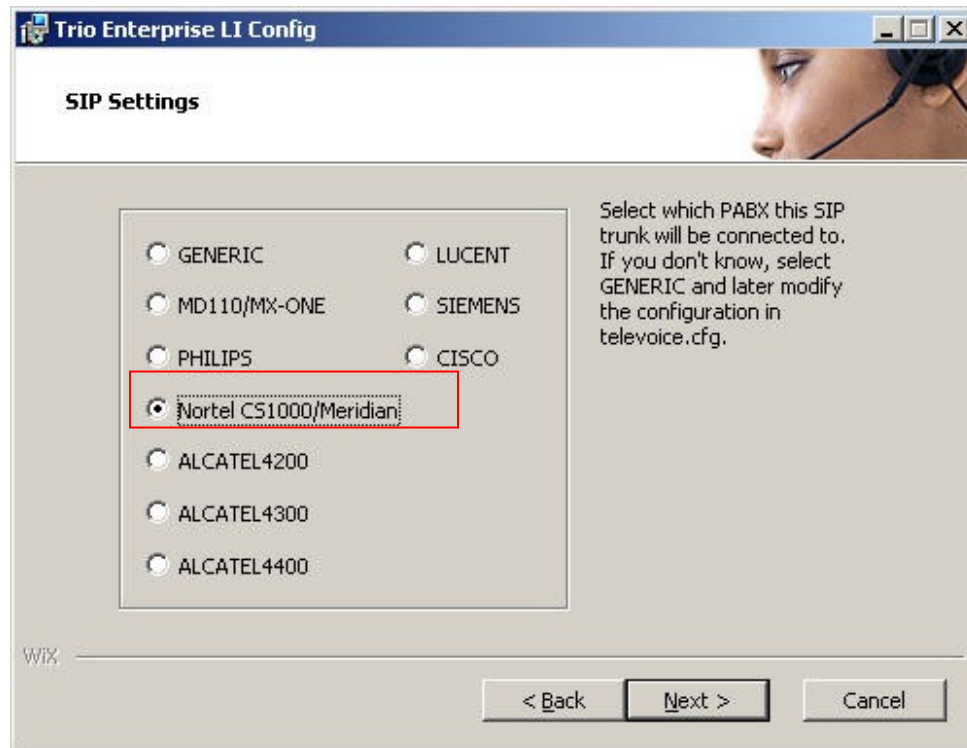
1 - 4 of 4 Routing Entry(ies) Page 1 of 1 First Previous Next Last

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.



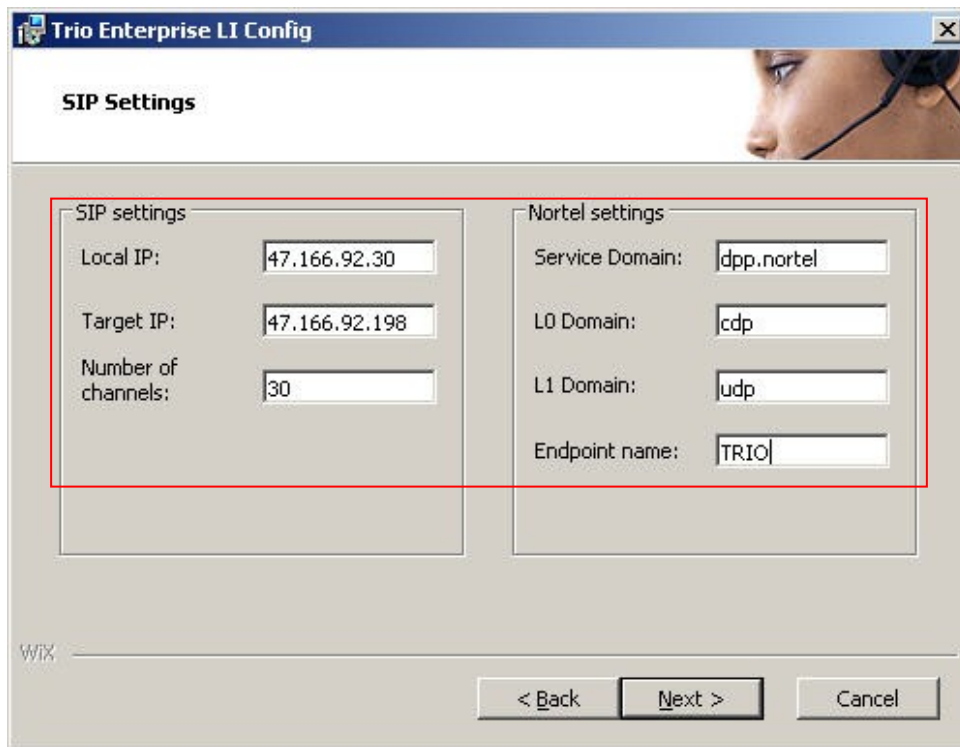
In the next screenshot (first page of SIP Settings), ensure the **Nortel CS1000/Meridian** radio button entry is selected. Click on **Next** when ready.



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

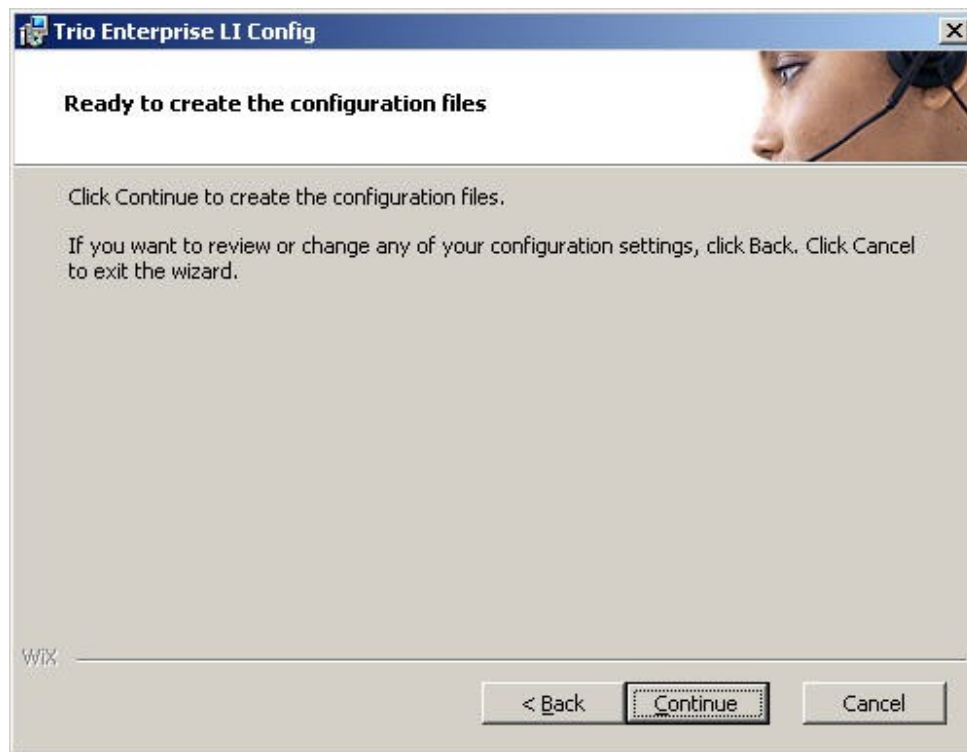
Click on **Next** when ready.



The screenshot shows the 'Trio Enterprise LI Config' window with the 'SIP Settings' tab selected. The window is divided into two main sections: 'SIP settings' and 'Nortel settings'. The 'SIP settings' section includes fields for 'Local IP' (47.166.92.30), 'Target IP' (47.166.92.198), and 'Number of channels' (30). The 'Nortel settings' section includes fields for 'Service Domain' (dpp.nortel), 'L0 Domain' (cdp), 'L1 Domain' (udp), and 'Endpoint name' (TRIO). A red rectangle highlights the 'SIP settings' and 'Nortel settings' sections. At the bottom of the window, there are three buttons: '< Back', 'Next >', and 'Cancel'. The 'Next >' button is highlighted.

Section	Field	Value
SIP settings	Local IP:	47.166.92.30
	Target IP:	47.166.92.198
	Number of channels:	30
Nortel settings	Service Domain:	dpp.nortel
	L0 Domain:	cdp
	L1 Domain:	udp
	Endpoint name:	TRIO

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



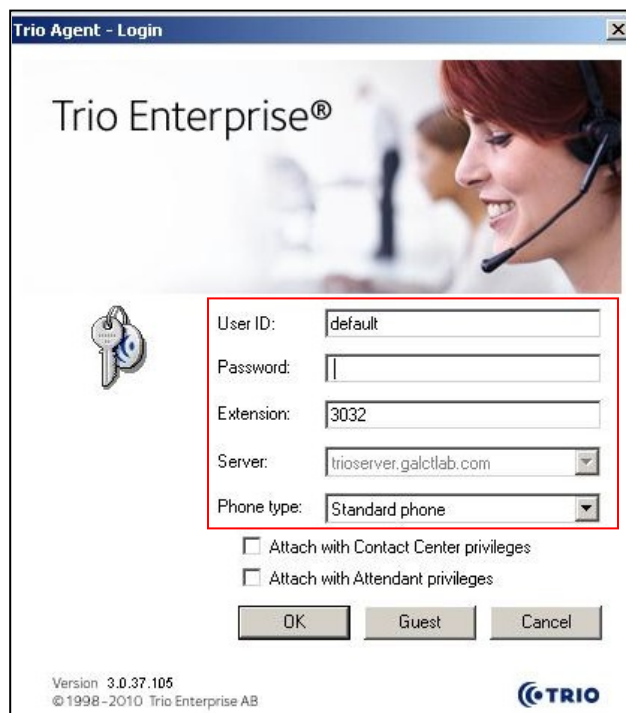
If the configuration wizard completes successfully, the following screen appears. Ensure the **Start TeleVoice service when finished** option is checked. Click on the **Finish** button to complete the configuration procedure. This completes the Trio Enterprise server side configuration steps. The Trio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



Trio Agent - Login

Trio Enterprise®

User ID: default

Password:

Extension: 3032

Server: trioserver.galctlab.com

Phone type: Standard phone

☐ Attach with Contact Center privileges

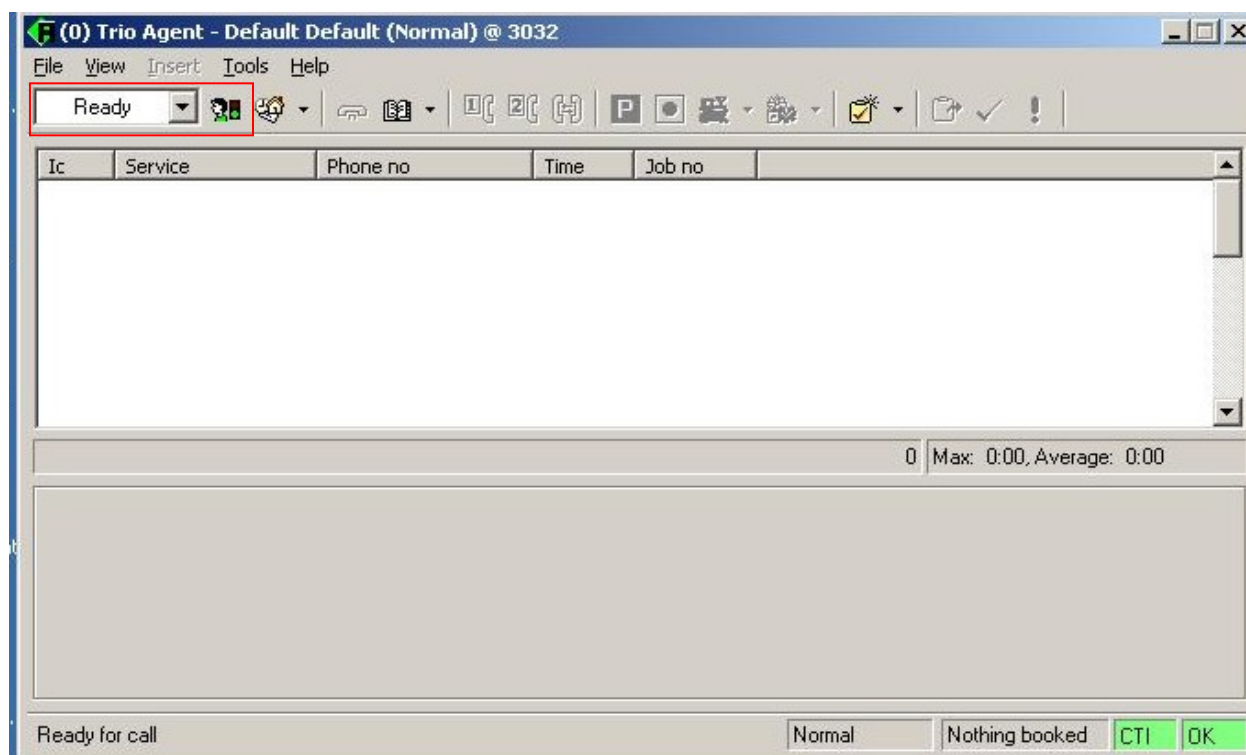
☐ Attach with Attendant privileges

OK Guest Cancel

Version 3.0.37.105
© 1998-2010 Trio Enterprise AB

TRIO

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.

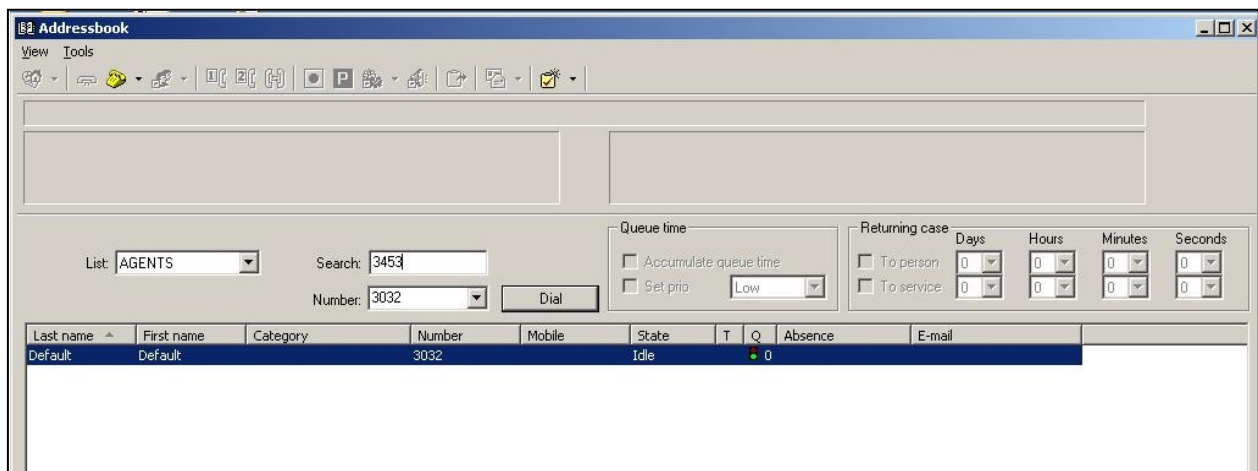
8. Verification Steps

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 5001 and 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	
c ppmUtil	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	
ftpkg	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.nt1

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	
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

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