

Avaya Solution & Interoperability Test Lab

Application notes for Colt Communication System with AvayaTM Communication Server 1000 release 6.0

Abstract

These Application Notes describe a solution comprised of Avaya[™] Communication Server 1000E Release 6.0 and Colt Communication SIP Trunk Product. The Primary focus of testing is the system verification of SIP trunk interoperability which includes the call scenario such as basic call, call forward (all calls, busy, no answer), call transfer (blind and consult) and conference. Calls should be placed in both directions and should involve various set types

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

This document provides a typical network deployment of Communication Server 1000 (CS1000) utilizing the Colt Communication SIP Trunking product offering. This document should serve as general guideline only, since it is not possible to document every possible variation of configuration. Further information may be obtained from your Nortel support representative. The CS1000E system is configured as a SIP gateway endpoint on the Colt Communication network. The enterprise customer will require an additional signaling server for each SIP gateway that will be deployed as SIP trunking to the carrier. In the diagram shown below, the signaling server is shown as the onboard CP-PM option, but it can also be the outboard, rack-mounted 1U server.

The CS1000, in this configuration, does not use SIP Redirect or Proxy for Carrier SIP trunking, the SIP Virtual Gateway is simply provisioned with the SBC as the static SIP endpoint of the SIP Trunk..

1.1. Interoperability Compliance Testing

System verification testing of SIP Trunking between CS1000 Rel. 6.0 and Carrier switch

- o General call processing between systems including:
 - Codec/ptime negotiation and transcoding (G.711 a-law and G.729 verification / 20ms)
 - Hold/Retrieve on both ends
 - CLID displayed
 - Ringback tone
 - Speech path
 - Dialing plan support
 - Advanced features (Call on Mute, Call Park, Call Waiting, use Feature Access Code)
 - Abandoned Call
- Call redirection verification: all supported methods (blind transfer, consultative transfer, call forward, and conference) including CLID. Call redirection is performed from both ends
- o FAX T38
- DTMF on both direction
- SIP Transport UDP
- Thru dialing via PBX Call Pilot
- Voice Mail Server (hosted on Nortel system)
- Early Media Transmission
- Inter-office tandem Call

1.2. Caveats

• The Fax/Modem pass through feature provides a modem pass through allowed (MPTA) class of service (CLS) for an analog phone TN. MPTA CLS dedicates an analog phone TN to a

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modem or a Fax machine terminal. A connection that initiates from the dedicated TN, and/or calls that terminate at the dedicated TN through a Digital Signal Processor (DSP), use a G711 NO VAD codec on the Call Server. To ensure proper functioning of the MPTA CLS, the Enable Modem/Fax pass through mode check box must be selected in the Gateways section of Element Manager. This check box is selected by default in Element Manager.

• The packet interval for G.711 codec is set to 20 ms in MPT. The maximum speed supported for modem and fax is 33.6 Kb/s. This limit is imposed by the analogue line card. When MPTA CLS is configured on a TN, the T.38 protocol is no longer supported for that particular TN.

1.3. Dependencies

. CS1000 R6.0 software and implementation of latest patches

. Colt Communications provides support to setup, configure, and troubleshoot on carrier switch for the duration of the testing.

1.4. Support

For technical support on Colt Communication system, please contact Colt technical support at:

• http://www.colt.net/UK-en/ContactUs/index.htm

2. Reference Configuration

Figure 1 illustrates the test configuration used during the compliant testing event between the Communication Server 1000E and Colt Communication System. This configuration is for a single Communication Server1000E deployment



Figure 1- Nettwork diagram for Nortel-Colt LAB setup

Figure 2 is the deployment option for 2 or more of Communication Server 1000E with the Colt communication system.

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Figure 2 - Network topology for Multi-System configuration for Tandem Calls

The following assumptions were made for this lab test configuration:

- 1. CS1000 R6.0 software and implementation of latest patches
- 2. Colt Communications provides support to setup, configure, and troubleshoot on carrier switch for the duration of the testing.

All test scenarios involving the establishment of calls will assume the following activities:

- 1. Calls will be checked for the correct call progress tones and cadences.
- 2. During the ringing state the ring back tone and destination ringing will be checked.
- 3. Calls will be checked in both hands-free and handset mode due to internal Nortel requirement.
- 4. Calls will be checked for speech path in both directions using spoken words to ensure clarity of speech.
- 5. The display(s) of the sets/clients involved will be checked for consistent and expected CLID, name and redirection information both prior to answer and after call establishment.
- 6. The speech path and messaging system will be observed for timely and quality End to End tone audio path generation and application responses.
- 7. The call server maintenance terminal window will be open during the test cases execution for the monitoring of BUG(s), ERR and AUD messages.
- 8. Speech path and display checked before and after call s are put on/off hold from each end.
- 9. Applicable of files will be screened on an hourly basis during the testing for message that may indicate technical issues. This refers to Nortel PBX files.
- 10. Calls will be checked to ensure that all resources such as Virtual trunks, TDM trunks, Sets and VGWs are released when a call scenario ends

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3. Equipment and Software Validated

Additional software and patch lineup for the configuration is as follows:

Patch ID	Issue	Title	Notes
		Ringback tone and speech path support in slow start CFNA	
MPLR28415	1	scenarios (activates feature in vtrk SU listed below)	
		Delete element removes all elements-services mapping of	
MPLR28774	1	associateroles	
		Unable to access overlays on inactive core when in split mode	
MPLR28797	1	with UCM	
MPLR27408	1	SIP: Disable SIP Session Timer on CS1K.	
MPLR25946	1	SIP GW patch to remove outbound MCDN from SIP messaging	
MPLR22968	1	Replace domain population in the FROM field	
MPLR25529	1	PI: SIP: Partial support of DIVERSION	
		Mandatory parameter "T38FaxRateManagement" isn't present in	
MPLR27159	1	T38 SDP	
nortel-			
cs1000-vtrk-			
6.00.18.065-			
024.i386.001	1	nortel-cs1000-vtrk-6.00.18.065-0124.i386.001	

Call Server: 6.00R plus latest DEPLIST **Signaling Server:** SSE 6.00.18 plus latest DEPLIST

Hardware system requirement and theirs soft/loadware version

System	Software/Loadware Version
Nortel CS1000E 6.0 (CPPM)	• Call Server: 6.00R
	• Signaling Server: 6.00.18
Nortel phones	• 2002 p2: 0604DCJ (Unistim)
	• 2004 p2: 0604DCJ (Unistim)
	• 1140: 0625C6O (Unistim)
	• 1120: 0624C6O (Unistim)
	• 2007: 0621C6M (Unistim)
	• 1220: 062AC6O (Unistim)
	• SIP 1140 i00v142
	• SIP 1120
	 SMC3456: Version 2.6 - RC14 build
	53715
Sonus PSX	• 7.2.4R1
Sonus GSX	• 7.2.4R1
Sonus DSI	• 7.2.3R1
Sonus EMS	• 7.2.4R0
	\bullet

4. Configure the Avaya Communication Server 1000E

4.1. Element Manager Configuration

4.1.1. Configure IP in CS1000 network

This section describes the steps for creating Node ID (1000) in CS 1000 network. Enter Element Manager through the IE browser (in IE address bar, type IP address of the Node IP or TLAN of Signalling Server).

- Input Node ID and press Save...
- Enter TLAN, ELAN IP address of Signalling Server.

Node 1000 was added to be configured as the SIP gateway to the carrier services.

NØRTEL	CS 1000 ELEMENT N	IANAGER			Help
- UCM Network Services - Home - Links - Virtual Terminals	System » IP Network » IP Teleph IP Telephony Nodes Click the Node ID to view or edit its propertie	ony Nodes			
+ Alarms - Maintenance	Add Import Export Delete			Print I	Refresh
+ Core Equipment - Peripheral Equipment	I <u>Node ID</u> ▲ Components	Enabled Applications	ELAN IP	TLAN IP 102 169 10 10	Status Supebranized
- IP Network	Show: V Nodes C Component Se	cvers and Cards	-	192.100.10.10	Synchronized
Maintenance and Reports Media Gateways Zones Host and Route Tables Network Address Translation OoS Thresholds Personal Directories Unicode Name Directory Interfaces Engineered Values Emergency Services Geographic Redundancy Software Customers Routes and Trunks Routes and Trunks DoChannels Digital Trunk Interface Dialing and Numbering Plans Electronic Switched Network Flexible Code Restriction Incoming Digit Translation					

Figure 3 – Adding a node

Figure 4 describes the Call server IP configuration:

The MILE	CO TOOD ELET		VEN			
- UCM Network Services	Sustan - ID Matu	ark D Telephony Neder	- Node Details			
- Home	System » IP Netw	OCO L TDO DD (» Node Details			
Links - Uitual Terminals - Virtual Terminals - Alarms - Maintenance Core Equipment - Peripheral Equipment - IP Network - Nodes: Servers, Media Cards - Media Gateways - Zones - Host and Route Tables - Network Address Translation - QoS Thresholds - Personal Directories - Unicode Name Directory + Interfaces - Engineered Values Emergency Services + Geographic Redundancy + Software - Customers - Routes and Trunks	Node Details (ID. 1 Node ID: Call Server IP Address: Telephony LAN (TLAN) Node IP Address: Subnet Mask: IP Telephony Node I	1000 - L TP 3, PD, C 192.168.10.5 192.168.10.10 255.255.255.0 PropertiesApplicat	Embedded LAN Gateway IP add Subnet Mask:	I (ELAN) Iress: 192.168.100. 255.255.255. ation)	1 <u> </u>	
	<u>Voice Gatewa</u> <u>Quality of Sen</u> * Required Value	v (VGW) and Cod vice (QoS)	ecs	• Terminal Provu S	anvar (TPS)	1 Ca
	Associated Signali	ng Servers & Car Remove Make Lead	rds er		Print E	<u>lefresh</u>
	I ☐ Hostname ▲	Type	Deployed Applications	ELAN IP	TLAN IP	Role
	nd1-car1	Signaling Server	LTPS, Gateway, PD	192.168.100.149	192.168.10.245	Leader
- Routes and Trunks - D-Channels - Digital Trunk Interface - Dialing and Numbering Plans	Note: Only server(s) that are available in the servers list .	e not part of any other IP te	lephony node and deployed applicat	ion(s) that match the servic	ce(s) selected for this i	iode are

Figure 4 - Call Server IP Configuration

Since the carrier does not support TLS or sRTP, we have them disable in our CS1K configuration. For the primary proxy enter the IP address of the Session Border Controller (SBC). Use UDP SIP transport, port 5060 for SIP communication. The NRS is not enabled as all calls are routed by the SBC.

M Notwork Sopricos		
me	System » IP Network » IP Telephony Nodes » Nod	e Details » Virtual Truck Gateway Configuration
iks	Nede ID: 1000 Virtual Trunk Cateway	Configuration Dataila
/irtual Terminals	Node ID. 1000 - Virtual Hulik Galeway C	computation Details
stem	General SIP Gateway Settings	SIP Gateway Services
arms	Vtrk Gateway Application: 🗹 Enable gateway service	ce on this Node
aintenance		
pre Equipment	General	Virtual Trunk Network Health Monitor
eripheral Equipment		
Network		
Maintenance and Reports	Vtrk Gateway Application: SIP Gateway (SIPGw)	
Media Gateways	OID Development 243 423 424 425	Information will be captured for the IP addresses listed below.
Zones	SIP Domain name: 213.123.124.123	Monitor IP
Host and Route Tables	Local SID Part 5060	526)
Network Address Translation	Local SIP Polt. 19000 [1-05	Monitor addresses
QoS Thresholds	Gateway endpoint name: 192.168.10.10	
Personal Directories	Gatoway enapoint name.	
terfaces	Gateway password: *	
igineered Values		Reinuve
mergency Services	Enable failsafe NRS: 🗖	
eographic Redundancy		
oftware		
tomers	SIP Gateway Settings	
ites and Trunks	Sir Galeway Settings	
outes and Trunks	TLS Security: Security Disabled	
-Channels	Port: 5061 (1 - 65535)	
ing and Numbering Plans		
ectronic Switched Network	Number of Byte Re-negotiation: 0	
exible Code Restriction	Options: Client Authentication	
coming Digit Translation	X509 certificate authority	
nes	Provy Or Redirect Server	
emplates		Cocondent TLANUD
eports	Primary TLAN IP Address: 213.123.124.125	Address: 0.0.0
gration	Dettor	Address.
s	POR. 5060 (1 - 65535)	Port: 5060 (1 - 65535)
ackup and Restore	Transport protocol: UDP	
all Server Initialization		Transport protocol: TCP
ate and Time	Options: Support registration	Options: Support registration
ogs and reports	Primary CDS Proxy	Secondary CDS Proxy
a series		

Figure 5 – Virtual Trunk Gateway configuration

4.1.2. Configure Voice Codec for Nortel IP Phone

This section describes the steps for administering a set of codecs in CS1000. This set of codecs is used in IP network for communication between Nortel IP Phones.

- Access EM by IE browser.
- Choose "IP Network", then choose "Nodes: Servers, Media Cards", select proper Node and press "Edit".

Figure 6 and 7 are showing how to change Codec profile for IP Phone, select "VGW and IP phone codec profile".

Uncheck Modem FAX pass through mode and check V.21. Codec T38 FAX does support on Colt. TN of sets with class of service = MPTD (Modem Pass Through Denied)

NORTEL	CS 1000 ELEN	MENTMANAGER		Help
- Maintenance and Reports	Managing: L	Username: ork » <u>IP Telephony Nodes</u> » <u>Node Details</u> »	VGW and Codecs	
- Host and Route Tables	Node ID: 1000 - Voi	ice Gateway (VGW) and Coo	decs	
- Network Address Translation	General	Voice Codecs		I <u>Fax</u>
OoS Thresholds OoS Thresholds Os Th	Echo Cancellation: VU V Dynamic attenua Voice Activity Detection Idle Noise Level: Signaling Options: VD Lc Voice Codecs Codec G711: Enable Voice Playout (jitter but Maximum delay may b Voice Activity det * Required Value.	Ise canceller, with tail delay: 128 ttion n Threshold: -17 -65 TMF Tone Detection w latency mode emove DTMF delay (squelch DTMF lodem/Fax pass-through .21 Fax Tone Detection d (required) (milliseconds per frame) ffer) delay: 40 80 (millise Nominal Maximum re automatically adjusted based on I tection (VAD) Note: Changes made on this	(-20 - +10 DBM) (-327 - +327 DBM) from TDM to IP) econds) Nominal settings.	Save C

Figure 6 – Voice Gateway and Codec settings

Figure 7 shows how to configure the Voice gateway and IP phone codec settings. The Colt Communication network supports both G.711 and G.729. The packet size is set to 20 to match the network also.

NØRTEL	CS 1000 ELEMENT MANAGER	Help ₌ogout
Maintenance and Reports Media Gateways Zones Host and Route Tables	Managing: 47.248.100.147 Username: admin System » IP Network » I <u>P Telephony Nodes</u> » <u>Node Details</u> » VGW and Codecs Node ID: 1000 - Voice Gateway (VGW) and Codecs	
 Network Address Translation QoS Thresholds Personal Directories Unicode Name Directory Interfaces Engineered Values Emergency Services Geographic Redundancy Software Customers Routes and Trunks Routes and Trunks Digital Trunk Interface Digital Trunk Interface Digital Trunk Interface Digital Code Restriction Incoming Digit Translation Phones Templates Reports Higration Tools Backup and Restore Call Server Initialization Incoming Digit Translation Phones Templates Reports Properties Migration Tools Backup and Restore Call Server Initialization Date and Time Logs and reports Security Passwords Policies Looin Options 	Voice activity detection (VAD) Codec G729: Finabled Voice payload size: 20 (milliseconds per frame) Voice Playout (jitter buffer) delay: 40 (milliseconds) Norminal Maximum Maximum delay may be automatically adjusted based on Nominal settings. Voice activity detection (VAD) Codec G723.1. Finabled Voice payload size: 30 (milliseconds per frame) Voice Playout (jitter buffer) delay: 60 (milliseconds) Norminal Maximum Maximum delay may be automatically adjusted based on Nominal settings. Codic G723.1. Enabled Voice Playout (jitter buffer) delay: 60 (milliseconds) Norminal Maximum Maximum delay may be automatically adjusted based on Nominal settings. Coding rate: 5.3 (kbps) Fax Codec name: Codec name: 138 FAX Maximum rate: 14400 (bps) Fax TCF method: 2 (0 (0 - 300 milliseconds)) FAX No Activity Timeout: 20 (10 - 32000 milliseconds) Packet size: 30 (bps) * Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved.	I Eax

rigure *i* – voice Galeway and codec settings

4.1.3. Configure Voice Codec for Media Gateways

This section describes the steps for administering a set of codecs in CS1000. This set of codec is used in IP network for communication through Media gateways.

- Access EM by IE browser.
- Choose "IP Network", then choose "Media gateways", select proper voice gateways •
- To change Codec profile for IP Phone, select "VGW and IP phone codec profile". •

Figure 8 shows how to configure the Voice Gateway and IP phone codec profile Uncheked Modem FAX pass through mode

TN of sets with class of service = MPTD (Modem Pass Through Denied) Voice gateway and IP phone codec settings.



Figure 8 – Voice Gateway and IP phone codec profile settings

4.1.4. Configure Quality of Service

This section describes the steps for administering QoS in CS1000.

- Access EM by IE browser
- Choose "IP Network", then choose "Nodes: Servers, Media Cards", select proper Node and press "Edit".
- To change Quality of Service, select "QoS".

The default Diffserv values are correct in figure 9.

NØRTEL	CS 1000 ELEMENT MANAGER	Help
- UCM Network Services - Home - Links Victual Terminals	System » IP Network » IP Telephony Nodes » Node Details » Quality of Service (QoS) Node ID: 1000 - Quality of Service (QoS)	
System Alarms Alarms Maintenance Core Equipment Peripheral Equipment Peripheral Equipment Peripheral Equipment Nodes Servers Media Cards Maintenance and Reports Media Gateways Zones Host and Route Tables Network Address Translation QoS Thresholds Personal Directories Unicode Name Directory Interfaces Application Module Link Value Added Server Property Management System Engineered Values Emergency Services	Diffserv Codepoint (DSCP) Enable Nortel Automatic QoS: □ Control Packets: 40 (0-63) Voice Packets: 46 (0-63) VLAN Tagging: □ 802. 1Q Support 802. 1Q Bits Value (802. 1P): 6 (0-7)	
+ Software - Customers - Routes and Trunks - Routes and Trunks - D-Channels - Digital Trunk Interface - Digital Trunk Interface - Dialing and Numbering Plans - Electronic Switched Network	* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved.	Save Can
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Figure 9 – Quality of Service settings.

4.1.5. Configure SIP URI

This section describes the steps for administering SIP URI configuration in CS1000.

- Access EM by IE browser
- Choose "IP Network", then choose "Nodes: Servers, Media Cards", select proper Node and press "Edit".
- To change SIP URI, select "SIP URI Map".

In figure 10, leave the SIP URI fields blank for E.164.

NØRTEL	CS 1000 I	ELEMENT MANA	GER			Help
- UCM Network Services	Managing: 47.248.1	100.147 Username: admin				
- Home	System	» IP Network » IP Telephony Nodes :	Node Details » Virtual Trunk	Gateway Con	figuration	
- Links	Node ID: 100	0 - Virtual Trunk Gatew	ay Configuration De	etails		
- Virtual Terminals	General	SIP Gateway Settings		SIP G	ateway Services	
- System	SIP LIRI Mar).				
- Maintenance	on oramap	-			Driveto Domain Namao	
+ Core Equipment		Public E 164 Domain Names	1000		Private Domain Marries	1
- Peripheral Equipment	National	-	- UDP:		ludp	
- IP Network	Nutronal.		- CDP:		cdp.udp	
- Maintenance and Reports	Subscriber:		Specia	al number	PrivateSpecial	1
- Media Gateways	Special num	ber: PublicSpecial				1
- Zones	Unknown	PublicUnknown	- Vacan	t number:	PrivateUnknown	1
- Host and Route Tables	Onitriown.	p doicemment	Unkno	wn:	UnknownUnknown	
- Network Address Translation	010.0					
- QOS Inresnoids	SIP Gateway Services					
- Unicode Name Directory	SIP Converged Desktop: C Enable CD service					
- Interfaces	Service DN:			Used for n	naking VTRK call from agent.	
- Application Module Link	Converged to	elephone call forward DN				
- Value Added Server	DANasuta fa	A second call forward Div.			Contract Lanex	
- Property Management System-	RANTOULETC	announce.		(route num	iber 0 - 511)	
+ Emergency Services	Wait time be	fore RAN queue:	1	(-1 - 3276	7 msec)	
+ Geographic Redundancy	Timeout for ringing indication:		10 (5 - 60 s		seconds)	
+ Software	Timeout for (CD server	5	(1 - 30 sec	conds)	
- Customers		25	r Ia	=		
- Routes and Trunks	Note: Changes made on this page will NOT be transmitted until the Node is also saved.				Save Car	
- Routes and Trunks	requires value.					
- Digital Trunk Interface						
- Dialing and Numbering Plans						
- Electronic Switched Network 🗾						
(4)						

Figure 10 – SIP Gateway Services Settings

4.1.6. Configure Zones and Bandwidth Management

This section describes the steps for administering Zone configuration in CS1000.

- Access EM by IE browser
- Choose "IP Network", then choose "Zones", select proper "Zone Basic Property and Bandwidth Management"

Figure 11 shows how to configure a zone for IP sets and bandwidth management. If it does not already exist, create a zone for IP sets. The bandwidth strategy can be adjusted to preference.

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Figure 12 shows how to configure a zone for new created SIP trunks.





Figure 12 – Zone Basic Property Settings for (virtual) SIP trunk

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4.1.7. Configure SIP trunk

This section describes the steps for establishing a SIP connection between CS 1000 switch and Carrier system.

1. Create D-channel (DCH)

- Launch Element Manager of CS 1000 6.0
- Choose D-Channels, enter D-channel number (i.e.: 100), select DCH for type

Click Add to create DCH 100



Figure 13 – D-Chanel Configurations

Also click on Basic Options and edit the Remote Capabilities (RCAP). Enable MWI if CS1K hosted voice mail will be used.

2. Create route: Create route 100 using DCH 100 for SIP trunks with figures 14 and 15



Figure 14 – Route Property Configuration



Figure 15 – Route Property Configuration (Cont..)

3. Create trunk: To create trunk using basic configuration in figure 16

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Figure 16 – Basic Trunk Configuration

Disable Media Security (sRTP) at the trunk level using figure 17 by editing the Class of Service (CLS) at the bottom basic trunk configuration page show in figure 16.



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Figure 17 – Class of Service

4. Create Special Number List:

a) Launch Element Manager of CS 1000 6.0

b) Select "Dialing and Numbering plans \rightarrow Electronic Switched Network \rightarrow Number Plan (Net) \rightarrow Access Code 1 (2) \rightarrow Special Number (SPN).

Create special number list for outgoing dialing plan using figure 18



Figure 18 – Special Number List

Create special number 001 (use RLI_11) to dial to North America in figure 19

NØRTEL	CS 1000 ELEMENT MANAGER	Help
- Maintenance + Core Equipment - Peripheral Equipment + IP Network	Special Number	
+ Interfaces - Engineered Values	Input Description	Input Value
Emergency Services Emergency Services Software Customers Routes and Trunks D-Channels Diction Took Indefedee	Special Number translation (SPN): 001 Flexible Length (FLEN): 13 - International Dialing Plan (INPL): Inhibit Time-out Handler (ITOH): Route List Index (PLI): 11] (0.24)
Digital From Interface Digital From Interface Dialing and Numbering Plans <u>Electronic Switched Network</u> Flexible Code Restriction Incoming Digit Translation	Type of call that is defined by the special number (CLTP): No ca	all type (NONE)
- Phones - Templates - Reports - Properties - Migration	Number to be Denied (DENY): (Items separated by a space)	*
- Tools + Backup and Restore - Call Server Initialization - Date and Time	Digit Manipulation Index for LDID Numbers (DMI):	<u></u>
+ Logs and reports - Security + Passwords + Policies + Login Options	- Local DID number to be recognized (LDID): (Items separated by a space) Copyright © 2002-2009 Nortel Networks. All rights reserved.	

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Figure 19 – Special Number for North America

Create special number SPN 1800 (Use RLI_50) for outgoing dialing plan to toll free calls



Figure 20 – Special Number for Tool Free Call

Create special number SPN 999 (Use RLI_99) for outgoing dialing plan to 999 service calls in figure 21.





Create special number 0044 (use RLI 23) to dial to UK in figure 22



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Figure 22 – Special Number for UK dialing



Create special number 0034 (use RLI_23) to dial to Spain, figure 23.

Figure 23 – Spain Special Number

Create special number 020 (use RLI_24) to dial to local UK, figure 24

NØRTEL	CS 1000 ELEMENT MANAGER	Help
- UCM Network Services - Home - Links - Virtual Terminals - System + Alarms	 Managing: <u>192,168.10.5</u> Dialing and Numbering Plans » <u>Electronic Switched Network (ESN)</u> » Customer 00 » Numbering Plan (NET) > Acces: List » Special Number Special Number 	s Code 2 » <u>Special Number</u>
+ Core Equipment	Input Description Input Value	
 Peripheral Equipment + IP Network 	Special Number translation (SPN): 020	
+ Interfaces – Engineered Values	Flexible Length (FLEN): 11 (0.24)	
+ Emergency Services	- International Dialing Plan (INPL): 🔲	
+ Geographic Redundancy + Software	Inhibit Time-out Handler (ITOH):	
- Customers	Basta List Index / PL In 24	
- Routes and Trunks	Route List index (RLI): 24	
- Routes and Trunks	Type of call that is defined by the special number (CLTP): No call type (NONE)	
- D-Channels Digital Trunk Interface		6
- Digital Hunk Interface		
- Electronic Switched Network	Number to be Denied (DENY):	
- Flexible Code Restriction	(Items separated by a space)	
- Incoming Digit Translation		3
- Phones		
- Reports	Digit Manipulation Index for LDID Numbers (DMI): 1 💉	
- Properties		~
- Migration		
- Tools	- Local DID number to be recognized (LDID):	
- Call Server Initialization	(items separated by a space)	
- Date and Time		
+ Logs and reports		1001
Foourity	Copyright @ 2002-2007 Nonel Networks: All rights reserved.	

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Figure 24 – Local UK Special Number

Create special number 077 (use RLI_24) to dial to UK mobile, figure 25



Figure 25 – UK Mobile Special Number

Create special number 44 (use RLI_1) for incoming call to UK.



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Figure 26 – UK Incoming Call Special Number

6. Create Route List Block

Create RLI_11 for outgoing calls to North America (Use route_100 and DMI_11), figure 27





Create RLI_24 for outgoing calls to Local UK and mobile phone (Use route_100 and DMI_24), figure 27.



Figure 28 – RLB for Outgoing Local UK Call and Mobile

Create RLI_16 for terminate the incoming calls (Use DMI_16 & SPN_44) or terminate to CS1000E_B, figure 29



Figure 29 – RLB for Terminating Incoming call on Local Network of CS1000

7. Create Digit Manipulation Block

DMI_11: Digit Manipulation Block configuration to North America, figure 30

Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » Digit Manipulati Manipulation Block	ion Block L
Digit Manipulation Block	
Input Description Input Value	
Digit Manipulation Index numbers (DMI): 11	
Number of leading digits to be Deleted (DEL):	
IP Special Number (ISPN):	
Call Type to be used by the manipulated digits (CTYP): Special number in International format (INTL)	
Submit Refresh Delete Cancel	
	Copyright © 2002-2009 Nortel Networks: All rights reserved. Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » Digit Manipulation Block Digit Manipulation Block Input Description Input Value Digit Manipulation Index numbers (DMI): Input Value Digit Manipulation Index numbers (DMI): Input Value Input Description Input Value Digit Manipulation Index numbers (DMI): Input Value Input Description Input Value Digit Manipulation Index numbers (DMI): Input Value Input Description Input Value Digit Manipulation Index numbers (DMI): Input Value Insert (INST): Insert (INST): IP Special Number (ISPN): Insert (INST): Call Type to be used by the manipulated digits (CTYP): Special number in International format (INTL)

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Figure 30 – Digit Manipulation for North America

DMI 23: Digit Manipulation Block configuration Outgoing to UK and Spain, figure 31



Figure 31 – Digit Manipulation outgoing to UK and Spain

DMI_24: Digit Manipulation Block configuration to Local UK, figure 32

NØRTEL	CS 1000 ELEMENT MANAGER
- UCM Network Services - Home - Links Vidual Torminals	Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » Digit Manipulation Block List » I Manipulation Block
- System + Alarms - Maintenance	Digit Manipulation Block
+ Core Equipment	Input Description Input Value
+ IP Network	Digit Manipulation Index numbers (DMI): 24
+ Interfaces	Number of leading digits to be Deleted (DEL):
+ Emergency Services	Insert (INST):
+ Geographic Redundancy + Software	
- Customers	
- Routes and Trunks	
- D-Channels - Digital Trunk Interface	Submit Refresh Delete Cancel
- Dialing and Numbering Plans - Electronic Switched Network - Flexible Code Restriction - Incoming Digit Translation	
- Phones	
- Templates - Reports	
- Properties	
- Tools	
+ Backup and Restore	Copyright © 2002-2009 Nortel Networks. All rights reserved.

Figure 32 – Digit Manipulation for UK

In the case of network CS1000 systems, calls are tandemed from one CS1000 to another then it is required to create DMI_17: Digit Manipulation Block to terminate calls on the local network CS1000E

NØRTEL	CS 1000 ELEMENT MANAGER	Help
- Home - Links - Virtual Terminals - System	Dialing and Numbering Plans » <u>Electronic Switched Network (ESN)</u> » Customer 00 » Network Control & Services » Manipulation Block	Digit Manipulation Block List
- Maintenance + Core Equipment - Peripheral Equipment	Digit Manipulation Block	
+ IP Network	Input Description Input Valu	e
- Engineered Values	Digit Manipulation Index numbers (DMI): 17	
+ Emergency Services	Number of leading digits to be Deleted (DEL): 8 (0-19)	
+ Geographic Redundancy + Software	Incost (INST) 0501	
- Customers		
- Routes and Trunks	IP Special Number (ISPN):	
- Routes and Trunks - D-Channels	Call Type to be used by the manipulated digits (CTYP): Call type will not be changed (NCHG)	T
- Digital Trunk Interface		
- Dialing and Numbering Plans	Submit Refresh Delete Cancel	
- Electronic Switched Network		
- Incoming Digit Translation		
- Phones		
- Templates		
- Properties		
- Migration		
+ Backup and Restore		
- Call Server Initialization		
- Date and Time		
- Security		
+ Passwords		
+ Policies		

Create DMI 16: Digit Manipulation Block for incoming call, figure 34.



Figure 34 – Digit Manipulation for Incoming Call

4.2. Configure on CS1000 Voicemail System (Call Pilot)

4.2.1. Configuration Details on CallPilot Manager

Configure CS1000E switch on Call Pilot configuration by entering:

- CS1000 Call Server IP address
- Create Multimedia Chanel for communication between CS1000 and Callpilot system

	TEL		CALLPILOT MAN	IAGER				E
							Preferences	s <u>Help</u> Logout
Home	User 🔻	System 🔻	Maintenance 🔻 Messag	jing 🔻 Tools 🔻	Help 🔻			
Location •	+ Configuratio	n Wizard 🕈 M1 Sv	witch Information					
Config	uration Wiz	ard: M1 Swite	ch Information					
Back	Next Ca	ancel Help						
Meridia	n 1 Switch	Information:						
				1 . I . N I	1			
Channel	i information	for each Link I	s displayed below. Click on a li	nk to update its chann	iel settings.			
STI Bo	ard 1 (201i ir	n slot 01)						
			Switch Type:	11				
Link	<u>k STIU1-UU1</u>		Onion (Jpc. On	11 Ontion 11				
Link	k STI01-002		Switch Customer					
			Number:			Enable Symposium Ca	all Center Server Integration	
					1 <u>1</u>	Symposium Call Cente	er Server 100	
			Switch III Address: 19/	168 111	5		197 168	10 50
			Switch IP Address: 192	[<u>168</u> [<u>10</u>	. 5	CLAN IP A	Address: 192 . 168	. 10 . 50
			Switch IP Address: 192	. 168 . 10	. 5	CLAN IP 4	Address: 192 . 168	. [10] . [50
			Switch IP Address: 192	. 168 . 10	5	CLAN IP A	Address: 192 . 168	. [10] . [50
			Stil Board 201i	. 168 . 10	. 5	CLAN IP A Board ID 68157440	Address: 192 . 168	. 10 . 50
			Switch IP Address: 192 STI Board 201i Link STI01-001	. 168 . 10]. [<u>5</u>]	CLAN IP 4 Board ID 68157440	Address. 192 168	. 10 . 50
			Switch IP Address: 192 STI Board 201i Link STI01-001 #Channel Name +	TN 0000	. 5 Key0	CLAN IP A Board ID 68157440	Channel Allocation	. 10 . 50
			Switch IP Address: 192 STI Board 201i Link STI01-001 #Channel Name 1 STI01-001-001 2 STI01-001-001	TN 0.0.9.0	Key0 3415 3416	CLAN IP 4 Board ID 68157440 Key1 3301 3406	Channel Allocation	Class ID
			Switch IP Address: 192 STI Board 201i Link STI01-001 #Channel Name 1 STI01-001-001 2 STI01-001-002 3 ST01-001-003	TN 0.0.9.0 0.0.9.1 0.0.9.2	Key0 3415 3416 3203	CLAN IP 4 Board ID 68157440 Key1 3301 3406 3303	Channel Allocation	Class ID
			Switch IP Address: 192 STI Board 201i Link STI01-001 #Channel Name 1 ST01-001-001 2 ST01-001-002 3 ST01-001-003 4 ST01-001-004	TN 0.0.9.0 0.0.9.1 0.0.9.2 0.0.9.3	Key0 3415 3416 3203 3204	CLAN IP 4 Board ID 68157440 Key1 3301 3406 3303 3304	Channel Allocation	Class ID
- Call Se	erver Initializa	ition	Switch IP Address: 192 STI Board 201i Link STI01-001 #Channel Name 1 STI01-001-001 2 STI01-001-002 3 STI01-001-003 4 STI01-001-004	TN 0.0.9.0 0.0.9.1 0.0.9.2 0.0.9.3	Key0 3415 3416 3203 3204	CLAN IP 4 Board ID 68157440 Key1 3301 3406 3303 3303 3304	Channel Allocation IVR IVR Multimedia Multimedia	Class ID
- Call Se - Date a	erver Initializa Ind Time and reports	tion	Switch IP Address: 192 STI Board 201i Link STI01-001 #Channel Name 1 STI01-001-001 2 STI01-001-002 3 STI01-001-003 4 STI01-001-004	TN 0.0.9.0 0.0.9.1 0.0.9.2 0.0.9.3	Key0 3415 3416 3203 3204	CLAN IP 4 Board ID 68157440 Key1 3301 3406 3303 3304	Channel Allocation IVR IVR Multimedia Multimedia	Class ID
- Call Se - Date a + Logs a Security	erver Initializa Ind Time and reports	tion	Switch IP Address: 192 STI Board 201i Link STI01-001 # Channel Name 1 STI01-001-001 2 STI01-001-002 3 STI01-001-003 4 STI01-001-004	TN 0.0.9.0 0.0.9.1 0.0.9.2 0.0.9.3	Key0 3415 3416 3203 3204	CLAN IP 4 Board ID 68157440 Key1 3301 3406 3303 3304	Channel Allocation IVR IVR Multimedia Multimedia	Class ID
- Call Se - Date a + Logs a Security + Passw	erver Initializa ınd Time and reports r rords	tion	Switch IP Address: 192 STI Board 201i Link STI01-001	TN 0.0.9.0 0.0.9.1 0.0.9.2 0.0.9.3	Key0 3415 3416 3203 3204	CLAN IP A Board ID 68157440 Key1 3301 3406 3303 3304	Channel Allocation IVR IVR Multimedia Multimedia	Class ID
- Call Se - Date a + Logs a Security + Passw + Policie	erver Initializa nd Time and reports / ords Scottops	tion	Switch IP Address: 192 STI Board 201i Link STI01-001 # Channel Name 1 STI01-001-001 2 STI01-001-002 3 STI01-001-003 4 STI01-001-004 Local DDD	TN 0.0.9.0 0.0.9.1 0.0.9.2 0.0.9.3	Key0 3415 3416 3203 3204 zed (LDDD):	CLAN IP A Board ID 68157440 ikey1 3301 3406 3303 3304	Channel Allocation IVR Multimedia Multimedia	Class ID
- Call Se - Date a + Logs a Security + Passw + Policie + Login 1	erver Initializa Ind Time and reports r roords rs Options	tion	Switch IP Address: 192 STI Board 201i Link STI01-001 # Channel Name 1 STI01-001-001 2 STI01-001-002 3 STI01-001-003 4 STI01-001-004 Local DDD Copyright @ 2002-2007 Nortel Network	TN 0.0.9.0 0.0.9.1 0.0.9.2 0.0.9.3 number to be recogniz rks. All rights reserved.	Key0 3415 3416 3203 3204 zed (LDDD):	CLAN IP 4 Board ID 68157440 Key1 3301 3406 3303 3304	Channel Allocation IVR Multimedia Multimedia	Class ID
- Call Se - Date a + Logs a Security + Passw + Policie + Login (erver Initializa nd Time and reports r ys Options	ation	Switch IP Address: 192 STI Board 201i Link STI01-001 Copyright © 2002-2007 Nortel Network	TN 0.0.9.0 0.0.9.1 0.0.9.3 0.0.9.3	 5 3415 3416 3203 3204 	CLAN IP 4 Board ID 68157440 Key1 3301 3406 3303 3304	Address: 192 168	Class ID

Figure 35 – CS1000 switch configuration on CallPilot Manager

Go to Maintenance pull down menu, select Channel Monitor to check status of the newly created multimedia channels on Call Pilot to see if the communication between Callpilot and CS1000 has been established, figure 36.

N@RTEL CALLPILOT MANAGER	^
Preferences Help Logout	
Home User System Maintenance Messaging Tools Help	
Location + Maintenance + Channel Monitor	
Start Courtesy Stop Stop Help	
Refresh Rate	
Delay between updates: 5 💌 seconds	
Channel Status	
M1 Select All 1 2 3 4 5 6 7 8 9 10 11 12	
Legend	
🝸 Active 📍 Idle 📙 In Test 🍸 Loading 📔 No Resources 💊 Not Configured 💆 Remote (Yellow) Alarm 🛛 ACCESS Channel	
🔓 Off Duty 💊 Remote Off Duty 🚡 Disabled 👃 Shutting Down ? Uninitialized 🐇 Local (Red) Alarm 🛛 IVR Channel	
Start Courtesy Stop Stop Help	
Copyright © 2008 Nortel and its licensors. All rights reserved.	~

Figure 36 – Channel Monitor

Create Service DN for Voice Messaging system, figure 37

NØRTEL		CALLPILOT MANAGER						
								Preferences Help Lo
Home	User 🔻	System 💌	Maintenance 🔻	Messaging 🔻	Tools - H	lelp 🔻		
Location +	System 🕈 Serv	ice Directory Numb)er					
Service	Directory N	lumber						
Service [Directory Nu	mber						
New	Delete S	Selected I	Help					
#	Service DN	+	App Name		Media Type	Min Channels	Max Channels	Comments
1	3111		Voice Messagin	g	Voice	0	Default Max.	
2	3222		Express Voice I	Messaging	Voice	0	Default Max.	
5		ND10	AMIS Networkin	ig.	Voice	0	Default Max.	
6		ND11	Remote Notifica	tion	Voice	0	Default Max.	
7		ND15	Multi-delivery to	Fax	Fax	o	Default Max.	
8		ND18	Desktop Teleph	ony Agent	Voice	0	Default Max.	
9 [ND23	SCCS VPE		Voice	0	Default Max.	
10		ND25	Conferencing O	utcalling	Voice	0	Default Max.	
11		ND55	Enterprise Diag	nostics	Voice	0	Default Max.	
12	OUTBOU	ND6	Admin Agent		Voice	0	Default Max.	
13		ND7	Delivery To Tele	phone	Voice	0	Default Max.	
14		ND8	Delivery To Fax		Fax	0	Default Max.	
15		ND88	SCCS IVR		Voice	0	Default Max.	
16		ND9	Enterprise Netw	orking	Voice	0	Default Max.	

Figure 37 – Service Directory Number Page

4.2.2. Voicemail System (CallPilot) configuration detail on CS1000E Call Server

Configure CS1000E for voicemail system Call Pilot

Configure Voice messaging service DN 3111 on CS1000E

>ld 23 ACD DNS REQ prt TYPE CDN CUST 0 CDN 3111 FRRT SRRT FROA NO UUI NO MURT CDSQ NO DFDN 3109 NAME NO CMB NO CEIL 2047 OVFL NO TDNS NO

AACQ NO CNTL NO VSID HSID Configure ACD Agent #1 3110: >ld 23 ACD DNS REQ prt TYPE ACD CUST 0 ACDN 3110 MWC YES MAXP 12 SDNB NO BSCW NO ISAP NO AACQ YES ASID 16 SFNB USFB 1 3 4 5 6 CALB 1 3 4 5 6 8 11 RGAI NO ACAA NO FRRT SRRT NRRT FROA NO CALP POS ICDD NO NCFW FNCF NO CWTT NONE HMSB YES ACPQ NO FORC NO RTQT 0 SPCP NO OBTN NO RAO NO CWTH 1 NCWL NO BYTH 0 OVTH 2047 TOFT NONE HPQ NO

OCN NO OVDN IFDN OVBU LNK LNK LNK LNK EMRT MURT RTPC NO NRAC NO RAGT 4 DURT 30 RSND 4 FCTH 20 **CRQS 100** CCBA NO IVR YES TRDN NONE ABR NO OBSC NO OBPT 5 CWNT NONE Configure ACD Agent#2 >ld 23 REQ prt TYPE acd CUST 0 ACDN 3109 TYPE ACD CUST 0 ACDN 3109 MWC NO DSAC NO MAXP 1 SDNB NO BSCW NO ISAP NO AACQ NO RGAI NO ACAA NO FRRT SRRT NRRT FROA NO CALP POS ICDD NO NCFW

```
FNCF NO
CWTT NONE
HMSB YES
ACPQ NO
FORC NO
RTQT 0
SPCP NO
OBTN NO
RAO NO
CWTH 1
NCWL NO
BYTH 0
OVTH 2047
TOFT NONE
HPQ NO
OCN NO
OVDN
IFDN
OVBU LNK LNK LNK LNK
EMRT
MURT
RTPC NO
NRAC NO
RAGT 4
DURT 30
RSND 4
FCTH 20
CRQS 100
CCBA NO
IVR NO
ABR NO
OBSC NO
OBPT 5
CWNT NONE
```

4.3. Output configuration details from CS1000 and Callpilot

Using the command line interface to output some of configured Customer Data Block and configuration record details, which have been created in section 4.2, for your reference

4.3.1. Overlay 15 - Customer Data Block

REQ: PRT TYPE CDB

QT; Reviewed: SPOC 03/05/2010 Solution & Interoperability Test Lab Application Notes ©2010 Avaya Inc. All Rights Reserved. CUST 00 AML_DATA OPT DNX VSID GP02 GP03 GP04 GP05 GP06 GP07 GP08 GP09 GP10 GP11 GP12 GP13 GP14 GP15 ANI_DATA ANAT 4227 ANLD 123 M911_PANI NO ATT_DATA OPT ABDD AHD BIND BIXA BLA BOHD DNCA DRE DNX DRE FACD IC1 XTG XDP XLF XBL FKA MCTD NCD CUI MWUD LOD PSD RECA REA SYD SLD SIAD THPD ATDA ATDN 7 NCOS 0 CWUP NO CWCL 0 0 CWTM 0 0 CWBZ NO NO EFLL 0 MATT NO RTIM 30 30 30 ATIM 0 AQTT 30 AODN SPVC 00 SBLF NO RTSA RSAD SACP NO ABDN NO **IRFR NO**

XRFR NO ADHT 0 AFNT 0 AFBT 0 IDBZ NO PBUZ 02 10 ICI 00 ICI 01 ICI 02 ICI 03 ICI 04 ICI 05 ICI 06 ICI 07 ICI 08 ICI 09 RICI PAGE 002 AWU_DATA AWU NO CAS_DATA CAS NO CCS_DATA CCRS UNR ECC1 UNR ECC2 UNR CNCS 0 PELK NO CDR_DATA CDR YES IMPH NO OMPH YES AXID YES TRCR NO CDPR NO ECDR NO BDI YES OTCR NO PORT CNI DGTS BCAP NO CHLN 1 FCAF NO FCR_DATA NFCR YES

OCB1 255 OCB2 255 OCB3 255 IDCA YES **DCMX 100** FFC_DATA CCRS UNR SCPL 0 FFCS NO STRL 0 STRG ADLD 0 MFAC * FTR_DATA **DAPC**PREFIX TABLE NO: 00 ** UNKN**INTL**NATL**ESPN**LOCL**ELOC**ECDP** UNKN* E164* 00 0 PRIV* E163* 00 0 TELX* X121* NATL* OPT ABDD AHD BIND BIXA BLA BOHD CFO CFRD COX CPA CTD DBD DNCA DNX DSX DRE DSTD FACD HTU HVD XBL IC1 XDP XLF IHD XTG FKA LOD LRA MCI MCTD CUI MWUD NCD PCMD PSD PVCA RECA REA RND RTR RTD ROX SBD SDDE SIAD SLD SYD THPD TTAD VOBD CCBD CWRD HLPD HRLD CXOD BWTD DGRP 0 **IRNG NO** PKND 1 DNDL NO SPRE PREO 0 BPSS NO SRCD 0000 EEST NO EESD NO

TTBL 0 MUS YES

QT; Reviewed:

SPOC 03/05/2010

MAXT 100
PAGE 003 MUSR 50 HCC NO ALDN RECD NO PORT 0 STCB NO NSCP NO TFDR NO RPA NO MCDC NO NAUT NO IDEF NO MTAR NO LEND NO MSCD NO CPCI NO ARDL_ATTEMPT 30 CONF_DSP **CNFFIELD NO** CNF_NAME CONF INTFIELD NO INT_NAME I EXTFIELD NO EXT_NAME E BSFE NO ASPCT 000 FXS NO DFLT_LANG ENG STS_MSG MSG01 Please leave message MSG02 Back to work MSG03 In a meeting MSG04 On a conference call MSG05 At lunch MSG06 Busy call MSG07 Out of the office today MSG08 On a business trip MSG09 Project deadline today MSG10 Will reply after VO_ALO NO PCA ON TPDN BFS_CFW YES

VO_CUR_ZONE_ZDM NO VO_CUR_ZONE_TD NO ICP_DATA ICP NO IMS_DATA IMS NO INT_DATA ACCD OVF OVF OVF ATN CTVN OVF OVF OVF ATN MBNR OVF OVF OVF ATN CTRC OVF NAP OVF NAP CLDN NAP OVF NAP NAP NINV OVF OVF OVF ATN NITR OVF OVF OVF ATN NRES OVF OVF OVF ATN NBLK OVF OVF OVF ATN MFVOOVF OVF OVF ATN MEVN OVE OVE OVE ATN MFCG OVF OVF OVF ATN PAGE 004 LCKT BSY BSY BSY BSY RCLE ATN OVF ATN ATN CONG OVF DLT OVF LLT OVF DNDT BSY ESAM OVF LDN_DATA OPT XLDN DLDN YES LDN0 2000 LDA0 LDN1 LDA1 LDN2 LDA2 LDN3 LDA3 LDN4 LDA4 LDN5 LDA5 LDBZ ICI 00 ICI 01

ICI 02 ICI 03 ICI 04 ICI 05 ICI 06 ICI 07 ICI 08 ICI 09 MON_DATA USBM NO MPO_DATA FMOP RGNA STD STD AOCS DIS ATN RCY1 06 RCY2 04 RALL NO CDTO 14 IFLS NO MHLD NO PCDS CNFD 1 TGLD 2 DISD 3 CCDO NO AFCO NO ACNS NO NET_DATA OPT RTD AC1 NPA SPN LOC AC2 INTL NXX FNP YES ISDN YES VPNI 1 PNI 1 PINX_DN MBG 0 PAGE 005 BSGC 65535 PFX1 PFX2 HLOC 521 LSC RCNT 5

PSTN NO TNDM 15 PCMC 15 SATD 1 OCLI NO TIDM NO DASC ROPT NRO DITI YES TRNX NO EXTT NO FTOP FRES APAD 0 0 VNR NO NIT 8 NAS_ATCL YES NAS_ACTV NO FOPT 6 CNDN CNAT PCAT CNIP YES DMWM NO MWNS NO CNTC NATC INTC NIT_DATA NIT1 TIM1 NIT2 TIM2 NIT3 TIM3 NIT4 TIM4 RPNS NO ENS NO OAS_DATA ODN0 ODN1 ODN2 ODN3 ODN4 ODN5

ODN6 ODN7 ODN8 ODN9 ASTM 30 HDOPT 0 HDTM 30 RDR_DATA OPT CFO CFRD DSTD PVCA CWRD MCI FNAD HNT FNAT HNT PAGE 006 FNAL HNT CFTA NO CCFWDN CFN0 3 CFN1 3 CFN2 3 DFN0 3 DFN1 3 DFN2 3 DNDH NO MDID NO NDID NO MWFB NO TRCL 0 DFNR 0 CRT0 00 00 00 00 CRT1 00 00 00 00 CRT2 00 00 00 00 CRT3 00 00 00 00 DAY0 DAY1 DAY2 DAY3 HOLIDAY0 HOLIDAY1 HOLIDAY2 HOLIDAY3 ROA_DATA OPT ROX RICI TIM_DATA FLSH 45 896 PHDT 30

DIND 30 32 30 DIDT 14 16 14 LDTT 6 DLAT 0 BOTO 14 DBRC 60 RTIM 30 30 30 ATIM 0 AQTT 30 ADLD 0 AFNT 0 NFNA 0 ADHT 0 **HWTT 300** NIT 8 FOPT 6 ARDL_ACCEPT 20 ARDL_RETRY 30 TST_DATA

4.3.2. Overlay 17 – Configuration Record

```
REQ PRT
TYPE CFN
ADAN HIST
SIZE 25000
USER MTC BUG
ADAN TTY 0
CTYP PTY
 DNUM 0
 PORT 0
 DES PTY0
 FLOW NO
 USER MTC TRF SCH BUG OSN
 XSM NO
TTYLOG
          0
 BANR NO
ADAN TTY 1
CTYP PTY
 DNUM 1
 PORT 1
 DES PTY1
 FLOW NO
 USER MTC TRF SCH BUG OSN
```

XSM NO TTYLOG 0 BANR NO ADAN TTY 2 CTYP PTY DNUM 2 PORT 2 DES PTY2 FLOW NO USER MTC TRF SCH BUG OSN XSM NO TTYLOG 0 BANR NO ADAN TTY 3 CTYP PTY DNUM 3 PORT 3 DES PTY3 FLOW NO USER MTC TRF SCH BUG OSN XSM NO TTYLOG 0 BANR NO ADAN TTY 4 CTYP CPSI DNUM 4 PORT 0 DES BPS 9600 BITL 8 STOP 1 PARY NONE FLOW NO USER MTC TRF SCH BUG OSN XSM NO TTYLOG 0 BANR NO ADAN TTY 5 CTYP CPSI DNUM 5 PAGE 001 PORT 1 DES BPS 9600 BITL 8

STOP 1 PARY NONE FLOW NO USER MTC TRF SCH BUG OSN XSM NO TTYLOG 0 BANR YES ADAN ELAN 16 (Configuration for CallPilot) CTYP ELAN DES CPilot N1 512 ADAN DCH 100 CTYP DCIP DES VolP USR ISLD ISLM 4000 SSRC 1800 OTBF 32 NASA YES IFC SL1 CNEG 1 RLS ID 5 RCAP ND2 MWI (Configuration for CallPilot) MBGA NO H323 OVLR NO OVLS NO ADAN DCH 101 CTYP DCIP DES Enterprise USR ISLD ISLM 4000 SSRC 1800 OTBF 32 NASA NO IFC SL1 CNEG 1 RLS ID 25 RCAP ND2 MWI MBGA NO H323 PAGE 002 OVLR NO OVLS NO PARM

LPIB 3500 HPIB 3500 500B 2000 SL1B 255 DTIB 35 DTOB 4 NCR 20000 MGCR 25 CSQI 255 CSQO 255 TUBO NO NCPU 2 CFWS NO PCML A ALRM YES ERRM ERR BUG AUD **DTRB 100** ABCD NO **TMRK 128** FCDR OLD PCDR NO TPO NO TSO NO CLID NO DUR5 NO MLDN NO MARP YES IPIE NO FRPT NEFR DCUS NULL DTDT NO MSCL 0 PMSI MANU PMS1 PMCR 0 PORT NONE NDIS 20 OCAC NO MTRO MR SBA_ADM_INS 000 SBA_USER 512 BCAP SPEECH IDLE_SET_DISPLAY ICON NO MSEC ON

MSSD MSBT NKEY 31 TKEY 24 CEQU MPED 8D TERM REMO TERD REMD TERQ REMQ SUPL V000 V096 V100 V200 SUPC PAGE 003 SUPF DDCS MG_CARD DTCS хст CONF MGTDS IPMG IPMG_TYPE 126 000 0 MGC MGCONF IPMG PORTS IPMG_TYPE 127 000 0 30 MGC MFSD * 126 APVL MISP MG_CARD SYNM 0 EXT0 3PE EXT1 3PE MCFN 011 MB OVLY SID 0 BKGD 044 PBXH X TODR 00 DROL 030 032 045 135 137 MID_SCPU NO CY45 00 MULTI_USER OFF VAS VSID 016 DLOP ELAN 016 SECU NO INTL 0001

MCNT 9999 VSID 022 DLOP ELAN 022 SECU YES INTL 0001 MCNT 9999 VSID 034 DLOP ELAN 034 SECU YES INTL 0001 MCNT 9999 VSID 035 DLOP ELAN 035 SECU NO INTL 0001 MCNT 9999 VSID 038 DLOP ELAN 038 SECU YES INTL 0001 MCNT 9999 PAGE 004 ATRN CODE 0 SOLR 12 ROLR +45.00 AOLR +45.00 TOLR -45.00 AGCD NO VOLR NO HRLR +42.00 HTLR -44.00 ESA LIS EXT/DM DYNAMIC_ELIN_TIMEOUT 180 DYNAMIC_ELIN_REUSE YES EXT_DM_UPDT_TIMEOUT 15

4.4. CS1K Tandem Configuration

This configuration is for the deployment model of 2 or more CS1000 with Colt communication system. Represent here is for 2 CS1000s configuration.

4.4.1. Configure CS1000E A

1. Create IP on CS1000E

This section describes the steps for creating Node ID (1001) in CS 1000 network.

Enter Element Manager through the IE browser (in IE address bar, type IP address of the Node IP or TLAN of Signalling Server).

- Input Node ID and press Add...
- Enter TLAN, ELAN IP address of Signalling Server.

Node 1001 was added to be configured as the SIP gateway to the Enterprise services.

NØRTEL	CS 1000 E	LEMENT M	IANAGER			Help
- UCM Network Services	System »	IP Network » IP Telepho	ony Nodes			
- Links - Virtual Terminals - System	IP Telephony Click the Node ID to vi	Nodes	k			
+ Alarms	Add Import	Export Delete			Print	Refresh
- Core Equipment	Node ID +	Components	Enabled Applications	ELAN IP	TLAN IP	Status
- Loops	□ <u>1001</u>	1	LTPS, Gateway (SIPGw)	-	192.168.10.12	Synchronized
Superloops MSDL/MISP Cards Conference/TDS/Multifrequen Tone Senders and Detectors Peripheral Equipment IP Network Nodes: Servers. Media Cards Maintenance and Reports Metwork Address Translation CoS Thresholds Personal Directories Unicode Name Directory Interfaces Emgineered Values Emergency Services Geographic Redundancy software Customers Portage and Trunke	Show: M Nodes	9 Nortel Networks: All ri	vers and Cards	U		

Figure 38 – Creating a node on CS1000 A

The node IP information is added. For the primary proxy enter the IP address of the SIP Proxy Server (SPS). Use UDP port 5060 for SIP communication, figure 39 Support registration

NØRTEL	CS 1000 ELE	MENT MANA	GER			Help			
- UCM Network Services - Home - Links	System » IP Network » IP Telephony Nodes » Node Details Node Details (ID: 1001 - LTPS, PD, Gateway (SIPGw))								
- Virtual Terminals - Virtual Terminals - System + Alarms - Maintenance + Core Equipment - Peripheral Equipment - IP Network - <u>Nodes: Servers. Media Cards</u> - Malintenance and Reports - Media Gateways - Zones - Host and Route Tables - Network Address Translation - QoS Thresholds - Personal Directories - Unicode Name Directory + Interfaces	Node ID: Call Server IP Address: Telephony LAN (TLAN) Node IP Address: Subnet Mask: IP Telephony Node • <u>Voice Gatewa</u> • <u>Quality of Ser</u> * Required Value.	1001 192.168.10.5 192.168.10.12 255.255.255.0 PropertiesApplicat av (VGW) and Cod vice (QoS) ing Servers & Cat	Embedded LAN (E Gateway IP addres Subnet Mask: tions (click to edit configurations) ecs	-AN) s. 192.168.100. [255.255.255 on) erminal Provy S	1 0 Convor (TDQ) Save	Can			
+ Emergency Services + Geographic Redundancy + Software	Select to add Add	Remove Make Leade	er er		Print E	<u>tefresh</u>			
- Customers - Routes and Trunks - Routes and Trunks - D-Channels - Digital Trunk Interface - Digital Trunk Interface - Digital Trunk Interface - Dialing and Numbering Plans - Electronic Switched Network - Flexible Code Restriction - Incoming Digit Translation - Phones	Hostname A nd1-car1 Note: Only server(s) that ar available in the servers list .	Type Signaling Server e not part of any other IP te	Deployed Applications LTPS, Gateway, PD lephony node and deployed application(ELAN IP 192.168.100.151 b) that match the service	TLAN IP 192.168.10.147 cc(s) selected for this r	Role Leader node are			

Figure 39 – CS1000 Node Detail Settings

In the signaling server properties, the Line TPS will be enabled if this signalling server will be used for IP set registration. If the role of this server is SIP gateway only then this can be left unchecked, figure 40.

NØRTEL	CS 1000 ELEMENT MANAGE	RHelp 1					
- UCM Network Services	System » IP Network » IP Telephony Nodes » Node Node ID: 1001 - Virtual Trunk Gateway C	Details » Virtual Trunk Gateway Configuration onfiguration Details					
- Links	General SIP Gateway Settings	SIP Gateway Services					
- Virtual Terminals	Vtrk Gateway Application 🔽 Enable gateway service on this Node						
+ Alarms - Maintenance - Core Equipment	General	Virtual Trunk Network Health Monitor					
- Loops - Superloops - MSDL/MISP Cards - Conference/TDS/Multifrequen - Tone Senders and Detectors - Perioheral Equiment	Vtrk Gateway Application: SIP Gateway (SIPGw) SIP Domain name: [interop.com +]	Monitor IP Addresses (listed below) Information will be captured for the IP addresses listed below. Monitor IP:Add					
IP Network Nodes: Servers, Media Cards Maintenance and Reports Media Gateways	Gateway endpoint name car1_ss2	Monitor addresses:					
 Zones Host and Route Tables Network Address Translation QoS Thresholds Personal Directories 	Enable failsafe NRS: SIP Gateway Settings						
- Unicode Name Directory + Interfaces - Engineered Values + Emergency Services + Geographic Redundancy Software	TLS Security Isecurity Disabled Port: 5061 (1 - 65535) Number of Byte Re-negotiation: 0 Options: Client Authentication						
- Customers	🗖 X509 certificate authority						
- Routes and Trunks - Routes and Trunks - D-Channels - Digital Trunk Interface	Proxy Or Redirect Server: Primary TLAN IP Address: 192.168.10.60	Secondary TLAN IP Address: 0.0.0.0					
Dialing and Numbering Plans Electronic Switched Network Flexible Code Restriction Incoming Digit Translation Phones	Port 5060 (1 - 65535) Transport protocol: UDP V Options Support registration	Port: 5060 (1 - 65535) Transport protocol: UDP 💌 Options: 🗆 Support registration					
- Templates - Reports - Properties	* Required Value. Note: Changes mad	Secondary CDS Proxy e on this page will NOT be transmitted until the Node is also saved. Save Cancel					

Figure 40 – SIP Gateway Settings

2. Create D-channel (DCH)

- Launch Element Manager of CS 1000 6.0
- Choose D-Channels, enter D-channel number (i.e.: 101), select DCH for type

Click Add to create DCH 101 in figure 41

Figure 41 – D-Channels Property Configuration



Also click on Basic Options and edit the Remote Capabilities (RCAP). Enable MWI if CS1K hosted voice mail will be used.

3. Create Route

Create route 101 using DCH 101 for SIP trunks figure 42





Configure Route 101 for SIP trunks continue, figure 43



Figure 43 – Route Property Configuration Details (cont.)

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4. Create Trunk (figure 44)



Figure 44 – Trunk Property Configuration

Since media security is not support under Colt system, Disable Media Security (sRTP) at the trunk level as show in figure 45.



Figure 45 – Class of Service Configuration

5. Create LOC

Create LOC 521 for outgoing calls to CS1000E_B (Use RLI_5), figure 46

NØRTEL	CS 1000 ELEMENT MANAGER	
UCM Network Services Home Links - Virtual Terminals	Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Numbering Plan (NET) > Access	s Code 2 » <u>Location Code</u>
System + Alarms - Maintenance	Location Code	
+ Core Equipment	Input Description Input Va	lue
+ IP Network	Location code (LOC): 521	
+ Interfaces	Elovible Length (ELEN): 7	
+ Emergency Services	(0.10)	
+ Geographic Redundancy	Route List Index (RLI): 5	
+ Soπware Customers	Maximum 7 digit NPA code allowed (NPA):	
Routes and Trunks	Maximum 7 digit NXX code allowed (NXX):	
- Routes and Trunks	Inhibit Time Out Llandler (ITOLI)	
- D-Channels - Digital Trunk Interface		
Dialing and Numbering Plans	Incoming Trunk group Exclusion Index (ITEI):	
- Electronic Switched Network	Listed Directory Number (LDN): 123456789	
– Flexible Code Restriction – Incoming Digit Translation	Direct Inward Dial (DID):	
Phones		
- Templates	Submit Refresh Delete Cancel	
- Properties		
- Migration		
Fools		
- Call Server Initialization		
- Date and Time		
+ Logs and reports		
+ Passwords		

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Figure 46 – Location code for outgoing call to CS1000E_B

6. Create SPN

Create SPN 511 for terminate from CS1000E_B.



7. Create Route List Block

Create RLI_5 for outgoing call to CS1000E_B (Use route 101, DMI_0), figure 48



Figure 48 – Create Route for outgoing call to CS1000E_B

Create RLI_4 to terminate calls from CS1000E_B (Use route 101, DMI_4), figure 49



Figure 49 – Create Route for call termination from CS1000E_B

8. Create DMI

DMI_4: DMB configuration (Delete: 3) and Insert 91613967 to terminate call to PSTN, figure 50



Figure 50 – Digit Manipulation to terminate call to PSTN

DMI_6: Digit Manipulation Block configuration (Delete: 3) to terminate call from CS1000E_B, figure 51

NØRTEL	CS 1000 ELEMENT MANAGER	Help
- UCM Network Services - - Home - Links	Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » Digit Manipulation Manipulation Block	<u>Block List</u> »
 Virtual Terminals System Alarms Maintenance 	Digit Manipulation Block	
+ Core Equipment	Input Description Input Value	
 Peripheral Equipment + IP Network 	Digit Manipulation Index numbers (DMI): 6	
+ Interfaces	Number of leading digits to be Deleted (DEL)	
- Engineered Values	(0-19)	
+ Emergency Services	Insert (INST):	
+ Software	IP Special Number (ISPN):	
- Customers		
- Routes and Trunks	Call Type to be used by the manipulated digits (CTYP): Call type will not be changed (NCHG)	
- Routes and Trunks	P	
- D-Channels - Digital Trunk Interface	Submit Refresh Delete Cancel	
Dialing and Numbering Plans Electronic Switched Network Flexible Code Restriction Incoming Digit Translation		
- Phones		
- Templates		
- Reports		
- Migration		
- Tools		
+ Backup and Restore		
- Call Server Initialization		
+Loos and reports		

Figure 51 – Digit Manipulation to terminate call from CS1000E_B

4.4.2. Configure CS1000E 6.0 B:

1. Create IP on CS1000E

This section describes the steps for creating Node ID (2001) in CS 1000 network. Enter Element Manager through the IE browser (in IE address bar, type IP address of the Node IP or TLAN of Signalling Server).

- Input Node ID and press Add...
- Enter TLAN, ELAN IP address of Signalling Server.

Node 2001 was added to be configured as the SIP gateway to the Enterprise services, figure 52

N@RTEL	CS 1000 EL	EMENT M	IANAGER			Help
UCM Network Services Home Links Virtual Terminals System Alarms	System » IP 1 IP Telephony No Click the Node ID to view	Network » IP Telepho odes or edit its properties	ny Nodes		Print	Refresh
- Maintenance		Componente	Enabled Applications	CLANUD		Statuc
- Loops	2001	1	LTPS, Gateway (SIPGw)	-	192 168 10 11	Synchronized
Superloops MSDL/MISP Cards Conference/TDS/Multifrequen Tone Senders and Detectors Peripheral Equipment IP Network Nodes: Servers. Media Cards Maintenance and Reports Media Gateways Zones Host and Route Tables Network Address Translation QoS Thresholds Personal Directories Unicode Name Directory Interfaces Engineered Values Emergency Services Geographic Redundancy Software Customers Porsonal Directory	Show: R Nodes	Component Ser	yers and Cards	÷		

Figure 52 – Node Configured as Enterprise Service SIP Gateway

The node IP information is added. For the primary proxy enter the IP address of the SIP Proxy Server (SPS). Use UDP port 5060 for SIP communication. Support registration, figure 53

	Condition in the Arrived Annual Constant Arrest	the transmission of the second second	e dest i see de dest						
- UCM Network Services	System » IP Net	twork » IP Telephony Nodes	» Node Details						
- Home	Node Details (ID:	2001 - I TPS PD (Sateway (SIPGw))						
- Virtual Terminals									
System	Node ID:	2001	* (0-9999)						
+ Alarms	Call Server ID Address	102 168 10 6							
- Maintenance	Gan Gerver II Address	5. [152.100.10.0							
+ Core Equipment - Peripheral Equipment - IP Network - <u>Nodes: Servers, Media Cards</u> Mainteegnee and Penede	Telephony LAN (ILAN	i) I	Embedded LAN (ELA	AN)	1				
	Node IP Address:	192.168.10.11	Gateway IP address:	192.168.100.	1 *				
	Subnet Mask:	255.255.255.0	Subnet Mask:	255.255.255	D *				
- Media Gateways	IP Telephony Node	IP Telephony Node Properties Applications (click to edit configuration)							
- Network Address Translation - QoS Thresholds - Personal Directories - Unicode Name Directory Interfaces - Engineered Values + Emergency Services	Voice Gatew Quality of Se Required Value. Associated Signa	vay (VGW) and Cod invice (QoS) ling Servers & Cal	• Te	rminal Provy S	on/or (TPS)	re Car			
+ Geographic Redundancy + Software	Select to add 💌 Ad	d Remove Make Lead	er		Print I	Refresh			
- Customers	 ☐ Hostname ▲	Type	Deployed Applications	ELAN IP	TLAN IP	Role			
- Routes and Trunks	nd1-car1	Signaling Server	LTPS, Gateway, PD	92.168.100.150	192.168.10.246	E Leader			
- Routes and Trunks - D-Channels - Digital Trunk Interface	Note: Only server(s) that a available in the servers list	are not part of any other IP te ·	elephony node and deployed application(s)	that match the servic	e(s) selected for this	s node are			

Figure 53 – Node Details Configuration

NØRTEL	CS 1000 ELEMENT MANAGER	Help Logo
JCM Network Services	System » IP Network » IP Telephony Nodes » Node Details	s Virtual Trunk Gateway Configuration
lome	Node ID: 2001 - Virtual Trunk Gateway Config	guration Details
inks	General SIP Gateway Settings	SIP Gateway Services
virtual Terminals system	Vtrk Gateway Application 🗹 Enable gateway service on t	this Node
Maintenance Core Equipment	General	Virtual Trunk Network Health Monitor
- Loops - Superloops	Vtrk Gateway Application: SIP Gateway (SIPGw)	Monitor IP Addresses (listed below)
- MSDL/MISP Cards - Conference/TDS/Multifrequen Tope Senders and Detectors	SIP Domain name interop.com	Information will be captured for the IP addresses listed below.
Peripheral Equipment	Local SIP Port: 5060 *(1 - 65535)	Monitor addresses:
- Nodes: Servers, Media Cards - Maintenance and Reports	Gateway endpoint name Car2_ss2 *	
– Media Gateways – Zones – Host and Route Tables	Gateway password:	Remove
- Network Address Translation - QoS Thresholds	SIP Gateway Settings	
- Personal Directories - Unicode Name Directory	TLS Security Security Disabled	
Engineered Values	Port: 5061 (1 - 65535)	
Geographic Redundancy Software	Options: Client Authentication	
stomers	X509 certificate authority	
utes and Trunks	Proxy Or Redirect Server:	
Outes and Trunks O-Channels Digital Trunk Interface	Primary TLAN IP Address: 192.168.10.60	Secondary TLAN IP 0.0.0.0
ling and Numbering Plans	Port. 5060 (1 - 65535)	Port: 5060 (1 - 65535)
Texible Code Restriction ncoming Digit Translation	Options Support registration	Transport protocol: UDP
Femplates Reports	Primary CDS Proxy	Secondary CDS Proxy
Properties ligration	* Required Value. Note: Changes made on thi	is page will NOT be transmitted until the Node is also saved. Save Cancel
ols Jackup and Restore		
ate and Time		

Figure 54 – Trunk Gateway Configuration Details

In the signalling server properties, the Line TPS will be enabled if this signalling server will be used for IP set registration. If the role of this server is SIP gateway only then this can be left unchecked.

2. Create D-channel (DCH)

- Launch Element Manager of CS 1000 6.0
- Choose D-Channels, enter D-channel number (i.e.: 101), select DCH for type

Click Add to create DCH 101, figure 55



Figure 55 – D-Channel Property Configuration

Also click on Basic Options and edit the Remote Capabilities (RCAP). Enable MWI if CS1K hosted voice mail will be used.

3. Create Route

Create route 101 using DCH 101 for SIP trunks, figure 56



60 of 88 Colt Sonus

Figure 56 – Route Property Configuration

Configure Route 101 for SIP trunks, figure 57



Figure 57 – Route Configuration

4. Create Trunk (figure 58)



Figure 58 – Trunk Property Configuration

Disable Media Security (sRTP) at the trunk level, figure 59

NØRTEL	CS 1000 ELEMENT MANAGER	R	Help Logout
- Home - Links - Virtual Terminals - Bookmarks	Managing: 192.168.10.6 Routes and Trunks » Routes and Trunks » <u>Customer I</u> Class of Service Configuration	0. Route 101, Trunk 1 Property Configuration » Class of Service Configuration	
- System + Alarms - Maintenance	- Class of Service		
- Peripheral Equipment	Input Description	Input Value	
- IP Network	- ACD Priority (CLS)	ACD Priority not required (APN) 👻	
 Nodes: Servers, Media Cards Maintenance and Reports 	- Analog Semi-Permanent Connections (CLS)	Analog Semi-Permanent Connections Denied (SPCD) 💌	
- Media Gateways	- ARF Supervised COT (CLS)		
- Zones	- Barring (CLS)	~	
 Host and Route Tables Network Address Translation 	Pattery Supervised COT (CLS)		
- QoS Thresholds	- Ballery Supervised COT (CLS)		
- Personal Directories	- Busy Tone Supervised COT (CLS)	×	
- Engineered Values	- Calling Line Identification (CLS)	~	
+ Emergency Services	- Calling party (CLS)	Calling party Denied (CND) 👻	
+ Software	- Central Office Ringback (CLS)		
- Customers	Centrey Switchbook Elash (CLS)	Centrey Switchhook Flach Denied (THED)	
- Routes and Trunks	PidDules (CLC)		
- D-Channels	- Diai Puise (CLS)		
- Digital Trunk Interface	- DTR PAD value (CLS)	×	
- Electronic Switched Network	- Echo Canceling (CLS)	Echo Canceling Denied (ECD) 😒	
- Network Routing Service	- Hong Kong DTI (CLS)	~	
- Flexible Code Restriction	- Loop Break Supervised COT (CLS)	V	
- Tools	- Make-break ratio for dial pulse (CLS)	10 pulses per second (P10)	
+ Backup and Restore	Manual harvester (CLC)		
- Call Server Initialization	- Manual Incoming (CLS)		
+ Logs and reports	-Media Security (CLS)	Media Security Never (MSNV)	
- Security	-Network Hook Flash Over M911P (CLS)	×	~
	Copyright @ 2002-2007 Nortel Networks. All rights reserved:		

Figure 59 – Class of Service Configuration

5. Create SPN

Create LOC 511 (Use RLI_5) for outgoing calls to CS1000E_B



Figure 60 – Special Number for Outgoing Call to CS1000E_B

6. Create HLOC

Create HLOC 521 (Use DMI_4) for incoming calls from CS1000E_A, figure 61

- UCM Network Services	
- Home - Links - Virtual Terminals - System	Dialing and Numbering Plans » <u>Electronic Switched Network (ESN)</u> » Customer 00 » Numbering Plan (NET) > Access Code 2 » <u>Home Location C</u> List » Home Location Code Home Location Code
+ Alarms - Maintenance	
+ Core Equipment	Input Description Input Value
 Peripheral Equipment IP Network 	Home Location code (HLOC): 521
+ Interfaces - Engineered Values	Digit Manipulation Index (DMI):
+ Emergency Services + Geographic Redundancy + Software	Submit Refresh Delete Cancel
- Customers	
- Routes and Trunks - Routes and Trunks - D-Channels - Digital Trunk Interface	
Dialing and Numbering Plans Electronic Switched Network Flexible Code Restriction	
- Incoming Digit Translation - Phones	
- Reports - Properties - Migration	
- Tools - Date and Time	
+ Logs and reports - Security	
+ Login Options	

7. Create Route List Block

Create RLI_5 for outgoing calls to CS1000E_B (Use DMI_0), figure 62

NØRTEL	CS 1000 ELEMENT MANAGER	Help
- UCM Network Services	Data Entry of a Route List Block	
- Home		
- LINKS	Double Link Dia de Jackey 5	
System	Koule List Block Index. 5	
+ Alarms	Input Vacua	
- Maintenance		
+ Core Equipment	Entry Number for the Route List (ENTR): 0	
 Peripheral Equipment + IP Network 	Local Termination entry (LTER):	
+ Interfaces	Route Number (ROUT): 101	
+ Emergency Services	Skip Conventional Signaling (SCNV):	
+ Geographic Redundancy + Software	Use Tone Detector (TDET):	
Customers	Time of Day Schedule (TOD): 0	
Routes and Trunks		
- Routes and Trunks	Entry is a VNS Route (VNS):	
- D-Channels	Conversion to LDN (CNV):	
- Digital Trunk Interface	Expansive Poute (EXP)	
Dialing and Numbering Plans		
- Electronic Switched Network	Facility Restriction Level (FRL): 0 (0-7)	
- Incoming Digit Translation	Digit Manipulation Index (DNN)	
Phones		
- Templates	ISL D-Channel Down Digit Manipulation Index (ISDM): 0 (0-999)	
- Reports		
- Properties	Free Calling Area Screening Index (FCI):	
- Migration	Free Special Number Screening Index (FSNI): 0	
Tools	Pusiness Natural's Extension Pouto (PNE)	
+ Backup and Restore		
- Date and Time	Strategy on Congestion (SBOC): No Reroute (NRR)	•
Security	- OSIG Alternate Routing Causes (COPT): OSIG Alternate Routing Cause 1	
+ Policies		
+ Login Options	ISDN Drop Back Busy (IDBB): Drop Back Disabled (DBD)	

Figure 62 – Route for Outgoing Call to CS1000E_B

8. Create DMI

DMI_5: for outgoing calls to CS1000E_A, figure 63



Figure 63 – Digit Manipulation for Outgoing Calls to CS1000E_A

4.4.3. Configure SIP Proxy Server (SPS)

Create gateway endpoints on SPS

N@RTEL	NETWO	RK ROUTING	SERVIO	CE MANAG	ER		Help.	<u>Loqout</u>
«Common Manager - System NRS Server	Managing: 💿	Active database Standby database	192.168.10 Numbering P	. 60 an_» Endpoints				^
Database System Wide Settings	Search for I	Endpoints						<u>Hide</u>
 Numbering Prains Domains Endpoints Routes Network Post-Translation Collaborative Servers 	Enter an endpo Endpoint ID: *	int ID (use * for all) and c	lick Search You	u may narrow the sea	arch by specifying a	a particular domair	n.	
 Tools SIP Phone Context 	Comit esuits to Domain. Monophoon 7 dop 7 dop							
H.323 SIP Backup Restore	Gateway En	dpoints (5) User E	Endpoints (C)			Refr	r <u>esh</u>
ONINKS Data upgrade		Supported F	Protocols	Call Signaling IP	Description	# of Routing Entries	Context	^
	1 DCM50	2 Static SIP e	ndpoint/NCS	47.248.100.215	BCM50r2	2	interop.com / udp / cdp	
	2 🔲 <u>OCS-M(</u>	Dynamic SI	P endpoint /	47.248.100.123	OCS	1	interop.com / udp / cdp	
	3 🔲 <u>car1 ss</u>	2 Dynamic SI NCS	P endpoint /	192.168.10.12	car1_ss2	1	interop.com / udp / cdp	
	4 🗌 <u>car2 ss</u>	2 Dynamic SI NCS	P endpoint /	192.168.10.11	car2_ss2	2	interop.com / udp / cdp	
	- 🗔 mn118	mee lier Dynamic SI	P endnoint	Not registered	hnh	1	interon.com (udn (cdn	
	1 - 5 of 5 Gatevva	vy Endpoint(s)		Page 1 o	of 1		First Previous Next	Last
	Copyright @ 2008	Nortel Networks. All rights re	eserved.					

Figure 64 – SIP Gateway Endpoint Creation

Create routing entries for each of gateway endpoints on SPS, figure 65

«Common Manager System NRS Server	Managing: Active data Standby data 	abase 192.168.1 atabase <u>Numbering</u>	0.60 <u>Plan »</u> Routes		
Database System Wide Settings	Search for Routing	Entries			Hi
Numbering Plans Domains Endpoints	Enter a DnPrefix and Dn T	ype (use * for all) and click §	earch.You may nar	row the search by specifying a	particular domain.
Routes Network Post-Translation	DN Prefix:	DN Type:	II DN Types	*	
Collaborative Servers	The second s	interon com	uda	, cdp	
SIP Phone Context	Limit results to Domain:	interop.com	uuh	, cab	
- Routing Tests	Endpoint	Name: All gateway endpoi	nts 💌		
H.323				D	sulte ner nage: 50 🔽 Search
Backup				110	sours per page
Restore	Routing Entries (7)	Default Routes (0)			
GI//MDC Data unarada		aart			Defeed
GK/NRS Data upgrade	Routing test Ex	3011			Refrest
GK/NRS Data upgrade	Routing test Ex	Type non code)	Route Cost	SIP URI Phone Context	Context
GK/NRS Data upgrade	DN Prefix DN Trive 0 0 0 521	fype non code) ite level 1 regional (UDP tion code)	Route Cost	SIP URI Phone Context	Context interop.com / udp / cdp / car2_ss2
GK/NRS Data upgrade	DN Prefix DN DN Prefix DN 3 521 613 Prive	Type ion code) ite level 1 regional (UDP iton code) ite level 1 regional (UDP iton code)	Route Cost 1 1	SIP URI Phone Context	Context interop.com / udp / cdp / car2_ss2 interop.com / udp / cdp / car2_ss2

Figure 65 – Routing Entries for Gateway Endpoints

4.4.4. CS1000E SIPLINE CONFIGURATION

In this section, it shows how to configure a SIP LINE system on CS1000E. Follow the bellow steps to set up the SIP LINE server.

4.4.4.1 Configure SIP LINE CS1000E in Element Manager

Figure 66 show hot to add SIP LINE Node 1002 under System -> IP Network -> IP Telephony Nodes

System »	IP Network » IP Telepho	ony Nodes			
P Telephony	Nodes				
lick the Node ID to vie	ew or edit its properties	h.			
Add Import E	xport Delete			Print I	Refresh
Node ID +	Components	Enabled Applications	ELAN IP	TLAN IP	<u>Status</u>
1002	1	SIP Line	0.00	192.168.100.14	Synchronized

Figure 66 – IP Telephony Nodes

Figure 67, 68 and 69 show how to set up the SIP LINE Node 1002 configuration details SAVE and SYNC are required – And then APPSTART RESTART on SLG server.

CS 1000 ELI	EMENT MANA	AGER			Help Logo
System » IP N Node Details (ID	etwork » <u>IP Telephony Nodes</u> : 1002 - SIP Line)	<u>s</u> » Node Details			ä
Node ID:	1002	* (0-9999)		
Call Server IP Addres	ss: 192.168.100.5	*			
Telephony LAN (TLA	N)		Embedded LAN (ELAI	V)	
Node IP Address:	47.248.100.13	*	Gateway IP address:	47.248.10.2	4
Subnet Mask:	255.255.255.0	*	Subnet Mask:	255.255.255.0	*
IP Telephony Nod	le PropertiesApplica	tions (click t	o edit configuration)		_
 Voice Gate Quality of S 	way (VGW) and Coc ervice (QoS)	iecs			
* Required Value.					Save Cancel

Figure 67 – Node Configuration Details

CS 1000 ELEMENT MANAGER

	111
leip	Logo

Toue ID. It	Joz - on Line Conng	uration Details	2
Seneral	<u>SIP Line Gateway Settin</u>	i <u>as</u>	SIP Line Gateway Service
SIP Line Gat	teway Application: 🗹 Enable	gateway service o	on this Node
General			Virtual Trunk Network Health Monitor
SIP D	omain name: interop.com		Monitor IP Addresses (listed below)
SLG en	ndpoint name: VRS14-SLS		Information will be captured for the IP addresses listed below.
S	LG Group ID: 1002		
SLG L	ocal Sip Port: 5070	(1 - 65535)	Monitor addresses.
SLG L	ocal TIs Port: 5071	(1 - 65535)	Remove
SIP Line G	ateway Settings		
Security Poli	icy: Security	/ Disabled 💌	
Number of B	yte Re-negotiation: 0	7	
Options: C C x	Client Authentication 509 Certificate Authenticatio	on Enabled	
SIP Line G Branch / G	ateway Service R Office Settings:		
* Required Valu	ie. Note: I	Changes made on this	page will NOT be transmitted until the Node is also saved. Save Cano

Figure 68 – Node Configuration Details (Cont...)

CS 1000 ELEMENT	MANAGER	Help Logout
System » IP Network » I <u>P Tel</u>	phony Nodes » Node Details » SIP Line Configuration	
Node ID: 1002 - SIP Line C	onfiguration Details	
Branch / GR Office Settings		
SLG Role: MO -		
SLG Mode: S1/S2 -		
	MO SLG IP: 192.168.100.13	
	MO SLG Port: 5070 (1 - 65535)	
MO SLG Transport: TCP 💌		
	GR SLG IP: 0.0.0.0	
	GR SLG Port: 5070 (1 - 65535)	
GR SLG Transport: TCP 💌		-
* Required Value.	Note: Changes made on this page will NOT be transmitted until the No	ode is also saved. Save Cancel

Figure 69 – Node Configuration Details (Cont...)

4.4.4.2 Configure CS1000E Call Server

For the configuration of SIP Line on Call Server, one needs to use command line to set it up. Follow the bellow steps to accomplish that.

Packages Required for SIP line on Call Server of CS1000E, these are keycode enablement

- 1. SLS_Package 417 SIP Line Service
- 2. FFC-139 Flexible Feature Codes
- 3. SIP_LINE_NT_PKG 415 Nortel SIP Line Package
- 4. SIP_LINE_3P_PKG 416 3rdParty SIP Line Package

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4.4.4.2.1 Configure SIPL service in LD15

LD 15 REQ CHG TYPE SLS CUST 0 SIPL_ON **YES** SIPD **INTEROP.COM** UAPR **222** - DN prefix used to auto-generate UADN for all SIPL clients of this customer NMME NO

4.4.4.2.2 Configure DCH for SIPL in LD 17

LD 17 **REQ CHG** TYPE ADAN ADAN new dch 11 ADAN DCH 11 CTYP DCIP DES SIPL USR ISLD **ISLM 4000** SSRC 1800 OTBF 32 NASA NO IFC SL1 CNEG 1 RLS ID 25 RCAP MBGA NO H323 OVLR NO OVLS NO

4.4.4.2.3 Configure ELAN AML link in LD 17

LD 17 REQ CHG TYPE ADAN ADAN new elan 32 ADAN ELAN **32** – new AML ELAN link, link number should be bigger or equal to 32 CTYP **ELAN** DES **SIPL** N1 512

4.4.4.2.4 Configure VAS ID for AML link in LD 17

LD 17 REQ CHG TYPE VAS VAS new VSID **32** – VAS ID number ELAN **32** – Defined in step 3

4.4.4.2.5 Configure SIPL route

LD 16 REQ new TYPE rdb CUST 0 ROUTE 11 DES SIPL TKTP TIE VTRK YES ZONE 10 - virtual trunk zone defined in LD117 PCID SIPL NODE 1002 - node ID of SIPL node DTRK NO ISDN YES MODE ISLD DCH 11 – DCH defined in step 2 IFC SL1 PNI 00001 NCNA YES NCRD YES TRO NO FALT NO CTYP UKWN INAC YES ISAR NO DAPC NO

ICOG IAO

ACOD 8011 - route access code

4.4.4.2.6 Configure SIPL trunks

LD 14 REQ **NEW 256** – e.g. create 256 trunks TYPE **IPTI** TN **124000** - starting TN for virtual trunks DES **SIPL** CUST 0 RTMB **111** – route number and member CHID 1 TGAR **0** STRI **IMM** STRO **IMM** CLS **UNR**

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4.4.4.2.7 Check status of the details configuration SIPL link is up on Call Server and SIP line Gateway

On Call Server >*ld 96 DCH 011 : OPER EST ACTV AUTO DES : SIPL_N1402

On SLG [nortel@vrf14-sls ~]\$ slgShow === VTRK ===

===== General ===== SLG State = AppReady Total User Registered = 1

===== AML Info ====== hAppBlk TaskName Tid LinkState NumRetry LinkNum Trace 0x1226c80 SLG 0xfb00 **Up** 0 32 0

4.4.4.2.8 Configure SIP Line Client

Setting password length for SIP line client using LD15

LD 15 REQ CHG TYPE: **FFC** TYPE FFC_DATA CUST 0

SCPL **4** – password length is 4

4.4.4.2.9 Configure UEXT for SIPL client

LD 11 REQ NEW TYPE UEXT

TN **10400011** - Virtual TN for SIPL client CUST **0** UXTY **SIPL** – UEXT type must be SIPL MCCL **YES** SIPN 1 SIP3 1 FMCL 0 TLSV 0

** Begin Note: Sigma phone: SIPN-SIP3-FMCL-TLSV = 1-0-0-0 SMC3456: SIPN-SIP3-FMCL-TLSV = 1-0-0-0 SipToneV: SIPN-SIP3-FMCL-TLSV = 0-1-0-0

***End Note

SIPU **4861** – SIPL userID, often set equal to DN of the phone NDID **1002** – NodeID of the SIPL node ZONE 001 – MO zone configured in LD 117 TGAR **0** – should be 0, if not we can dial to SipToneV

SCPW 1234 - password for SIPL client to login

CLS UNR

```
KEY 00 SCR 4861 – DN of the phone
CPND NEW – in case you want to set CLID for phone
NAME set4861
XPLN 20
DISPLAY_FMT FIRST,LAST
01 HOT U 2224861 – autogenerate when you enter information for KEY 0
```

4.4.4.2.10 Check current status set registration on SLG

[nortel@vrf14-sls~]\$ slgSetShowAll

=== VTRK === UserID TN Clients Calls SetHandle 4861 104-00-00-11 1 0 0xb7d8a0c8

4.4.5. SMC3456 softphone

Link to download: <u>http://livelink-</u> <u>ott.ca.nortel.com/livelink/livelink.exe?func=ll&objId=34471954&objAction=browse&sort=nam</u> <u>e&viewType=1</u>.

After installation on the PC and apply the Licence key which is required for activate the SMC to be used. Run the SMC3456, you will see figure

SMC 3456 - 4863	
File View Contacts Actio	ns Help
Available -	III • 🖂
	~)))
Enter name or number	
	(Options -
Address Book	
Contacts History	
<u>ି</u> ୍ ×	
S Friends	
4861	
4862	
Home	
Work	
NORTEL	Q COUNTERPATH

Figure 70 SMC Client

On the top menu bar, go to FILE -> PREFERENCES -> ADVANCED -> LOGIN SERVER → No login server available

Preferences		_ ×
Application	Advanced	
Alerts & Sounds	General Options	
Privacy	Reduce echos from speakers (AEC)	
Quick Transfer	Use slower but more compatible video format	
Devices		
Network	Auto Answer Calls	
Audio Codecs	Auto answer incoming calls after ringing first	
Video Codecs	Auto answer incoming calls without ringing	
Quality of Service	Media:	
DAP	Auto answer with audio	
	Auto answer with audio and video	
Contact Storage	Los Donard	
Diagnostics	Login Server	
Advanced		
	Cogin server is required to login	
	Mumber of times a user can skip login to the server if it cannot be contacted	
	-1 (-1 denotes unlimited).	
	Reset to Default OK	Cancel

Figure 71 – Advanced Options Menu

4.4.5.1 Add a SIP Account on SMC3456
In order to create a SIP account for SMC3456 to be able to register to CS1000E SIP line server, From the top menu bar go to FILE -> ACCOUNT SETTINGS -> Add New SIP Account, see figure 72.

abled	Account name	Status	Protocol	User ID	Add >	
<u>×</u>	4863	Ready	sip	4863@interop.com	New SID Accou	unt
	56108	Disabled	sip	56108@interop.com	New Str Accou	
	CoRes PN 3005	Disabled	sip	3005@bell.ca	New XMPP Ac	count
					Clear	
					Move Up	
					Move Down	
1981-238	ater March Theorem	the set of the	18-1811			
e first enab	led SIP account in th	e list is the de	fault account. Ph	one calls will be made on this account if no		
i pian appii	es to the dialed numb	ber.				

Figure 72 – Accounting Settings

The created account is appeared as figure 73.

Account	
ccount Voicemail Topology Presence Sto	rage Security Advanced
Firewall Traversal	
IP Address	
Discover global address	
Use local IP address	
STUN server	
Discover server	
Use this server:	
Manual Enable ICE	
Port must be the s	ame, e.g. 5060
Range of ports used on local computer: 5060 -	5060

Figure 73 – Topology SIP Account Settings

1.1	nooman	Topology Presence Storage	Security Advanced
Acc	ount name:	4863	-
	Protocol:	SIP	
User Details			-
User ID:		4863@interop.com	e.g. joseph@domain.com
Password:		****	
Display name:		4863	
Authorization name:		4863	
Domain Pro:	ку ———		
Register	with domain	and receive calls	
Send outbou	ind via:		
Domain			
Prove	Address 19	92.168.100.13:5070	

Figure 74 shows how to set SIP account details by clicking on the Account menu tap.

Figure 74 – SIP Account Details Setting

Figure 75 shows the newly created SIP account

4863 5610 CoRe	R 8 D 8 PN 3005 D	eady isabled	sip sip	4863@interop.com 56108@interop.com	New SIP Account	
5610 CoRe	8 D 8 PN 3005 D	isabled isabled	sip	56108@interop.com	New XMPD Accourt	
CoRe	es PN 3005 D	isabled	1000			
		in a start	sip	3005@bell.ca		1t
					Clear	
					Move Up	
					Move Down	

Figure 75 – New Created SIP Account

4.4.6. Provisioning SIP Phone Sets 1140 and 1120 on CS1000E Call Server

On CS1000E Call Server, use Command Line interface to configure the 1120 and 1140 phone sets.

TN 104000 UXTY DATE PAGE DES DES SIPL TN 104 0 00 00 VIRTUAL TYPE UEXT CDEN 8D CTYP XDLC CUST 0 UXTY SIPL MCCL YES SIPN 1 SIP3 0 SIPU 4861 NDID 1002 SUPR NO SUBR DFLT MWI RGA CWI MSB UXID NUID NHTN CFG_ZONE 010

CUR ZONE 010 ERL 0 ECL 0 FDN TGAR 0 LDN NO NCOS 7 SGRP 0 RNPG 0 SCI 0 SSU XLST SCPW 1234 SFLT NO CAC MFC 0 CLS UNR FBD WTA LPR MTD FND HTD TDD HFD CRPD MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1 POD DSX VMD SLKD CCSD SWD LND CNDD CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD ICDD CDMD LLCN MCTD CLBD AUTU GPUD DPUD DNDD CFXD ARHD CLTD ASCD CPFA CPTA ABDD 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 VOLA VOUD CDMR ICRD MCDD T87D MSNV FRA PKCH CPND LANG ENG HUNT PLEV 02 DANI NO AST IAPG 0 AACS NO ITNA NO DGRP MLWU LANG 0 MLNG ENG DNDR 0 KEY 00 SCR 496855 0 MARP 01 HOT U 2224861 MARP 0

5. Colt Communication System configuration

Colt will have to provide this configuration notes.

6. General Test Approach and Test Results

The focus of this interoperability compliant testing was to verify the SIP trunk connectivity between the Colt Communication systems and Avaya Communication Server 1000E release 6.0. The testing verified SIP signaling and media of the basic telephony features are communicating correctly. The following features were covered; basic calls, busy, music on hold, blind and consultative transfers, DTMF, MWI, codec negotiations, conference.

6.1. General test approach

The general test approach was to have Colt Sonus system connected to CS1000E SIP Gateway using Sonus IP address. The SIP trunk communication should be established between CS1000E and Colt Sonus system. Calls can be made from end to end, i.e. PSTN phone can call through created route from Colt Communication system to CS1000Es' analog, IP, SIP phones via SIP trunk. The main objectives were to verify the basic SIP trunk features:

- Basic call from PSTN phone to CS1000E phones
- Perform basic call operation: DTMF transmission, voicemail with MWI notification, busy, hold.
- Redirect call between users/clients/endpoints: blind/consultative transfers, call forward all call, busy and no answer.
- Perform codec negotiation
- Perform conferencing

6.2. Test Results

The objectives outlined in section 6.1 were verified and met. The following observations were made during the compliance testing:

- Dial to telephone number which begins with "*", i.e. *xxxxx does not match required format on Avaya CS1000E.
- CPND, Call Party Name Display, does not support on test set up. Telephone number is displayed instead.
- Music on hold is not enabled on CS1000E. i.e. User won't hear music when call is put on hold
- Media Security is not enabled on this test configuration.
- Fax G711 is not supported on Colt Communication system.
- Basic call from Colt to CS1000; CS1000 phone displays CLID of 8100-1 instead of Colt DID number because Colt's INVITE header is Unavailable "Restricted".
- Original SMC & SIPLINE1140 can not blind transfer to PSTN Q02083694

7. Verification Steps

This section includes some steps that can be followed to verify the solution is working.

7.1. Verify that calls are established with two-way voice path when making calls from one CS1000E phone to another on the local CS1000E.

Verify that IP phones, digital, analog (Fax) register successfully show as below:

Verify status of IP phone registered

[nortel@nd1-car1 ~]	\$ isetShow			
==== TPS ====				
1. Set Informati	ion			
IP Address NAT	Model Name Type Reg	gType State	e Regd-TN	FWVsn
47.248.101.117	IP Phone 1120E	1120	Regular online	096-00-01-24 C60
47.248.101.120	IP Phone 2002 Phase 2	2002P2	Regular online	096-00-01-06 DCJ
47.248.101.116	IP Phone 1140E	1140	Regular online	096-00-01-26 C60
47.248.101.115	IP Phone 1220	1220	Regular online	096-00-01-05 C6O

Verify status of digital phone registred:

LD 32 Stat 4 0 7 >ld 32 .stat 4 0 7 00 = UNIT 00 = IDLE (3904) 01 = UNIT 01 = IDLE (3902)

.....

Verify status of Analog (Fax machine registered): LD 32 .stat 4 0 8 00 = UNIT 00 = IDLE (L500) 01 = UNIT 01 = IDLE (L500)

Verify the following basic calls in local CS1000E:

IP phone	call	IP phone
IP phone	call	SIP Line Client
IP Phone	call	Analog/Fax phone
IP Phone	call	Digital phone
SIP Line Client	call	Analog/Fax phone
SIP Line Client	call	Digtal Phone
Analog/Fax phone	call	Digital Phone

QT; Reviewed: SPOC 03/05/2010 Solution & Interoperability Test Lab Application Notes ©2010 Avaya Inc. All Rights Reserved. User can verify the same for calls from oposite direction.

Verify that calls are established with two-way voice path and busy status under CS1000E call server as below:

Verify status of IP phones which are busy

[nortel@nd1-car1 ~]\$ isetShow === TPS ===

Set Information

IP Address	NAT Model Name	Type RegT	ype State	Regd-TN	UNIStimVsn
47.248.101.117	IP Phone 1120E	1120	Regular busy	096-00-01-24	C6O
47.248.101.120	IP Phone 2002 Pha	se 2 2002F	2 Regular b	usy 096-00-01	-06 DCJ
47.248.101.116	IP Phone 1140E	1140	Regular busy	096-00-01-26	C6O
47.248.101.115	IP Phone 1220	1220 R	Legular busy	096-00-01-05	C6O

Verfify status of digital phone is busy

LD 32 .stat 4 0 7 000 = UNIT 00 = BUSY (3904) 01 = UNIT 01 = BUSY (3902)

Verify status analog phone is busy

LD 32 .stat 4 0 8 00 = UNIT 00 = BUSY (L500) 01 = UNIT 01 = BUSY (L500)

Verify status of voice gateway if calls are established between IP phone/SIP line Clients to Analog/Digital phones or call to voice message

>>ld 32		
NPR000		
.stat 4 0 11		
00 = UNIT 00 = BUSY	(TRK)(IPTN REG)
01 = UNIT 01 = BUSY	(TRK)(IPTN REG)
02 = UNIT 02 = BUSY	(TRK)(IPTN REG)
03 = UNIT 03 = BUSY	(TRK)(IPTN REG)

During the call, use pcap tool (ethereal/wireshark) at the TLAN media gateway card, RTP streams are going for call relate to analog, digital or voice message.

QT; Reviewed:	Solution & Interoperability Test Lab Application Notes
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7.2. Verify that calls are established with two-way voice path when making calls from PSTN phone to Avaya phones on the CS1000 through Colt Communication system via configured SIP trunk.

- Verify basic call between PSTN phones and Avaya phones. At the CS1000E SIP Gateway during the call, use pcap tool (ethereal/wireshark) to make sure that all SIP request/response messages
- Verify Codec, SIP trunk status when call is established under CS1000E call server by tracing DID number

LD 80 .trac 0 496856

ACTIVE VTN 096 0 01 06 ORIG VTN 096 0 01 06 KEY 0 SCR MARP CUST 0 DN 496856 TYPE 2002P2 SIGNALLING ENCRYPTION: INSEC MEDIA ENDPOINT IP: 47.248.101.120 PORT: 5200 TERM VTN 100 0 00 31 VTRK IPTI RMBR 100 32 OUTGOING VOIP GW CALL FAR-END SIP SIGNALLING IP: 217.110.230.98 FAR-END MEDIA ENDPOINT IP: 217.110.230.97 PORT: 6478 FAR-END VendorID: Not available MEDIA PROFILE: **CODEC G.711 A-LAW** PAYLOAD 20 ms VAD OFF RFC2833: RXPT 101 TXPT 101 DIAL DN 916139675258 MAIN_PM ESTD TALKSLOT ORIG 21 TERM 53 QUEU NONE CALL ID 511 941

---- ISDN ISL CALL (TERM) ----CALL REF # = 416 BEARER CAP = VOICE HLC = CALL STATE = 10 ACTIVE CALLING NO = 442033496856 NUM_PLAN:E164 TON:INTERNATIONAL ESN:UNKNOWN CALLED NO = 16139675258 NUM_PLAN:E164 TON:INTERNATIONAL ESN:UNKNOWN

• Verify SIP Trunk is released when DID number is released the call by tracing that DID number under CS1000E call server

LD 80

.trac 0 496856 (DID number)

• **IDLE** VTN 096 0 01 06 MARP

8. Conclusion

All of the executed test cases have passed and met the objectives outlined in **Section 6.1**, with some exceptions outlined in **Section 6.2**. The outstanding issues are being investigated by Colt and Avaya design teams. Some of these issues are considered as exceptions. The Colt Communication System is considered compliant with Communication Server 1000E release 6.0.

9. Additional References

Product documentation for Avaya products may be found at: <u>http://support.nortel.com/go/main.jsp</u> [1] *Communication Server 1000E Overview Release 6.0, Revision 03.04, October 2009, Document Number NN43041-110*

[2] Product Compatibility Matrix release 5.0/5.5/6.0, Revision 01.07, February 2010, Document Number NN43001-140

[3] Communication Server 1000 Network Routing Service Fundamentals, Release 6.0, Revision 01.04, Jun 2009, Document Number NN43001-130

[4] Communication Server 1000 Unified Communications Management Common Services Fundamentals, Revision 03.05, February 2010, Document Number NN43001-116

[5] Communication Server 1000 SIP Line Fundamentals, Release 6.0, Revision 01.08, February 10, Document Number NN43001-508

[6] Communication Server 1000 Dialing Plans Reference, Release 6.0, Revision 03.09, June 2009, Document Number NN43001-283

10. Appendixes

Appendix A: CS1000E CPPM Call Server RIs 6.00R Patches Installed

>ld 143 CCBR000 .mdp issp

VERSION 4021 RELEASE 6

ISSUE 00 R +

DepList 1: core Issue: 01 (created: 2009-07-14 16:05:05 (est)) ALTERED

IN-SERVICE PEPS

PAT S	# CR # I	PATCH REF #	NAME D	DATE FI	LENAME SF	PECIN
000	Q00349046-03	ISS1:10F1	p17588_1	05/01/2010	0 p17588_1.cpr	n NO
001	Q01680019	ISS1:10F1	p24307_1	05/01/2010	p24307_1.cpm	NO
002	Q01900523	ISS1:10F1	p26666_1	05/01/2010	p26666_1.cpm	NO
003	Q01983521-04	ISS1:10F1	p27616_1	05/01/2010	0 p27616_1.cpr	n NO
004	Q01849803	ISS1:10F1	p28064_1	05/01/2010	p28064_1.cpm	YES
005	Q01976701-01	ISS1:10F1	p28211_1	05/01/2010	0 p28211_1.cpr	n NO
006	Q02017013-01	ISS1:10F1	p28313_1	05/01/2010	0 p28313_1.cpr	n NO
007	Q02024135-04	ISS1:10F1	p28381_1	05/01/2010	0 p28381_1.cpr	n YES
008	Q02014044	ISS1:10F1	p28461_1	05/01/2010	p28461_1.cpm	NO
009	Q02029209	ISS1:10F1	p28469_1	05/01/2010	p28469_1.cpm	NO
010	Q02023636	ISS1:10F1	p28475_1	05/01/2010	p28475_1.cpm	NO
011	Q02022264	ISS1:10F1	p28486_1	05/01/2010	p28486_1.cpm	NO
012	Q02030977	ISS1:10F1	p28507_1	05/01/2010	p28507_1.cpm	NO
013	Q02020526	ISS1:10F1	p28537_1	05/01/2010	p28537_1.cpm	NO
014	Q02031323-01	ISS1:1of1	p28546_1	05/01/2010	p28546_1.cpm	NO
015	Q02034083	ISS1:10F1	p28553_1	05/01/2010	p28553_1.cpm	YES
016	Q02030235	ISS1:10F1	p28557_1	05/01/2010	p28557_1.cpm	NO
017	Q02028560-04	ISS1:10F1	p28564_1	05/01/2010	0 p28564_1.cpr	n NO
018	Q02034835	ISS1:10F1	p28569_1	05/01/2010	p28569_1.cpm	YES
019	Q02034040	ISS1:10F1	p28577_1	05/01/2010	p28577_1.cpm	NO
020	Q02033951	ISS1:10F1	p28579_1	05/01/2010	p28579_1.cpm	NO
021	Q02033139	ISS1:10F1	p28582_1	05/01/2010	p28582_1.cpm	NO
022	Q02032850	p28472	p28592_1 05	5/01/2010 p	28592_1.cpm	NO
023	Q02018384	ISS1:10F1	p28598_1	05/01/2010	p28598_1.cpm	NO
025	Q02033201	ISS1:10F1	p28631_1	05/01/2010	p28631_1.cpm	YES
026	Q02032155	p28538	p28638_1 05	5/01/2010 p	28638_1.cpm	YES
027	Q02040712	ISS1:10F1	p28653_1	05/01/2010	p28653_1.cpm	NO
028	Q02040015	ISS1:10F1	p28657_1	05/01/2010	p28657_1.cpm	NO
029	Q02038675	ISS1:10F1	p28665_1	05/01/2010	p28665_1.cpm	YES
030	Q02020734-02	2 ISS1:10F1	p28668_1	05/01/2010	0 p28668_1.cpr	n NO
031	Q02038440	ISS1:10F1	p28674_1	05/01/2010	p28674_1.cpm	NO

032	Q02035396	ISS1:10F1	p28675_1 05/01/2010 p28675_1.cpm NO
033	Q02031118	ISS1:10F1	p28680_1 05/01/2010 p28680_1.cpm NO
034	Q02029228-01	ISS1:10F1	p28681_1 05/01/2010 p28681_1.cpm YES
035	Q02038482	ISS1:10F1	p28682_1 05/01/2010 p28682_1.cpm NO
036	Q02039994	ISS1:10F1	p28690_1 05/01/2010 p28690_1.cpm NO
037	Q02024455-01	ISS1:10F1	p28717_1 05/01/2010 p28717_1.cpm NO
038	Q02041981	p28695_1	p28719_1 05/01/2010 p28719_1.cpm NO
039	Q02043226	ISS1:10F1	p28722_1 05/01/2010 p28722_1.cpm NO
040	Q02031359	p28679	p28725_1 05/01/2010 p28725_1.cpm YES
041	Q02031959	ISS1:10F1	p28728_1 05/01/2010 p28728_1.cpm NO
042	Q02033000	ISS1:1of1	p28736_1 05/01/2010 p28736_1.cpm NO
043	Q02039217-03	ISS1:10F1	p28760_1 05/01/2010 p28760_1.cpm NO
044	Q02043669	ISS1:10F1	p28771_1 05/01/2010 p28771_1.cpm NO
045	Q02033321	ISS1:10F1	p28801_1 05/01/2010 p28801_1.cpm NO
046	Q02035555	p28544 p288	13 p28814_1 05/01/2010 p28814_1.cpm NO
047	Q02038393	ISS1:10F1	p28820_1 05/01/2010 p28820_1.cpm NO
048	BV12345	67890 ts	stpatch 05/01/2010 vcm_diag.cpm NO

Appendix B: CS1000E CPPM Signaling Server RIs 6.00.18 Patches Installed

Prod	Product Release: 6.00.18.00						
In sy	stem patche	s: 7					
PAT	CH# NAME	IN <u>.</u>	_SERVICE DATE	SPEC	INS TYPE RPM		
13	p28774_1	Yes	15/10/09 NO	FRU	nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386		
14	p28797_1	Yes	15/10/09 NO	FRU	nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386		
15	p27408_1	Yes	29/10/09 NO	FRU	nortel-cs1000-pi-control-1.00.00.00-00.noarch		
16	p22968_1	Yes	29/10/09 NO	FRU	nortel-cs1000-pi-control-1.00.00.00-00.noarch		
17	p25529_1	Yes	29/10/09 NO	FRU	nortel-cs1000-pi-control-1.00.00.00-00.noarch		
18	p25946_1	Yes	29/10/09 NO	FRU	nortel-cs1000-pi-control-1.00.00.00-00.noarch		
19	p28415_1	Yes	30/11/09 NO	FRU	nortel-cs1000-pi-control-1.00.00.00-00.noarch		

In System service updates: 15

PAT	CH# IN_S	ERVICE	DATE	SPECIN	S REMOVABLE NAME
0	Yes	15/10/09	NO	YES	nortel-cs1000-auth-6.00.18.04-00.i386.000
1	Yes	15/10/09	NO	YES	nortel-cs1000-ftrpkg-6.00.18.04-00.i386.000

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2	Yes	15/10/09	NO	YES	nortel-cs1000-ISECSH-6.00.18.04-00.i386.000
3	Yes	15/10/09	NO	YES	nortel-cs1000-bcc_6-0-6.00.18.09-00.i386.000
4	Yes	15/10/09	NO	YES	nortel-cs1000-dbcom-6.00.18.09-00.i386.000
5	Yes	15/10/09	YES	YES	nortel-cs1000-pd-6.00.18.10-00.i386.000
6	Yes	15/10/09	NO	YES	nortel-cs1000-emWeb_6-0-6.00.18.11-00.i386.000
7	Yes	15/10/09	YES	NO	nortel-cs1000-patchWeb-6.00.18.13-00.i386.000
8	Yes	15/10/09	YES	YES	nortel-cs1000-tps-6.00.18.14-00.i386.000
9	Yes	15/10/09	NO	YES	nortel-cs1000-csmWeb-6.00.18.14-00.i386.000
10	Yes	15/10/09	NO	YES	nortel-cs1000-sps-6.00.18.17-00.i386.000
11	Yes	04/01/10	NO	YES	nortel-cs1000-shared-general-6.00.18.62-00.i386.000
12	Yes	15/10/09	YES	YES	nortel-cs1000-linuxbase-6.00.18.20-00.i386.000
21	Yes	04/01/10	NO	YES	nortel-cs1000-shared-pbx-6.00.18.62-00.i386.000
22	Yes	25/02/10	NO	YES	nortel-cs1000-vtrk-6.00.18.065-016.i386.001

Appendix C: Configure SIP trunk in CS1000 using overlays

Procedure summary

This information is provided as a simple summary of tasks to complete when configuring IP Peer Networking, but it does not replace the full details provided in the IP Peer Networking Guide.

No.	Overlay	Element Management	Action
1	LD 97		Define a virtual super loop
2	LD 17	Select Configuration/D-Channel link	Create a virtual D-channel
3	LD 15	Select Configuration/Customer Explorer link	Define the customer to support ISDN
4	LD 16	Select Configuration/Customer Explorer /Add Route	Create a virtual service route
5	LD 14	Select Configuration/Customer Explorer /Add Trunk	Create virtual trunks

Define a virtual superloop

Use Overlay 97

Prompt	Response	Description
REQ	CHG	
TYPE	SUPL	Configuration data block

SUPL	V100	Virtual	superloop	number	(96	-	112	and	multiple	of	4	for	11C
		system	s.)//CS 1000)E not vlo	op100)							

Create a virtual D-channel

Use Overlay 17

Prompt	Response	Description
REQ	CHG	
TYPE	ADAN	Configuration data block
ADAN	NEW DCH 100	Add a primary D-Channel port 100
СТҮР	DCIP	D-channel is over IP
DES	VIRTUAL_TRK	Description
USR	ISLD	Integrated services signaling link dedicated
IFC	SL1	Interface type is Meridian 1 – Meridian 1
ISLM	4000	Integrated services signaling link maximum
SIDE	USR	Slave to the controller (USR).
RLS	25	X11 software release of far-end.//not need
RCAP	ND2	Name display format 2//not need

Define a customer with ISDN support

Use Overlay 15

Prompt	Response	Description
REQ	NEW	
TYPE	CDB	Customer data block
CUST	0	Customer number
ANAT	1111	ANI Attendant billing number for making ANI calls
ANLD	111	ANI listed directory number
ISDN	YES	Customer is equipped with ISDN.
VPNI	1	Virtual private network identifier//important
PNI	1	Private network identifier.//important

Define a virtual service route

Use Overlay 16

Prompt	Response	Description
REQ	NEW	
TYPE	RDB	Route data block
CUST	0	Customer number
ROUT	100	Route number
DES	VTRK	Designator field for trunk
ТКТР	TIE	TIE trunk only, allowed between SL-1
ICOG	IAO	Incoming and outgoing
VTRK	YES	Virtual trunk route
ZONE	0	Zone for codec selection and bandwidth management
NODE	2000	Node ID of signaling server of this route.
PCID	SIP	Protocol ID for this route
ISDN	YES	ISDN option
MODE	ISLD	Route uses ISDN signaling link
DCH	100	D-channel number for this route
PNI	1	Customer private network identifier.
IFC	SL 1	Interface type : Meridian 1 to Meridian 1
NCNA	YES	Network calling name allowed.
NCRD	YES	Network call redirection.
CHTY	ВСН	B-channel type.
СТҮР	CDP	Coordinated dialing plan

Define virtual trunks

Use Overlay 14

Prompt	Response	Description
REQ	NEW 32	
TYPE	IPTI	IP trunk

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TN	100 0 0 0	Virtual card and channel number
DES	VTRK	Designator field for trunk
CUST	0	Customer number
RTMB	100 1	Route number and member number.
STRI	IMM	Start arrangement incoming
STRO	IMM	Start arrangement outgoing
TGAR	1	Trunk group access restriction.
CHID	1	Channel ID for trunk

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