Abstract

These Application Notes describe a solution comprised of Avaya™ Communication Server 1000E Release 6.0 and Paetec Communication SIP Trunk Product. The Primary focus of testing is the system verification of SIP trunk interoperability which includes the call scenarios 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 Paetec 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 Paetec 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 CPPM Cores 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

- General call processing between systems including:
  - Codec/ptime negotiation and transcoding (G.711 u-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
  - Fax G711 Pass Through (Fax T38 does not support)
  - MedSec is not supported.
  - 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 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
• Paetec Communications provides support to setup, configure, and troubleshoot on carrier switch for the duration of the testing.

1.4. Support
For technical support on Paetec Communication system, please contact Paetec technical support at:

- Toll Free: 800.967.2233
- E-mail: datatac@paetec.com

2. Reference Configuration
Figure 1 illustrates the test configuration used during the compliant testing event between the Communication Server 1000E and Paetec Communication System. This configuration is for a single Communication Server1000E deployment
Figure 1 - Network diagram for Nortel-Paetec LAB setup

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

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. Paetec 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 calls 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

3. Equipment and Software Validated

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

**Call Server:** 6.00R plus latest DEPLIST

**Signaling Server:** SSE 6.00.18 plus latest DEPLIST

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<th>Title</th>
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<td>Ringback tone and speech path support in slow start CFNA scenarios</td>
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<td></td>
<td>Delete element removes all elements-services mapping of associateroles</td>
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<td>MPLR28774</td>
<td>1</td>
<td>Unable to access overlays on inactive core when in split mode with UCM</td>
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<td>MPLR28797</td>
<td>1</td>
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<td>SIP GW patch to remove outbound MCDN from SIP messaging</td>
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<td>Replace domain population in the FROM field</td>
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<td>MPLR25529</td>
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<td>PI: SIP: Partial support of DIVERSION</td>
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Hardware system requirement and theirs soft/loadware version

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<tr>
<td></td>
<td>Signaling Server: 6.00.18</td>
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<td>2004 p2: 0604DCJ (Unistim)</td>
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<td></td>
<td>1220: 062AC6O (Unistim)</td>
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<tr>
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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.

![Figure 3 – Adding a node](image)

Figure 4 describes the 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.
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".

Enable Modem FAX pass through mode for G711 and check V.21 Fax tone Detection (enabled as default) for MPT.

TN of sets with class of service = MPTD (Modem Pass Through Denied)
Figure 6 – Voice Gateway and Codec settings

Figure 7 shows how to configure the Voice gateway and IP phone codec settings. The Paetec Communication network supports both G.711 and G.729. The packet size is set to 20 to match the network also. FAX codec with T.38 FAX is as default in system.
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

Check Modem FAX pass through mode
TN of sets with class of service = MPTD (Modem Pass Through Denied)
Voice gateway and IP phone codec 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.
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.

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.

Figure 12 shows how to configure a zone for new created SIP trunks.
Figure 11 – Zone Basic Property Setting for IP phones

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

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](image-url)
3. **Create trunk**: To create trunk using basic configuration in figure 16
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 shown in figure 17.
Since Media security is not supported under Broadsoft system, Disable Media Security (MSNv) at the Trunk level as show in figure 17.
4. Create Dialing Plan:

Create Special number list:
Create special number list for outgoing dialing plan using figure 18
Launch Element Manager of CS 1000 6.0
Select “Dialing and Numbering plans → Electronic Switched Network → Number Plan (Net) → Access Code 1 (2) → Special Number (SPN).
Create special number SPN 011 (Use RLI_10) for outgoing dialing plan to International calls

![Figure 19 – Special Number for International Calls](image1.png)

Create special number SPN 1800 (Use RLI_10) for outgoing dialing plan to toll free calls

![Figure 20 – Special Number for Tool Free Call](image2.png)
Create special number SPN 411 (Use RLI_10) for outgoing dialing plan to 411 service calls in figure 21

![Image of CS 1000 Element Manager](image1)

**Figure 21 – Special Number for 411 Service Call**

Create special number 911 (use RLI_10) to dial to Emergency service in figure 22

![Image of CS 1000 Element Manager](image2)

**Figure 22 – Special Number for Emergency 911 dialing**
Create Numbering Plan Area Code:

Create NPA numbers for outgoing.

NPA_1713: Create NPA_1713 for outgoing calls to numbers beginning with; Figure 23

NPA_1613: Create NPA 1613 to dial to national DID numbers beginning with 613; Figure 24

Figure 23 – Create NPA_1713 for outgoing calls

Figure 24 – Create NPA_1613 for outgoing calls to national numbers
Create RLI_10 for outgoing calls (Use route_100 and DMI_10), figure 25

![Data Entry of a Route List Block](image1)

**Figure 25 - RLB for Outgoing calls**

Create Local Steering Code

Create LSC_713 to terminate the incoming calls (Use DMI_3); Figure 26

![Local Steering Code](image2)

**Figure 26 – Create LCS_713 for incoming call**
Create Digit Manipulation Block

DMI_10: Digit Manipulation Block configuration for Outgoing calls; figure 27

![Digit Manipulation Block for Outgoing calls](image1)

**Figure 27 – Digit Manipulation for Outgoing calls**

DMI_3: Digit Manipulation Block configuration for incoming calls; figure 28

![Digit Manipulation Block for Incoming calls](image2)

**Figure 28 – Digit Manipulation for Incoming calls**
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 Channel for communication between CS1000 and Callpilot system

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 30.
Create Service DN for Voice Messaging system, Figure 31
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
SDBN 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
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
CROS 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
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
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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
  MFVN OVF OVF OVF ATN
  MFCG OVF OVF OVF ATN
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RCLE ATN OVF ATN ATN
CONG OVF
DLT OVF
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**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
- EXT 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 VoIP
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
CFSW NO
PCML A
ALRM YES
ERRM ERR BUG AUD
DTRB 100
ABCD NO
TMRK 128
FCDR OLD
PCDR NO
TPO  NO
TSE  NO
CLID NO
DURS 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
XCT
CONF
MGTD SD 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
4.4 CS1K Tandem Configuration

This configuration is for the deployment model of 2 or more CS1000 with Paetec 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.
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 33

Support registration
**Figure 33 – 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 34.
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 35; Also click on Options and edit the Remote Capabilities (RCAP). Enable MWI if CS1K hosted voice mail will be used.

---

**Figure 34 – SIP Gateway Settings**
Figure 35 – D-Channels Property Configuration
3. **Create Route**

Create route 101 using DCH 101 for SIP trunks figure 36

![Diagram](image)

**Figure 36 – Route Property Configuration**

Configure Route 101 for SIP trunks continue, figure 37
Figure 37 – Route Property Configuration Details (cont.)

4. Create Trunk (figure 38)

Since Media security is not support under Paetec system, Disable Media Security (SRTP) at the Trunk level as show in figure 38

Figure 38 – Trunk Configuration Details
Figure 39 – Class of Services configuration Details (cont.)

- Configure Dialing Plan for CS1000E_A

Create Location Code:

Create LOC 521 for basic outgoing calls to CS1000E_B (Use RLI_5; DMI_0); Figure 40
Create Home Location Code

Create HLOC_613 for incoming calls from CS1000E_B and outgoing calls to PSTN; Figure 41

**Figure 40 – LOC_521 for basic outgoing calls to CS1000EB**

Create HLOC_511 for basic incoming call from CS1000E_B (DMI_4); Figure 42

**Figure 41 – HLOC: 613 to tandem calls from CS1000B to PSTN**

Create HLOC_511 to terminate calls from CS1000E_B

**Figure 42 – HLOC_511 to terminate calls from CS1000E_B**
Create Distant Steering Code

Create DSC_713 (RLI_6) to receive Calls from PSTN and tandem to CS1000E_B; Figure 43

Figure 43 – DSC_713 to receive calls from PSTN and tandem to CS1000E_B
Create DSC_711 to tandem calls from CS1000E_A to CS1000E_B; Figure 44

Figure 44 – DSC_711 to tandem calls to CS1000E_B.
Create Route List Block

Create RLI_5 for basic outgoing call to CS1000E_B (Use route 101, DMI_0); Figure 45

Figure 45 – Create RLI_5 for basic outgoing calls to CS1000E_B
Create RLI_6 to for incoming calls from PSTN and outgoing calls to CS1000E_B (Use route 101, DMI_6); Figure 46

Figure 46 – RLI_6 to tandem calls from PSTN to CS1000E_B
Create Digit Manipulation Block

DMI_13 for Incoming calls from CS1000EB and Outgoing to PSTN; Figure 47

![Digit Manipulation Block](image)

Figure 47 – DMI_13 for incoming from CS1000E_B and outgoing calls to PSTN
Create DMI_4 for incoming calls from CS1000E_B; Figure 48

Figure 48 – DMI_4 to terminate calls from CS1000E_B

Create DMI_6: (Delete: 6) for incoming calls from PSTN and tandem calls to CS1000E_B; Figure 49

Figure 49 – DMI_6 for incoming calls from PSTN and tandem to 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 50.

![Figure 50 – Node Configured as Enterprise Service SIP Gateway](image-url)
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 51.
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 53
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 54

Configure Route 101 for SIP trunks, figure 55
4. Create Trunk (figure 56)

Figure 56 – Trunk Property Configuration
Disable Media Security (sRTP) at the trunk level, figure 57

![Class of Service Configuration](image)

**Figure 57 – Class of Service Configuration**
5. Create Dialing Plan

Create Location Code

Create LOC 511 (Use RLI_5) for outgoing calls to CS1000E_A; Figure 58

Figure 58 – LOC_511 for outgoing calls to CS1000E_A

Create Numbering Plan Area Code

Create NPA_613 (RLI_70) for outgoing calls to PSTN through CS1000E_A; Figure 59

Figure 59 – NPA_613 for outgoing calls to PSTN
Create Home Location Code

Create HLOC_521 (Use DMI_4) for incoming calls from CS1000E_A; Figure 60

Figure 60 – Create HLOC 521 for incoming calls from CS1000E_A
Create Route List Block

Create RLI_5 for outgoing calls to CS1000E_A (Use DMI_0)

Figure 61 – RLI_5 (use DMI_0) for Outgoing calls to CS1000E_A
4.4.2 Configure SIP Proxy Server (SPS)

Create gateway endpoints on SPS

Figure 62 – SIP Gateway Endpoint Creation

Create routing entries for each of gateway endpoints on SPS, figure 63

Figure 63 – Routing Entries for Gateway Endpoints
4.4.3 CS1000E SIPLINE CONFIGURATION

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

4.4.3.1 Configure SIP LINE CS1000E in Element Manager

Figure 64 show how to add SIP LINE Node 1002 under System -> IP Network -> IP Telephony Nodes

Figure 64 – IP Telephony Nodes

Figure 65, 66 and 67 show how to set up the SIP LINE Node 1002 configuration details. SAVE and SYNC are required – And then APPSTART RESTART on SLG server.

Figure 65 – Node Configuration Details
4.4.3.2 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 below 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
4.4.3.2.1 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

4.4.3.2.2 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

4.4.3.2.3 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

4.4.3.2.4 Configure VAS ID for AML link in LD 17

LD 17
REQ CHG
4.4.3.2.5 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

4.4.3.2.6 Configure SIPL trunks

LD 14
REQ NEW 256 – e.g. create 256 trunks
TYPE IPTI
TN 124 0 0 0 - starting TN for virtual trunks
DES SIPL
CUST 0
RTMB 11 1 – route number and member
CHID 1
TGAR 0
STRI IMM
STRO IMM
CLS UNR
4.4.3.2.7 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

```bash
=== VTRK ===

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

====== AML Info ======

<table>
<thead>
<tr>
<th>hAppBlk</th>
<th>TaskName</th>
<th>Tid</th>
<th>LinkState</th>
<th>NumRetry</th>
<th>LinkNum</th>
<th>Trace</th>
</tr>
</thead>
<tbody>
<tr>
<td>0x1226c80</td>
<td>SLG</td>
<td>0xfb00</td>
<td>Up</td>
<td>0</td>
<td>32</td>
<td>0</td>
</tr>
</tbody>
</table>
```

4.4.3.2.8 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

4.4.3.2.9 Configure UEXT for SIPL client

LD 11
REQ NEW
TYPE UEXT

TN 104 0 00 11 - 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

4.4.3.2.10 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

4.4.4 SMC3456 softphone

Link to download:

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
On the top menu bar, go to FILE -> PREFERENCES -> ADVANCED -> LOGIN SERVER

⇒ No login server available

4.4.4.1 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 70.

![Account Settings](image)

**Figure 70 – Accounting Settings**

The created account is appeared as figure 71.

![SIP Account](image)

Port must be the same, e.g. 5060

**Figure 71 – Topology SIP Account Settings**
Figure 72 shows how to set SIP account details by clicking on the Account menu tap.

**Figure 72 – SIP Account Details Setting**

Figure 73 shows the newly created SIP account.
4.4.5 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  104 0 0 0
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
SUBLR DFLT MWI RGA CWI MSB
UXTY
NUID

Figure 73 – New Created SIP Account
5  Paetec Communication System configuration

Paetec 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 Paetec 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.

6.4 General test approach

The general test approach was to have Paetec Sonus system connected to CS1000E SIP Gateway using Sonus IP address. The SIP trunk communication should be established between CS1000E and Paetec Sonus system. Calls can be made from end to end, i.e. PSTN phone can call through created route from Paetec 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

6.5 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.
- CS1000E_PHONE1 calls CS1000E_PHONE2. Call is established when CS1000E_PHONE2 answers. The CS1000E_PHONE1 does blind transfer to PSTN1. User expects that PSTN1 rings; CS1000E_PHONE2 hears ringback tone. Call is established two ways speech path between CS1000E_PHONE2 and PSTN1. However, Paetec does not support UPDATE, this will cause lack of ring back tone or early media. Following are some scenarios where the lack of ring back tone or early media will apply

Scenario 1: PSTN 1 calls CS1K set 2. CS1K set 2 does a blind transfer to PSTN 3.
- Issue: when CS1K set 2 completes the blind transfer and PSTN 3 is still ringing, due to lack of UPDATE support; user at PSTN 1 won’t be able to hear ring back tone or early
media because CS1K cannot send SDP of PSTN 1 to PSTN 3 via UPDATE (call not yet connected on 2nd call leg between CS1K set 2 and PSTN 3; so we must send the SDP via UPDATE).

Scenario 2: CS1K set A calls CS1K set B. Set B does a blind transfer to PSTN
- Issue: when CS1K set completes blind transfer; and PSTN is still ringing; user on set A won't be able to hear ring back tone - same technical issue as Scenario 1 above.

Scenario 3: PSTN calls CS1K set A. Set A does a blind transfer to CS1K set C
- Issue: PSTN won't hear ring back tone when CS1k set A completes the transfer (and set C is still ringing).

Scenario 4: PSTN 1 calls 1k set A...set A does a local blind transfer to set B..B does CFNA to PSTN 2
- Issue: PSTN won't hear ring back tone when CS1K set A completes the transfer; as well as when PSTN 2 is ringing.
We have observed Paetec send 183+sdp when calls are made to PSTN...The workaround patch on the CS1000 SipGW has the CS1K treats an incoming 183+sdp as 180 Ringing and generates a local ring back tone.

- **NOTE**: With this workaround, **early media support is not possible**...ie. CS1K will treat an **incoming** 183+sdp for ANY call as a 180 Ringing without sdp. Therefore, even for a simple call such as: CS1K set A calls PSTN...PSTN sends 183+sdp providing an announcement. The CS1K user will just hear the ring back tone and not the announcement.

**Results** with the workaround patch applied:
In all the above 4 scenarios; when blind transfer is completed; the originating user will hear ring back tone....and once the call has been picked up on far end; there will be 2 way speech. Please refer to CR Q02128833 which blocks test case

- PSTN1 calls CS1000E_PHONE which is set up to call forward all call to PSTN2. User expects that call will be forwarded to PSTN2 right after PSTN1 calls CS1000E_PHONE; PSTN2 will ring and PSTN1 will ringback tone but PSTN2 does not ring, PSTN1 shows “Release and Try again”. Issue is resolved after changing configuration on Paetec’s SBC.

- PSTN1 calls CS1000E_PHONE which is set up to call forward no answer to PSTN2. User expects that call will be forwarded to PSTN2 after CS1000E_PHONE rings three times and then PSTN2 will ring and PSTN1 will ring back tone. Actually, PSTN2 does not ring, PSTN1 shows “Release and Try again”. Issue is resolved after changing configuration on Paetec’s SBC.

- PSTN1 calls CS1000E_PHONE which is set up to call forward busy to PSTN2. User expects that call will be forwarded to PSTN2 if PSTN1 calls to CS1000E_PHONE which is busy. PSTN2 will ring and PSTN1 will ring back tone. Actually, PSTN2 does not ring, PSTN1 shows “Release and Try again”. Issue is resolved after changing configuration on Paetec’s SBC.

- PSTN1 calls CS1000E_PHONE which is set up call waiting. Call is established successfully when CS1000E_PHONE answers PSTN1. PSTN2 calls the same CS1000E_PHONE which is busy with PSTN1. CS1000E_PHONE will receive a call
waiting tone/indication but there is no waiting tone/indication on CS1000E_PHONE. Please ensure DID numbers also are enabled the call waiting feature on Paetec. Issue is resolved after call waiting is enabled on Paetec.

- At this moment, SIPLINE clients testing result based on SU nortel-cs1000-vtrk-6.00.18.065-016.i386.001.ntl is installed on SS_Carrier and nortel-cs1000-vtrk-6.00.18.63-06.i386.001.ntl is installed on SLG, as CS1000E designer’s suggestion. Please refer to Appendix B for details of patches installation. If SU “nortel-cs1000-vtrk-6.00.18.065-016.i386.001.ntl” is installed on both SS_carrier and SLG, all simple calls relate to SIPLINE will be failed between Unistim calls SIPLINE, PSTN calls SIPLINE, SIPLINE1 calls SIPLINE2 as one hangs up call. Another is NOT released. Please refer to CR Q02129692. 80% SIPLINE test cases are also failed with this patch as basic SIPLINE features do not work such as conference, blind transfer, consult transfer, call forward no answer.

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

7.4 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 Information
IP Address   NAT  Model Name  Type  RegType  State        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  C60

Verify status of digital phone registered:

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----------------------Digital Phone
Analog/Fax phone-------------call----------------------Digital Phone
User can verify the same for calls from opposite 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
------------
<table>
<thead>
<tr>
<th>IP Address</th>
<th>NAT</th>
<th>Model Name</th>
<th>Type</th>
<th>RegType</th>
<th>State</th>
<th>Regd-TN</th>
<th>UNIStimVsn</th>
</tr>
</thead>
<tbody>
<tr>
<td>47.248.101.117</td>
<td></td>
<td>IP Phone 1120E</td>
<td>1120</td>
<td>Regular busy</td>
<td>096-00-01-24</td>
<td>C6O</td>
<td></td>
</tr>
<tr>
<td>47.248.101.120</td>
<td></td>
<td>IP Phone 2002 Phase 2</td>
<td>2002P2</td>
<td>Regular busy</td>
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Verify 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.

7.5 Verify that calls are established with two-way voice path when making calls from PSTN phone to Avaya phones on the CS1000 through Paetec 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 Paetec and Avaya design teams. Some of these issues are considered as exceptions. The Paetec Communication System is considered compliant with Communication Server 1000E release 6.0.

9 Additional References

Product documentation for Avaya products may be found at:
[http://support.nortel.com/go/main.jsp](http://support.nortel.com/go/main.jsp)

[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*

10 Appendixes

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

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QT; Reviewed: Solution & Interoperability Test Lab Application Notes 88 of 95
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### Appendix B: CS1000E CPPM Signaling Server Rls 6.00.18

**Patches Installed**

Product Release: 6.00.18.00

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SIPLINE Gateway patches:

```
[admin@sl-node1 ~]$ pstat
```

Product Release: 6.00.18.00

In system patches: 2

<table>
<thead>
<tr>
<th>PATCH#</th>
<th>NAME</th>
<th>IN_SERVICE</th>
<th>DATE</th>
<th>SPECINS TYPE</th>
<th>RPM</th>
</tr>
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<tbody>
<tr>
<td>9</td>
<td>p28774_1</td>
<td>Yes</td>
<td>25/03/10</td>
<td>FRU</td>
<td>nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386</td>
</tr>
<tr>
<td>10</td>
<td>p28797_1</td>
<td>Yes</td>
<td>25/03/10</td>
<td>FRU</td>
<td>nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386</td>
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</table>

In System service updates: 9

<table>
<thead>
<tr>
<th>PATCH#</th>
<th>IN_SERVICE</th>
<th>DATE</th>
<th>SPECINS</th>
<th>REMOVABLE</th>
<th>NAME</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>Yes</td>
<td>25/03/10</td>
<td>YES</td>
<td>YES</td>
<td>nortel-cs1000-linuxbase-6.00.18.63-02.i386.000</td>
</tr>
<tr>
<td>1</td>
<td>Yes</td>
<td>25/03/10</td>
<td>YES</td>
<td>YES</td>
<td>nortel-cs1000-patchWeb-6.00.18.63-01.i386.000</td>
</tr>
</tbody>
</table>
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.

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

Define a virtual superloop

Use Overlay 97

<table>
<thead>
<tr>
<th>Prompt</th>
<th>Response</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>REQ</td>
<td>CHG</td>
<td></td>
</tr>
<tr>
<td>TYPE</td>
<td>SUPL</td>
<td>Configuration data block</td>
</tr>
<tr>
<td>SUPL</td>
<td>V100</td>
<td>Virtual superloop number (96 - 112 and multiple of 4 for 11C systems.)/CS 1000E not vloop100</td>
</tr>
</tbody>
</table>
Create a virtual D-channel

*Use Overlay 17*

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

Define a customer with ISDN support

*Use Overlay 15*

<table>
<thead>
<tr>
<th>Prompt</th>
<th>Response</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>REQ</td>
<td>NEW</td>
<td></td>
</tr>
<tr>
<td>TYPE</td>
<td>CDB</td>
<td>Customer data block</td>
</tr>
<tr>
<td>CUST</td>
<td>0</td>
<td>Customer number</td>
</tr>
<tr>
<td>ANAT</td>
<td>1111</td>
<td>ANI Attendant billing number for making ANI calls</td>
</tr>
<tr>
<td>ANLD</td>
<td>111</td>
<td>ANI listed directory number</td>
</tr>
<tr>
<td>ISDN</td>
<td>YES</td>
<td>Customer is equipped with ISDN.</td>
</tr>
<tr>
<td>VPNI</td>
<td>1</td>
<td>Virtual private network identifier//important</td>
</tr>
<tr>
<td>PNI</td>
<td>1</td>
<td>Private network identifier.//important</td>
</tr>
</tbody>
</table>
Define a virtual service route

Use Overlay 16

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

*Use Overlay 14*

<table>
<thead>
<tr>
<th>Prompt</th>
<th>Response</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>REQ</td>
<td>NEW 32</td>
<td></td>
</tr>
<tr>
<td>TYPE</td>
<td>IPTI</td>
<td>IP trunk</td>
</tr>
<tr>
<td>TN</td>
<td>100 0 0 0</td>
<td>Virtual card and channel number</td>
</tr>
<tr>
<td>DES</td>
<td>VTRK</td>
<td>Designator field for trunk</td>
</tr>
<tr>
<td>CUST</td>
<td>0</td>
<td>Customer number</td>
</tr>
<tr>
<td>RTMB</td>
<td>100 1</td>
<td>Route number and member number</td>
</tr>
<tr>
<td>STRI</td>
<td>IMM</td>
<td>Start arrangement incoming</td>
</tr>
<tr>
<td>STRO</td>
<td>IMM</td>
<td>Start arrangement outgoing</td>
</tr>
<tr>
<td>TGAR</td>
<td>1</td>
<td>Trunk group access restriction</td>
</tr>
<tr>
<td>CHID</td>
<td>1</td>
<td>Channel ID for trunk</td>
</tr>
</tbody>
</table>
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