Nortel Communication Server 1000M Release 4.0 using SIP Trunk to Cisco CallManager Release 4.1(3)

Introduction

- This is an application note for interoperability connectivity of Nortel Communication Server 1000 (formerly known as Succession 1000) PBX with Cisco CallManager Release 4.1(3)SR2 via SIP trunk.
- The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability with the Nortel CS1000 PBX configured as a SIP Trunk with the Cisco CallManager 4.1(3)SR2 release.
- On Cisco CallManager SIP Trunk Configuration web page defined for the Nortel PBX, please ensure the following “Media Termination Point Required” box is checked.

Network Topology

Figure 1. Network Topology or Test Setup

SIP Call Setup End-to-End Configuration
Limitations

- Cisco CallManager and Nortel CS1K systems use different method of passing the name and number information across the SIP trunk. Cisco CallManager use Remote-Party-Id (RPID) field in the SIP header and Nortel use P-Asserted-Id field (PAI) in the SIP header instead. Since both parties do not understand each other method, they used the name and number information in the SIP FROM: header instead.

- For features such as CLIR, CNIR, COLR and CONR, both systems set the SIP FROM: header to be ”Anonymous” and the privacy setting on the Remote-Party-Id or P-Asserted-Id field set to be restricted. Since both parties do not understand each other method, they used name and number information in the SIP FROM: header instead.

- Cisco CallManager support the both Alerting/Connected Name and Number using the Remote-Party-Id tag in the SIP Header. Nortel support the Connected Name and Number by using the P-Asserted-Id tag in the SIP Header. However, Nortel PBX do not support Alerting name and number information (there is no P-Asserted-Id tag within the SIP 180/183 alerting message).

- Call Transfer and Call Forward features work but the phone’s name and number display update capability do not since the two systems use different method of passing the name and number across the SIP trunk.

- Call Completion (Callback) Feature is not supported on either systems (Cisco CallManager or Nortel CS1000M) using standard SIP protocol.

- MWI ON/OFF messages doesn’t work across the SIP Trunk connection between the two systems.

- End-to-end DTMF relay signaling doesn’t work between the two systems. Cisco CallManager use RFC2833 method of passing the DTMF digits across SIP trunk and Nortel use SIP INFO method of passing the DTMF digits across SIP Trunk.

- For the Cisco CallManager SIP Trunk configuration, “Media Termination Point Required”, box must be checked in order for the two systems to communicate.

System Components

Hardware Requirements

- Cisco CallManager MCS server, Unity server, and Cisco 7960 and 7940 phones
- Nortel Communication System 1000M (which includes Call Server, Signaling Server and Media gateway) and Nortel’s i2004/i2002 IP phones

Software Requirements

- Cisco CallManager Release 4.1(3)xr2
- Nortel Succession 4.0 Release

Features

- CLIP-Calling Line (Number) Identification Presentation (Please see the Limitation section)
- CLIR-Calling Line (Number) Identification Restriction (Please see the Limitation section)
- CNIP-Calling Name Identification Presentation (Please see the Limitation section)
- CNIR-Calling Name Identification Restriction (Please see the Limitation section)
- Alerting Name (Please see the Limitation section)
- CT-Call Transfer by Join (Please see the Limitation section)
- CFU-Call Forwarding Unconditional (Please see the Limitation section)
- CFB-Call Forwarding Busy (Please see the Limitation section)
- COLP-Connected Line (Number) Identification Presentation (Please see the Limitation section)
• COLR- Connected Line (Number) Identification Restriction (Please see the Limitation section)
• CONP-Connected Name Identification Presentation (Please see the Limitation section)
• CONR-Connected Name Identification Restriction (Please see the Limitation section)

**Features Not Supported**

• MWI- Message Waiting Indication (lamp ON, lamp OFF) across the SIP Trunk
• Call Completion (Callback; Automatic Callback)
• Call Completion
• End-to-End DTMF signaling

**Configuration**

**Nortel Communication Server 1000 PBX Configuration Sequence and Tasks**

**Call Server Setup via SSC card console**

1. LD 17 – Configure the IP D-channel (signaling channel) between the Call Server and the Signaling Server
2. LD 97 – Configure the Super-loop for the Virtual Trunks
3. LD 14 – Configure the SIP Virtual Trunks to the Signaling Server
4. LD 14 – Configure the Virtual Gateway Trunks
5. LD 11 – Configure for the Virtual lines for the Nortel IP phone (i200x series)
6. LD 16 – Configure the SIP route
7. LD 86 – Configure the Route List Block for the Virtual Trunk route
8. LD 87 – Configure CDP steering codes

**Signaling Server Setup via the Nortel Element Manager**

9. Configure the Zones
10. Configure a new IP Telephony Node summary
11. Configure the Node section
12. Configure the VGW and IP phone codec profile section
13. Configure the Quality of Service (QoS) section
14. Configure LAN Configuration section
15. Configure the SIP GW Setting section
16. Configure the Card section for the MC-32 VGMC card section
17. Configure the Signaling Server section

**NRS (Network Routing Server)**

18. Configure the System Wide Settings
19. Configure the NRS Server Settings
20. Configure a Service Domain
21. Configure a L1 Domain (UDP)
22. Configure a L0 Domain (CDP)
23. Configure a SIP gateway
24. Configure the Routing Entries

**Cisco CallManager Setup**

25. Create the Media Resource Group and Media Resource Group List for the MTP requirement
26. Add an SIP Trunk for the Nortel CS1000 PBX under the Device pull-down menu
27. Add a Route Pattern to reach the Nortel’s phone DN extensions
28. Add a Route Pattern to reach the Nortel CallPilot VoiceMail system
29. Add a Route Pattern to PSTN via Nortel CS1000M
30. Configure Cisco 7960 phone and line DN

**Configuration Menus and Commands**

Nortel Communication Server 1000 (CS1000) Configuration

**Call Server Setup:**

1. LD 17 – Configure the IP D-channel (signaling channel) between the Call Server and the Signaling Server

   ```
   >ld 22
   PT2000
   
   REQ prt
   TYPE adan dch 3
   
   ADAN       DCH 3
   CTYP DCIP
   DES IP_Trunk_DCH
   **USR ISLD**
   ISLM 4000
   SSRC 1800
   OTBF 32
   NASA NO
   **IFC SL1**
   CNEG 1
   RLS ID 4
   **RCAP ND2**
   MBGA NO
   H323
   OVLR NO
   OVLS NO
   ```

2. LD 97 – Configure the Super-loop for the Virtual Trunks

   ```
   >ld 97
   SC SYS000
   MEM AVAIL: (U/P): 2854769  USED U P: 182454 59352  TOT: 3096575
   DISK RECS AVAIL: 1152
   REQ prt
   TYPE supl
   SUPL
   
   SUPL  SUPT SLOT XPEC0  XPEC1
   
   000  STD LEFT 01 0 1  -- - -
   004  STD LEFT 02 0 1  -- - -
   008  STD LEFT 03 0 1  -- - -
   012  STD LEFT 04 0 1  -- - -
   016  STD LEFT 05 0 1  -- - -
   032  STD LEFT 06 0 3  -- - -
   036  STD LEFT 07 0 3  -- - -
   040  STD LEFT 08 0 3  -- - -
   ```
3. **LD 14 – Configure the SIP Virtual Trunks to the Signaling Server (One trunk = one line connection)**

>ld 20
REQ: prt
TYPE: tnb
TN   62 0 0 0

[**SIP Virtual trunk to Signaling Server**]

DES  SIP_IP_VTRK
TN   062 0000 VIRTUAL
TYPE IPTI
CDEN 8D
CUST 0
XTRK VTRK
ZONE 000
LDOP BOP
TIMP 600
BIMP 600
AUTO_BIMP NO
TRK ANLG
NCOS 0
RTMB 101
CHID 1
TGAR 1
STRI/STRO IMM IMM
SUPN YES
AST NO
IAPG 0
CLS  CTD DTN WTA LPR APN THFD
   P10 NTC MID
TKID
AACR NO
DATE 25 FEB 2005

NACT
4. LD 14 – Configure the Virtual Gateway Trunks (upto 32 trunks per MC-32)

```
>ld 20
REQ: prt
TYPE: tnb
TN 3
CDEN
CUST
DATE
PAGE
DES

DES 192.168.1.2
TN 003 000 00 ➤ 1st channel define on the gateway
TYPE VGW
CUST 0
XTRK MC32
ZONE 000

DES 192.168.1.2
TN 003 000 01 ➤ 2nd channel define on the gateway
TYPE VGW
CUST 0
XTRK MC32
ZONE 000
```

5. LD 11 – Configure for the Virtual lines for the Nortel IP phones (phone A and phone B)

Phone A (i2004)

```
>ld 11
SL1000
MEM AVAIL: (U/P): 2854769 USED U P: 182454 59352 TOT: 3096575
DISK RECS AVAIL: 1152
DIGITAL TELEPHONES AVAIL: 6 USED: 2 TOT: 8
IP USERS AVAIL: 6 USED: 2 TOT: 8
BASIC IP USERS AVAIL: 7 USED: 1 TOT: 8
ACD AGENTS AVAIL: 10 USED: 0 TOT: 10
PCA AVAIL: 0 USED: 0 TOT: 0
AST AVAIL: 1 USED: 0 TOT: 1
TNS AVAIL: 2405 USED: 95 TOT: 2500
DATA PORTS AVAIL: 2500 USED: 0 TOT: 2500

REQ: prt
TYPE: tnb

TN 61 0 0 02
DATE
PAGE
DES
```
DES I2004
TN 061 0002 VIRTUAL
TYPE I2004
CDEN 8D
CUST 0
ZONE 000
FDN 2500
TGAR 1
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
LNRS 16
XLST
CLS  CTD FBA WTA LPR MTD FNA HTA TDD HFD CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRA NID OLD VCE DRG1
POD DSX VMD CMSD SLKD CCSD SWD LNA CNDA
CFTA SFD MRD DDV CNIA CDCA MSID DAPA BFED RCBD
ICDD CDMD LLCN MCTD CLBD AUTU
GPUD DPUD DNDA CFXA ARHD CLTD ASCD
CFLA CPF ABD CFHD FICD NAIM BUZZ AGRD MOAD
AHDD DDGA NAMA
DRDD EXR0
USRD ULAD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
VOLA VOUD CDMR
CPND_LANG ENG
RCO 0
EFD 2500
HUNT 2500
EHT 2500
LHK 0
LPK 1
PLEV 02
CSDN
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
DNDR 0
KEY 00 SCR 2201 0 MARP
CPND
NAME ZEUS_2201
XPLN 8
DISPLAY_FMT FIRST, LAST
01
02
03 MIK
04 MCK
05
06
07
08
09
10
11
12
13
14
15
16 MWK 2500
17 TRN
18 AO6
19 CFW 16 2500
20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN
27
28
29
30
31
DATE 30 NOV 2005
NACT

Phone B (i2002):

REQ: prt
TYPE: tnb
TN 61 0 0 01
DATE
PAGE
DES

DES I2002
TN 061 0 00 01 VIRTUAL
TYPE I2002
CDEN 8D
CUST 0
ZONE 000
FDN 3690
TGAR 1
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
LNRS 16
XLST
CLS CTD FBD WTA LPR MTD FNA HTA TDD HFD CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRA NID OLD VCE DRG1
POD DSX VMD CMSD SLKD CCSD SWD LNA CNDA
CFTA SFD MRD DDV CNIA CDCA MSID DAPA BFED RCBR
ICDD CDMD LLCP MCTD CLBD AUTU
GPUD DPUD DNDA CFJA ARHD CLTD ASCD
CFPA CPFA ABDD CFHD FICD NAID BUZZ AGRD MOAD AHD
DDGA NAMA
DRDD EXR0
USRD ULAU RTDD RBDD RBHD PGND OCBN FTLD FTTC DNDY DNO3 MCBN
VOLA VOUU CDMR
CPND_LANG ENG
RCO 0
EFD 3690
HUNT 3690
EHT 3690
LHK 0
LPK 1
PLEV 02
CSDN
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
DNDR 0
KEY 00 SCR 2201 0 MARP
CPND
NAME ZEUS_2201
XPLN 8
DISPLAY_FMT FIRST,LAST
01
02
03 MIK
04 MCK
05
06
07
08
09
10
11
12
13
14
15
16 MWK 3690
17 TRN
18 AO6
19 CFW 16 3690

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6. LD 16 – Configure the SIP route

```plaintext
>ld 21
PT1000

REQ: prt
TYPE: rdb
CUST 0
ROUT 10

TYPE RDB
CUST 00
DMOD
ROUT 10
DES SIP_TIE
TKTP TIE
NPI.TBL_NUM 0
ESN NO
CNVT NO
SAT NO
RCLS EXT
VTRK YES
ZONE 000
PCID SIP
CRID YES
NODE 101
DTRK NO
ISDN YES
MODE ISLD
DCH 3
IFC SL1
PNI 00001
NCNA YES
NCRD YES
TRO NO
FALT NO
CTYP UKWN
INAC NO
```
7. LD 86 – Configure the Route List Block for the Virtual Trunk route

> ld 86
ESN000

MEM AVAIL: (U/P): 2819994 USED U/P: 223389 69576 TOT: 3112959
DISK RECS AVAIL: 1152
REQ prt
CUST 0
FEAT rlb
RLI 10

RLI 10
ENTR 0
LTER NO
ROUT 10
TOD 0 ON 1 ON 2 ON 3 ON
 4 ON 5 ON 6 ON 7 ON
VNS NO
SCNV NO
CNV NO
EXP NO
FRL 0
DMI 0
ISDM 0
FCI 0
FSNI 0
SBOC NRR
8. LD 87 – Configure CDP steering codes

>ld 87
ESN000

MEM AVAIL: (U/P): 2819994 USED U P: 223389 69576 TOT: 3112959
DISK RECS AVAIL: 1152
REQ prt
CUST 0
FEAT cdp
TYPE dsc
DSC
DSC 35 ➔ 35xx Route to Unity VM send out via SIP route
FLEN 0
DSP LSC
RLI 10 ➔ SIP Route List
NPA
NXX

DSC 36 ➔ 36xx Route to CCM extension, sent out via SIP route
FLEN 0
DSP LSC
RLI 10 ➔ SIP Route List
NPA
NXX

9. LD 90 – Configure BARS/NARS route to PSTN

>ld 90
ESN000

MEM AVAIL: (U/P): 2819994 USED U P: 223389 69576 TOT: 3112959
DISK RECS AVAIL: 1152
REQ prt
CUST 0
FEAT net
TRAN ac1
TYPE npa

NPA

NPA 1212
RLI 2
SDRR NONE
ITEI NONE

NPA 1408
RLI 2
SDRR NONE
ITEI NONE

NPA 1800
RLI 2
SDRR NONE
ITEI NONE

NPA 1808
RLI 7
SDRR NONE
ITEI NONE

MEM AVAIL: (U/P): 2819994   USED U/P: 223389 69576   TOT: 3112959
DISK RECS AVAIL: 1152
REQ
Signaling Server Setup:

10. Configure the Zones

<table>
<thead>
<tr>
<th>Zone Basic Property and Bandwidth Management</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Input Description</strong></td>
</tr>
<tr>
<td>Zone Number (ZONE):</td>
</tr>
<tr>
<td>Intrazone Bandwidth (INTRA_BW):</td>
</tr>
<tr>
<td>Intrazone Strategy (INTRA_STGY):</td>
</tr>
<tr>
<td>Interzone Bandwidth (INTER_BW):</td>
</tr>
<tr>
<td>Interzone Strategy (INTER_STGY):</td>
</tr>
<tr>
<td>Resource Type (RES_TYPE):</td>
</tr>
<tr>
<td>Branch Office Support (ZURN):</td>
</tr>
<tr>
<td>Description (ZDES):</td>
</tr>
</tbody>
</table>

[Submit] [Refresh] [Delete] [Cancel]
11. Configure a new IP Telephony Node summary
12. Configure the Node section
13. Configure the VGW and IP phone codec profile section
<table>
<thead>
<tr>
<th>Codec</th>
<th>G711</th>
<th>Select</th>
</tr>
</thead>
<tbody>
<tr>
<td>Codec Name</td>
<td>G711</td>
<td></td>
</tr>
<tr>
<td>Voice payload size (ms/frame)</td>
<td>0.00</td>
<td></td>
</tr>
<tr>
<td>Voice playout jitter buffer nominal delay</td>
<td>0.00</td>
<td></td>
</tr>
<tr>
<td>Modifications may cause changes to dependent settings</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Voice playout jitter buffer maximum delay</td>
<td>0.00</td>
<td></td>
</tr>
<tr>
<td>Modifications may cause changes to dependent settings</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

VAD

<table>
<thead>
<tr>
<th>Codec</th>
<th>G729A</th>
<th>Select</th>
</tr>
</thead>
<tbody>
<tr>
<td>Codec Name</td>
<td>G729A</td>
<td></td>
</tr>
<tr>
<td>Voice payload size (ms/frame)</td>
<td>0.00</td>
<td></td>
</tr>
<tr>
<td>Voice playout jitter buffer nominal delay</td>
<td>0.00</td>
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<tr>
<td>Modifications may cause changes to dependent settings</td>
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<td></td>
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<tr>
<td>Voice playout jitter buffer maximum delay</td>
<td>0.00</td>
<td></td>
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<td>Modifications may cause changes to dependent settings</td>
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VAD

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<tr>
<th>Codec</th>
<th>G723.1</th>
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<tbody>
<tr>
<td>Codec Name</td>
<td>G723.1</td>
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</tbody>
</table>

<table>
<thead>
<tr>
<th>Codec</th>
<th>T38 FAX</th>
<th>Select</th>
</tr>
</thead>
<tbody>
<tr>
<td>Codec Name</td>
<td>T38 FAX</td>
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</tbody>
</table>
14. Configure the QoS section

<table>
<thead>
<tr>
<th>System Status</th>
<th>Call Server</th>
<th>IP Telephony</th>
<th>Configuration</th>
<th>Call Server</th>
<th>IP Telephony</th>
<th>Network Numbering Plan</th>
<th>Call Server</th>
<th>Network Routing Service</th>
<th>Software Upgrade</th>
<th>Patching</th>
<th>System Utility</th>
<th>Administration</th>
<th>Support</th>
<th>Tools</th>
<th>Logout</th>
</tr>
</thead>
<tbody>
<tr>
<td>QoS</td>
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<td></td>
<td>QoS</td>
<td></td>
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<tr>
<td>Codec: G711</td>
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<tr>
<td>Codec: G729A</td>
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<tr>
<td>Codec: G723.1</td>
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<tr>
<td>Codec: T.30 FAX</td>
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</tr>
</tbody>
</table>

- **Diffserv Codepoint (DSCP) Control packets**: 40 (Range: 0 to 63)
- **Diffserv Codepoint (DSCP) Voice packets**: 46 (Range: 0 to 63)
- **Enable 802.1Q support**: 
- **802.1Q bits value (802.1p)**: 5 (Range: 0 to 7)
15. Configure LAN Configuration section

<table>
<thead>
<tr>
<th>Code:</th>
<th>Select</th>
</tr>
</thead>
<tbody>
<tr>
<td>VLAN</td>
<td>140 FAX</td>
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<tr>
<td>QoS</td>
<td>40</td>
</tr>
<tr>
<td>Diffserv Codepoint(DSCP) Control packets</td>
<td>46</td>
</tr>
<tr>
<td>Diffserv Codepoint(DSCP) Vo ice packets</td>
<td>40</td>
</tr>
<tr>
<td>Enable 802.1Q support</td>
<td>0</td>
</tr>
<tr>
<td>802.1Q inner value (802.1p)</td>
<td>5</td>
</tr>
</tbody>
</table>

**LAN configuration**

<table>
<thead>
<tr>
<th>Management LAN (ELAN) configuration</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call server IP address</td>
</tr>
<tr>
<td>Survivable Succession Media Gateway IP address</td>
</tr>
<tr>
<td>Signaling port</td>
</tr>
<tr>
<td>Broadcast port</td>
</tr>
</tbody>
</table>

**Voice LAN (TELAN) configuration**

| Signaling port | 5000 | Range: 16244 to 65535 |
| Voice port | 5280 | Range: 16244 to 65535 |

<table>
<thead>
<tr>
<th>Routes</th>
<th>IP address</th>
<th>Subnet mask</th>
</tr>
</thead>
<tbody>
<tr>
<td>Add</td>
<td>172.20.216.1</td>
<td>255.255.255.0</td>
</tr>
</tbody>
</table>
16. Configure the SIP GW Setting section

<table>
<thead>
<tr>
<th><strong>SIP GW Settings</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Primary Proxy / Re-direct IP address</td>
<td>172.20.216.103</td>
</tr>
<tr>
<td>Primary Proxy / Re-direct IP Port</td>
<td>5060</td>
</tr>
<tr>
<td>Primary Proxy Supports Registration</td>
<td>☑</td>
</tr>
<tr>
<td>Primary CDS Proxy or Re-direct server flag</td>
<td>☑</td>
</tr>
<tr>
<td>Secondary Proxy / Re-direct IP address</td>
<td>0.0.0.0</td>
</tr>
<tr>
<td>Secondary Proxy / Re-direct IP Port</td>
<td>5060</td>
</tr>
<tr>
<td>Secondary Proxy Supports Registration</td>
<td>☑</td>
</tr>
<tr>
<td>Secondary CDS Proxy or Re-direct server flag</td>
<td>☑</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>SIP URI Map</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Public E.164/Notional domain name</td>
<td>+1</td>
</tr>
<tr>
<td>Public E.164/Subscriber domain name</td>
<td>+1314</td>
</tr>
<tr>
<td>Public E.164/Unknown domain name</td>
<td></td>
</tr>
<tr>
<td>Public E.164/Special Number domain name</td>
<td></td>
</tr>
<tr>
<td>Private10P domain name</td>
<td>1p</td>
</tr>
<tr>
<td>PrivateCOP domain name</td>
<td>intersip</td>
</tr>
<tr>
<td>Private/Special Number domain name</td>
<td>1sip</td>
</tr>
<tr>
<td>Private/Unknown (vacant number routing) domain name</td>
<td></td>
</tr>
<tr>
<td>Unknown/Unknown domain name</td>
<td></td>
</tr>
</tbody>
</table>

**SIP CD Services**

**Cards**