Application Note

Avaya S8500 Communications Manager 3.0 Using T1 QSIG to Cisco Unified CallManager Express Release 4.0(3)

November 1, 2007 Revision 5

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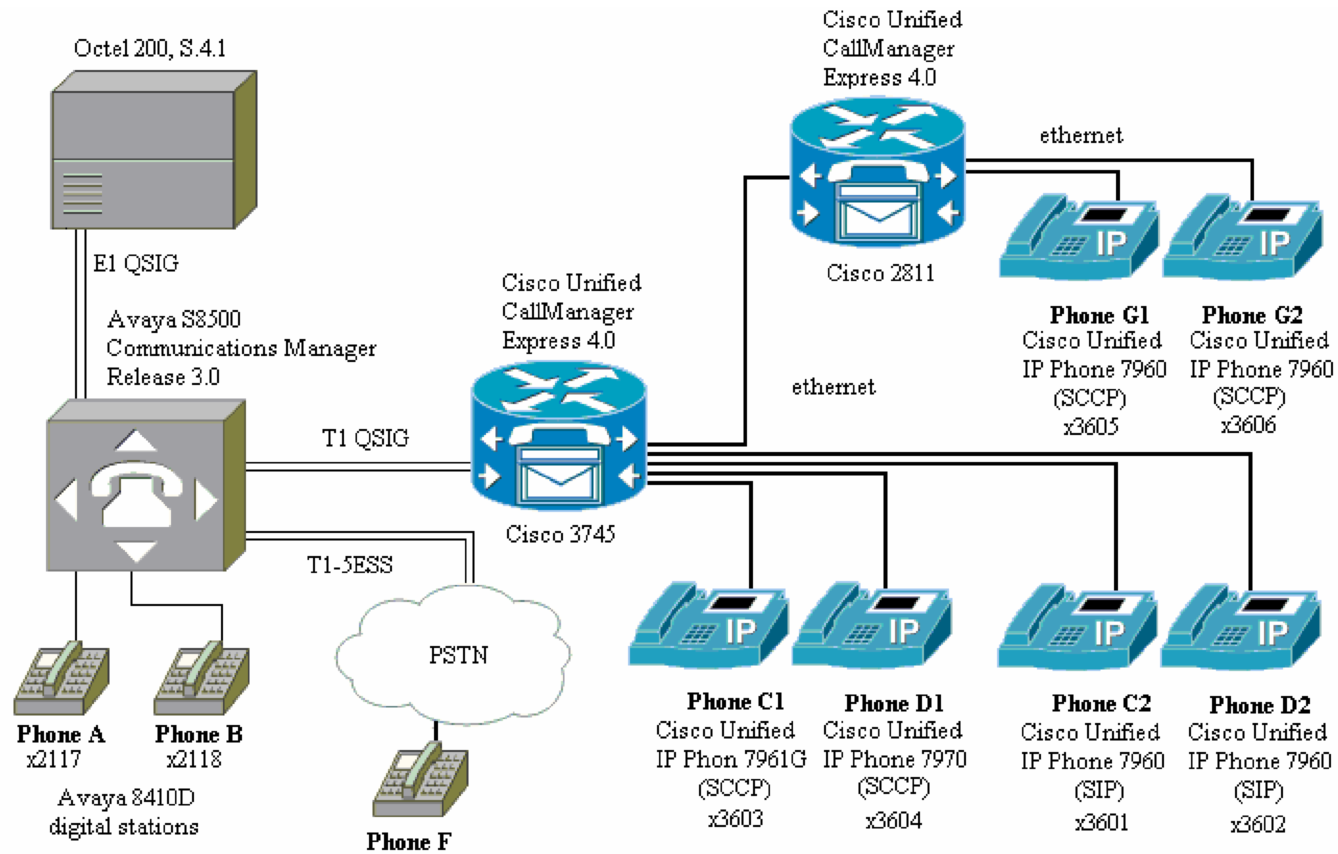
Introduction

- This is an Application Note for connectivity between an Avaya S8500 Communications Manager Release 3.0 PBX and Cisco Unified CallManager Express Release 4.0(3) using a Cisco 3745 voice gateway with QSIG protocol.
- Voice mail testing was performed with an Octel 200 (S.4.1) using QSIG integration (E1-DTIC).
- The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability with Cisco Unified CallManager Express Release 4.0(3) connected to the PBX via the 3745 T1 QSIG link. The 3745 IOS voice gateway was connected via H.323 to a Cisco 2811 IOS voice gateway. The two gateways were running Cisco Unified CallManager Express 4.0(3). Cisco Unified IP phones (models 7960, 7961G, and 7970) were connected to the 2 Cisco Unified CallManager Express gateways via SIP and SCCP, as per the figure. A NM-HDV and VWIC-1MFT-T1 were used for the T1 QSIG interface. Calls were made to test basic call, caller ID, conference, transfer, forward, call back, reroute, and MWI features.
- This Application Note uses the 3745 voice gateway. However, the use of other Cisco voice gateways is also an option since Cisco Unified Call Manager Express QSIG implementation does not depend on the physical interface.
- The inclusion of Cisco SIP phones in this application note is for reference only. Cisco Unified Communications Manager Express 4.0(3) supports SIP end-points with limited number of features.

Network Topology

Figure 1. Test Network Topology.



Limitations

Basic Calls

- Cisco Unified CallManager Express does not support overlap sending. It supports overlap receiving.
- Connected Name and Alerting Name are not supported on calls between PBX and Cisco Unified IP Phone running SIP.
- Calling Name Restriction is not supported for calls originated from Cisco Unified CallManager Express 4.0(3).
- Connected Number/Name Restriction is not supported from Cisco Unified CallManager Express 4.0(3).

Call Transfers

- The Avaya PBX will not perform a true blind transfer. It can perform a consultation transfer or early attended transfer.
- A transfer originated from a call placed from a phone on the remote Cisco Unified CallManager Express to a SIP phone on the local Cisco Unified CallManager Express, and then transferred to a PBX phone (e.g., G1 calls C2, and C2 transfers to A) does not complete.
- For most call transfers, the original calling name and number are not displayed on the final destination. Specifically, this applies to all consultation and early attended network/external transfers, and all consultation and early attended local transfers that involve a transfer from a SCCP phone to a SIP phone. The remaining local transfers and all blind transfers result in the original calling name and number information displaying properly.
- For many call transfers, the called (connected) name and number are not updated on the original phone after the transfer.

Call Forwards

- Generally, for forwarded calls where the final destination is a PBX station, the original calling name, but not the number, is displayed on the final destination station.
- For many call forwards, the forwarding called name and number are not displayed on the final destination.
- For many call forwards, the called (connected) name and number are not updated on the original phone.
- Forwarded calls originated from a PBX extension to a remote Cisco Unified CallManager Express SCCP extension, and forwarded to a local Cisco Unified CallManager Express extension (e.g., A calls G1, and G1 forwards to C2), Cisco Unified CallManager Express performs a QSIG reroute, even though a QSIG reroute is not in order (i.e., there is no QSIG ”hairpin” or ”trombone”).
- Forwarded calls hairpinned at a SIP extension (PBX phone calls Cisco Unified CallManager Express 4.0(3) SIP phone that forwards back to another PBX phone), the call completes, but Cisco Unified CallManager Express 4.0(3) does not perform a reroute, even if reroute is enabled.
- Forwarded calls originated from a PBX extension to a local Cisco Unified CallManager Express SCCP extension, and forwarded to another local Cisco Unified CallManager Express extension (e.g., A calls C1, and C1 forwards to D1 or D2), Cisco Unified CallManager Express performs a reroute, and even though a reroute is not in order (i.e., there is no ”hairpin” or ”trombone”).
- For calls that are hairpinned at a SIP extension (PBX phone calls Cisco Unified CallManager Express 4.0(3) SIP phone that forwards unconditionally back to another PBX phone) when a CFNR number was set up resulted in a 3rd SETUP message from CME. The timeout is set under the CFNR command. If enough time passes before the final destination (B) answers, the CFNR is invoked, and the 3rd SETUP is sent from CME. A new (3rd) B-chan is set up. The 2nd one is then torn down.
- Forwarded ”trombone” (or ”hairpin”) calls originated from a PBX extension to a Cisco Unified CallManager Express 4.0(3) extension, and forwarded back to another PBX extension (e.g, A calls C1, C2, or G1, which forwards to B), ”joined” calls (i.e., no Reroute or Path Replacement) could not be performed, because the PBX initiates Path Replacement after the call is joined. This feature can not be turned off. The only exception is when the forwarding is unconditional (CFU) and the forwarding phone is a SIP phone (e.g., C2). Then, there is not enough information in the 2nd SETUP message for the PBX to recognize it as a forwarded call, so there is no Path Replacement proposal, and the call is “joined”. There are 2 B-channels are in use. However, if CFNR is configured and enough time passes before the final destination answers for CFNR to be invoked, Cisco Unified CallManager Express 4.0(3) sends an additional (3rd) SETUP message. A new (3rd) B-chan is set up, and the 2nd one is then torn down, following the scenario in the previous bullet. This 3rd SETUP message does have the call fwd diverting leg info. and Path Replacement does occur.

- Forwarded calls that are initiated by overlap dialing from a PBX extension to a Cisco Unified CallManager Express 4.0(3) extension, the call completes, but Cisco Unified CallManager Express does not perform a reroute, even if reroute is enabled and the call is eligible for a reroute.

MWI

- Cisco Unified Communications Manager Express 4.0(3) supports Cisco Unity integration with QSIG. However, in this instance, no testing was performed with Cisco Unified Communications Manager Express 4.0(3) as the message center PINX.
- MWI was not tested for SIP extensions on Cisco Unified CallManager Express 4.0(3) with the PBX as the message center PINX.   It was tested for SCCP extensions only.

System Components

Hardware Requirements

- Cisco 3745 IOS voice gateway
  - NM-HDV
  - VWIC-2MFT-T1
- Cisco 2811 IOS voice gateway
- (4) Cisco Unified IP phone 7960s
- (1) Cisco Unified IP phone 7961G
- (1) Cisco Unified IP phone 7970
- (1) Avaya S8500 PBX
  - (2) Avaya 8410D digital station phones
  - (1) TN464F T1 trunk card (for PSTN link)
  - (1) TN464GP T1 trunk card (for QSIG trunk)
- (1) Octel 200 voice mail system
  - (2) E1-DTIC

Software Requirements

- Cisco Unified CallManager Express Release 4.0(3)
- Cisco IOS Software, 3700 Software (C3745-IPVOICE-M), Version 12.4(4)XC4
- Cisco IOS Software, 2800 Software (C2800NM-IPVOICE-M), Version 12.4(4)XC4
- Avaya Communications Manager Release 3.0
- Octel S.4.1 voice mail

G1, G2 – 7960 – SCCP

- Cisco7960 IP phone version 7.2(T0.23)
- Cisco 7960 IP phone app load P0030702T023
- Cisco 7960 IP phone boot load PC0303010200

C2, D2 – 7960 - SIP

- Cisco7960 DSP load ID PS03AT46
- Cisco 7960 IP phone app load P0S3-07-5-00
- Cisco 7960 IP phone boot load PC030301

C1 – 7961G – SCCP

- Cisco7961G IP phone load file: TERM61.DEFAULT
- Cisco 7961G IP phone app load ID: Jar41.2-9-1-45.sbn
- Cisco 7961G IP phone boot load ID: 7961G_64-020704128Amd64meg.bin

D1 – 7970 – SCCP

- Cisco7970 IP phone load file: SCCP70.8-0-3S
- Cisco 7970 IP phone app load ID: jar70sccp.8-0-2.25.sbn
- Cisco 7970 IP phone boot load ID: 7970_64060118.bin

Features

Features Supported

- Basic Call, ENBLOC
- Basic Call, Overlap (From PBX to Cisco Unified CallManager Express only)
- CLIP-Calling Line (Number) Identification Presentation on Basic Calls
- CLIR-Calling Line (Number) Identification Restriction on Basic Calls
- CNIP-Calling Name Identification Presentation on Basic Calls
- CNIR-Calling Name Identification Restriction on Basic Calls (From PBX to Cisco Unified CallManager Express only)
- COLP-Connected Line (Number) Identification Presentation on Basic Calls
- CONP-Connected Name Identification Presentation (for calls between PBX and Cisco Unified IP Phones running SCCP)
- Alerting Name (for calls between PBX and Cisco Unified IP Phones running SCCP)
- Tandem PSTN call
- Consultation Transfer – Local
- Consultation Transfer – Network/External (See Limitations Section)
- Early Attended Transfer – Local
- Early Attended Transfer – Network/External (See Limitations Section)
- Blind Transfer – Local (See Limitations Section)
- Blind Transfer – Network/External (See Limitations Section)
- Call Forward Unconditional by Join – Local (See Limitations Section)
- Call Forward Unconditional by Join – Network/External (See Limitations Section)
- Call Forward Busy by Join – Local (See Limitations Section)
- Call Forward Busy by Join – Network/External (See Limitations Section)
- Call Forward No Reply by Join – Local (See Limitations Section)
- Call Forward No Reply by Join – Network/External (See Limitations Section)
- Call Forward Unconditional by Reroute – Network/External (See Limitations Section)
- Call Forward Busy by Reroute – Network/External (See Limitations Section)
- Call Forward No Reply by Reroute – Network/External (See Limitations Section)
- MWI (See Limitations Section)

Features Not Supported

- Overlap dialing from Cisco Unified CallManager Express 4.0(3) to PBX
- CNIR-Calling Name Identification Restriction from Cisco Unified CallManager Express 4.0(3) to PBX
- COLR- Connected Line (Number) Identification Restriction
- CONR- Connected Name Identification Restriction
- CONP-Connected Name Identification Presentation (for calls between PBX and Cisco Unified IP Phones running SIP)
- Alerting Name (for calls between PBX and Cisco Unified IP Phones running SIP)
- Blind Transfers initiated from PBX
- H323/QSIG tandem transfers via SIP phone
- CLIP-Calling Line (Number) Identification Presentation on Transferred Calls
- CNIP-Calling Name Identification Presentation on Transferred Calls
- COLP-Connected Line (Number) Identification Presentation on Transferred Calls
- CONP-Connected Name Identification Presentation on Transferred Calls
- CLIP-Calling Line (Number) Identification Presentation on Forwarded Calls to a PBX station.
- COLP-Connected Line (Number) Identification Presentation on Forwarded Calls
- CONP-Connected Name Identification Presentation on Forwarded Calls
- Call Forward by Reroute for QSIG ”trombone” from a Cisco Unified CallManager Express SIP extension
- Call Forward by Reroute with overlap dialing
- Call Completion to Busy Subscriber (Call Back when Free)
- Call Completion on No Reply (Call Back Next Used)
- Path Replacement for Call Transfer by Join
- Path Replacement for Trombone Connection
- Path Replacement for Call Diversion by Forward Switch

Configuration

Configuring the sequence for the Avaya S8500 Communications Manager 3.0 PBX

1. Check the system-parameter customer-option screen to insure the proper QSIG optional features are installed
2. Configure DS1 circuit pack.
3. Configure Signaling Group
4. Configure Trunk Group
5. Configure Route Pattern
6. Configure ISDN Public-Unknown numbering screen
7. Configure Uniform-Dialplan screen
8. Configure AAR analysis screen

Configuring the Avaya S8500 Communications Manager 3.0 Screens

Avaya S8500 Communications Manager 3.0 Configuration

GLOBAL PARAMETERS

Figure 2. QSIG Options – 1 of 1.

```
display system-parameters customer-options                      Page   8 of  10
                              QSIG OPTIONAL FEATURES

                                        Basic Call Setup? y
                              Basic Supplementary Services? y
                                    Centralized Attendant? y
                                    Interworking with DCS? y
                     Supplementary Services with Rerouting? y
                              Transfer into QSIG Voice Mail? y
                                      Value-Added (VALU)? y









       (NOTE: You must logoff & login to effect the permission changes.)
```

Figure 3. Software Version – 1 of 1.

```
list configuration software-versions

                                 SOFTWARE VERSIONS

    SOFTWARE VERSION
       Memory Resident: R013x.00.0.340.3
         Disk Resident: R013x.00.0.340.3

    TRANSLATION DATE
       Memory Resident: 10:00 pm  SUN OCT 22, 2006
         Disk Resident: 10:00 pm  SUN OCT 22, 2006
      Disk Second Copy: good

Command successfully completed
```

CONFIGURATION FOR TRUNKS

Figure 4. Circuit Pack for T1-QSIG trunk to Cisco Unified CallManager Express – 1 of 1.

```
display ds1 1a14                                                Page   1 of   2
                                DS1 CIRCUIT PACK

            Location: 01A14                               Name: T1 QSIG
            Bit Rate: 1.544                        Line Coding: b8zs
   Line Compensation: 1                           Framing Mode: esf
      Signaling Mode: isdn-pri
             Connect: pbx                            Interface: peer-master
   TN-C7 Long Timers? n                          Peer Protocol: Q-SIG
Interworking Message: PROGress                            Side: a
Interface Companding: mulaw                               CRC? n
           Idle Code: 11111111
                              DCP/Analog Bearer Capability: 3.1kHz

                                           T303 Timer(sec): 4


      Slip Detection? n                 Near-end CSU Type: other

   Echo Cancellation? n
```

Figure 5. Trunk Group for T1-QSIG trunk to Cisco Unified CallManager Express – 1 of 3.

```
display trunk-group 14                                          Page   1 of  19
                                TRUNK GROUP

Group Number: 14                  Group Type: isdn          CDR Reports: y
  Group Name: Chris CME Testing          COR: 1        TN: 1        TAC: 814
   Direction: two-way       Outgoing Display? y        Carrier Medium: PRI/BRI
 Dial Access? y               Busy Threshold: 255      Night Service:
Queue Length: 0
Service Type: tie                 Auth Code? n            TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
TRUNK PARAMETERS
         Codeset to Send Display: 0     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: b     Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: descend                        QSIG Value-Added? y
                                                    Digital Loss Group: 13
Incoming Calling Number - Delete:      Insert:             Format: unk-unk
              Bit Rate: 1200       Synchronization: async     Duplex: full
 Disconnect Supervision - In? y  Out? y
 Answer Supervision Timeout: 0
```

Figure 6. Trunk Group for T1-QSIG trunk to Cisco Unified CallManager Express – 2 of 3.

```
display trunk-group 14                                          Page   2 of  19
TRUNK FEATURES
          ACA Assignment? n            Measured: none        Wideband Support? n
                                 Internal Alert? n         Maintenance Tests? y
                               Data Restriction? n       NCA-TSC Trunk Member: 1
                                      Send Name: y        Send Calling Number: y
            Used for DCS? n             Hop Dgt? y
   Suppress # Outpulsing? n    Format: unknown
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider

                                             Replace Restricted Numbers? y
                                            Replace Unavailable Numbers? y
                                       Send Called/Busy/Connected Number: y
                                              Hold/Unhold Notifications? y
             Send UUI IE? y                Modify Tandem Calling Number? n
              Send UCID? n
 Send Codeset 6/7 LAI IE? y                         Ds1 Echo Cancellation? n

 Path Replacement with Retention? y

                     SBS? n  Network (Japan) Needs Connect Before Disconnect? n
```

Figure 7. Trunk Group for T1-QSIG trunk to Cisco Unified CallManager Express – 3 of 3.

```
display trunk-group 14                                          Page   3 of  19
                                 TRUNK GROUP
                                     Administered Members (min/max):   1/10
GROUP MEMBER ASSIGNMENTS                 Total Administered Members:  10

       Port     Code Sfx Name         Night             Sig Grp
  1: 01A1401   TN464  G                                   14
  2: 01A1402   TN464  G                                   14
  3: 01A1403   TN464  G                                   14
  4: 01A1404   TN464  G                                   14
  5: 01A1405   TN464  G                                   14
  6: 01A1419   TN464  G                                   14
  7: 01A1420   TN464  G                                   14
  8: 01A1421   TN464  G                                   14
  9: 01A1422   TN464  G                                   14
 10: 01A1423   TN464  G                                   14
 11:
 12:
 13:
 14:
 15:
```

Figure 8. Signalling Group for T1-QSIG trunk to Cisco Unified CallManager Express – 1 of 1.

```
display signaling-group 14
                               SIGNALING GROUP

 Group Number: 14               Group Type: isdn-pri
                        Associated Signaling? y         Max number of NCA TSC: 10
                           Primary D-Channel: 01A1424    Max number of CA TSC: 10
                                                       Trunk Group for NCA TSC: 14
       Trunk Group for Channel Selection: 14
          Supplementary Service Protocol: b
```

Figure 9. Circuit Pack for T1-5ESS trunk to PSTN – 1 of 1.

```
display ds1 1a13                                                    Page   1 of   2
                               DS1 CIRCUIT PACK

            Location: 01A13                             Name: 5ESS
            Bit Rate: 1.544                      Line Coding: b8zs
   Line Compensation: 1                         Framing Mode: esf
      Signaling Mode: isdn-pri
             Connect: pbx                          Interface: user
   TN-C7 Long Timers? n                     Country Protocol: 1
Interworking Message: PROGress              Protocol Version: a
Interface Companding: mulaw                             CRC? n
           Idle Code: 11111111
                              DCP/Analog Bearer Capability: 3.1kHz

                                            T303 Timer(sec): 4


      Slip Detection? n                     Near-end CSU Type: other
```

Figure 10. Trunk Group for T1-5ESS trunk to PSTN – 1 of 3.

```
display trunk-group 13                                          Page   1 of  19
                                TRUNK GROUP

Group Number: 13                   Group Type: isdn          CDR Reports: y
  Group Name: Chris T1 ISDN PRI test      COR: 1        TN: 1        TAC: 807
   Direction: two-way        Outgoing Display? y        Carrier Medium: PRI/BRI
 Dial Access? y                Busy Threshold: 255      Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n             TestCall ITC: rest
                          Far End Test Line No:
TestCall BCC: 4
TRUNK PARAMETERS
         Codeset to Send Display: 0     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: a     Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: ascend                        QSIG Value-Added? n
                                                    Digital Loss Group: 13
Incoming Calling Number - Delete:     Insert:                Format: unk-unk
              Bit Rate: 1200        Synchronization: async     Duplex: full
 Disconnect Supervision - In? y  Out? y
 Answer Supervision Timeout: 0
```

Figure 11. Trunk Group for T1-5ESS trunk to PSTN – 2 of 3.

```
display trunk-group 13                                          Page   2 of  19
TRUNK FEATURES
          ACA Assignment? n            Measured: none       Wideband Support? n
                                 Internal Alert? n       Maintenance Tests? y
                             Data Restriction? n     NCA-TSC Trunk Member: 1
                                     Send Name: y      Send Calling Number: y
            Used for DCS? n
   Suppress # Outpulsing? n    Format: unknown
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider

                                          Replace Restricted Numbers? y
                                         Replace Unavailable Numbers? y
                                               Send Connected Number: y
                                           Hold/Unhold Notifications? n
             Send UUI IE? y              Modify Tandem Calling Number? n
               Send UCID? n
 Send Codeset 6/7 LAI IE? y                          Ds1 Echo Cancellation? n

                                    US NI Delayed Calling Name Update? n

                     SBS? n  Network (Japan) Needs Connect Before Disconnect? n
```

Figure 12. Trunk Group for T1-5ESS trunk to PSTN – 3 of 3.

```
display trunk-group 13                                          Page   3 of  19
                                  TRUNK GROUP
                                      Administered Members (min/max):   1/10
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  10

       Port     Code Sfx Name          Night            Sig Grp
  1: 01A1301   TN464  F                                   13
  2: 01A1302   TN464  F                                   13
  3: 01A1303   TN464  F                                   13
  4: 01A1304   TN464  F                                   13
  5: 01A1305   TN464  F                                   13
  6: 01A1319   TN464  F                                   13
  7: 01A1320   TN464  F                                   13
  8: 01A1321   TN464  F                                   13
  9: 01A1322   TN464  F                                   13
 10: 01A1323   TN464  F                                   13
 11:
 12:
 13:
 14:
 15:
```

Figure 13. Signalling Group for T1-5ESS trunk to PSTN – 1 of 1.

```
display signaling-group 13                                      Page   1 of   5
                                 SIGNALING GROUP

 Group Number: 13                Group Type: isdn-pri
                         Associated Signaling? y          Max number of NCA TSC: 10
                            Primary D-Channel: 01A1324     Max number of CA TSC: 10
                                                        Trunk Group for NCA TSC: 13
       Trunk Group for Channel Selection: 13
          Supplementary Service Protocol: a
```

DIAL PLANS AND ROUTE PATTERNS

Figure 14. Uniform Dial Plan – 1 of 1.

display uniform-dialplan 36 Page 1 of 2

UNIFORM DIAL PLAN TABLE

Percent Full: 0

Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num	Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num
36	4	0	214	aar	n		5050	4	0	202	aar	n	
37	4	0	213	aar	n		60	4	0	201	aar	n	
40	4	0	201	aar	n							n	
4131	4	0	201	aar	n							n	
4132	4	0	201	aar	n							n	
4149	4	0	217	aar	n							n	
4150	4	0	217	aar	n							n	
4152	4	0	202	aar	n							n	
4154	4	0	201	aar	n							n	
4155	4	0	201	aar	n							n	
4156	4	0	201	aar	n							n	
45	4	0	201	aar	n							n	
5003	4	0	213	aar	n							n	
5004	4	0	215	aar	n							n	
5005	4	0	215	aar	n							n	
5008	4	0	201	aar	n							n	

Figure 15. AAR Analysis – 1 of 1.

display aar analysis 213 Page 1 of 2

AAR DIGIT ANALYSIS TABLE

Percent Full: 1

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
213	7	7	13	aar		n
214	7	7	14	aar		n
215	7	7	15	aar		n
216	7	7	6	aar	1	n
217	7	7	6	aar	6	n
224	7	7	224	aar		n
3	7	7	999	aar		n
4	7	7	999	aar		n
5	7	7	999	aar		n
6	7	7	999	aar		n
7	7	7	999	aar		n
8	7	7	999	aar		n
9	7	7	999	aar		n
						n
						n

Figure 16. Route Pattern for T1-QSIG trunk to Cisco Unified CallManager Express – 1 of 1.

```
display route-pattern 14                                        Page   1 of   3
                    Pattern Number: 14  Pattern Name: CCM 5.0
                             SCCAN? n      Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                          DCS/ IXC
    No          Mrk Lmt List Del  Digits                            QSIG
                             Dgts                                   Intw
 1: 14   0                    3                                      n   user
 2:                                                                  n   user
 3:                                                                  n   user
 4:                                                                  n   user
 5:                                                                  n   user
 6:                                                                  n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature BAND  No. Numbering LAR
    0 1 2 3 4 W     Request                                Dgts Format
                                                        Subaddress
 1: y y y y y n  y  as-needed rest                                        none
 2: y y y y y n  n            rest                                        none
 3: y y y y y n  n            rest                                        none
 4: y y y y y n  n            rest                                        none
 5: y y y y y n  n            rest                                        none
 6: y y y y y n  n            rest                                        none
```

Figure 17. Route Pattern for T1-5ESS trunk to PSTN – 1 of 1.

```
display route-pattern 13                                        Page   1 of   3
                    Pattern Number: 13  Pattern Name:
                             SCCAN? n      Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                          DCS/ IXC
    No          Mrk Lmt List Del  Digits                            QSIG
                             Dgts                                   Intw
 1: 13   0                    3                                      n   user
 2:                                                                  n   user
 3:                                                                  n   user
 4:                                                                  n   user
 5:                                                                  n   user
 6:                                                                  n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature BAND  No. Numbering LAR
    0 1 2 3 4 W     Request                                Dgts Format
                                                        Subaddress
 1: y y y y y n  y  as-needed rest                                        none
 2: y y y y y n  n            rest                                        none
 3: y y y y y n  n            rest                                        none
 4: y y y y y n  n            rest                                        none
 5: y y y y y n  n            rest                                        none
 6: y y y y y n  n            rest                                        none
```

CONFIGURATIONS FOR PHONES

Figure 18. Digital Station Configuration – 1 of 2.

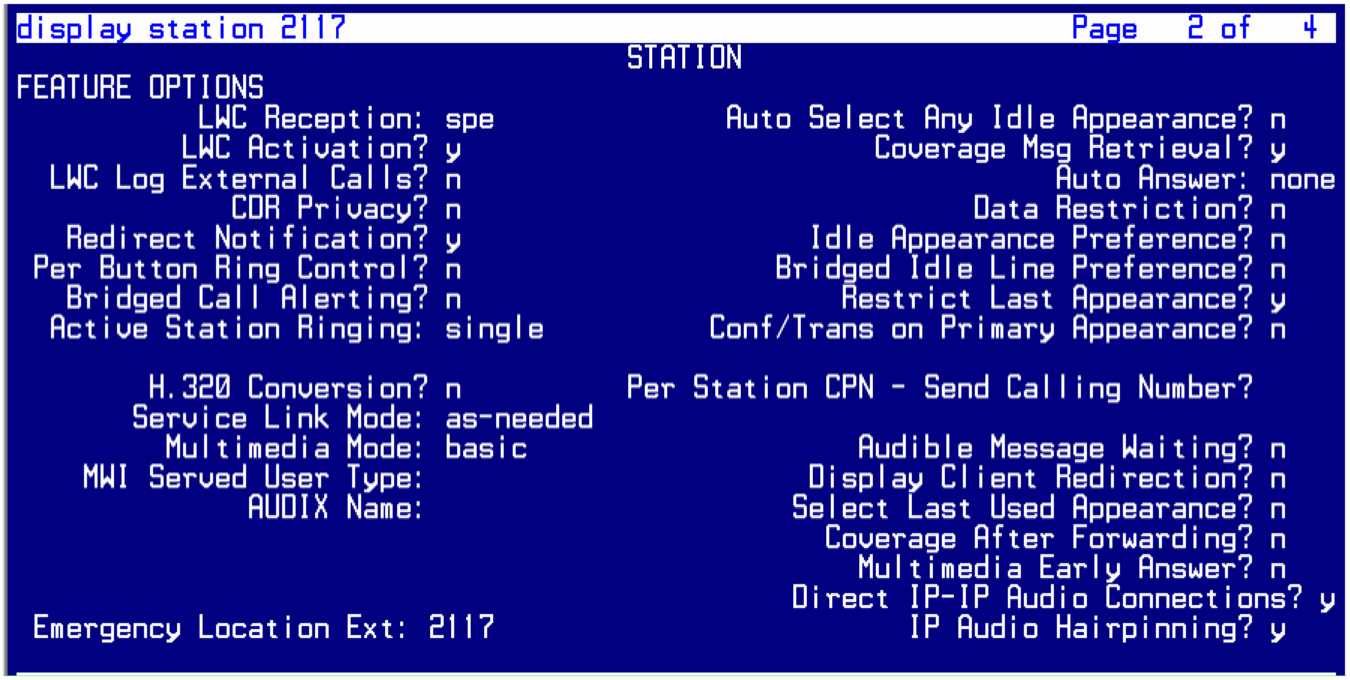
```
display station 2117                                            Page   1 of   4
                                     STATION

Extension: 2117                          Lock Messages? n               BCC: 0
     Type: 8410D                         Security Code:                  TN: 1
     Port: 01A0415                     Coverage Path 1: 1               COR: 1
     Name: Chris-A1                    Coverage Path 2:                 COS: 1
                                       Hunt-to Station:

STATION OPTIONS
              Loss Group: 2            Personalized Ringing Pattern: 1
             Data Module? n                        Message Lamp Ext: 2117
            Speakerphone: 2-way                 Mute Button Enabled? y
        Display Language: english

                                                  Media Complex Ext:
                                                       IP SoftPhone? n
                                                Remote Office Phone? n
```

Figure 19. Digital Station Configuration – 2 of 2.



```
display station 2117                                              Page   2 of   4
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe              Auto Select Any Idle Appearance? n
          LWC Activation? y                         Coverage Msg Retrieval? y
  LWC Log External Calls? n                                    Auto Answer: none
             CDR Privacy? n                               Data Restriction? n
  Redirect Notification? y                    Idle Appearance Preference? n
 Per Button Ring Control? n                 Bridged Idle Line Preference? n
   Bridged Call Alerting? n                     Restrict Last Appearance? y
  Active Station Ringing: single          Conf/Trans on Primary Appearance? n

       H.320 Conversion? n     Per Station CPN - Send Calling Number?
      Service Link Mode: as-needed
        Multimedia Mode: basic                    Audible Message Waiting? n
   MWI Served User Type:                       Display Client Redirection? n
             AUDIX Name:                      Select Last Used Appearance? n
                                                Coverage After Forwarding? n
                                                  Multimedia Early Answer? n
                                            Direct IP-IP Audio Connections? y
 Emergency Location Ext: 2117                          IP Audio Hairpinning? y
```

CLIR

For Calling Line ID Restriction (CLIR, CNIR) to be implemented, the associated trunk group must be modified.

- On page 2 of the Trunk Group screen, "Send Name" field and "Send Calling Number" field must be changed to "r" for restricted.

Figure 20. Trunk Group for T1-QSIG trunk to Cisco Unified CallManager Express – modified for CLIR – 1 of 1.

```
display trunk-group 14                                          Page   2 of  19
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                  Internal Alert? n      Maintenance Tests? y
                                Data Restriction? n     NCA-TSC Trunk Member: 1
                                       Send Name: r      Send Calling Number: r
            Used for DCS? n              Hop Dgt? y
   Suppress # Outpulsing? n    Format: unknown
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider

                                             Replace Restricted Numbers? y
                                            Replace Unavailable Numbers? y
                                      Send Called/Busy/Connected Number: y
                                              Hold/Unhold Notifications? y
             Send UUI IE? y                  Modify Tandem Calling Number? n
              Send UCID? n
 Send Codeset 6/7 LAI IE? y                          Ds1 Echo Cancellation? n

 Path Replacement with Retention? y

                    SBS? n  Network (Japan) Needs Connect Before Disconnect? n
```

CALL FORWARD BY JOIN

For diversion (CFU, CFB) to be accomplished by join instead of reroute, a coverage path must be assigned to the forwarding station.

- On page 1 of the station form associated with the forwarding station, “Coverage Path 1" must be set to 1.   See Figure 21.
- On page 2 of the station form associated with the forwarding station, "Coverage after Forwarding" must be set to "y". See Figure 22.

Some system parameters also must be enabled:

- On page 1 of the system parameters / coverage forwarding form, "QSIG VALU Coverage Overrides QSIG Diversion with Rerouting" must to be set to "y". See 0
- On page 1 of the system parameters / coverage forwarding form, "Call Forward Override” must be set "y". See 0
- On page 1 of the system parameters / coverage forwarding form, "Coverage After Forwarding" also must be set to "y". See 0
- On page 2 of the system parameters / coverage forwarding form. "Coverage of Calls Redirected Off-net Enabled" needs to be set to "y".Figure 24.

Figure 21. Screen shot of station form for Call Forward by Join – 1 of 2.

```
display station 2117                                              Page    1 of    4
                                     STATION

Extension: 2117                           Lock Messages? n             BCC: 0
     Type: 8410D                          Security Code:                TN: 1
     Port: 01A0415                      Coverage Path 1: 1             COR: 1
     Name: Chris-A1                     Coverage Path 2:               COS: 1
                                        Hunt-to Station:

STATION OPTIONS
              Loss Group: 2             Personalized Ringing Pattern: 1
             Data Module? n                         Message Lamp Ext: 2117
            Speakerphone: 2-way                  Mute Button Enabled? y
        Display Language: english

                                                      Media Complex Ext:
                                                           IP SoftPhone? n
                                                    Remote Office Phone? n
```

Figure 22. Screen shot of station form for Call Forward by Join – 2 of 2.

```
display station 2117                                            Page   2 of   4
                                       STATION
FEATURE OPTIONS
           LWC Reception: spe               Auto Select Any Idle Appearance? n
          LWC Activation? y                          Coverage Msg Retrieval? y
  LWC Log External Calls? n                                    Auto Answer: none
             CDR Privacy? n                               Data Restriction? n
   Redirect Notification? y                     Idle Appearance Preference? n
 Per Button Ring Control? n                   Bridged Idle Line Preference? n
   Bridged Call Alerting? n                       Restrict Last Appearance? y
  Active Station Ringing: single         Conf/Trans on Primary Appearance? n

        H.320 Conversion? n            Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: basic                    Audible Message Waiting? n
    MWI Served User Type:                       Display Client Redirection? n
              AUDIX Name:                      Select Last Used Appearance? n
                                                 Coverage After Forwarding? y
                                                   Multimedia Early Answer? n
                                               Direct IP-IP Audio Connections? y
 Emergency Location Ext: 2117                            IP Audio Hairpinning? y
```

Figure 23. Screen shot of system parameters / coverage forwarding form for Call Forward by Join – 1 of 2.

```
display system-parameters coverage-forwarding                   Page   1 of   2
                SYSTEM PARAMETERS CALL COVERAGE / CALL FORWARDING

CALL COVERAGE/FORWARDING PARAMETERS

          Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings): 4
        Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings): 4
                                Coverage - Caller Response Interval (seconds): 4
     Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls: 5

COVERAGE
                                         Keep Held SBA at Coverage Point? y
       External Coverage Treatment for Transferred Incoming Trunk Calls? y
        Immediate Redirection on Receipt of PROGRESS Inband Information? n
                                               Maintain SBA At Principal? y
             QSIG VALU Coverage Overrides QSIG Diversion with Rerouting? y
                                            Station Hunt Before Coverage? n

FORWARDING
                                                   Call Forward Override? y
                                               Coverage After Forwarding? y
```

Figure 24. Screen shot of system parameters / coverage forwarding form for Call Forward by Join – 2 of 2.

```
display system-parameters coverage-forwarding                   Page   2 of   2
              SYSTEM PARAMETERS CALL COVERAGE / CALL FORWARDING

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)

                       Coverage Of Calls Redirected Off-Net Enabled? y
 Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point? y
                                  Ignore Network Answer Supervision? n
                  Disable call classifier for CCRON over ISDN trunks? n
                   Disable call classifier for CCRON over SIP trunks? n
```

Configuring the Local Cisco Unified CallManager Express (Cisco 3745)

LOCAL-3745#sho ver

Cisco IOS Software, 3700 Software (C3745-IPVOICE-M), Version 12.4(4)XC4, RELEAS)

Synched to technology version 12.4(5.13)T

Technical Support: http://www.cisco.com/techsupport

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Compiled Mon 24-Jul-06 19:48 by ealyon

ROM: System Bootstrap, Version 12.2(8r)T2, RELEASE SOFTWARE (fc1)

ROM: Cisco IOS Software, 3700 Software (C3745-IPVOICE-M), Version 12.4(4)XC4, R)

LOCAL-3745 uptime is 2 weeks, 4 days, 1 hour, 22 minutes

System returned to ROM by reload

System image file is "flash:c3745-ipvoice-mz.124-4.XC4.bin"

Cisco 3745 (R7000) processor (revision 2.0) with 241664K/20480K bytes of memory.

Processor board ID JMX0813L0Z3

R7000 CPU at 350MHz, Implementation 39, Rev 3.3, 256KB L2, 2048KB L3 Cache

2 FastEthernet interfaces

48 Serial interfaces

2 Channelized T1/PRI ports

2 Voice FXS interfaces

2 Voice DID interfaces

DRAM configuration is 64 bits wide with parity enabled.

151K bytes of NVRAM.

62720K bytes of ATA System CompactFlash (Read/Write)

Configuration register is 0x0

```
LOCAL-3745#wr t
Building configuration...

Current configuration : 5340 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname LOCAL-3745
!
boot-start-marker
boot system flash:c3745-ipvoice-mz.124-4.XC4.bin
boot-end-marker
!
logging buffered 99999999 debugging
enable password cisco
!
no aaa new-model
!
resource policy
!
no network-clock-participate slot 1
no network-clock-participate slot 3
voice-card 1
 no dspfarm
!
```

```
voice-card 3
 dspfarm
!
ip cef
!
!
no ip dhcp use vrf connected
!
ip dhcp pool ephone3
   host 172.20.15.203 255.255.255.0
   client-identifier 0100.170e.c858.d4
   default-router 172.20.15.1
   option 150 ip 172.20.15.196
!
ip dhcp pool ephone4
   host 172.20.15.204 255.255.255.0
   client-identifier 0100.15f9.c856.1a
   default-router 172.20.15.1
   option 150 ip 172.20.15.196
!
ip dhcp pool ephone1
   host 172.20.15.201 255.255.255.0
   client-identifier 0100.15fa.0cb1.dc
   default-router 172.20.15.1
   option 150 ip 172.20.15.196
!
ip dhcp pool ephone2
   host 172.20.15.202 255.255.255.0
   client-identifier 0100.15fa.0cb5.d9
```

```
   default-router 172.20.15.1
   option 150 ip 172.20.15.196
!
ip dhcp pool ephone7
   host 172.20.15.207 255.255.255.0
   client-identifier 0100.15c6.96dd.6b
   default-router 172.20.15.1
   option 150 ip 172.20.15.196
!
!
no ip domain lookup
ip dhcp-server query lease retries 5
ip dhcp-server 172.20.15.196
isdn switch-type primary-qsig
!
!
voice call carrier capacity active
!
voice service pots
 <supplementary-service qsig call-forward>[1]
!
voice service voip
 qsig decode
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 supplementary-service h450.12
```

[1] Omitted to force QSIG call forward by join (no reroute).

< **no supplementary-service h450.2**> [2]

<**no supplementary-service h450.3** > [2]

h323

sip

registrar server expires max 600 min 60

!

!

voice register global

mode cme

source-address 172.20.15.196 port 5060

max-dn 100

load 7960-7940 P0S3-07-5-00

tftp-path flash:

create profile sync 001139502554208A

!

voice register dn 1

number 3601

< call-forward b2bua busy 2118>[3]

<call-forward b2bua noan 2118 timeout 7>[4]

name Local IP1

huntstop

!

voice register dn 2

number 3602

name Local IP2

huntstop

!

---

[2] Inserted to force IP call forward by join (no reroute).
[3] Inserted for call forward busy from SIP extension.
[4] Inserted for call forward no reply from SIP extension.

```
voice register dn  3
 call-forward b2bua busy 3015
!
voice register pool  1
 id mac 0015.FA0C.B1DC
 type 7960
 number 1 dn 1
 max registrations 42
 dtmf-relay rtp-nte
 description Cisco7960
 codec g711ulaw
!
voice register pool  2
 id mac 0015.FA0C.B5D9
 type 7960
 number 1 dn 2
 max registrations 42
 dtmf-relay rtp-nte
 description Cisco7960
 codec g711ulaw
!
!
controller T1 3/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
controller T1 3/1
 framing esf
```

```
 linecode b8zs
 pri-group timeslots 1-24
!
interface FastEthernet0/0
 ip address 172.20.15.196 255.255.255.0
 duplex auto
 speed auto
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface Serial3/0:23
 no ip address
 encapsulation hdlc
 isdn switch-type primary-qsig
 isdn overlap-receiving
 isdn incoming-voice voice
 no cdp enable
!
interface Serial3/1:23
 no ip address
 encapsulation hdlc
 isdn switch-type primary-qsig
 isdn overlap-receiving
 isdn protocol-emulate network
 isdn incoming-voice voice
```

```
 isdn T310 120000
 no cdp enable
!
ip route 0.0.0.0 0.0.0.0 172.20.15.1
!
ip http server
ip http authentication local
ip http path flash:
!
!
!
tftp-server flash:P003-07-5-00.bin
tftp-server flash:P003-07-5-00.sbn
tftp-server flash:P0S3-07-5-00.bin
tftp-server flash:P0S3-07-5-00.sb2
tftp-server flash:P0S3-07-5-00.loads
```

< tftp-server flash: any load file that is not on the phone and is needed >

< tftp-server slot0: any load file that is not on the phone and is needed>

```
!
control-plane
!
!
!
voice-port 1/0/0
 timing digit 75
 timing inter-digit 65
!
voice-port 1/0/1
```

```
!
voice-port 1/1/0
!
voice-port 1/1/1
!
voice-port 3/0:23
!
voice-port 3/1:23
!
!
dial-peer voice 3023 pots
 destination-pattern 2...
 incoming called-number ....
```

<clid restrict> [5]

< supplementary-service qsig call-forward > [6]

```
 direct-inward-dial
 port 3/0:23
 forward-digits all
!
dial-peer voice 1 voip
 preference 1
 destination-pattern 36..
 session target ipv4:172.20.15.159
 dtmf-relay h245-alphanumeric
 codec g711ulaw
 no vad
```

[5] Inserted for CLID restrict cases only.
[6] Omitted to force QSIG call forward by join (no reroute).

```
!
dial-peer voice 5050 pots
 destination-pattern 5050
 direct-inward-dial
 port 3/0:23
 forward-digits all
!
dial-peer voice 5 pots
 destination-pattern 5...
 direct-inward-dial
 port 3/0:23
 forward-digits all
!
dial-peer voice 3700 pots
 destination-pattern 37..
 direct-inward-dial
 port 3/0:23
 forward-digits all
!
!
sip-ua
!
!
telephony-service
 load 7960-7940 P003-07-5-00
 load 7961 Jar41.2-9-1-45.sbn
 load 7970 jar70sccp.8-0-2.25.sbn
 max-ephones 25
 max-dn 50
```

ip source-address 172.20.15.196 port 2000

max-conferences 8 gain -6

call-forward pattern .T

transfer-system full-consult

transfer-pattern .... <**blind**> [7]

create cnf-files version-stamp 7960 Sep 11 2006 16:53:04

!

!

ephone-dn 3 dual-line

number 3603

name Local IP3

< call-forward busy 2118> [8]

<call-forward noan 2118 timeout 7>[9]

huntstop channel

!

!

ephone-dn 4 dual-line

number 3604

name Local IP4

huntstop channel

!

!

ephone-dn 5

call-forward busy 2118

call-forward noan 2118 timeout 7

!

!

---

[7] Inserted to enable blind transfers, as opposed to early attended transfers.
[8] Inserted for call forward busy from SCCP extension.
[9] Inserted for call forward no reply from SCCP extension.

```
ephone-dn  7  dual-line
 number 3017
 name Local IP7
 huntstop channel
!
!
ephone  3
 mac-address 0017.0EC8.58D4
 type 7961
 keep-conference
 button  1:3
!
ephone  4
 mac-address 0015.F9C8.561A
 type 7970
 keep-conference
 button  1:4
!
ephone  7
 mac-address 0015.C696.DD6B
 type 7970
 keep-conference
 button  1:7
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 exec-timeout 0 0
```

```
 password cisco
 login
 transport input telnet
!
end
LOCAL-3745#
```

Configuring the Remote Cisco Unified CallManager Express (Cisco 2811)

REMOTE-2811#

REMOTE-2811#sho ver

Cisco IOS Software, 2800 Software (C2800NM-IPVOICE-M), Version 12.4(4)XC4, RELE)

Synched to technology version 12.4(5.13)T

Technical Support: http://www.cisco.com/techsupport

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Compiled Mon 24-Jul-06 18:33 by ealyon

ROM: System Bootstrap, Version 12.4(1r) [hqluong 1r], RELEASE SOFTWARE (fc1)

ROM: Cisco IOS Software, 2800 Software (C2800NM-IPVOICE-M), Version 12.4(4)XC4,)

REMOTE-2811 uptime is 7 weeks, 4 days, 23 hours, 19 minutes

System returned to ROM by power-on

System restarted at 16:23:28 UTC Thu Sep 7 2006

System image file is "flash:c2800nm-ipvoice-mz.124-4.XC4.bin"

Cisco 2811 (revision 53.51) with 251904K/10240K bytes of memory.

Processor board ID FHK0946F0MZ

2 FastEthernet interfaces

2 Voice FXS interfaces

DRAM configuration is 64 bits wide with parity enabled.

239K bytes of non-volatile configuration memory.

62592K bytes of ATA CompactFlash (Read/Write)

Configuration register is 0x2

```
REMOTE-2811#wr t
Building configuration...

Current configuration : 3617 bytes
!
! Last configuration change at 15:42:37 UTC Tue Oct 31 2006
! NVRAM config last updated at 15:42:38 UTC Tue Oct 31 2006
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname REMOTE-2811
!
boot-start-marker
boot system flash:c2800nm-ipvoice-mz.124-4.XC4.bin
boot-end-marker
!
enable password cisco
!
no aaa new-model
!
resource policy
!
!
!
ip cef
no ip dhcp use vrf connected
```

```
!
ip dhcp pool ephone5
   host 172.20.15.205 255.255.255.0
   client-identifier 0100.15fa.0cb7.46
   default-router 172.20.15.1
   option 150 ip 172.20.15.159
!
ip dhcp pool ephone6
   host 172.20.15.206 255.255.255.0
   client-identifier 0100.15fa.63bf.84
   default-router 172.20.15.1
   option 150 ip 172.20.15.159
!
!
no ip domain lookup
ip dhcp-server query lease retries 5
ip dhcp-server 172.20.15.159
!
!
voice-card 0
 no dspfarm
!
!
!
voice service voip
 qsig decode
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
```

allow-connections sip to sip

supplementary-service h450.12

< **no supplementary-service h450.2** inserted here to force call by join>[10]

<**no supplementary-service h450.3** inserted here to force call by join> [10]

h323

sip

!

interface FastEthernet0/0

ip address 172.20.15.159 255.255.255.0

duplex auto

speed auto

!

interface FastEthernet0/1

no ip address

shutdown

duplex auto

speed auto

!

ip route 0.0.0.0 0.0.0.0 172.20.15.1

!

ip http server

!

tftp-server flash:P0030702T023.bin

tftp-server flash:P0030702T023.loads

tftp-server flash:P0030702T023.sb2

tftp-server flash:P0030702T023.sbn

< tftp-server flash: any load file that is not on the phone and is needed >

< tftp-server slot0: any load file that is not on the phone and is needed>

[10] Inserted to force IP call forward by join (no reroute).

```
!
control-plane
!
voice-port 0/1/0
!
voice-port 0/1/1
!
dial-peer voice 1 voip
 destination-pattern 2...
 session target ipv4:172.20.15.196
 dtmf-relay h245-alphanumeric
 codec g711ulaw
!
dial-peer voice 3011 voip
 destination-pattern 3011
 session target ipv4:172.20.15.196
 dtmf-relay h245-alphanumeric
 codec g711ulaw
!
dial-peer voice 3014 voip
 destination-pattern 3014
 session target ipv4:172.20.15.196
 dtmf-relay h245-alphanumeric
 codec g711ulaw
!
dial-peer voice 3012 voip
 destination-pattern 3012
 session target ipv4:172.20.15.196
```

```
 dtmf-relay h245-alphanumeric
 codec g711ulaw
!
dial-peer voice 3013 voip
 destination-pattern 3013
 session target ipv4:172.20.15.196
 dtmf-relay h245-alphanumeric
 codec g711ulaw
!
dial-peer voice 4300 voip
 destination-pattern 43..
 session target ipv4:172.20.15.196
 dtmf-relay h245-alphanumeric
 codec g711ulaw
!
dial-peer voice 5214 voip
 destination-pattern 5...
 session target ipv4:172.20.15.196
 dtmf-relay h245-alphanumeric
 codec g711ulaw
!
dial-peer voice 2 voip
 destination-pattern 36..
 session target ipv4:172.20.15.196
 dtmf-relay h245-alphanumeric
 codec g711ulaw
!
dial-peer voice 5 voip
 destination-pattern 5...
```

```
 session target ipv4:172.20.15.196
!
dial-peer voice 3700 voip
 destination-pattern 37..
 session target ipv4:172.20.15.196
 dtmf-relay h245-alphanumeric
 codec g711ulaw
!
sip-ua
!
telephony-service
 load 7960-7940 P0030702T023
 max-ephones 25
 max-dn 50
 ip source-address 172.20.15.159 port 2000
 max-conferences 8 gain -6
 call-forward pattern .T
 transfer-system full-consult
transfer-pattern .... <blind> [11]
 create cnf-files version-stamp Jan 01 2002 00:00:00
!
ephone-dn  5  dual-line
 number 3605
 name Remote IP5
 <call-forward busy 3603>[12]
< call-forward noan 3603 timeout 7>[13]
!
```

[11] Inserted to enable blind transfers, as opposed to early attended transfers.
[12] Inserted for call forward busy from SCCP extension.
[13] Inserted for call forward no reply from SCCP extension.

```
ephone-dn  6  dual-line
 number 3606
 name Remote IP6
!
ephone  5
 mac-address 0015.FA0C.B746
 type 7960
 keep-conference
 button  1:5
!
ephone  6
 mac-address 0015.FA63.BF84
 type 7960
 keep-conference
 button  1:6
!
line con 0
line aux 0
line vty 0 4
 password cisco
 login
!
scheduler allocate 20000 1000
!
end

REMOTE-2811#
```

Acronyms

Acronym	Definitions
BRI	Basic Rate ISDN
CAMA	Centralized Automatic Message Accounting
CAS	Channel Associated Signaling
CFB	Call Forward when Busy
CFNR	Call Forward when No Reply
CFU	Call Forward Unconditional
CO	Central Office
FGD	Feature Group “D”
FXO	Foreign Exchange – Office
FXS	Foreign Exchange – Station
IOS	Internetworking Operating System
MCID	Malicious Caller ID
MGCP	Media Gateway Control Protocol
MoH	Music on Hold
MWI	Message Waiting Indication
PBX	Private Branch Exchange
PRI	Primary Rate ISDN
PSAP	Public Service Access Point
SIP	Session Initiation Protocol
ToH	Tone on Hold

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