Application Note

Alcatel 4400 Release 6.0 using E1 QSIG to Cisco Unified Communications Manager Express Release 4.0(2)

November 1, 2007 Revision 2

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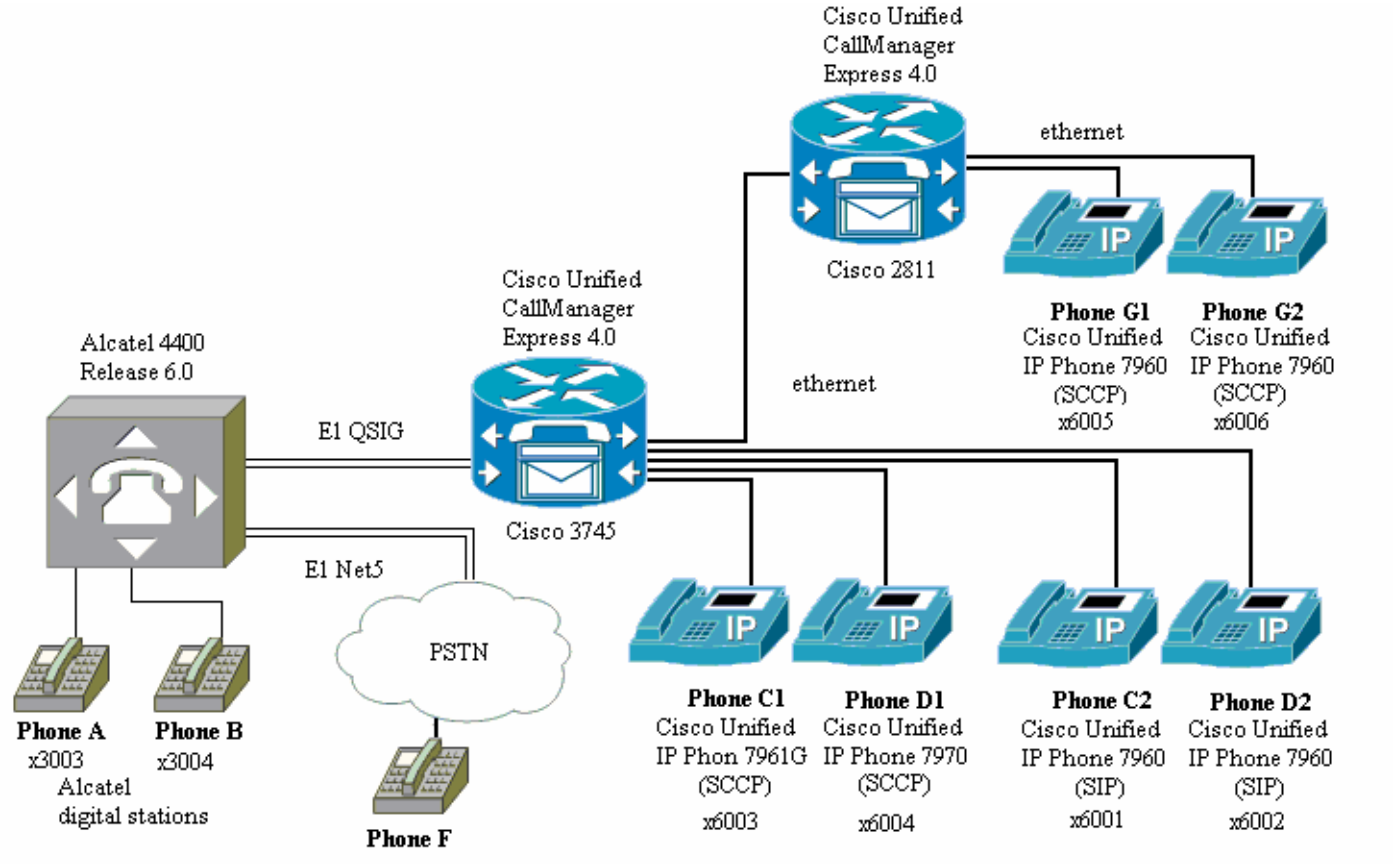
Introduction

- This is an Application Note for connectivity between an Alcatel 4400 Release 6.0 PBX and Cisco Unified Communications Manager Express Release 4.0(2) using a Cisco 3745 voice gateway with QSIG protocol.
- The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability with Cisco Unified Communications Manager Express Release 4.0(2) connected to the PBX via the 3745 E1 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 Communications Manager Express 4.0(2). Cisco Unified IP phones (models 7960, 7961G, and 7970) were connected to the 2 Cisco Unified Communications Manager Express gateways via SIP and SCCP, as per the figure. A NM-HDV and VWIC-1MFT-E1 were used for the E1 QSIG interface. Calls were made to test basic call, caller ID, transfer, forward, and reroute features.
- This Application Note uses the 3745 voice gateway. However, the use of other Cisco voice gateways is also an option since Cisco Unified Communications 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(2) supports SIP end-points with limited number of features.

Network Topology

Figure 1. Test Network Topology.



Limitations

Basic Calls

- Overlap dialing is not supported from Cisco Unified Communications Manager Express.
- Connected Name is not supported on calls between PBX and Cisco Unified IP Phone running SIP.
- Alerting Name is not supported on calls between PBX and Cisco Unified IP Phone running SIP.
- If CLIR is configured for outbound calls on Cisco Unified Communications Manager Express 4.0(2), it treats name and number on an incoming call as restricted, even if they are not set as restricted on the PBX.
- Connected Number/Name Restriction are not supported.

Call Transfers

- The Alcatel PBX will not perform a true blind transfer. It can perform a consultation transfer or early attended transfer.
- An early-attended local transfer that originates from a PBX extension to a SIP extension, and is then transferred to another SIP extension (e.g., A calls C2 and C2 transfers to D2), the transferring phone (C2 in example) stays in the call until the final destination (D2 in example) answers.
- An early-attended network/external transfer that originates from a SIP extension to another SIP extension, and is then transferred to a PBX extension (e.g., C2 calls D2 and D2 transfers to A), the transferring phone (D2 in example) stays in the call until the final destination (A in example) answers.
- A blind local transfer originated from a call placed from a PBX station to a SCCP phone on the local Cisco Unified Communications Manager Express, and then transferred to a SIP phone on the same Cisco Unified Communications Manager Express (e.g., A calls C1, and C1 transfers to D2) does not complete. The call drops as soon as the number is dialed.
- A consultation or early-attened transfer originated from a call placed from a phone on the remote Cisco Unified Communications Manager Express to a SIP phone on the local Cisco Unified Communications Manager Express, and then transferred to a PBX phone (e.g., G1 calls C2, and C2 transfers to A) does not complete properly.
- For consultation network/external transfers and early attended network/external transfers, and all consultation local transfers and early attended local transfers that involve a transfer from a SCCP phone to a SIP phone, the original calling name and number are not displayed on the final destination. 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

- The Alcatel PBX will not perform call forward by join. It always performs a reroute in calls where a PBX phone is in the middle.
- For calls from Cisco Unified Communications Manager Express 4.0(2) to a PBX phone that is then forwarded locally to another PBX phone (e.g., Phone C1 calls Phone A, and Phone A forwards to Phone B), the Alcatel PBX sends a reroute proposal, and the Cisco Unified Communications Manager Express responds with a second SETUP message (i.e., a new call). This feature is inherent to the PBX and can not be turned off.
- For local CFNR from a SIP phone to a PBX phone to another PBX phone (e.g., phone C2 calls phone A, and phone A forwards on no response to phone B), the call connects, but has 1-way voice.
- For ”trombone” or ”hairpin” calls from a PBX phone to a Cisco Unified Communications Manager Express SIP phone to a PBX phone (e.g., phone A calls phone C2, which forwards to phone B), the call completes, but Cisco Unified Communications Manager Express does not perform a reroute, even when reroute is enabled.
- For ”trombone” or ”hairpin” calls involving overlap dialing from a PBX phone to a Cisco Unified Communications Manager Express (SCCP or SIP) phone to a PBX phone (e.g., phone A calls phone C1/C2 using overlap dialing, and C1/C2 forwards to phone B), the call completes, but there is no reroute. If the Cisco Unified Communications Manager Express phone is a SIP phone, there is no reroute proposal. If the Cisco Unified Communications Manager Express phone is a SCCP phone, there is a reroute propsal, but the PBX ignores it. It is believed that this is because overlap dialing on the Alcatel PBX is accomplished by a direct trunk select (as opposed to

ARS), and this causes the PBX to ignore the reroute proposal because the first trunk is ”nailed up”. This is believed to be an inherent limitation of the Alcatel PBX.

- Forwarded calls originated from a PBX extension to a remote Cisco Unified Communications Manager Express SCCP extension, and forwarded to a local Cisco Unified Communications Manager Express extension (e.g., A calls G1, and G1 forwards to C1), Cisco Unified Communications Manager Express performs a QSIG reroute, even though a QSIG reroute is not in order (i.e., there is no QSIG ”hairpin” or ”trombone”).
- 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. In some instances, the correct connected number is sent by the PBX and replaced by Cisco Unified Communications Manager Express with the forwarding number.

MWI

- Cisco Unified Communications Manager Express 4.0(2) supports Cisco Unity integration with QSIG. However, in this instance, no testing was performed with Cisco Unified Communications Manager Express 4.0(2) as the message center PINX.
- There was no PBX voice mail system present at the time of testing. Therefore, no testing was performed with the PBX as the message center PINX.

System Components

Hardware Requirements

- Cisco 3745 IOS voice gateway
  - NM-HDV
  - VWIC-2MFT-E1
- Cisco 2811 IOS voice gateway
- (4) Cisco Unified IP phone 7960s
- (1) Cisco Unified IP phone 7961G
- (1) Cisco Unified IP phone 7970
- (1) Alcatel 4400 PBX
  - (2) Alcatel 4035 *Advanced Reflexes* digital phones
  - (2) PRA2 trunk cards

Software Requirements

- Cisco Unified Communications Manager Express Release 4.0(2)
- Cisco IOS Software, 3700 Software (C3745-IPVOICE-M), Version 12.4(11)T
- Cisco IOS Software, 2800 Software (C2800NM-IPVOICE-M), Version 12.4(11)T
- Alcatel 4400 software release 6.0

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: SCCP41.8-0-4SR2S
- Cisco 7961G IP phone app load ID: jar41sccp.8-0-3-32.sbn
- Cisco 7961G IP phone boot load ID: 7961G_64-020704128Amd64meg.bin

D1 – 7970 – SCCP

- Cisco7970 IP phone load file: SCCP70.8-0-4SR2S
- Cisco 7970 IP phone app load ID: jar70sccp.8-0-3-32.sbn
- Cisco 7970 IP phone boot load ID: 7970_64060118.bin

Features

Features Supported

- Basic Call, ENBLOC dialing
- Basic Call, Overlap dialing from PBX to Cisco Unified Communications Manager Express
- CLIP-Calling Line (Number) Identification Presentation on Basic and Forwarded Calls
- CNIP-Calling Line (Name) Identification Presentation on Basic Calls and Forwarded Calls
- CLIP-Calling Line (Number) Identification Presentation on Transferred Calls (See Limitations Section)
- CNIP-Calling Line (Name) Identification Presentation on Transferred Calls (See Limitations Section)
- CLIR-Calling Line (Number) Identification Restriction (See Limitations Section)
- CNIR-Calling Line (Name) Identification Restriction (See Limitations Section)
- COLP-Connected Line (Number) Identification Presentation on Basic Calls (See Limitations section.)
- CONP-Connected Line (Name) Identification Presentation on Basic Calls (See Limitations section.)
- Alerting Name (See Limitations section.)
- Tandem PSTN call
- Consultation Transfer – Local (See Limitations Section)
- Consultation Transfer – Network/External (See Limitations Section)
- Early Attended Transfer – Local (See Limitations Section)
- 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)

Features Not Supported

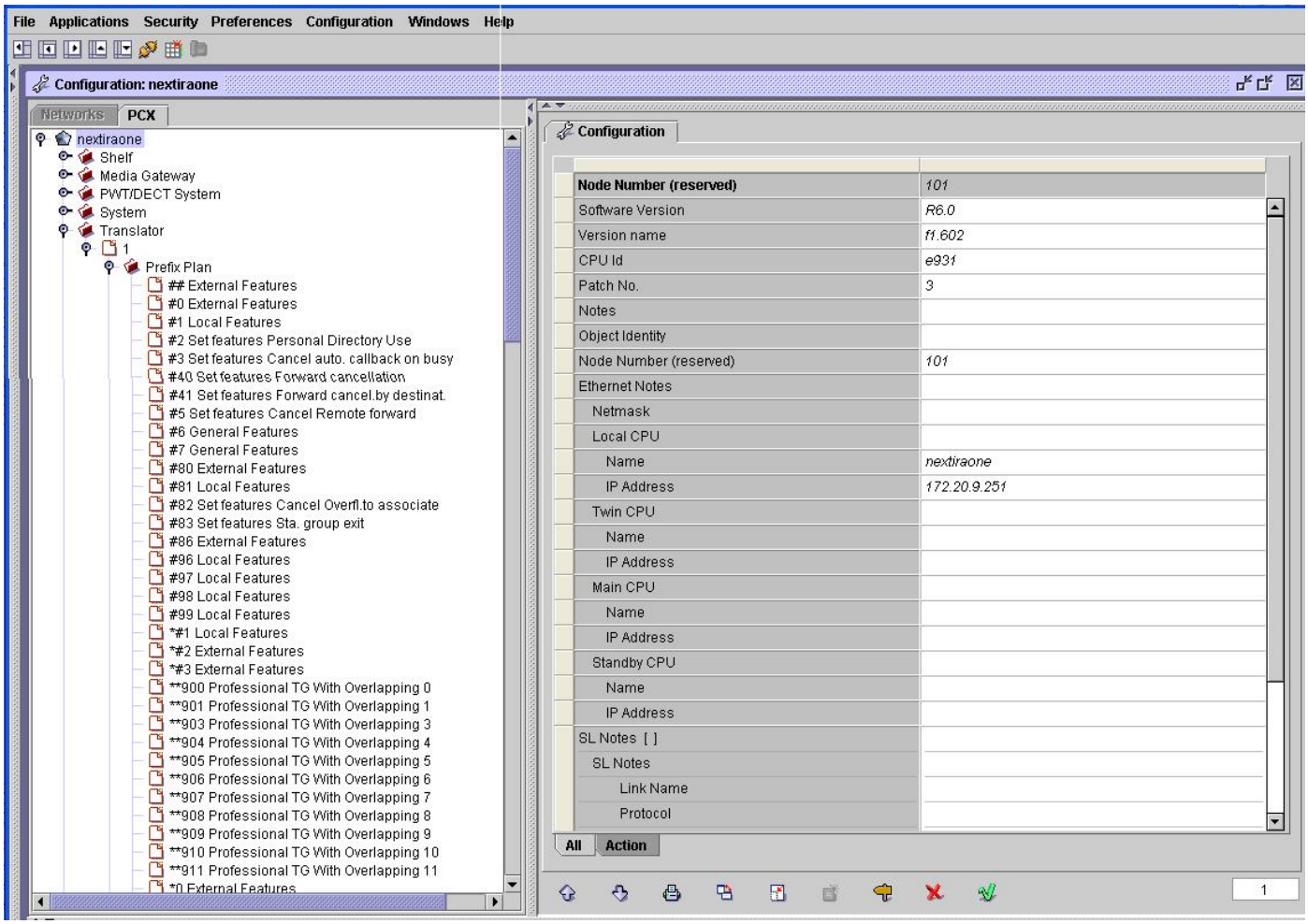
- Overlap dialing from Cisco Unified Communications Manager Express
- COLR- Connected Line (Number) Identification Restriction
- CONR- Connected Line (Name) Identification Restriction
- Blind Transfers initiated from PBX
- COLP-Connected Line (Number) Identification Presentation on Transferred Calls
- CONP-Connected Line (Name) Identification Presentation on Transferred Calls
- COLP-Connected Line (Number) Identification Presentation on Forwarded Calls
- CONP-Connected Line (Name) Identification Presentation on Forwarded Calls
- 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 Alcatel 4400 Release 6.0

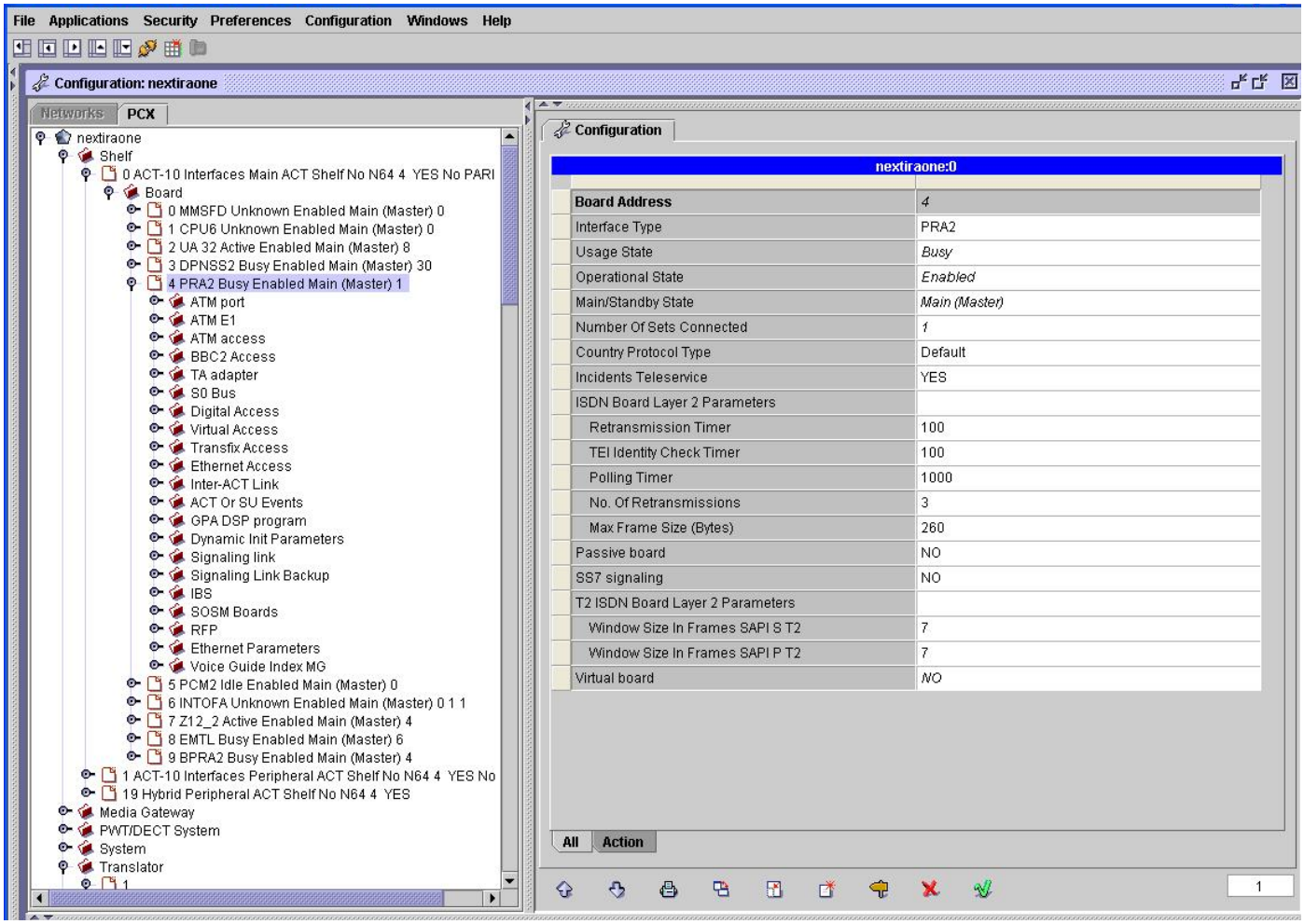
System Software

Figure 2. PBX system software – 1 of 1.



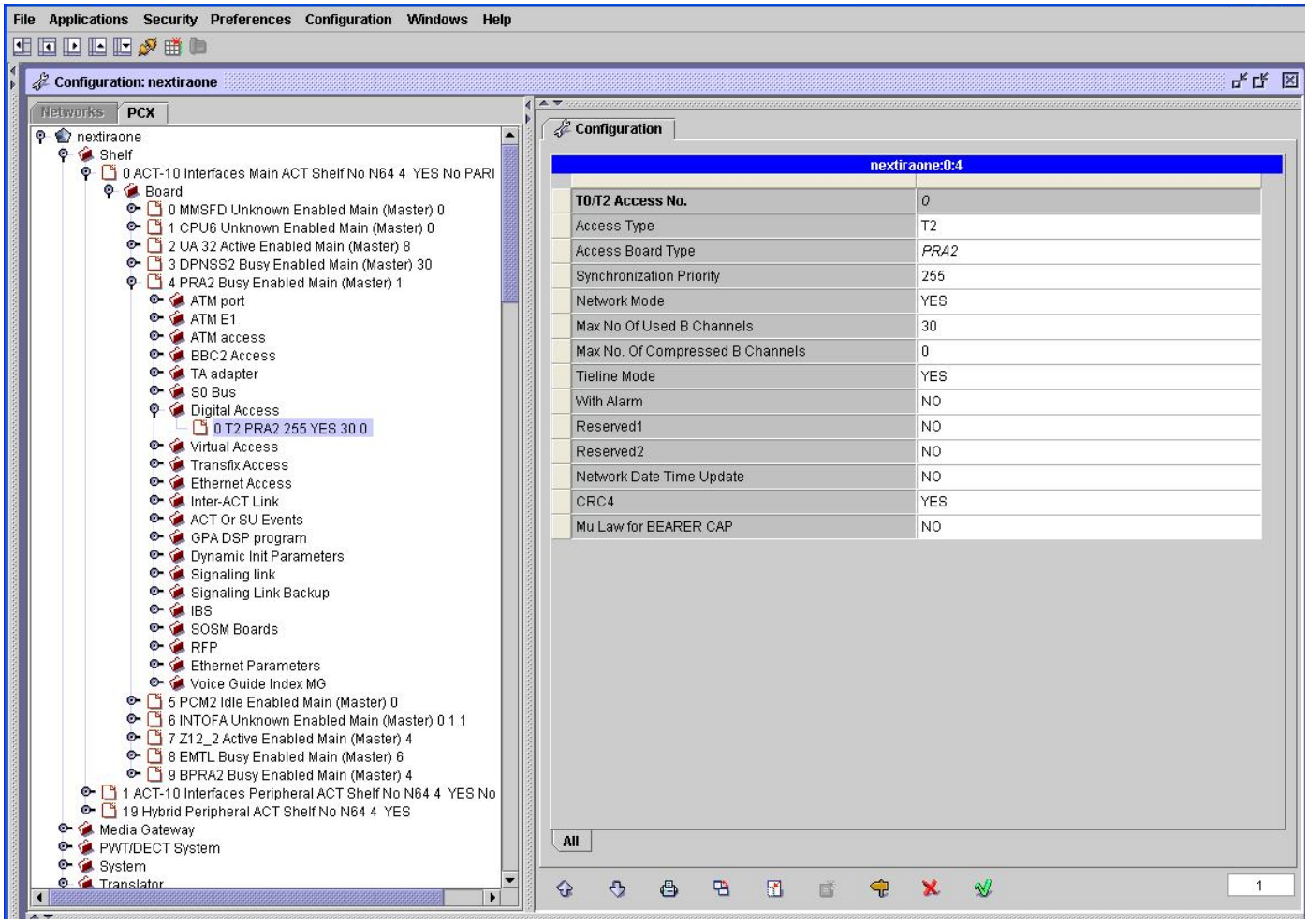
Circuit Board

Figure 3. Circuit board configuration – 1 of 1.



Digital Access

Figure 4. Digital Access – 1 of 1.



“ISO Function” System Parameter

Figure 5. ISO function system parameter configuration – 1 of 3.

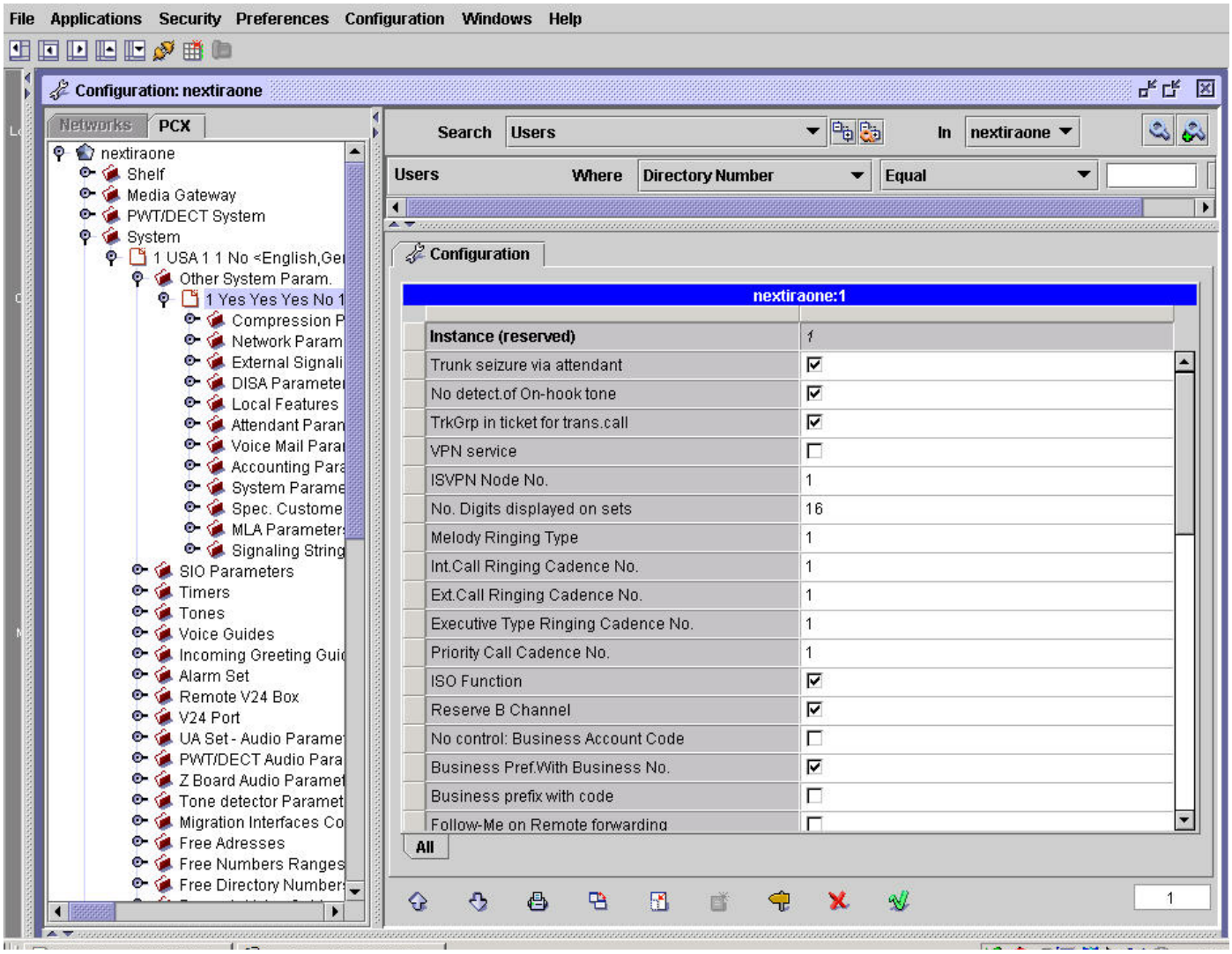


Figure 6. ISO function system parameter configuration – 2 of 3.

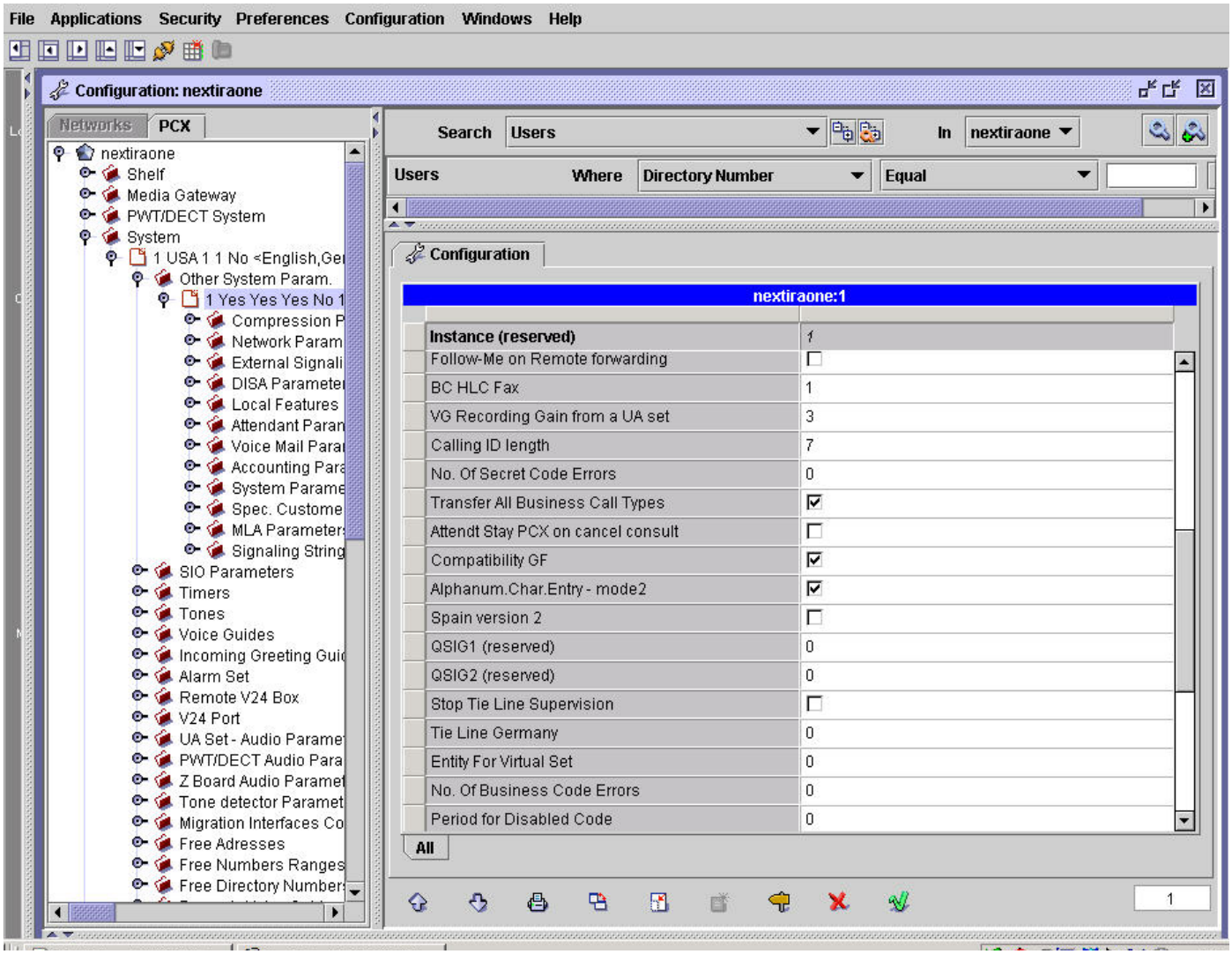
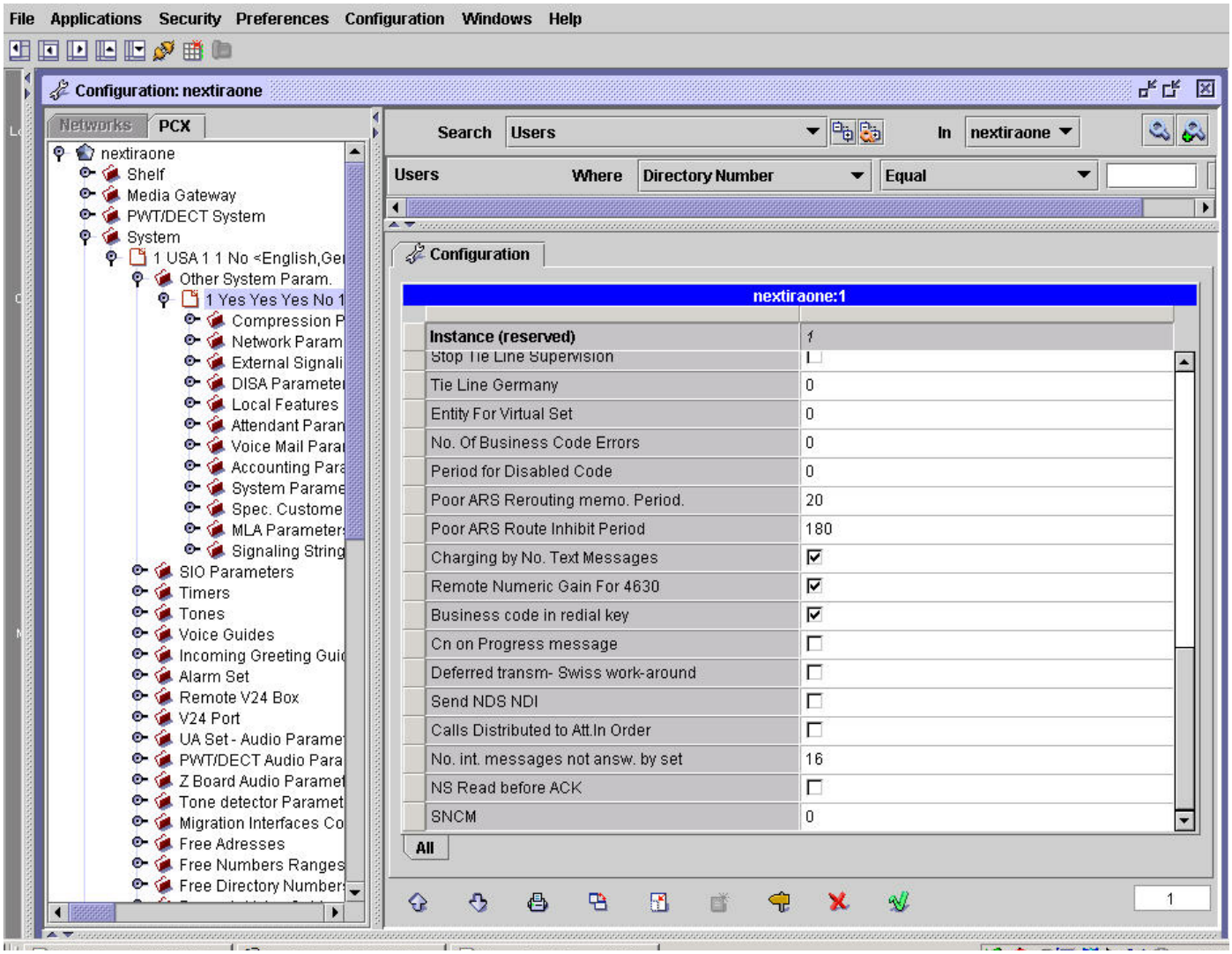


Figure 7. ISO function system parameter configuration – 3 of 3.



PRI ABC_F Trunk Group

Figure 8. PRI ABC_F Trunk configuration – 1 of 5.

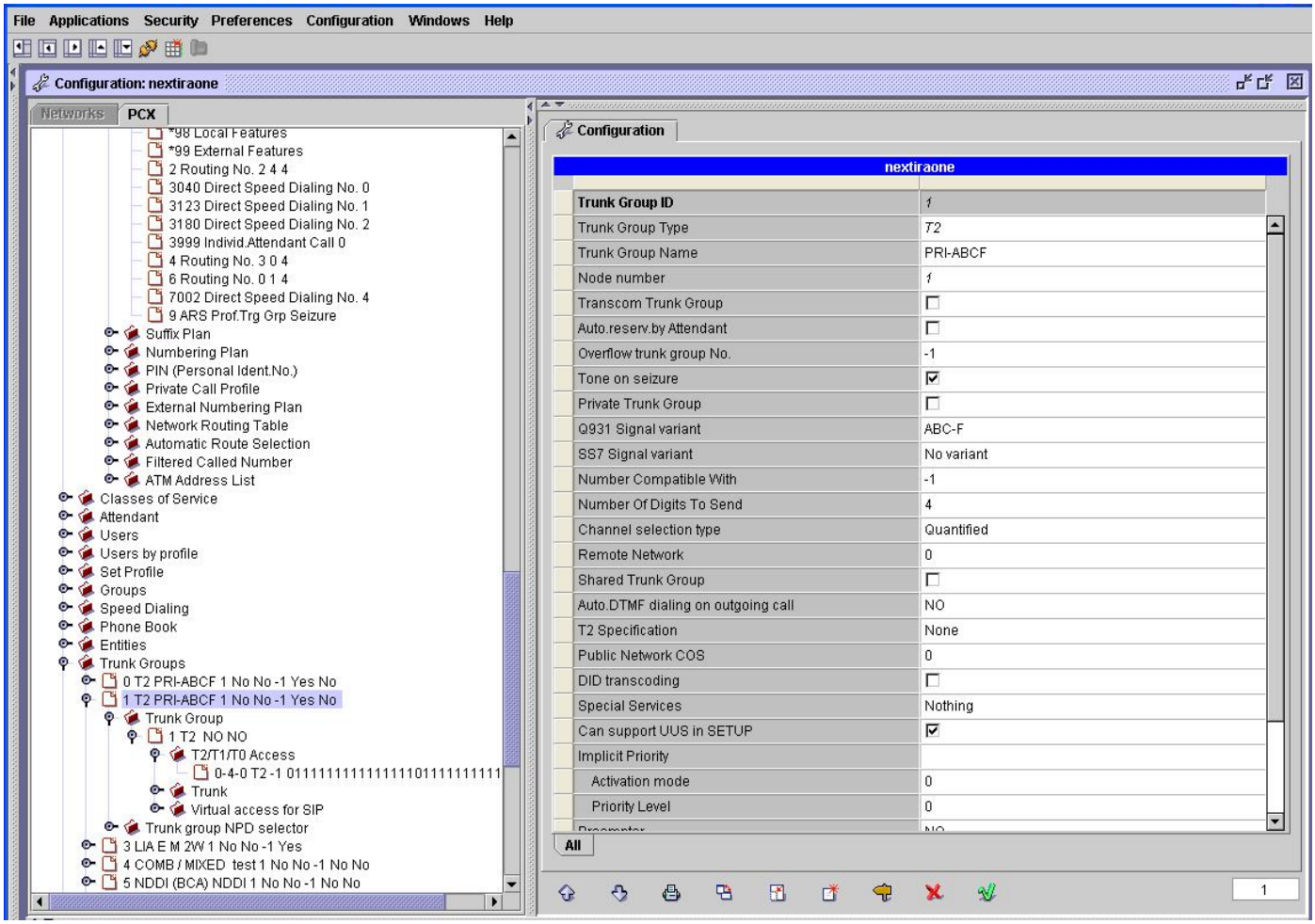


Figure 9. PRI ABC_F Trunk configuration – 2 of 5.

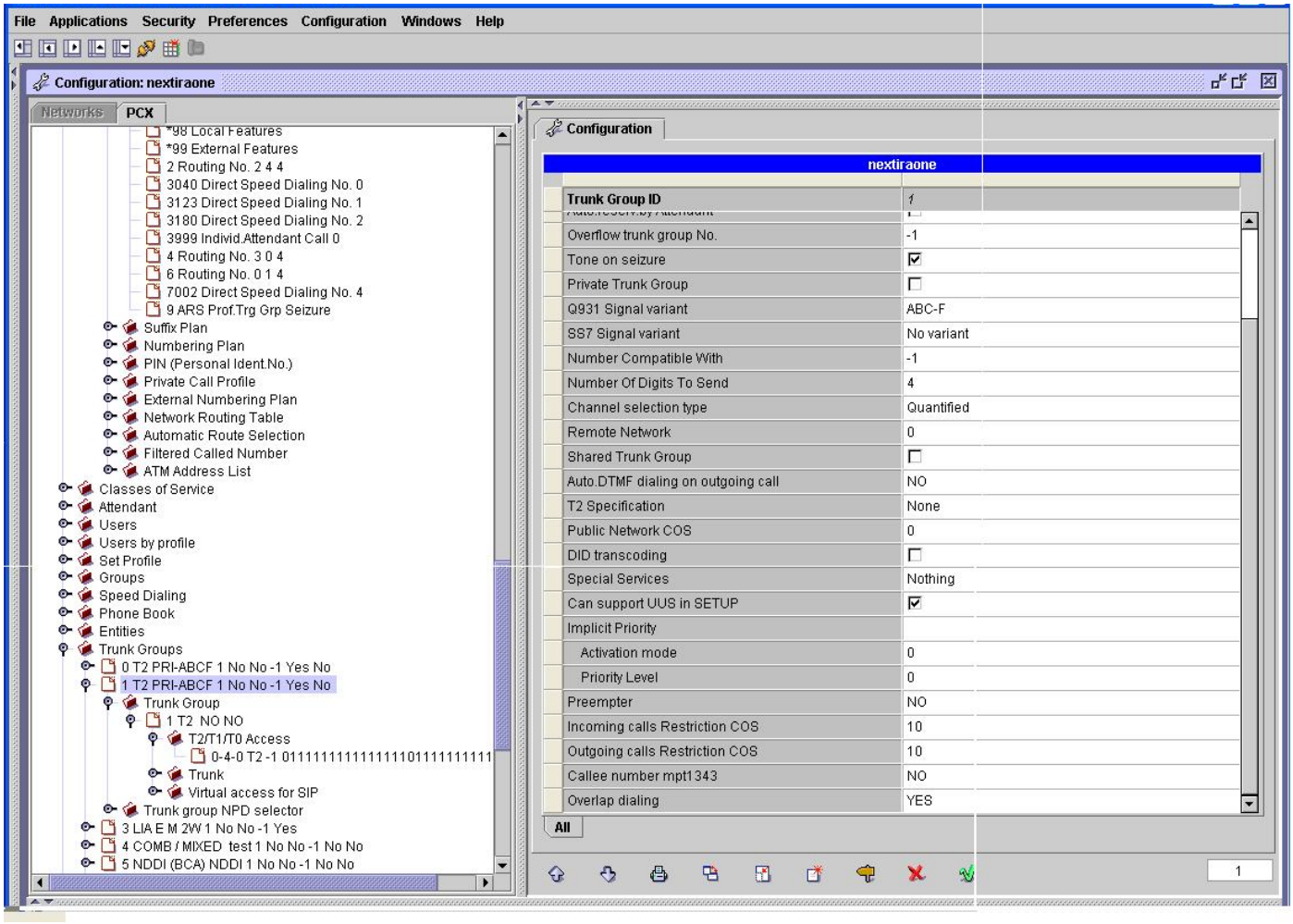


Figure 10. PRI ABC_F Trunk configuration – 3 of 5.

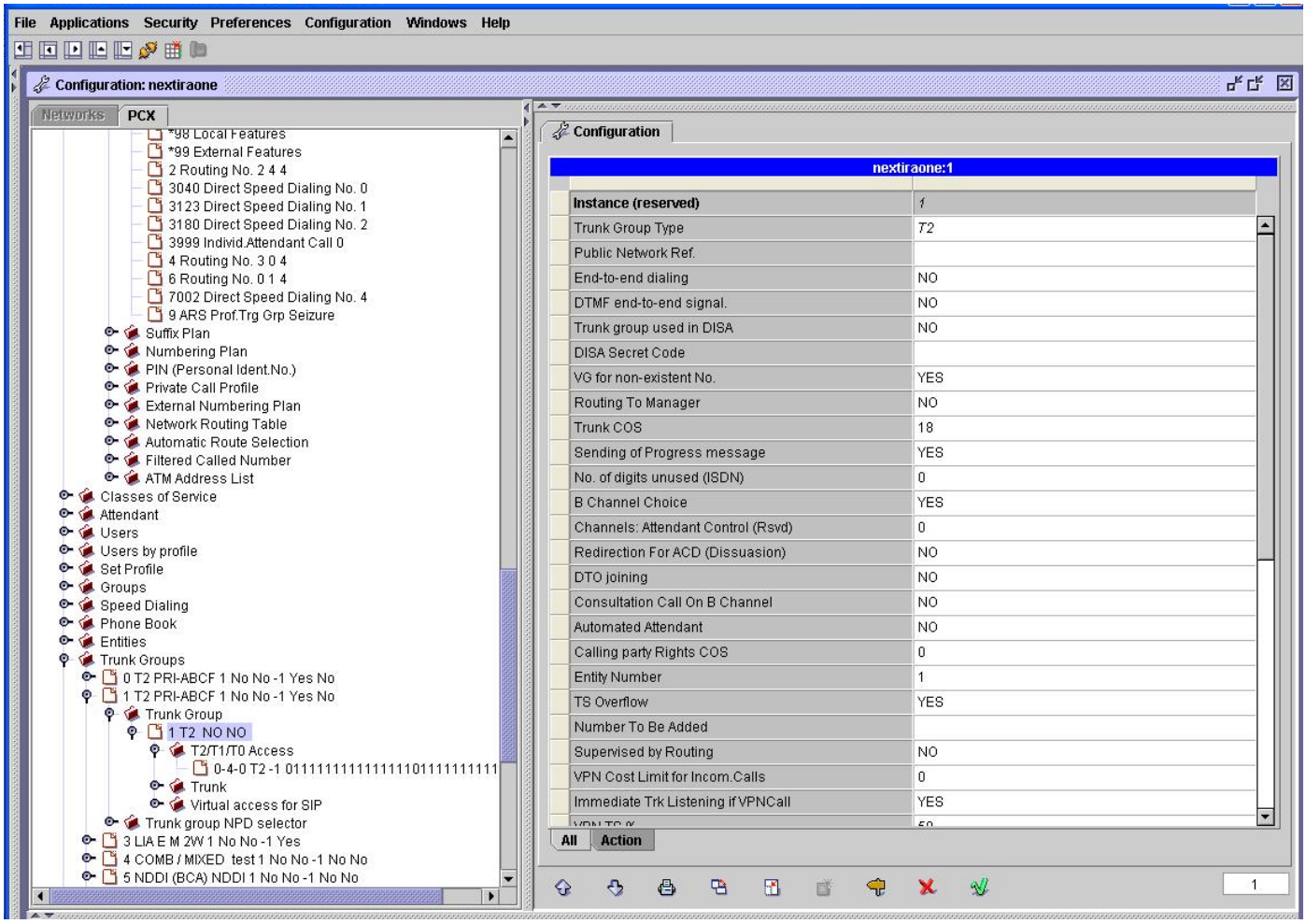


Figure 11. PRI ABC_F Trunk configuration – 4 of 5.

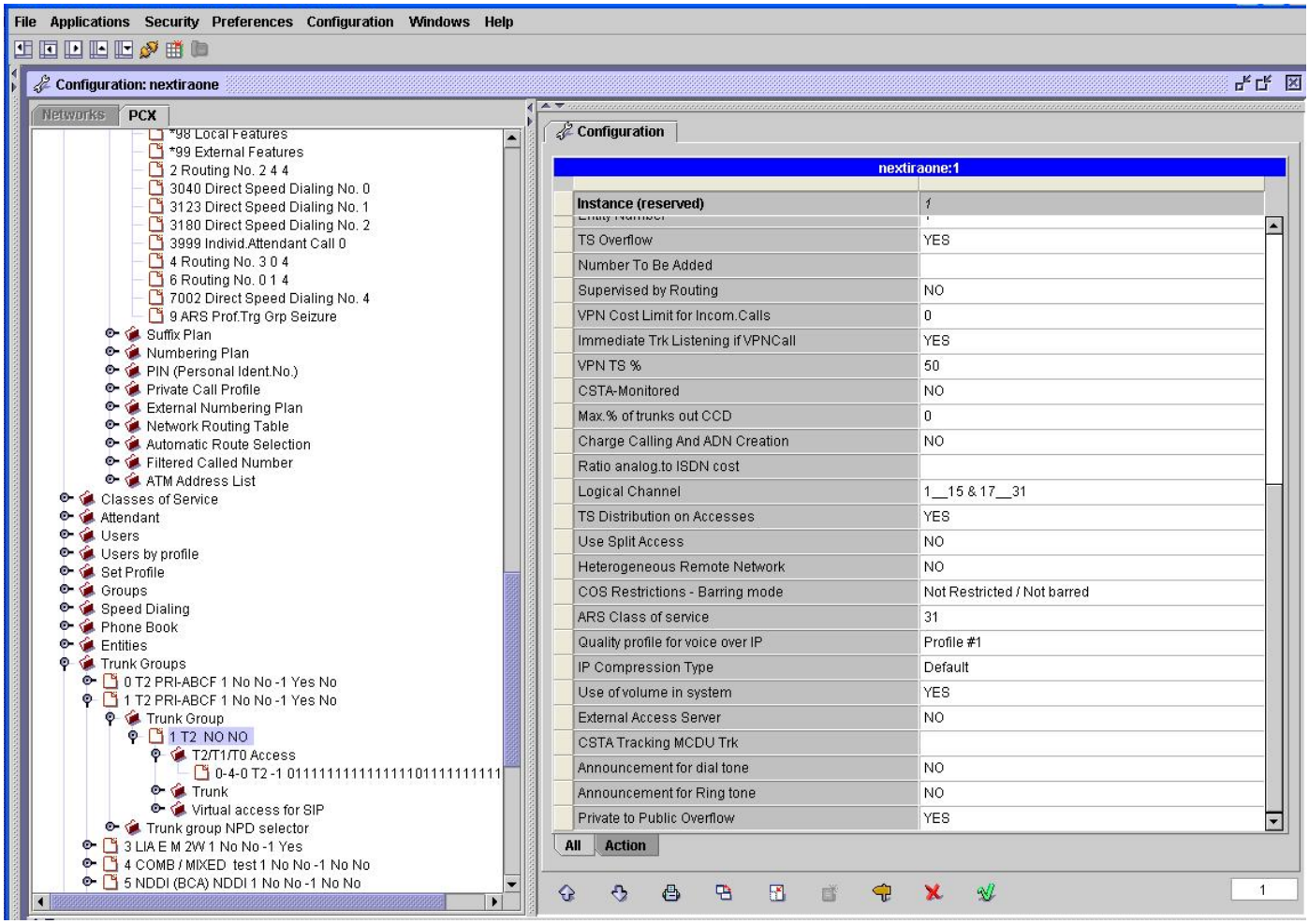
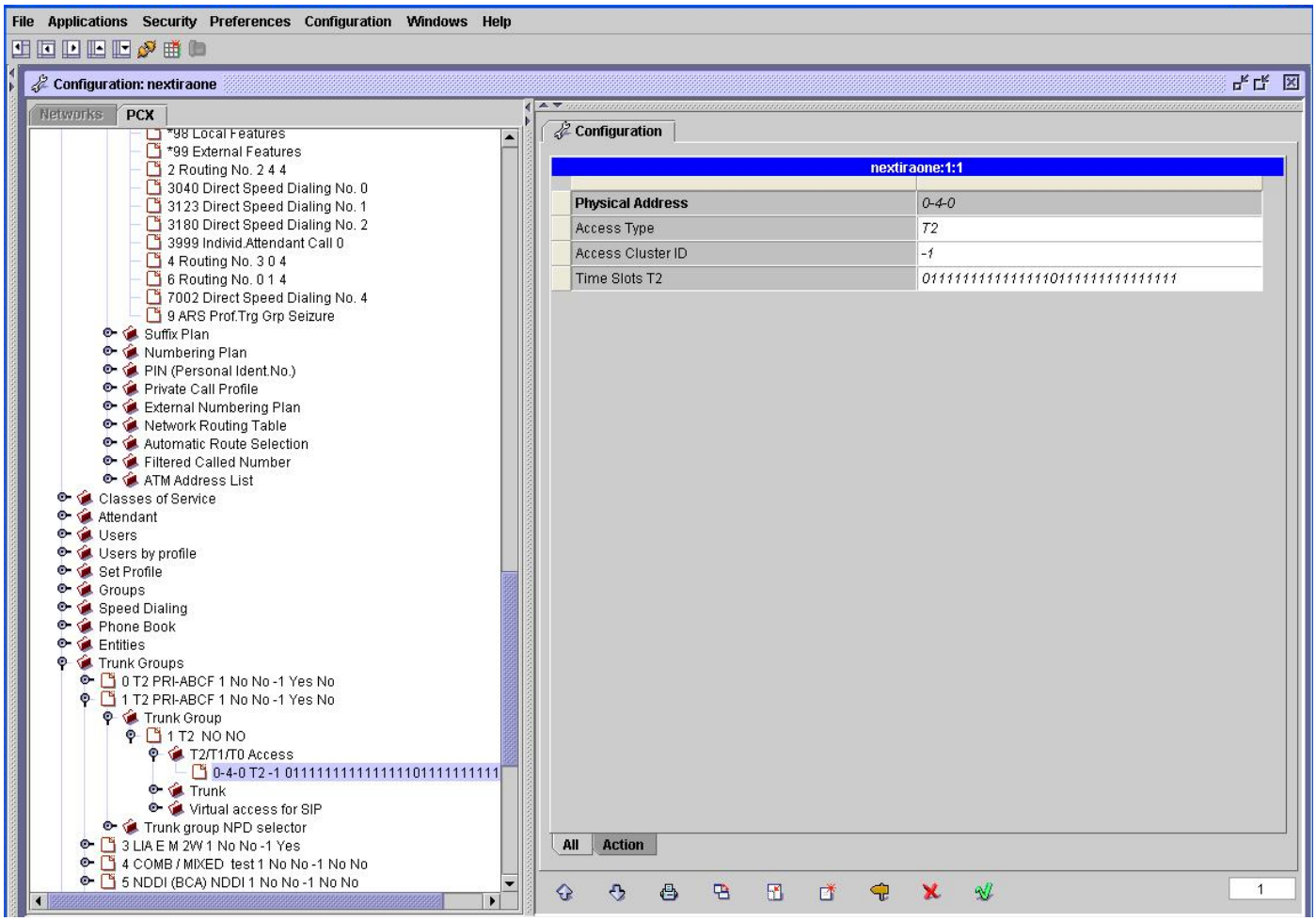


Figure 12. PRI ABC_F Trunk configuration – 5 of 5.



Routing Prefixes

Figure 13. Routing prefix configuration – 1 of 2 (ENBLOC dialing).

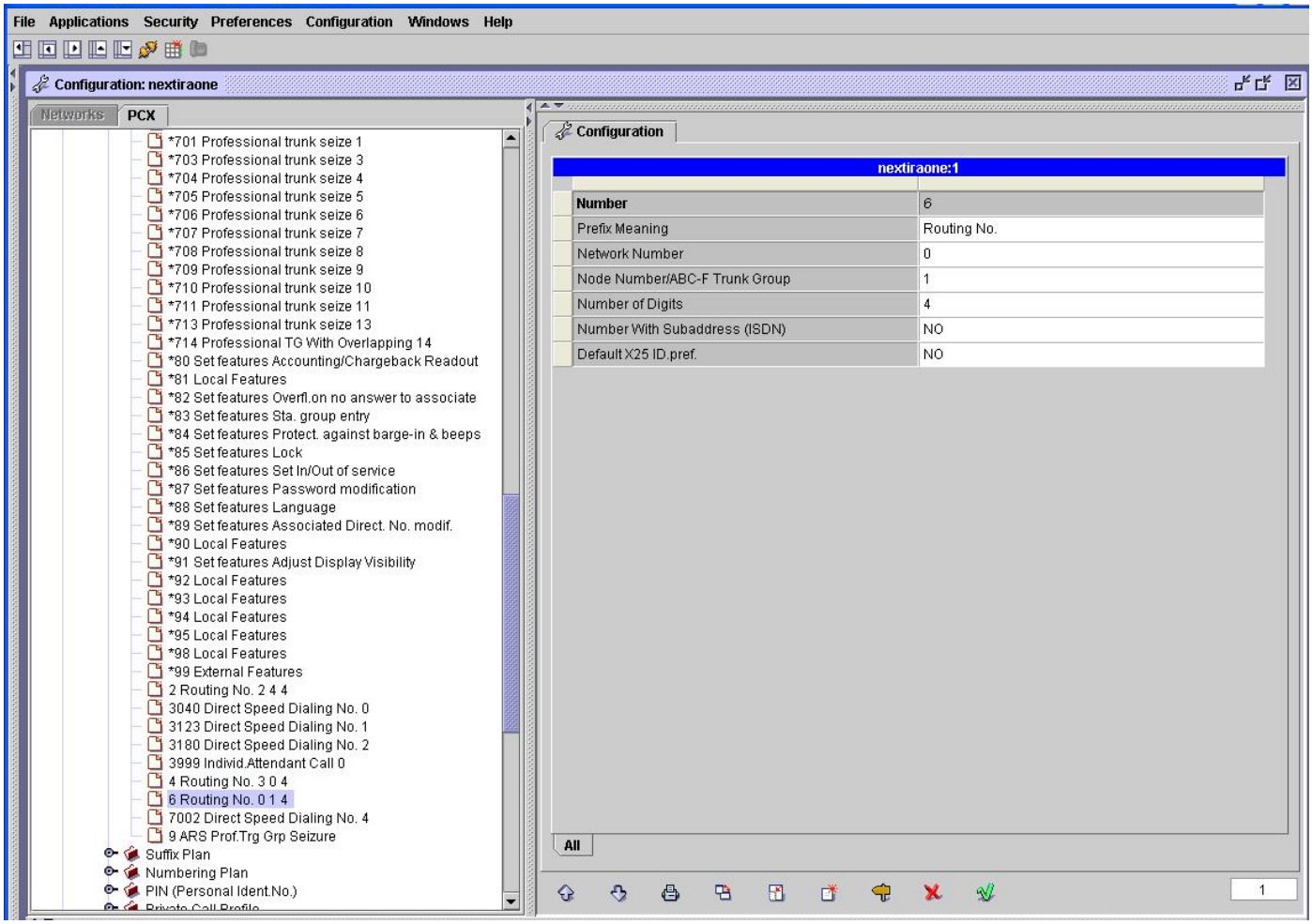
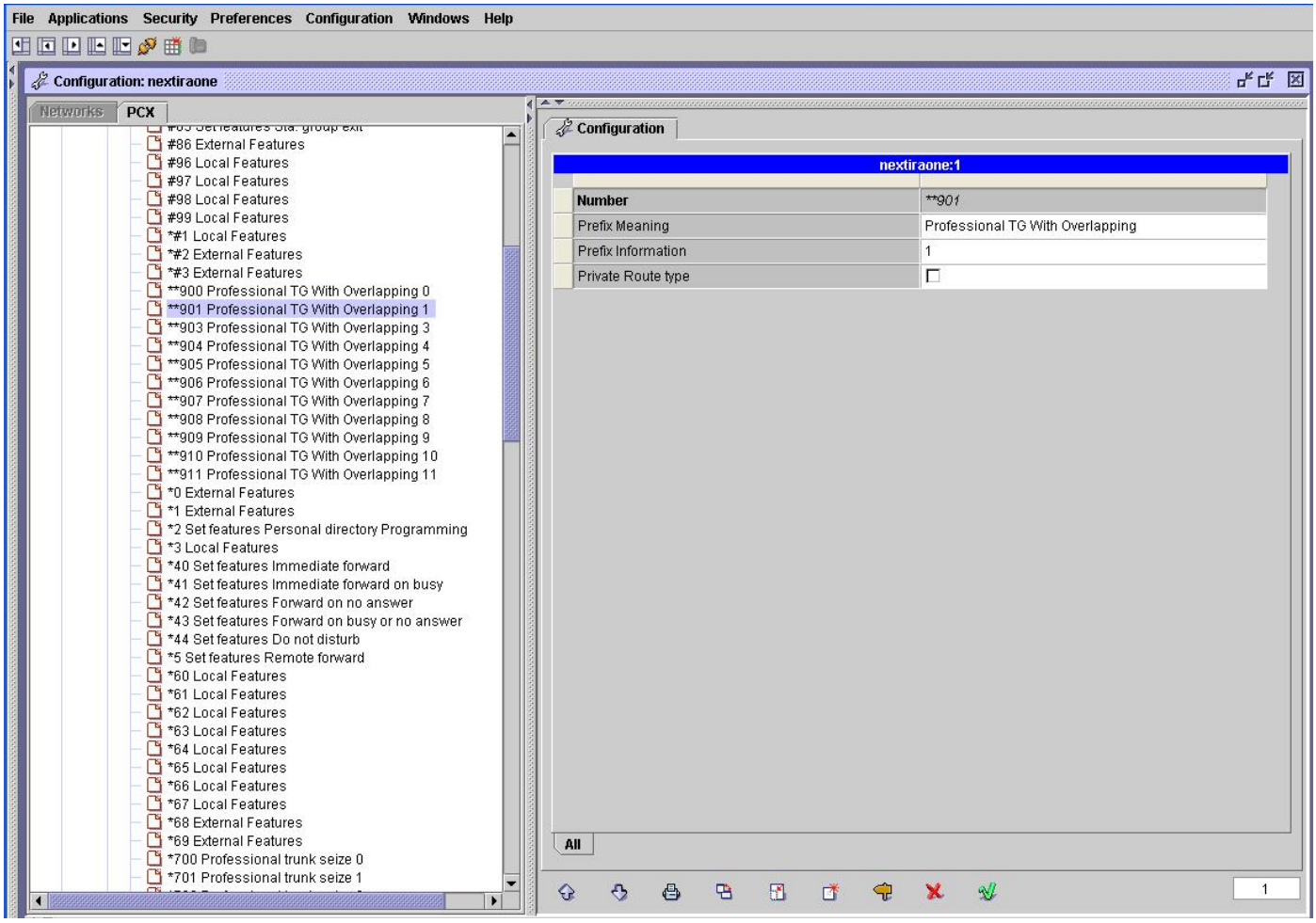


Figure 14. Routing prefix configuration – 2 of 2 (overlap dialing).



Digital Station

Figure 15. Digital station configuration – 1 of 6.

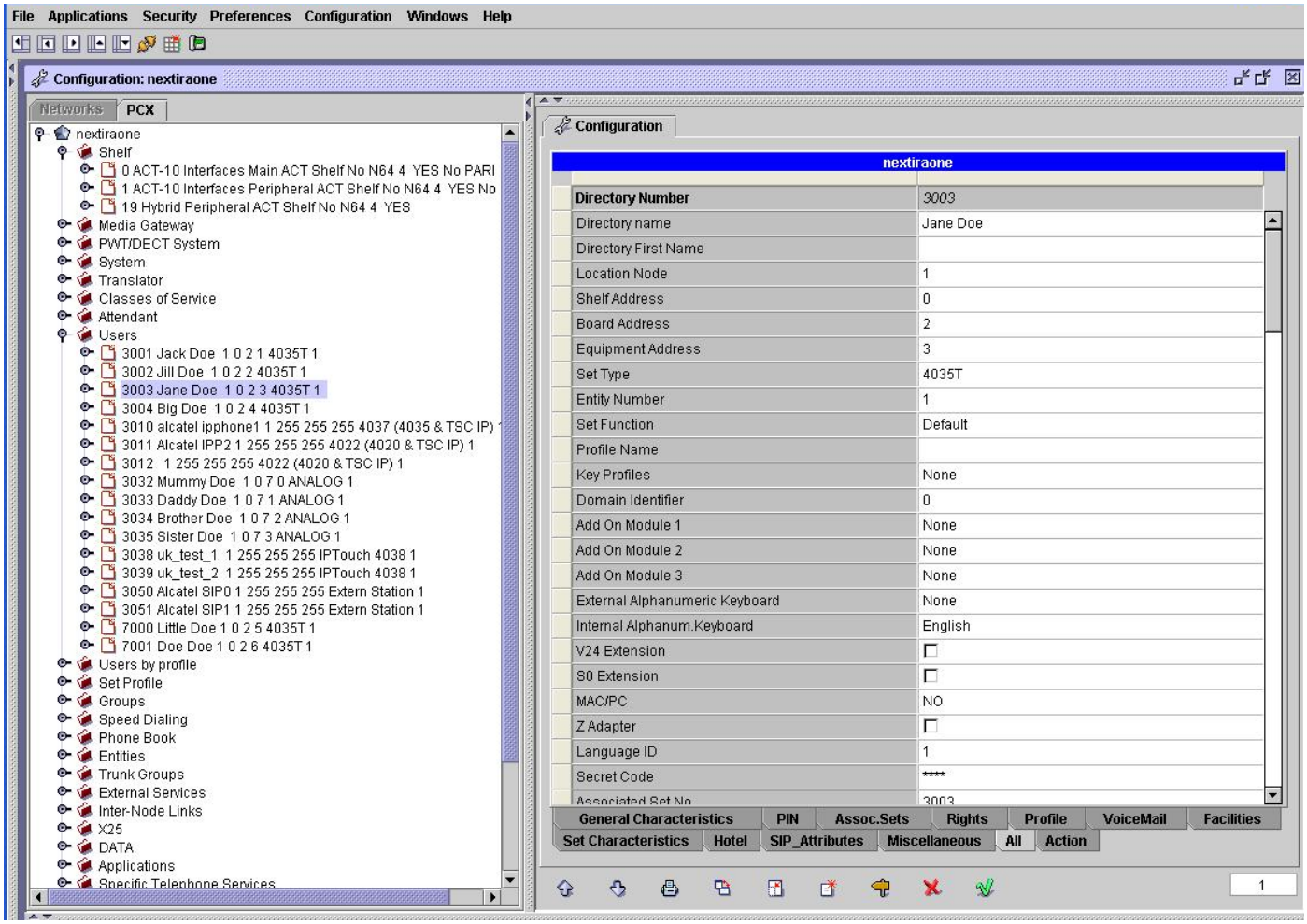


Figure 16. Digital station configuration – 2 of 6.

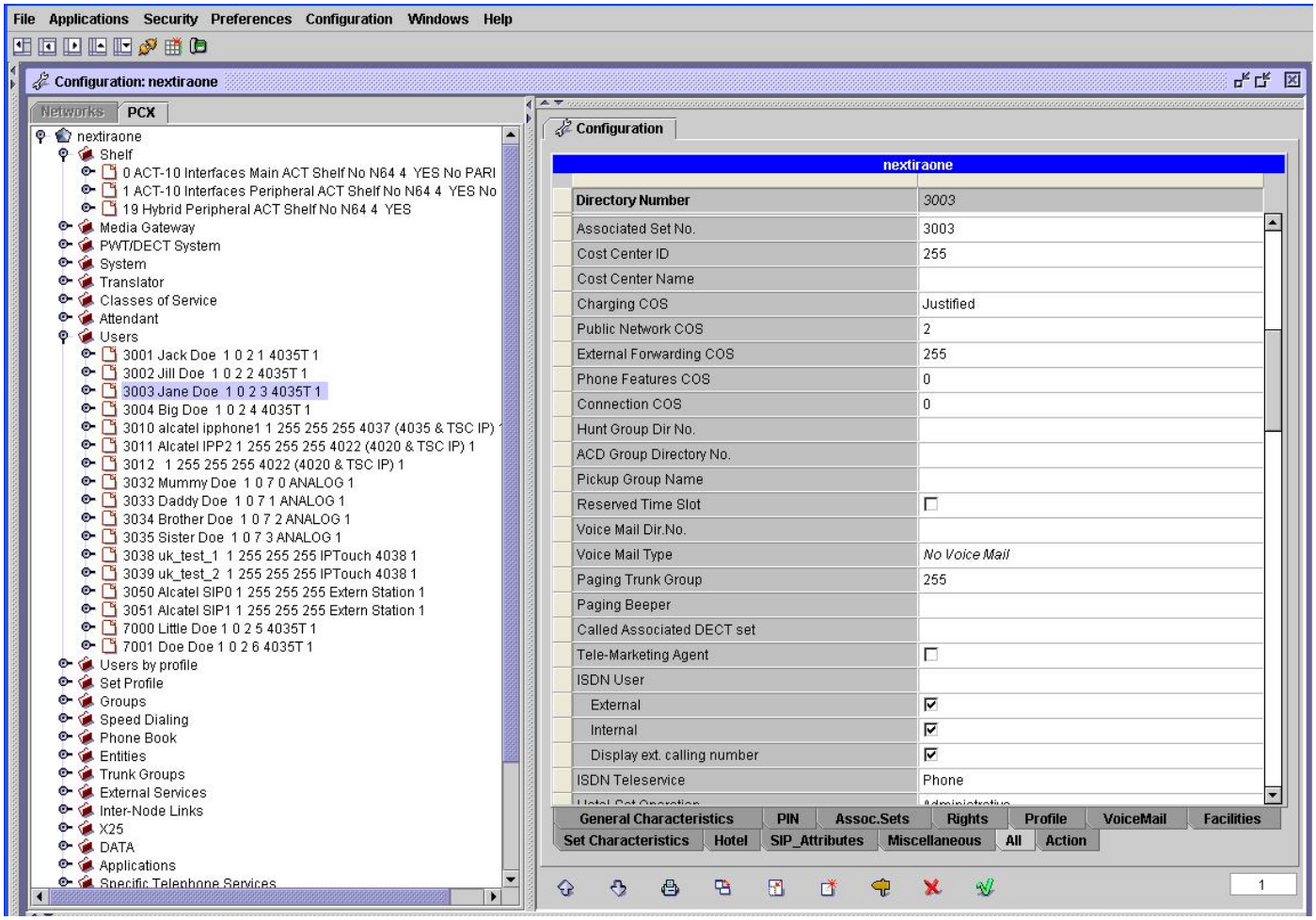


Figure 17. Digital station configuration – 3 of 6.

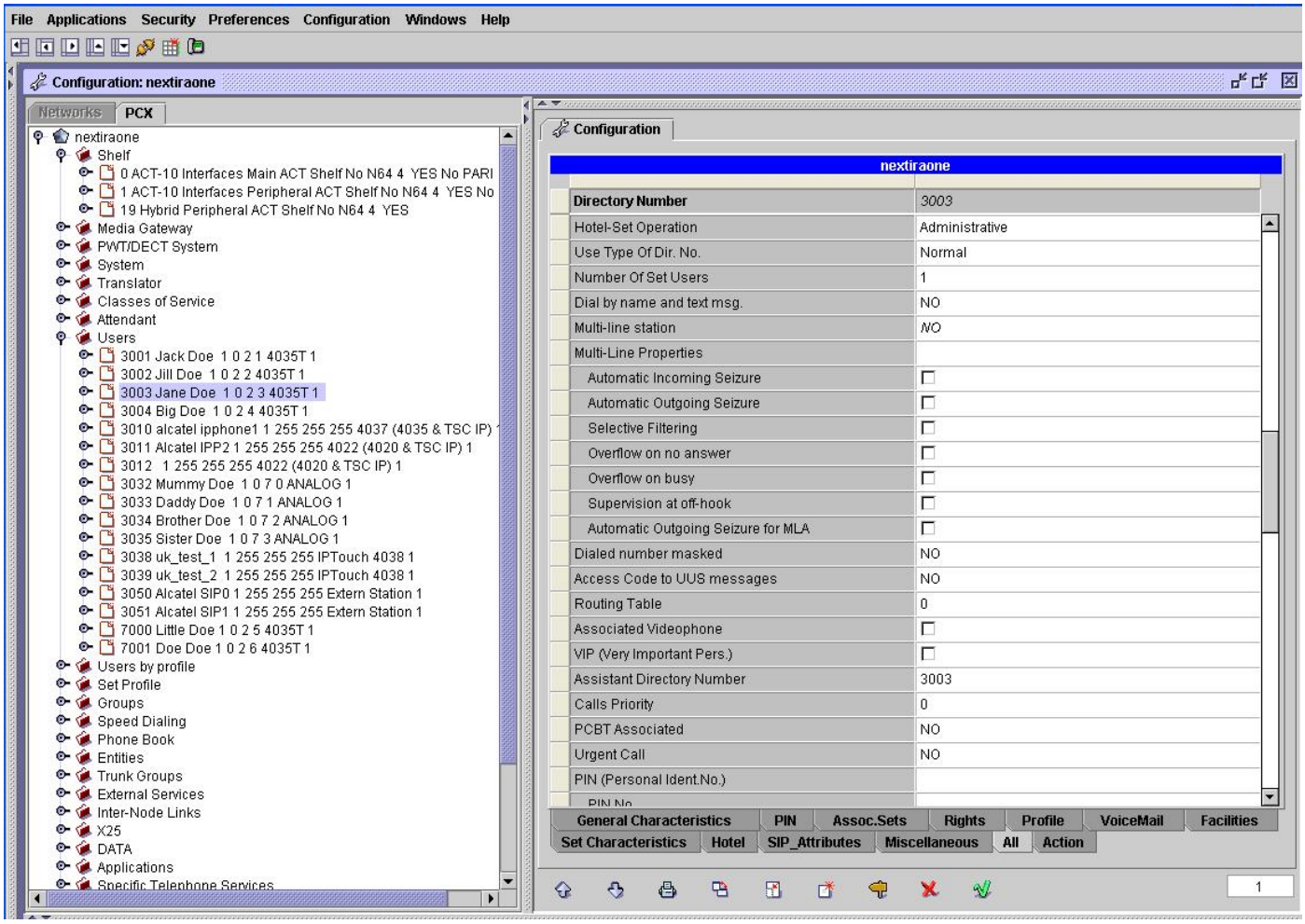


Figure 18. Digital station configuration – 4 of 6.

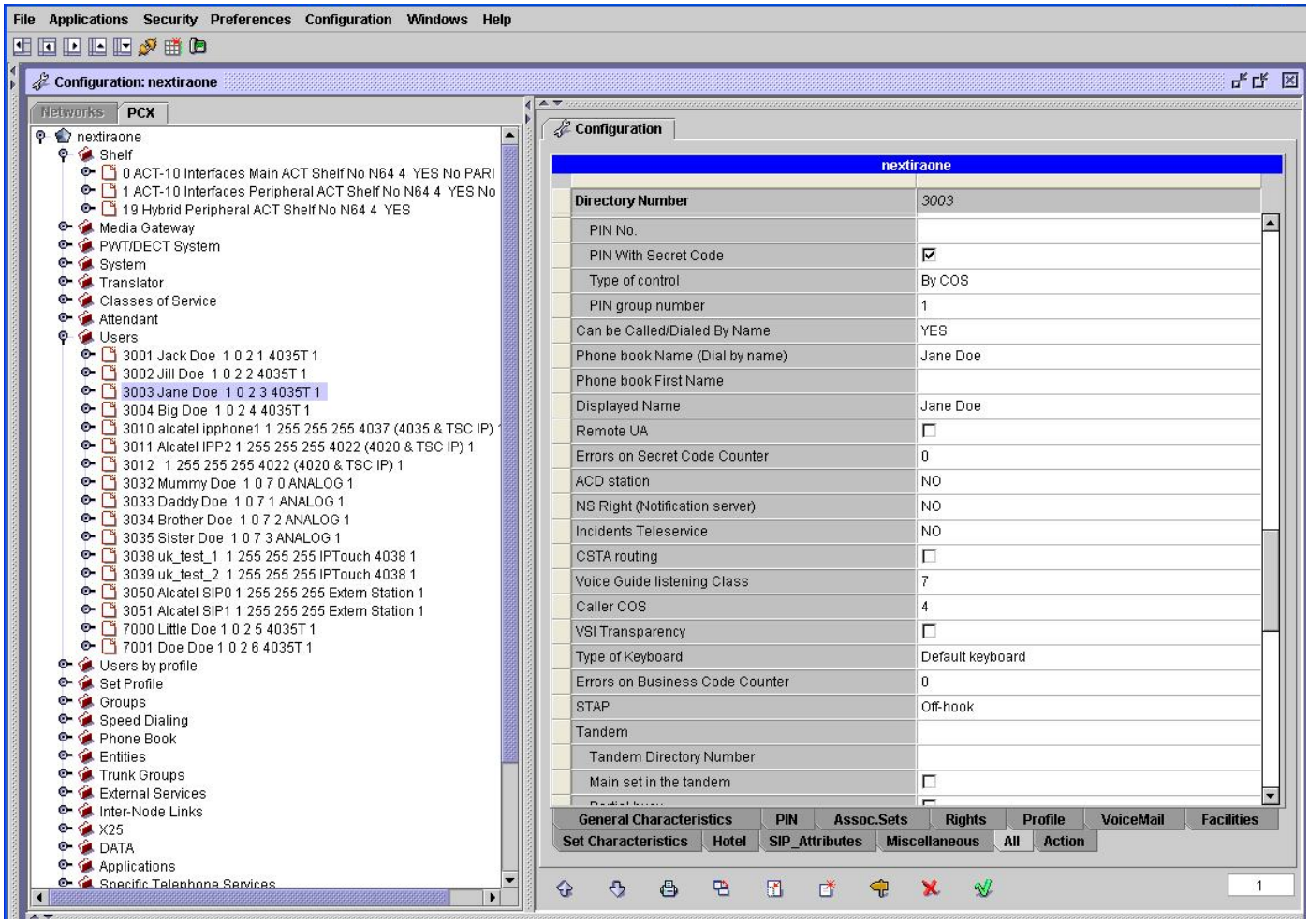


Figure 19. Digital station configuration – 5 of 6.

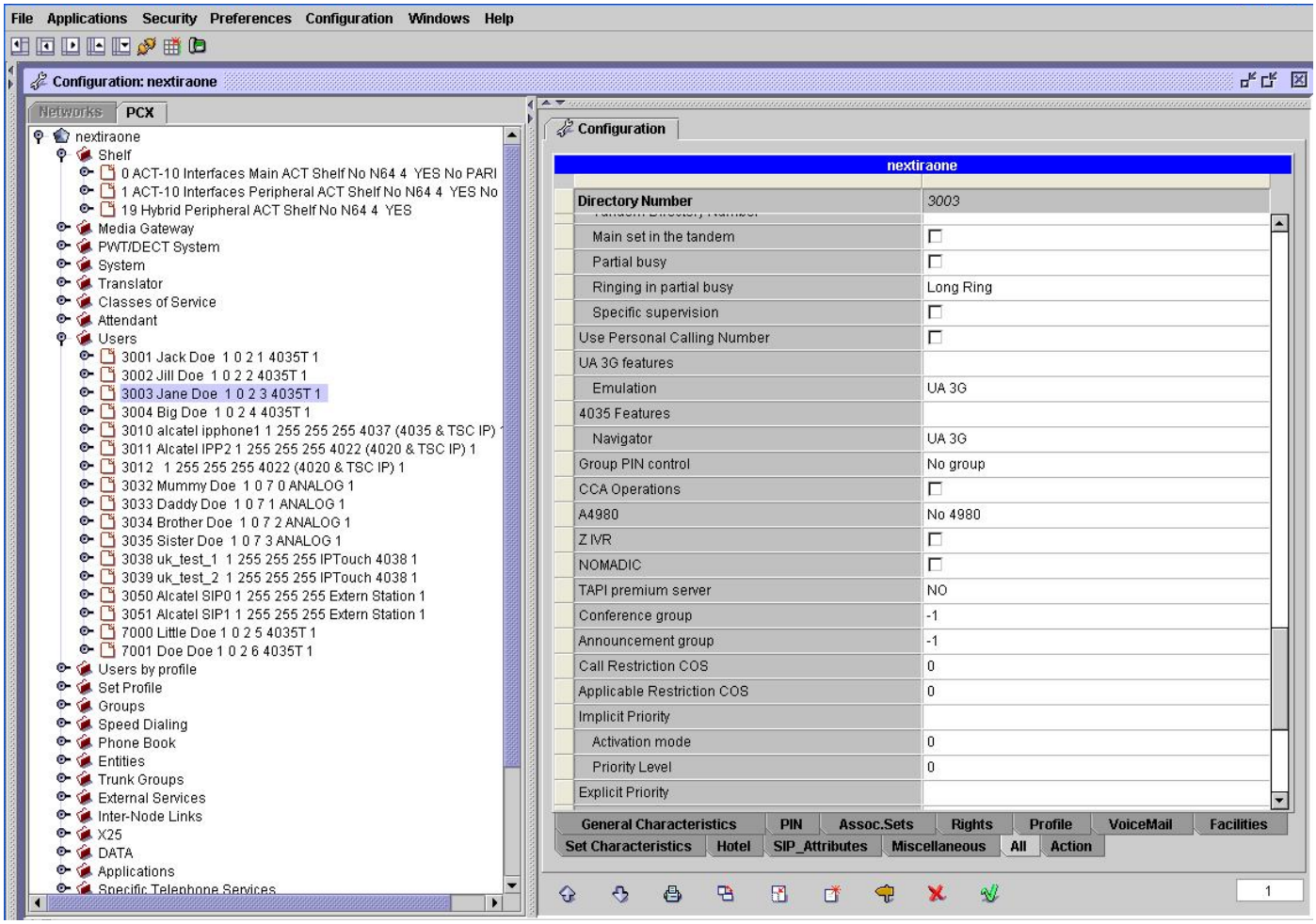
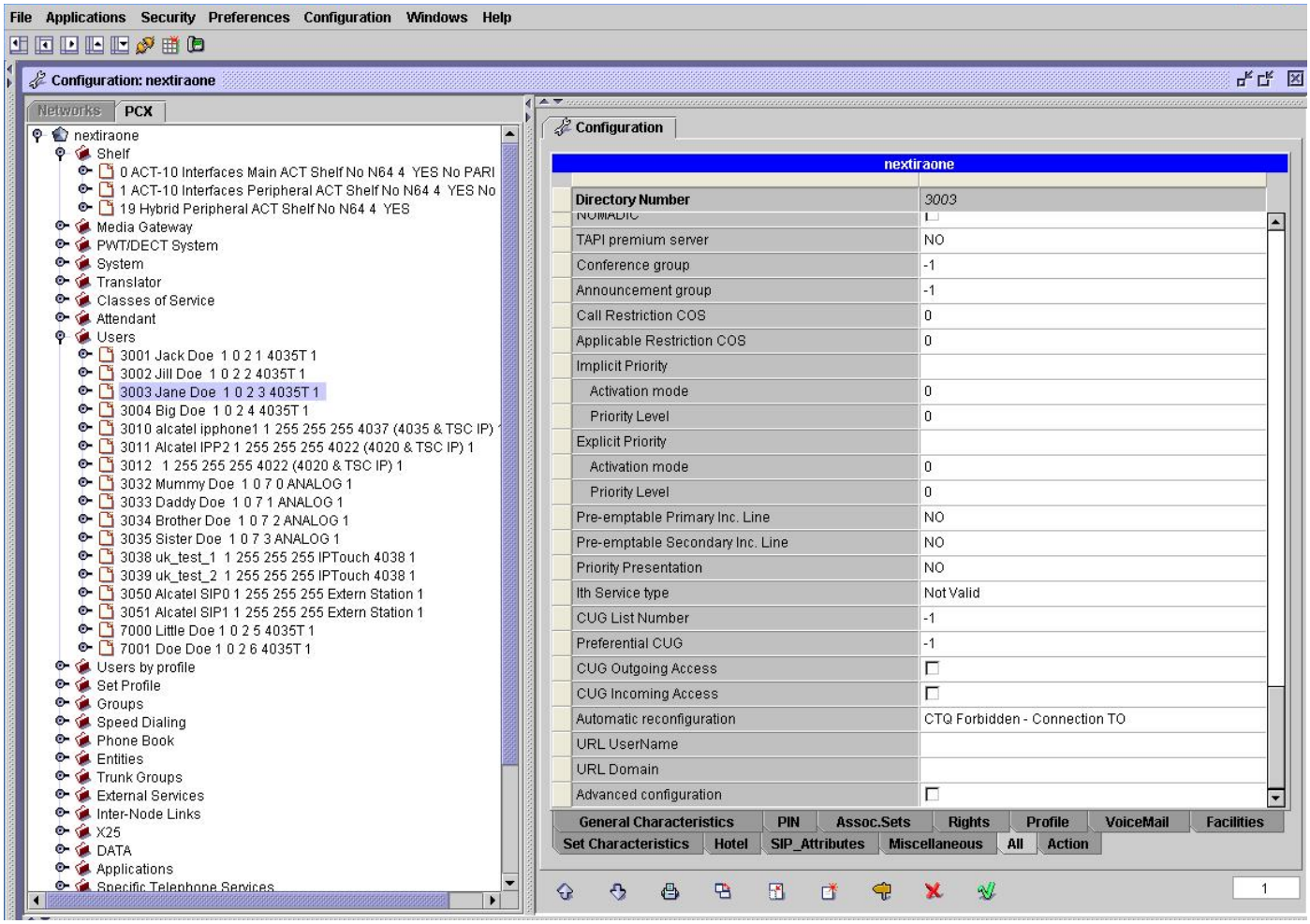


Figure 20. Digital station configuration – 6 of 6.



Digital Station Phone Facilities

Figure 21. Digital station facilities configuration – 1 of 9.

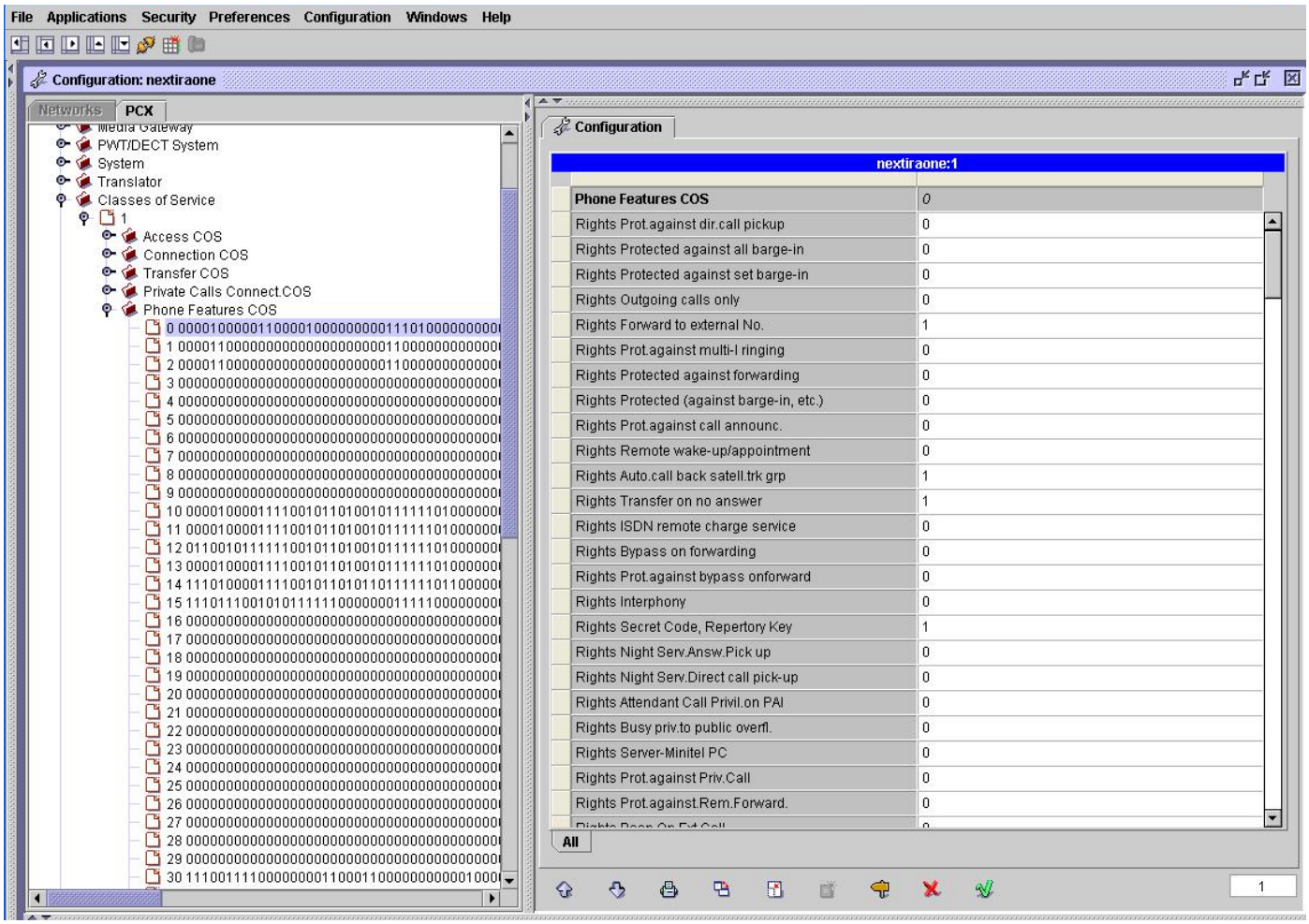


Figure 22. Digital station facilities configuration – 2 of 9.

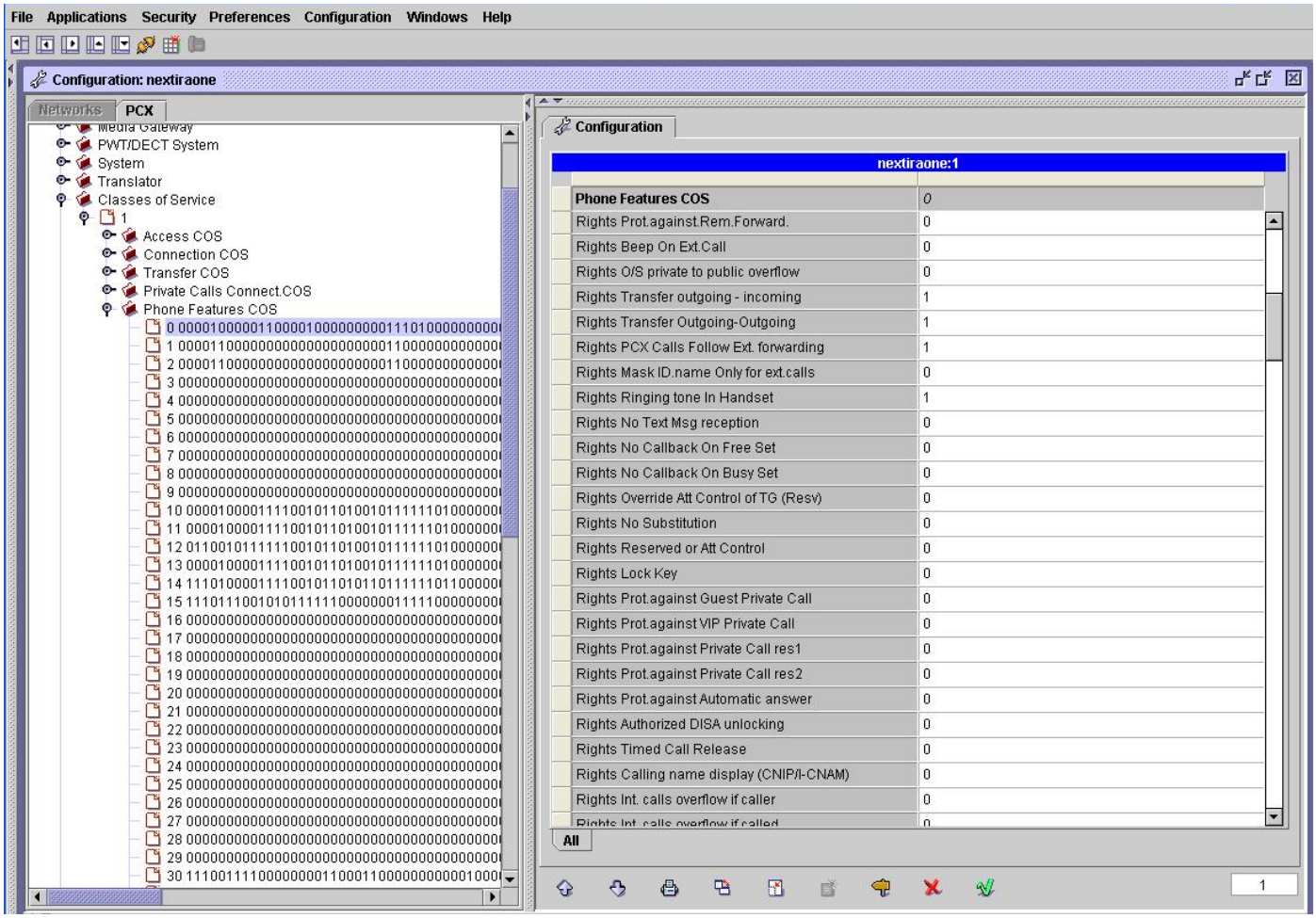


Figure 23. Digital station facilities configuration – 3 of 9.

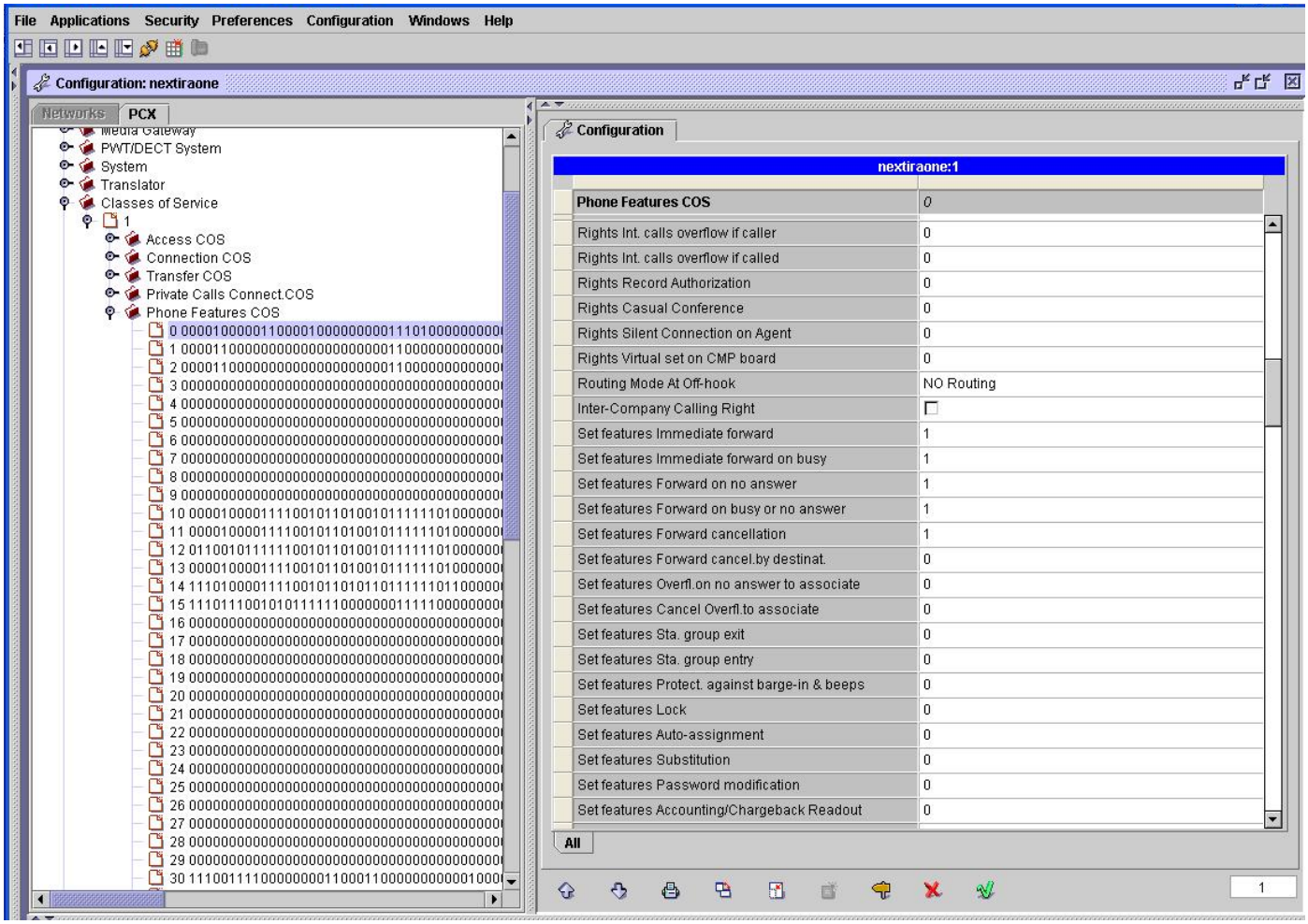


Figure 24. Digital station facilities configuration – 4 of 9.

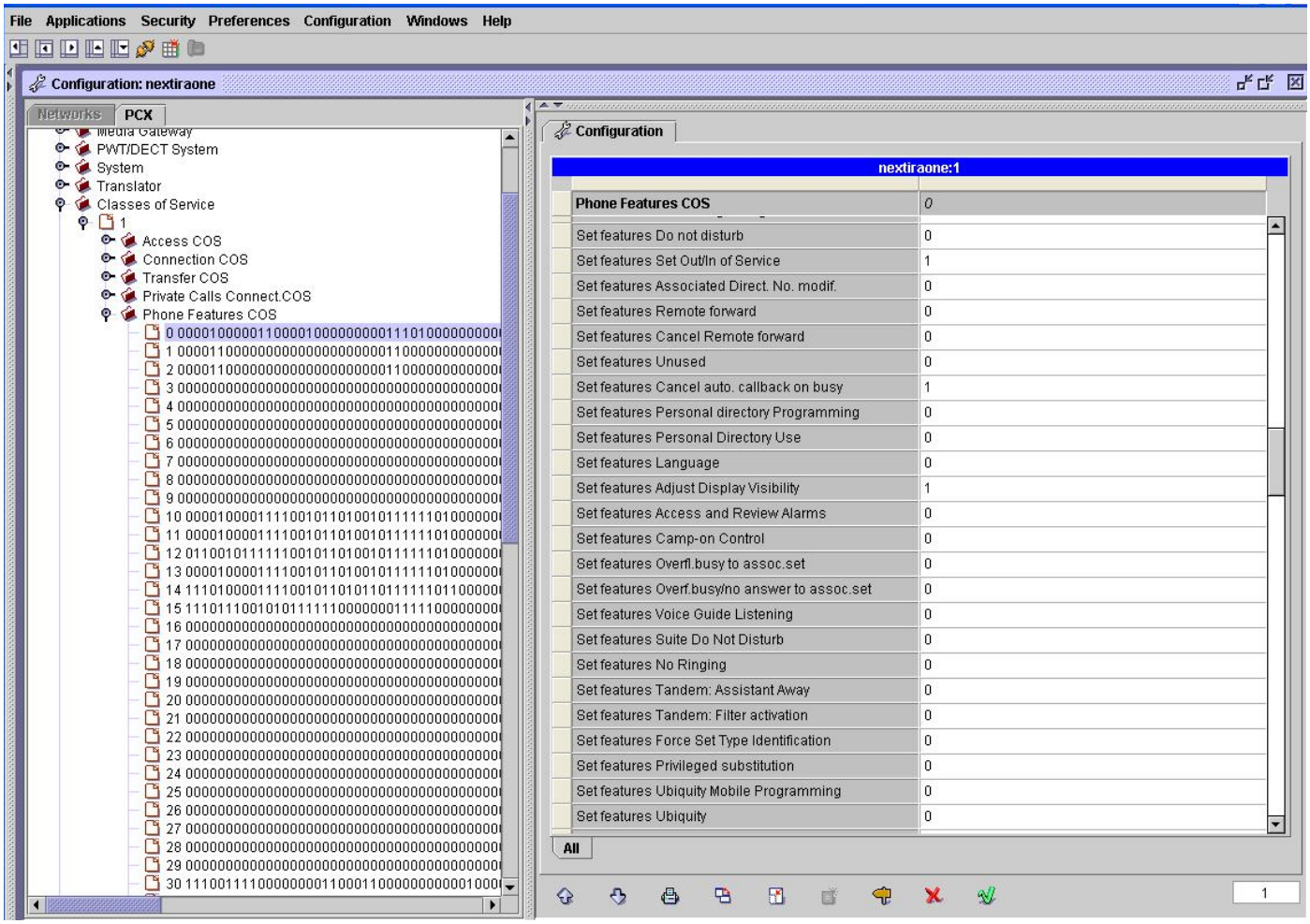


Figure 25. Digital station facilities configuration – 5 of 9.

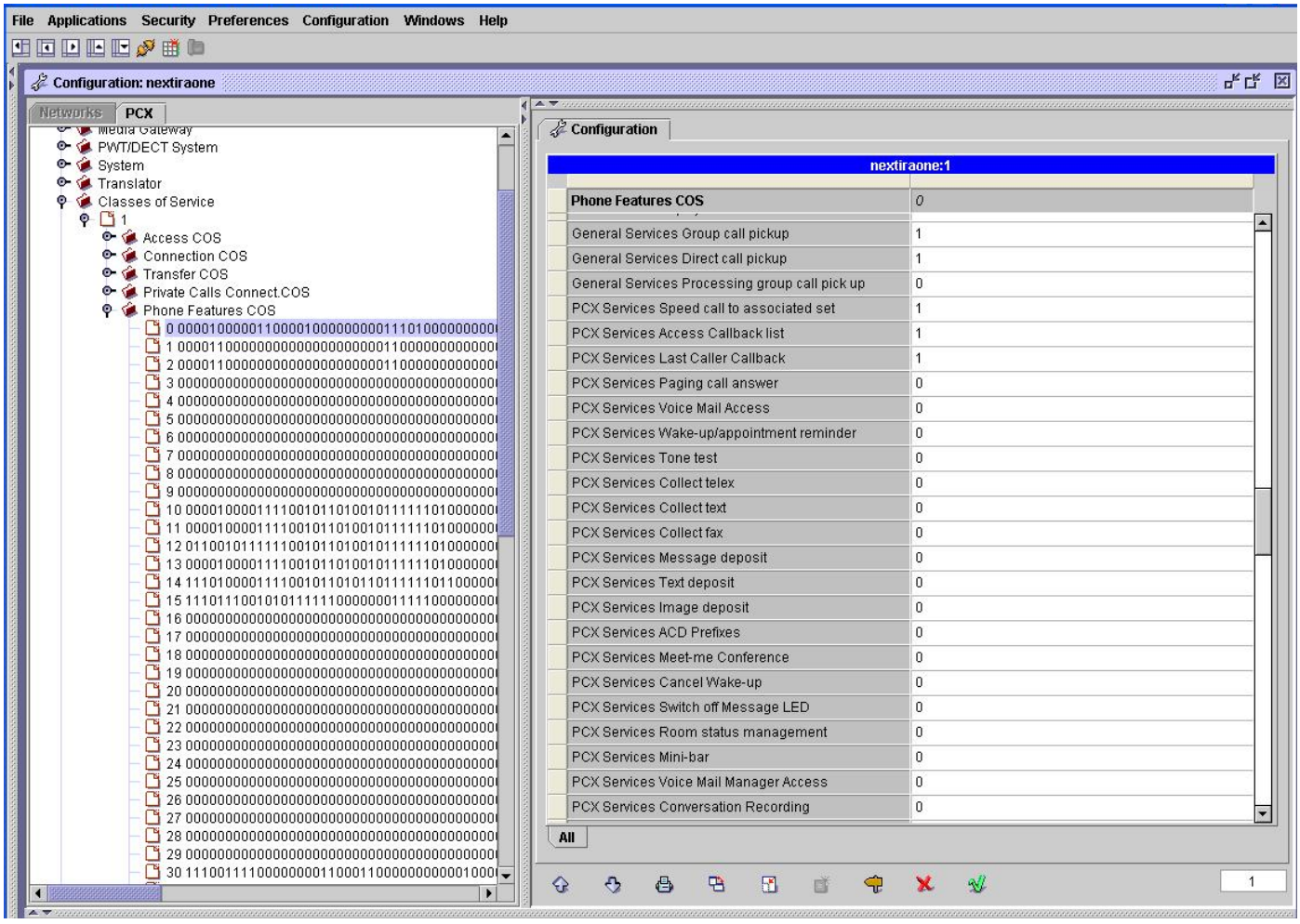


Figure 26. Digital station facilities configuration – 6 of 9.

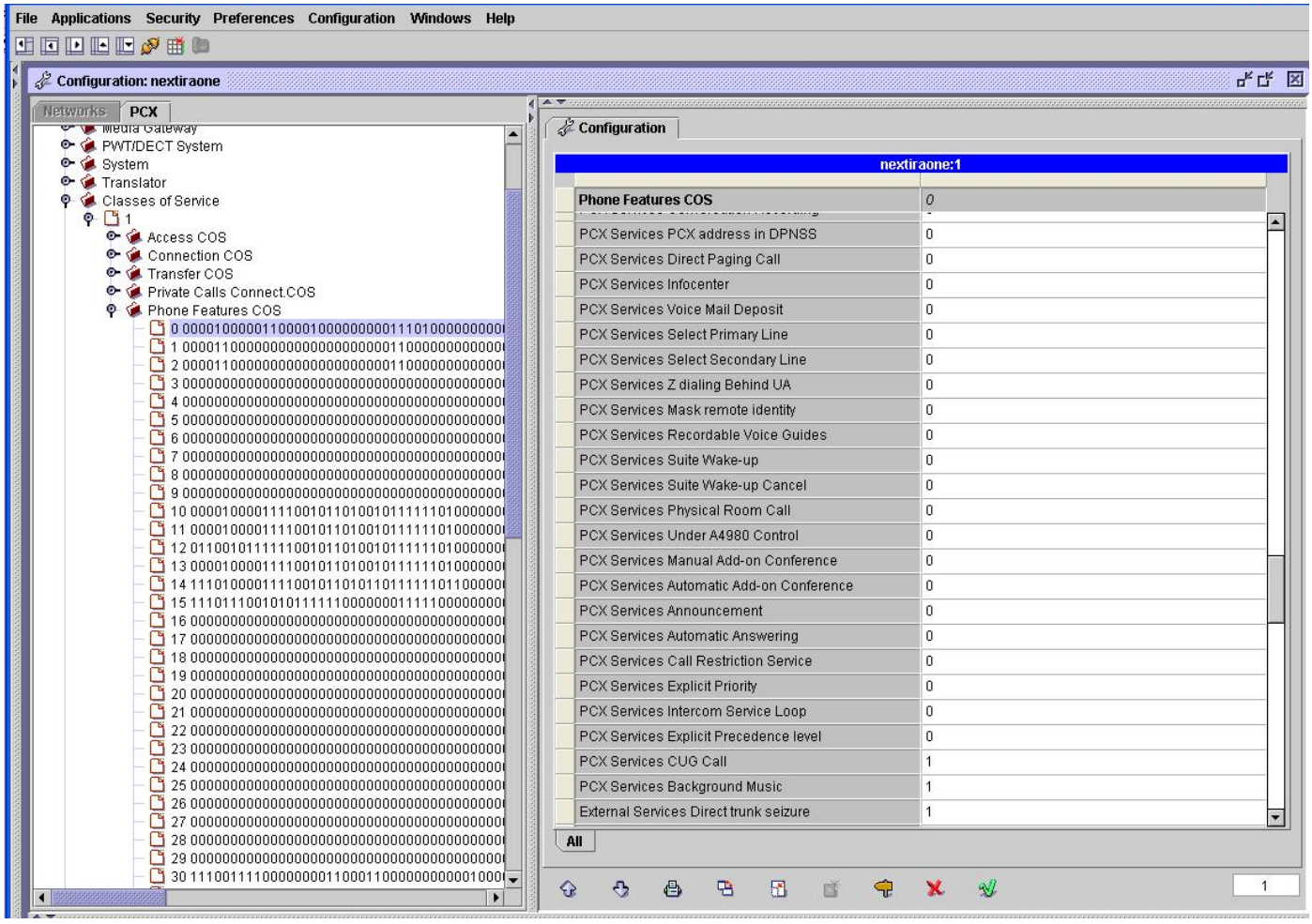


Figure 27. Digital station facilities configuration – 7 of 9.

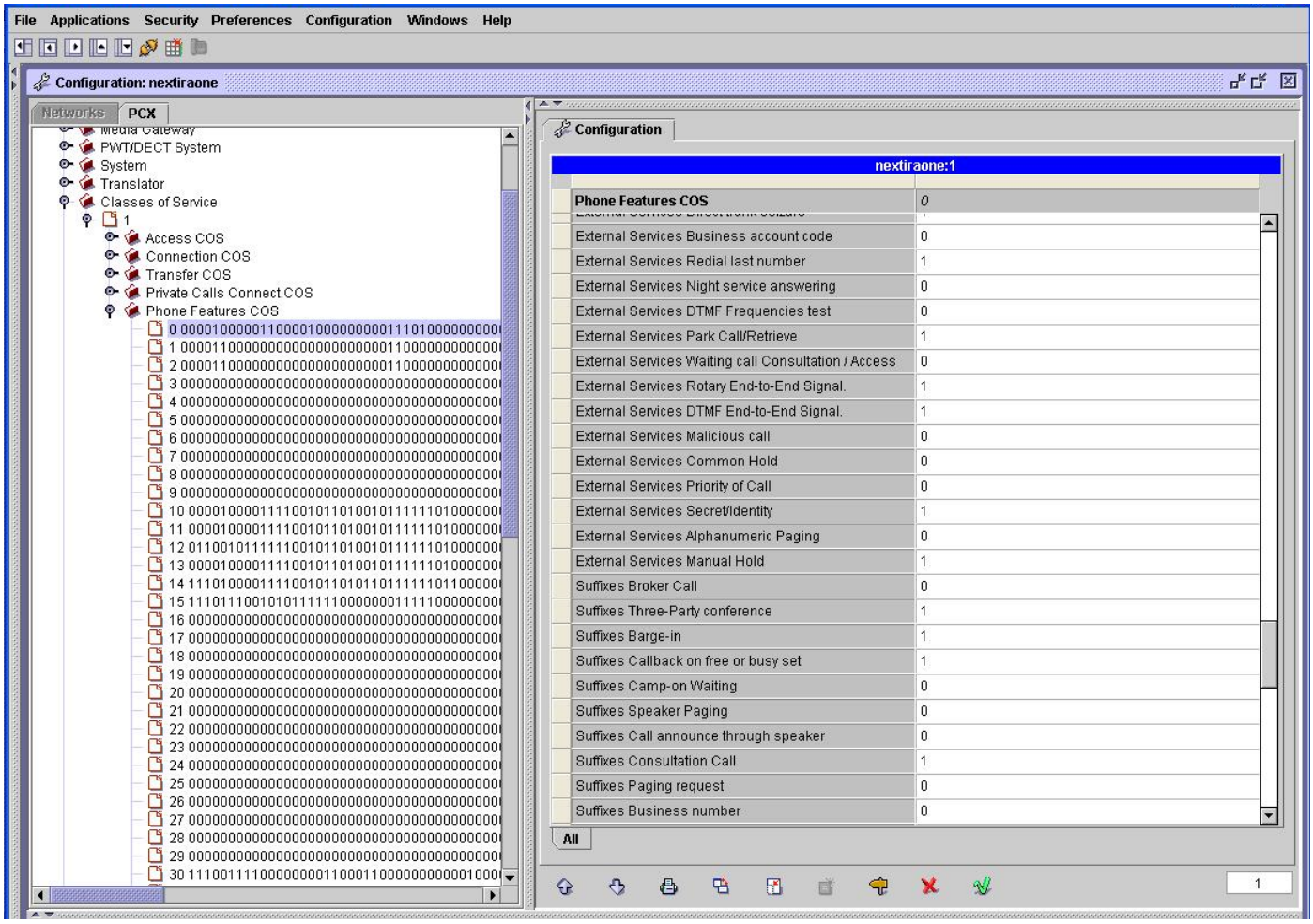


Figure 28. Digital station facilities configuration – 8 of 9.

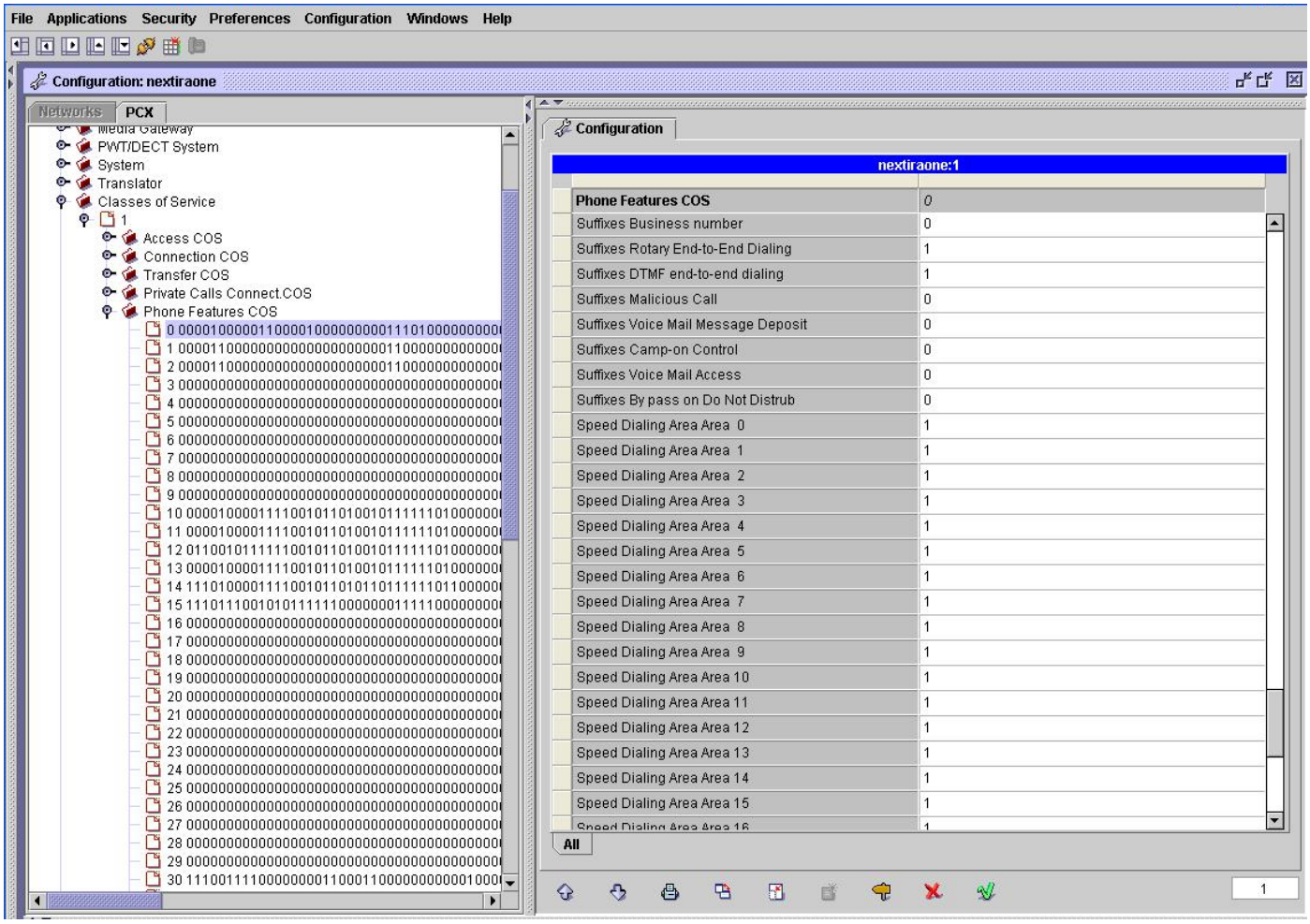
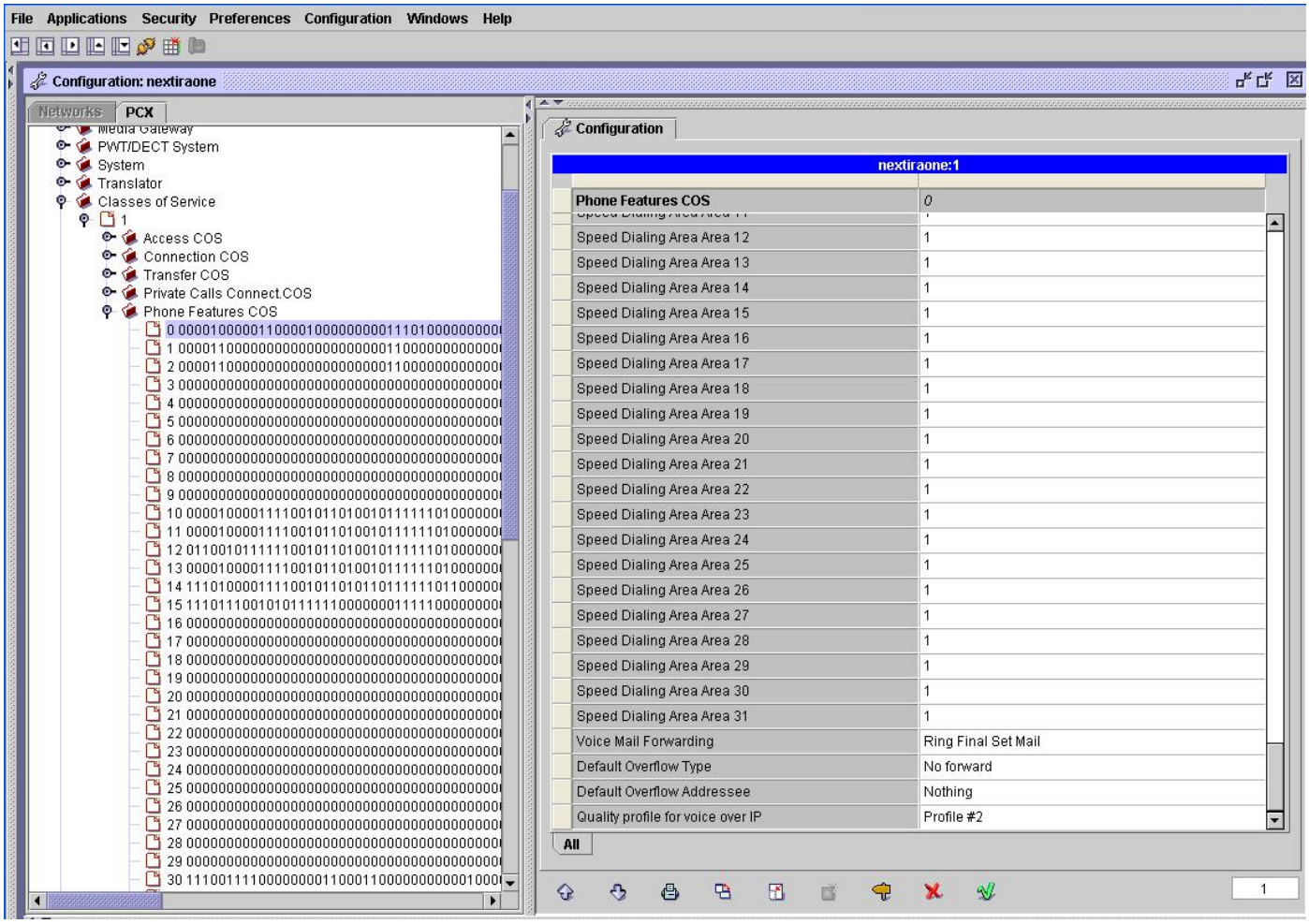


Figure 29. Digital station facilities configuration – 9 of 9.



Configuring the Local Cisco Unified Communications Manager Express 1 (Cisco 3745)

LOCAL-3745#sho ver

Cisco IOS Software, 3700 Software (C3745-IPVOICE-M), Version 12.4(11)T, RELEASE)

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

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Compiled Sat 18-Nov-06 22:37 by prod_rel_team

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

LOCAL-3745 uptime is 10 weeks, 4 days, 5 hours, 53 minutes

System returned to ROM by reload

System image file is "flash:c3745-ipvoice-mz.124-11.T.bin"

Cisco 3745 (R7000) processor (revision 2.0) with 243712K/18432K bytes of memory.

Processor board ID JMX0813L0Z3

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

2 FastEthernet interfaces

31 Serial interfaces

2 Channelized E1/PRI ports

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#

LOCAL-3745#

LOCAL-3745#wr t

Building configuration...

Current configuration : 4768 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.124-12.4.PI6a

boot-end-marker

!

logging buffered 10000000

enable password cisco

!

no aaa new-model

network-clock-participate slot 3

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
multilink bundle-name authenticated
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
< no supplementary-service h450.2> [2]
 <no supplementary-service h450.3 > [2]
 h323
 sip
```

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

```
 registrar server expires max 600 min 60
!
!
!
voice register global
 mode cme
 source-address 172.20.15.196 port 5060
 max-dn 100
 max-pool 192
 load 7960-7940 P0S3-07-5-00
 tftp-path flash:
 create profile sync 1037810039062866
!
voice register dn  1
 number 6001
 name Local IP1
< call-forward b2bua busy 6005>[3]
<call-forward b2bua noan 6005 timeout 7>[4]
 huntstop
!
voice register dn  2
 number 6002
 name Local IP2
 huntstop
!
voice register pool  1
 id mac 0015.FA0C.B1DC
```

[3] Inserted for call forward busy from SIP extension.
[4] Inserted for call forward no reply from SIP extension.

```
 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 E1 3/0
 pri-group timeslots 1-31
!
controller E1 3/1
!
!
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 Serial3/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-qsig
 isdn overlap-receiving
 isdn incoming-voice voice
 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:apps41.1-1-3-30.sbn
tftp-server flash:apps70.1-1-3-30.sbn
tftp-server flash:cnu41.3-1-3-30.sbn
```

tftp-server flash:cnu70.3-1-3-30.sbn

tftp-server flash:cvm41sccp.8-0-3-32.sbn

tftp-server flash:cvm70sccp.8-0-3-32.sbn

tftp-server flash:dsp41.1-1-3-30.sbn

tftp-server flash:dsp70.1-1-3-30.sbn

tftp-server flash:jar41sccp.8-0-3-32.sbn

tftp-server flash:jar70sccp.8-0-3-32.sbn

tftp-server flash:SCCP41.8-0-4SR2S.loads

tftp-server flash:SCCP70.8-0-4SR2S.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 3/0:15

!

!

dial-peer voice 3023 pots

 destination-pattern 3...

 incoming called-number ....

<clid restrict> [5]

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

 direct-inward-dial

 port 3/0:15

 forward-digits all

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

```
!
dial-peer voice 1 voip
 preference 1
 destination-pattern 6...
 session target ipv4:172.20.15.159
 dtmf-relay h245-alphanumeric
 codec g711ulaw
 no vad
!
dial-peer voice 4 pots
 destination-pattern 4...
 direct-inward-dial
 port 3/0:15
 forward-digits all
!
!
sip-ua
 retry options 0
!
!
telephony-service
 load 7960-7940 P003-07-5-00
 load 7961 SCCP41.8-0-4SR2S
 load 7970 SCCP70.8-0-4SR2S
 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 Jan 01 2002 00:00:00

!

!

ephone-dn 3 dual-line

number 6003

name Local IP3

< call-forward busy 6005> [8]

<call-forward noan 6005 timeout 7>[9]

huntstop channel

!

!

ephone-dn 4 dual-line

number 6004

name Local IP4

huntstop channel

!

!

ephone 3

mac-address 0017.0EC8.58D4

type 7961

keep-conference

button 1:3

!

!

!

---

[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  4
 mac-address 0015.F9C8.561A
 type 7970
 keep-conference
 button  1:4
!
!
!
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 Communications Manager Express (Cisco 2811)

REMOTE-2811#sho ver

Cisco IOS Software, 2800 Software (C2800NM-IPVOICE-M), Version 12.4(11)T, RELEA)

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

Copyright (c) 1986-2006 by Cisco Systems, Inc.

Compiled Sat 18-Nov-06 17:16 by prod_rel_team

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

REMOTE-2811 uptime is 13 weeks, 1 hour, 13 minutes

System returned to ROM by reload at 23:23:17 UTC Fri Jan 5 2007

System image file is "flash:c2800nm-ipvoice-mz.124-11.T.bin"

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

Processor board ID FHK0946F0MZ

2 FastEthernet interfaces

24 Serial interfaces

1 Channelized T1/PRI port

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#

Building configuration...

Current configuration : 3046 bytes

!

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.124-11.T.bin

boot-end-marker

!

logging buffered 10000000

no logging console

enable password cisco

!

no aaa new-model

network-clock-participate wic 0

!

!

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
multilink bundle-name authenticated
!
isdn switch-type primary-qsig
!
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
!
!
controller T1 0/0/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
!
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
!
interface Serial0/0/0:23
 no ip address
 encapsulation hdlc
 isdn switch-type primary-qsig
```

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

```
 isdn timer T310 120000
 isdn overlap-receiving
 isdn protocol-emulate network
 isdn incoming-voice voice
 no cdp enable
!
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>

!
control-plane
!
!
voice-port 0/0/0:23
!
voice-port 0/1/0
!
voice-port 0/1/1
!
```

```
!
!
dial-peer voice 2 voip
 destination-pattern 3...
 session target ipv4:172.20.15.196
 dtmf-relay h245-alphanumeric
 codec g711ulaw
!
dial-peer voice 6 voip
 destination-pattern 6...
 session target ipv4:172.20.15.196
 dtmf-relay h245-alphanumeric
 codec g711ulaw
!
dial-peer voice 4 voip
 destination-pattern 4...
 session target ipv4:172.20.15.196
 dtmf-relay h245-alphanumeric
 codec g711ulaw
!
!
sip-ua
 retry options 0
!
!
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 6005
 name Remote IP5
<call-forward busy 3004>[12]
< call-forward noan 3004 timeout 7>[13]
!
!
ephone-dn  6  dual-line
 number 6006
 name Remote IP6
!
!
ephone  5
 mac-address 0015.FA0C.B746
 type 7960
 keep-conference
 button  1:5
!
!
```

[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  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
CFB	Call Forward when Busy
CFNR	Call Forward when No Reply
CFU	Call Forward Unconditional
CO	Central Office
FXO	Foreign Exchange – Office
FXS	Foreign Exchange – Station
IOS	Internetworking Operating System
MWI	Message Waiting Indication
PBX	Private Branch Exchange
PRI	Primary Rate ISDN

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Corporate Headquarters

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters

Cisco Systems International BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

Americas Headquarters

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

Asia Pacific Headquarters

Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
www.cisco.com
Tel: +65 317 7777
Fax: +65 317 7799

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