Application Note

Avaya S8500 Communications Manager 4.0 to Cisco IOS Voice Gateway using H.323 with E1 NET5

January 28, 2008

Table of Contents

Introduction

- This Application note provides basic call interoperability and documented steps and configurations necessary for H.323 integration between Avaya S8500 Communications Manager (CM) 4.0 to Cisco IOS Voice Gateway providing E1 NET5 PSTN connectivity.
- The H323 protocol is used between Cisco IOS Voice gateway and Avaya S8500 CM 4.0. The connection between Cisco IOS Voice Gateway and PSTN uses E1-PRI with switch-type NET5 protocol.
- Features tested include Basic call, Call Transfer supervised, Call Transfer blind, Call Forward (All, Busy and No Answer), Three-way Conference, DTMF tones, Caller ID functionality between Avaya S8500 CM 4.0 users and the PSTN.
- The Cisco IOS Voice Gateway offers the advantage of providing connectivity between Avaya S8500 CM 4.0 and PSTN by offering H323 to ISDN inter-working functionality.
- The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability with the Cisco IOS Voice Gateway connected to the Avaya S8500 CM 4.0 and connected to the PSTN via E1 NET5 ISDN.
- This Application Notes uses the Cisco IAD2432 IOS-voice-gateway, however other Cisco voice gateways are also an option to use since the voice gateway implementation does not depend on the platform. Below is a list of Cisco platforms capable of voice gateway functionality: Care must be taken when selecting a voice gateway platform depending on the capacity and capability required for the intended deployment.

Cisco 1861 Integrated Services Router
Cisco IAD2430 Integrated Access Device
Cisco 2800 Series Integrated Services Routers
Cisco 3700 Series Routers
Cisco 3800 Series Integrated Services Routers
Cisco AS5350XM Universal Gateway
Cisco AS5400XM Universal Gateway

Network Topology

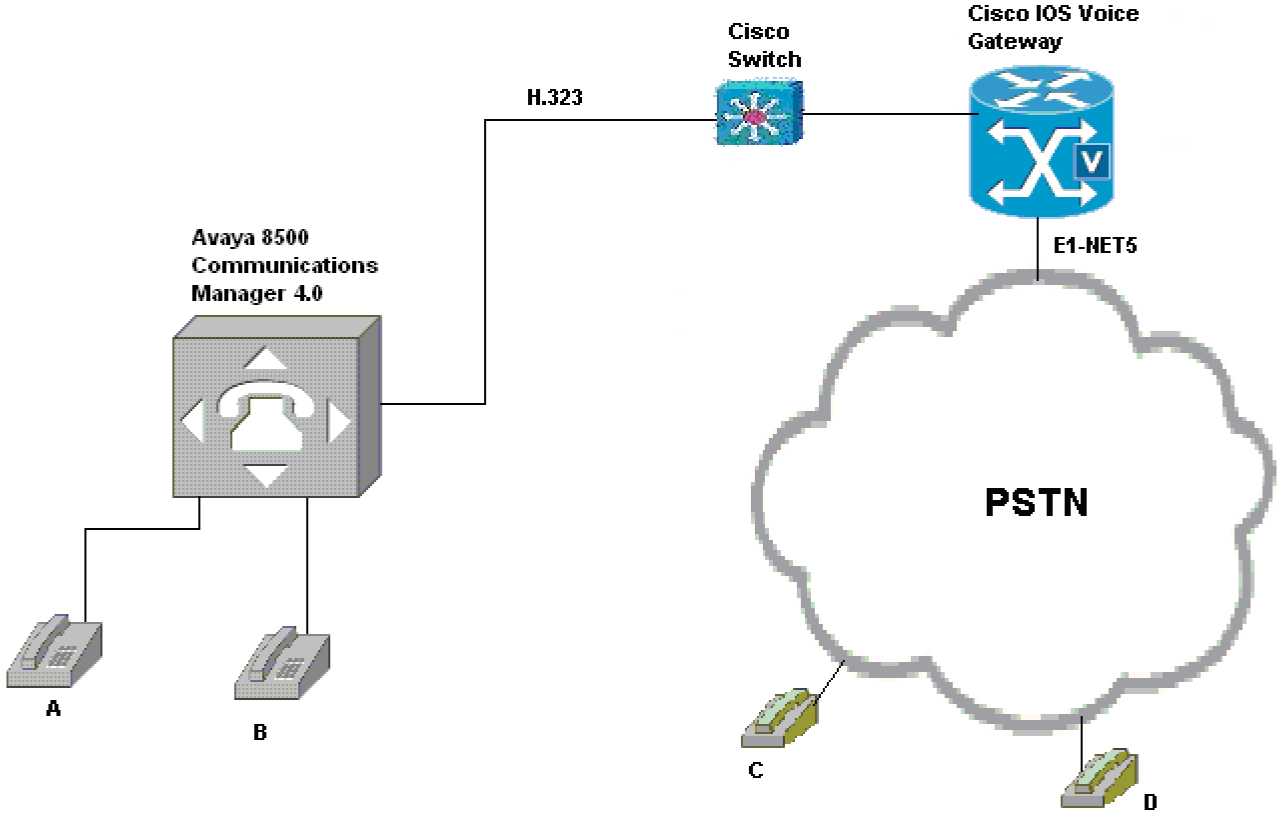


Figure 1. **Network Topology**

Limitations

- Basic call using G.726A – 32K and G.722-64K failed. Avaya rejects G.726 and G.722 codec, even when the Avaya is set for G.726 and G.722.

System Components

Hardware requirements

Cisco equipment

- Cisco 2432 (Cisco IOS Voice Gateway)
- Cisco 3560 powered Ethernet switch

Avaya equipment

- Avaya S8500
- TN2312BP IPSI
- TN799DP C-LAN
- TN2302AP IP Media Processor
- TN746B Analog
- TN2224B 2-wire Digital
- 2 - Digital stations 8410D

Software Requirements

- IOS Gateway: Cisco IOS Release – Cisco IAD2432: c2430-ik9o3s-mz.124-15.T3.bin
- Avaya S8500: Communications Manager 4.0

Features

Features Supported

- Basic Calls with G.711alaw, G.711ulaw, G.723 (5.3K and 6.3K), G.729(A and B)
- Calling name and Calling number
- Call Transfer Blind and Call Transfer Supervised
- Call Conference
- Caller ID restriction
- Call Forward All
- Call Forward No Reply
- Call Forward Busy
- DTMF using RFC2833
- DTMF in-band
- DTMF using H245 Alphanumeric or Signal
- Digit Translation

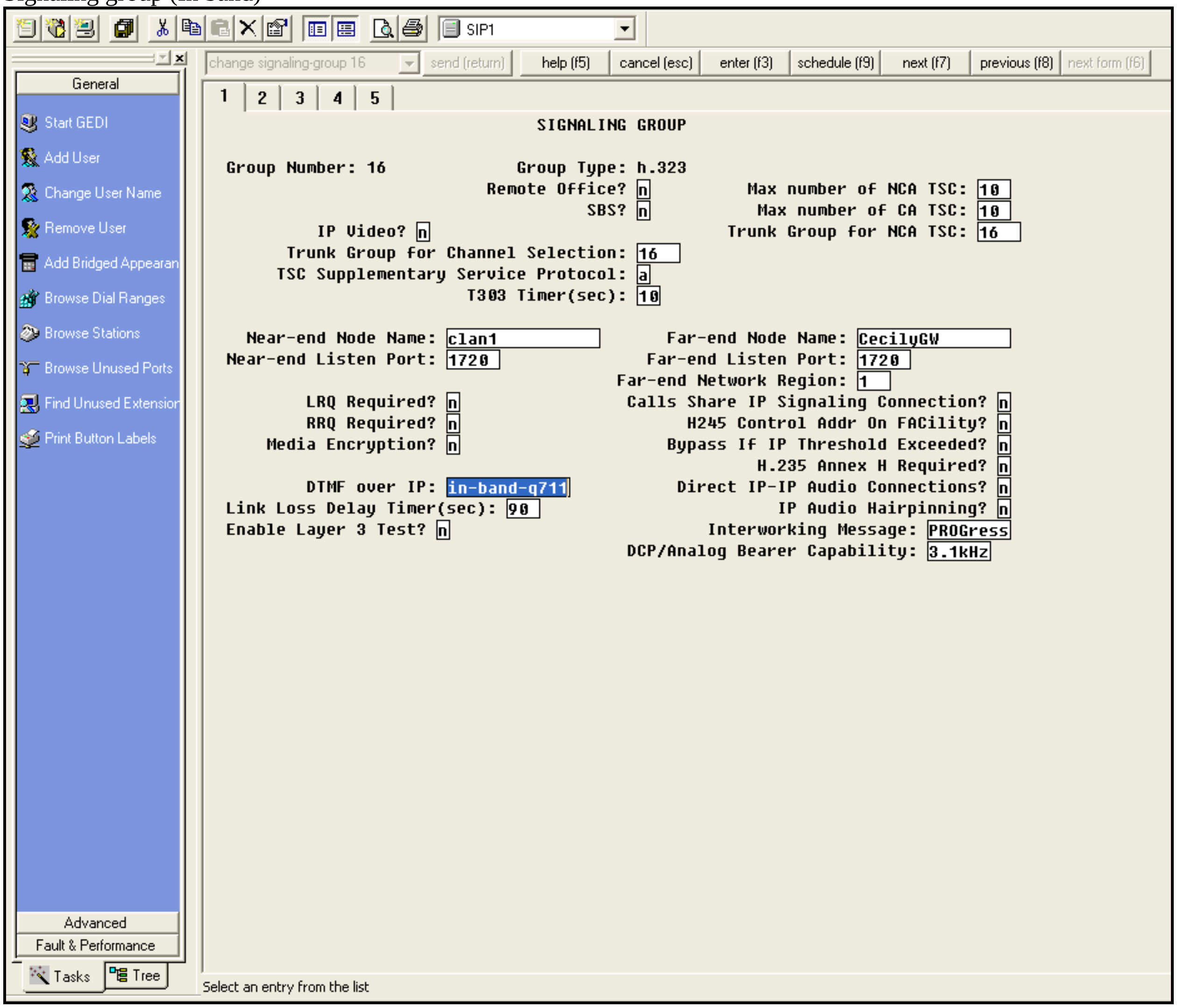
Features Not Supported

- G726 codec
- G.722 codec

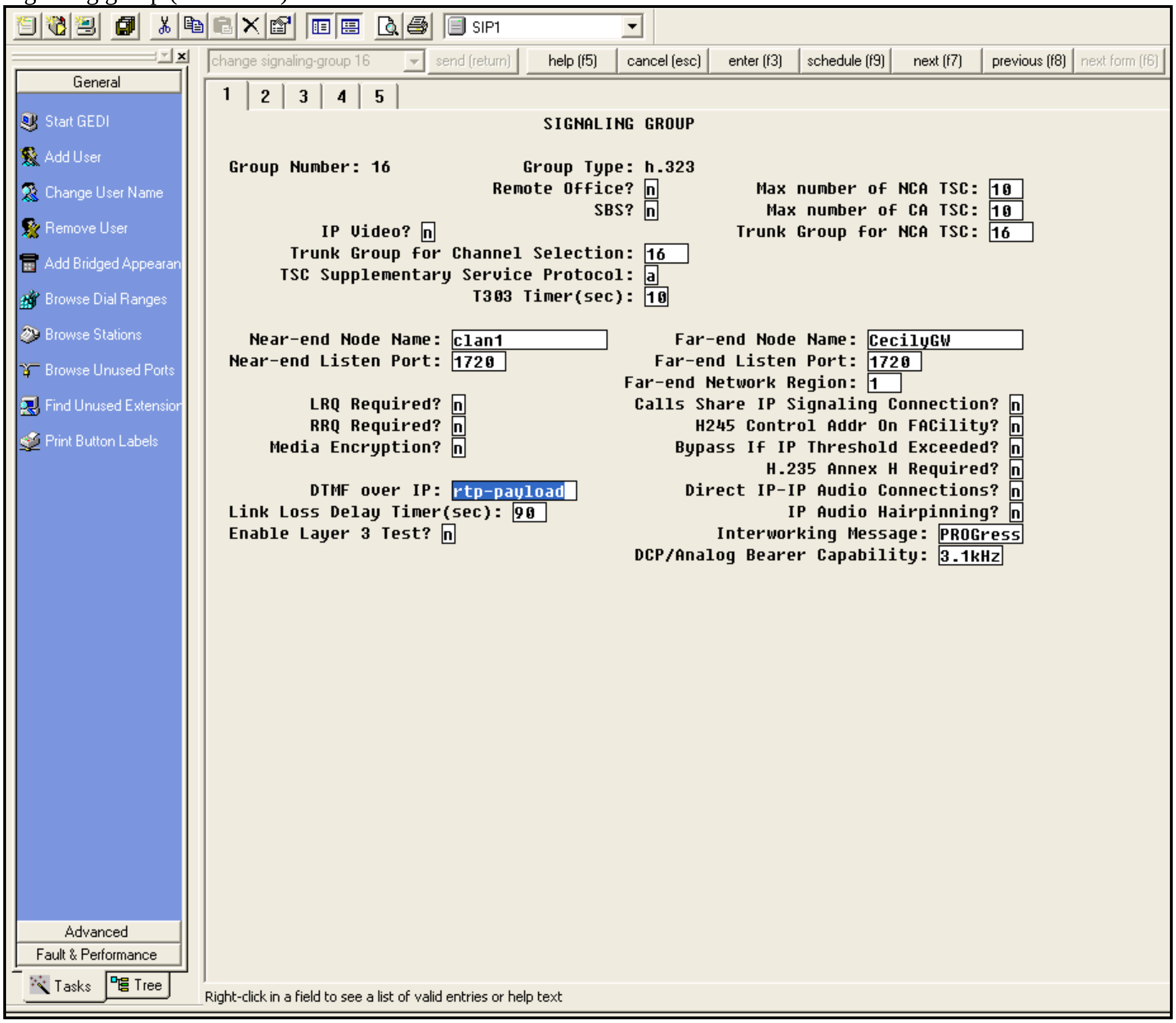
Configuration

Configuring Avaya S8500 CM 4.0

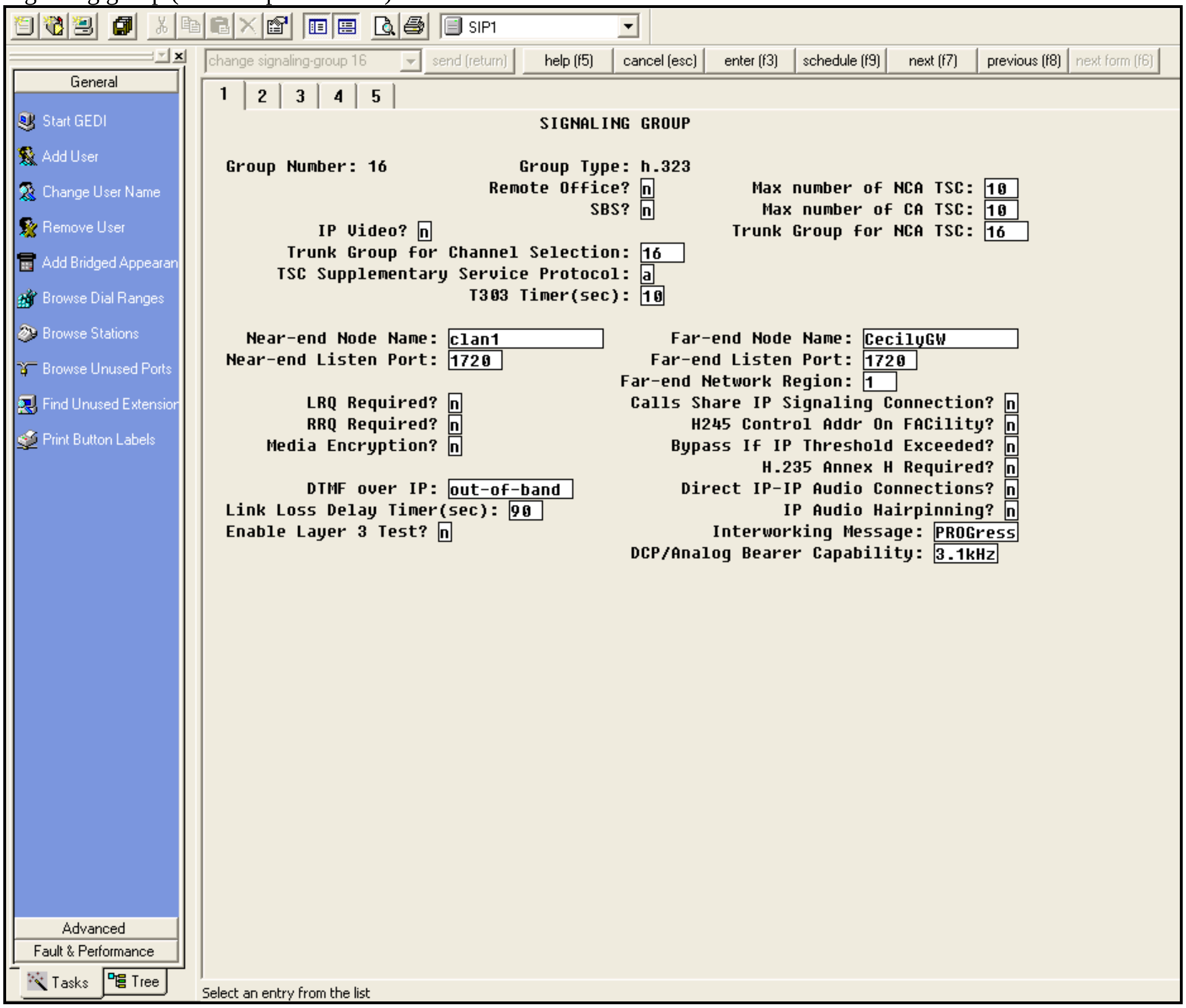
Signaling group (In band)



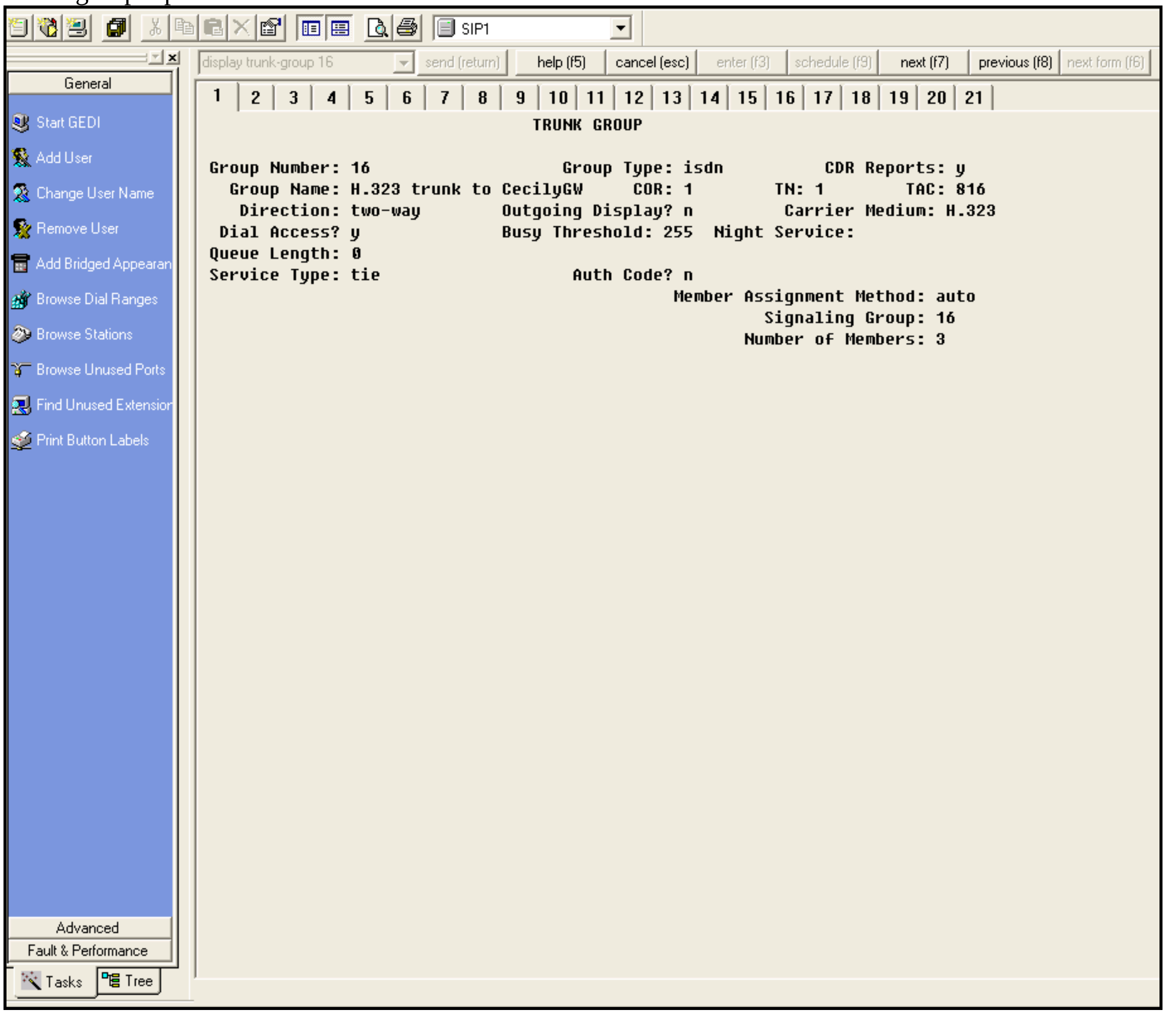
Signaling group (RFC 2833)



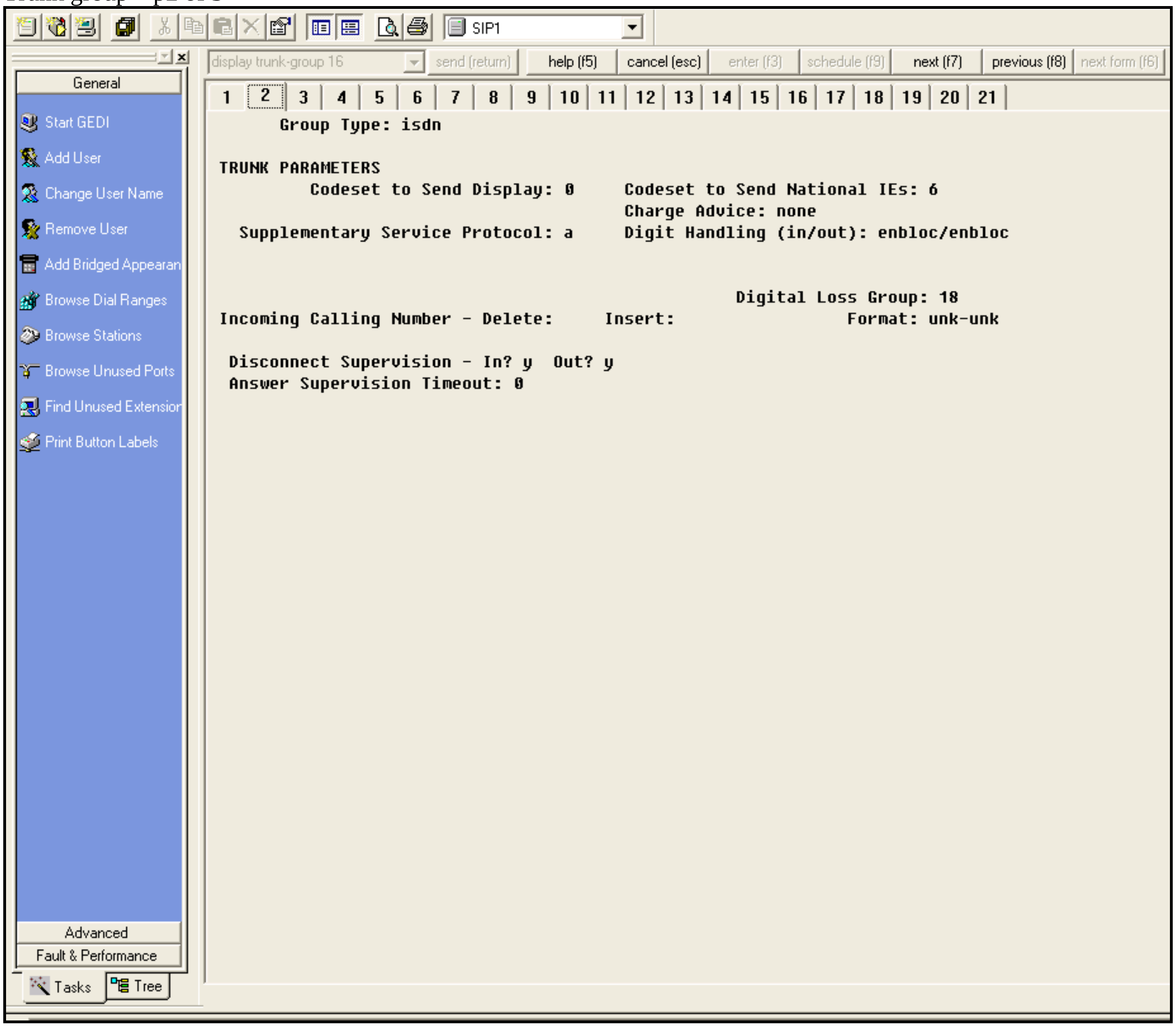
Signaling group (H245 Alphanumeric)



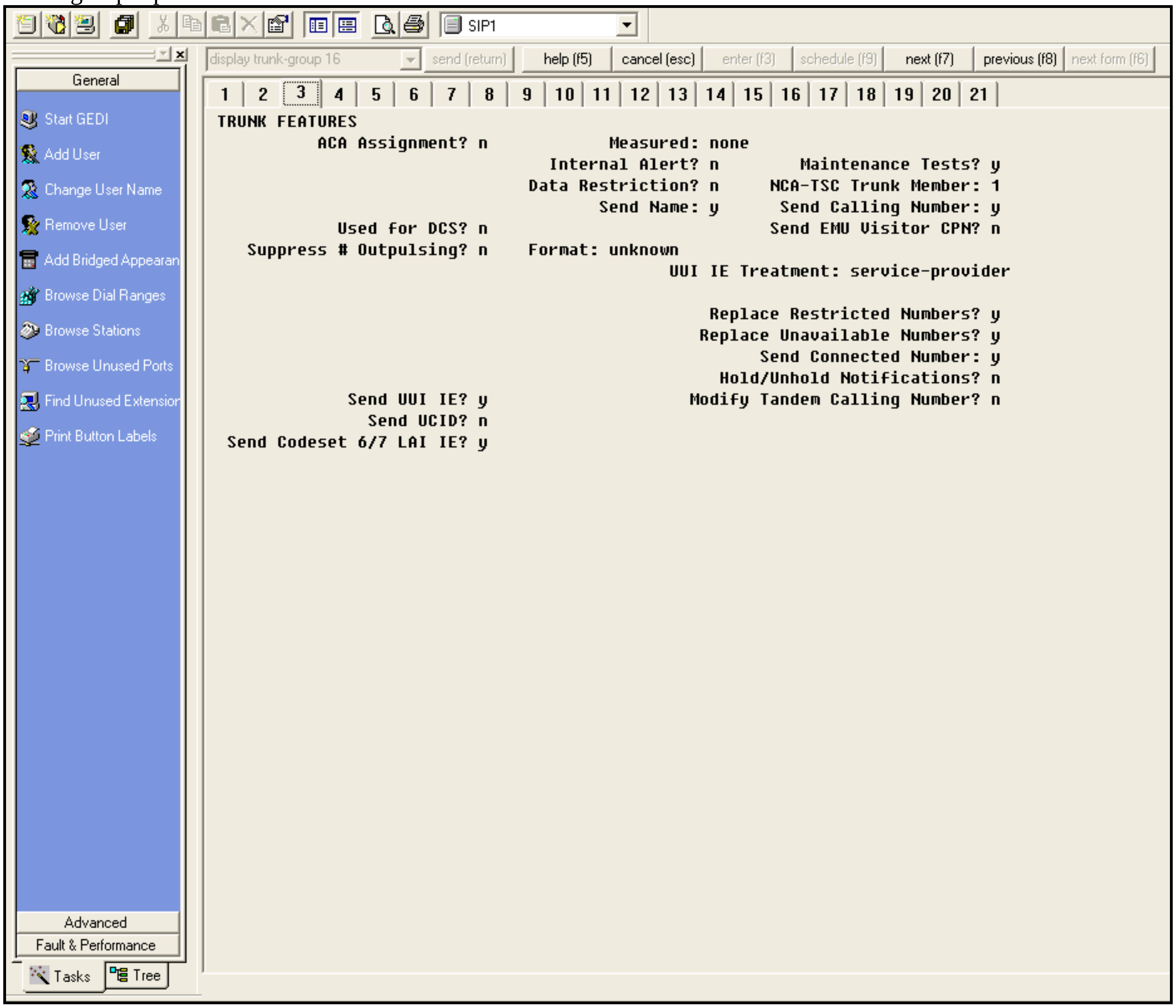
Trunk group – p1 of 3



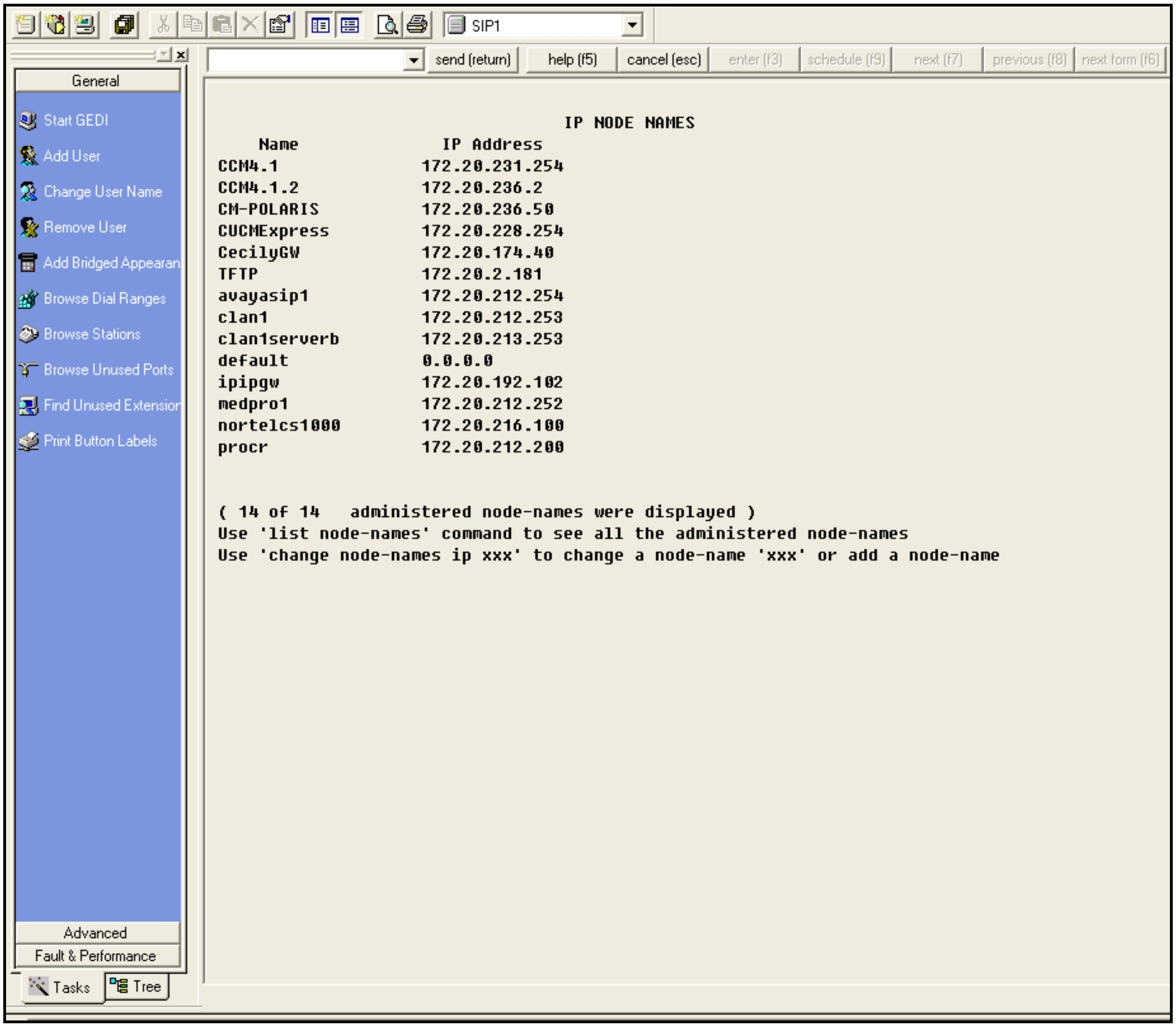
Trunk group – p2 of 3



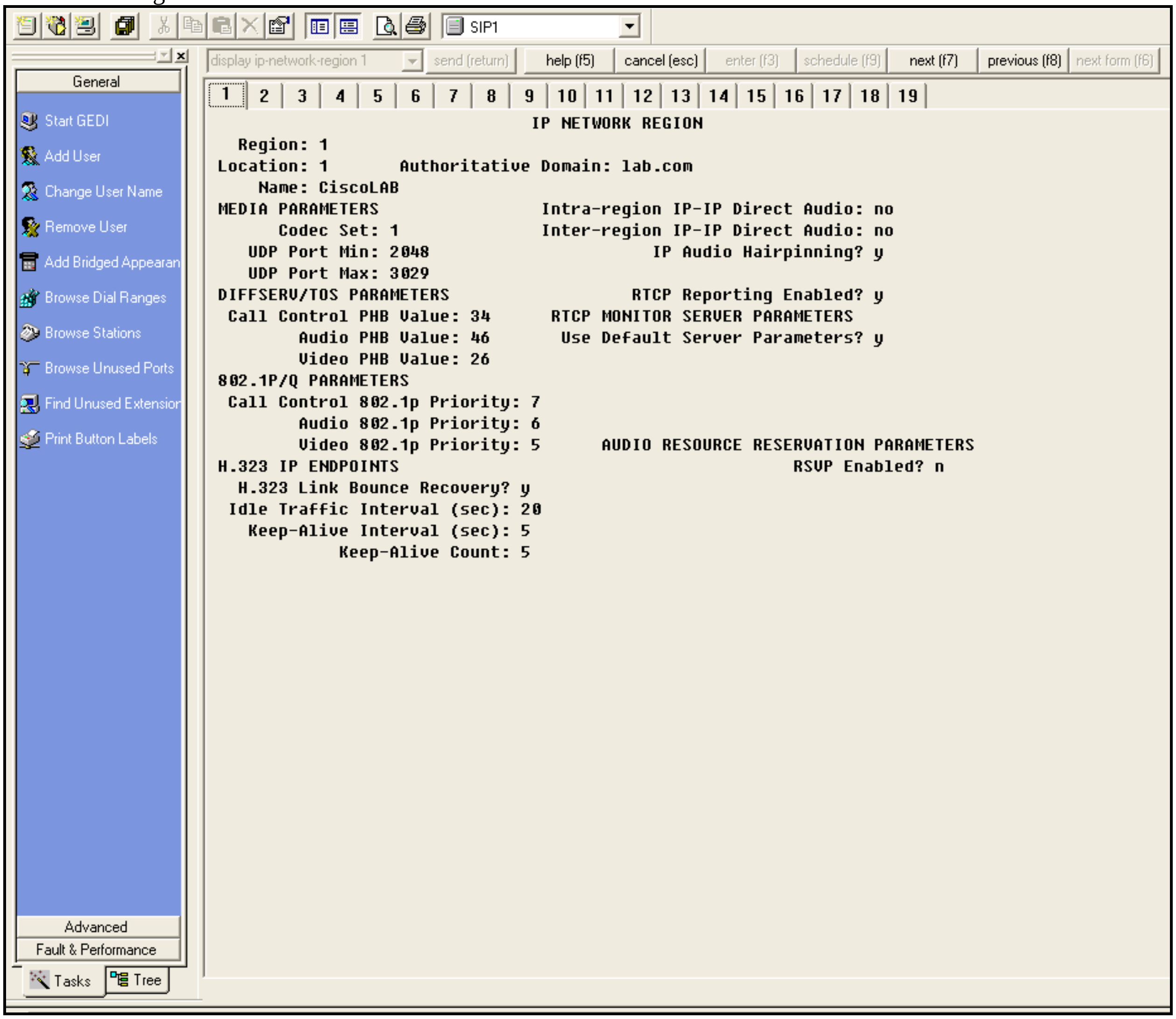
Trunk group – p3 of 3



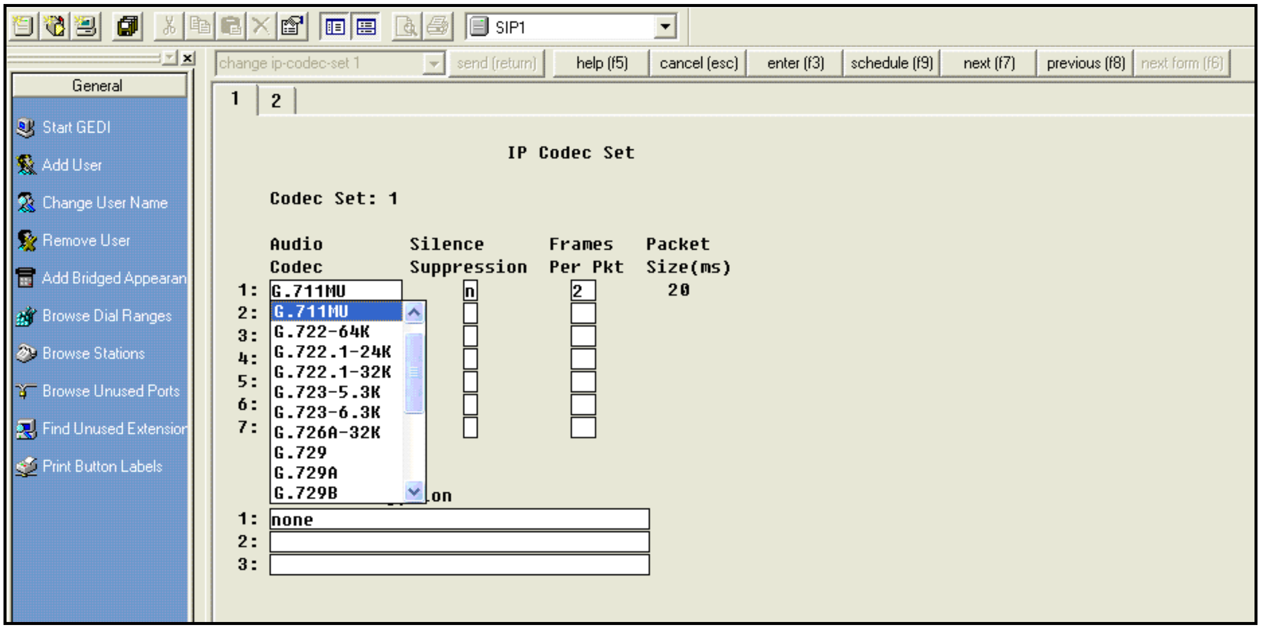
Node-names IP



IP Network Region

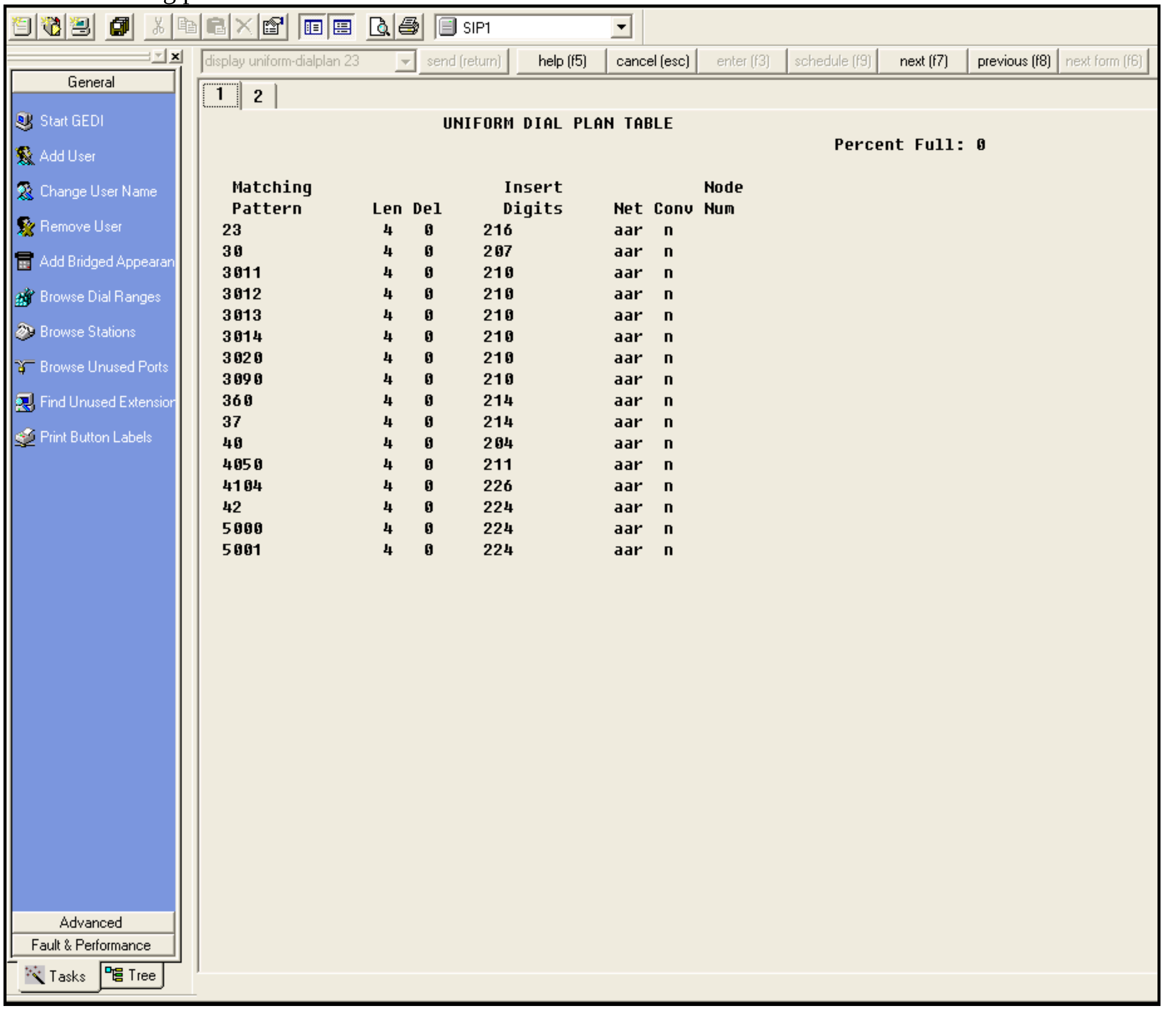


IP codec set



SIP1
change ip-codec-set 1
send (return)
help (f5)
cancel (esc)
enter (f3)
schedule (f9)
next (f7)
previous (f8)
next form (f6)
General
Start GEDI
Add User
Change User Name
Remove User
Add Bridged Appearan
Browse Dial Ranges
Browse Stations
Browse Unused Ports
Find Unused Extension
Print Button Labels
1
2
IP Codec Set
Codec Set: 1
Audio Codec
Silence Suppression
Frames Per Pkt
Packet Size(ms)
1: G.711MU n 2 20
G.711MU
G.722-64K
G.722.1-24K
G.722.1-32K
G.723-5.3K
G.723-6.3K
G.726A-32K
G.729
G.729A
G.729B
1: none

Uniform dialing plan

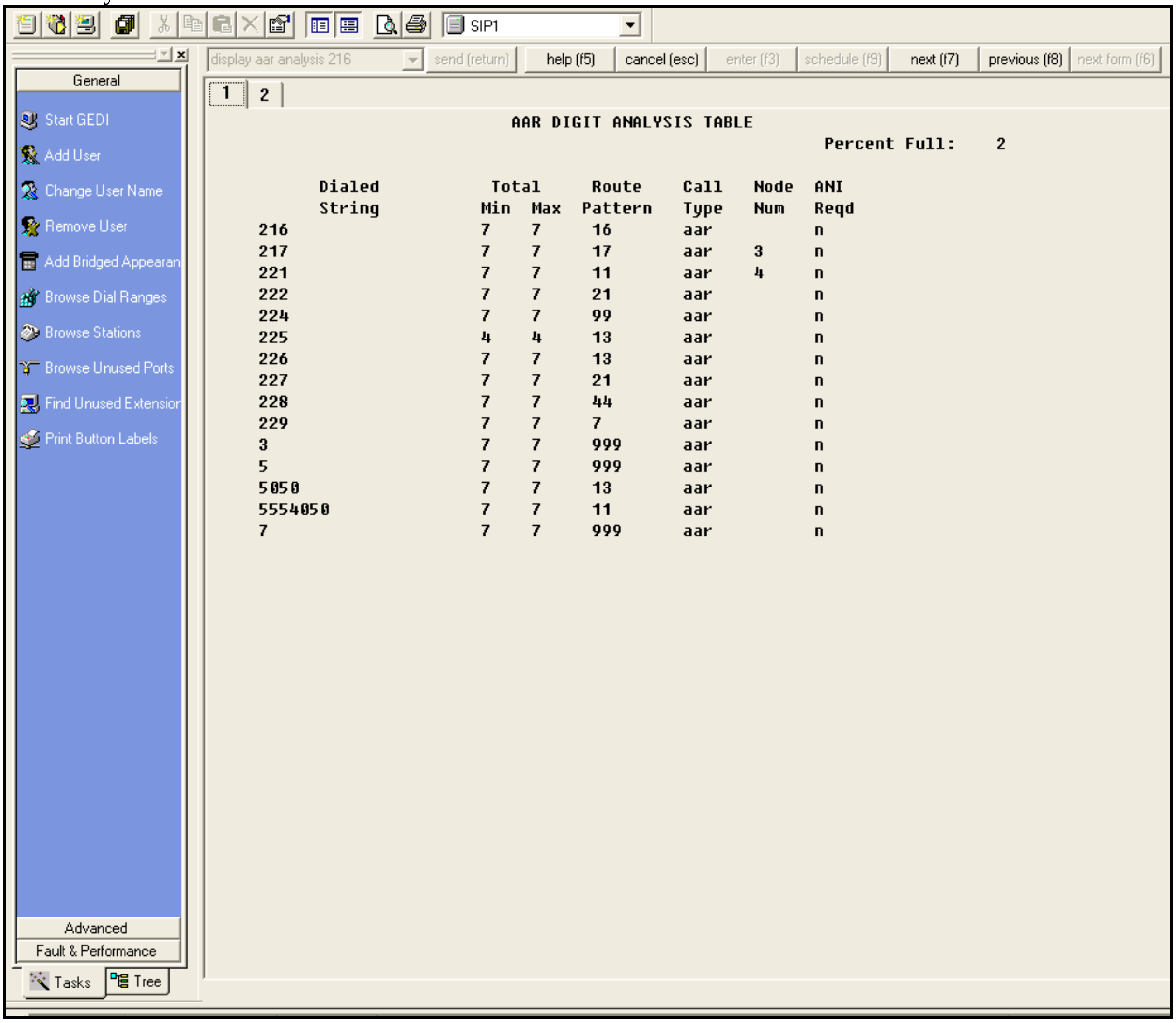


UNIFORM DIAL PLAN TABLE

Percent Full: 0

Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num
23	4	0	216	aar	n	
30	4	0	207	aar	n	
3011	4	0	210	aar	n	
3012	4	0	210	aar	n	
3013	4	0	210	aar	n	
3014	4	0	210	aar	n	
3020	4	0	210	aar	n	
3090	4	0	210	aar	n	
360	4	0	214	aar	n	
37	4	0	214	aar	n	
40	4	0	204	aar	n	
4050	4	0	211	aar	n	
4104	4	0	226	aar	n	
42	4	0	224	aar	n	
5000	4	0	224	aar	n	
5001	4	0	224	aar	n	

AAR analysis

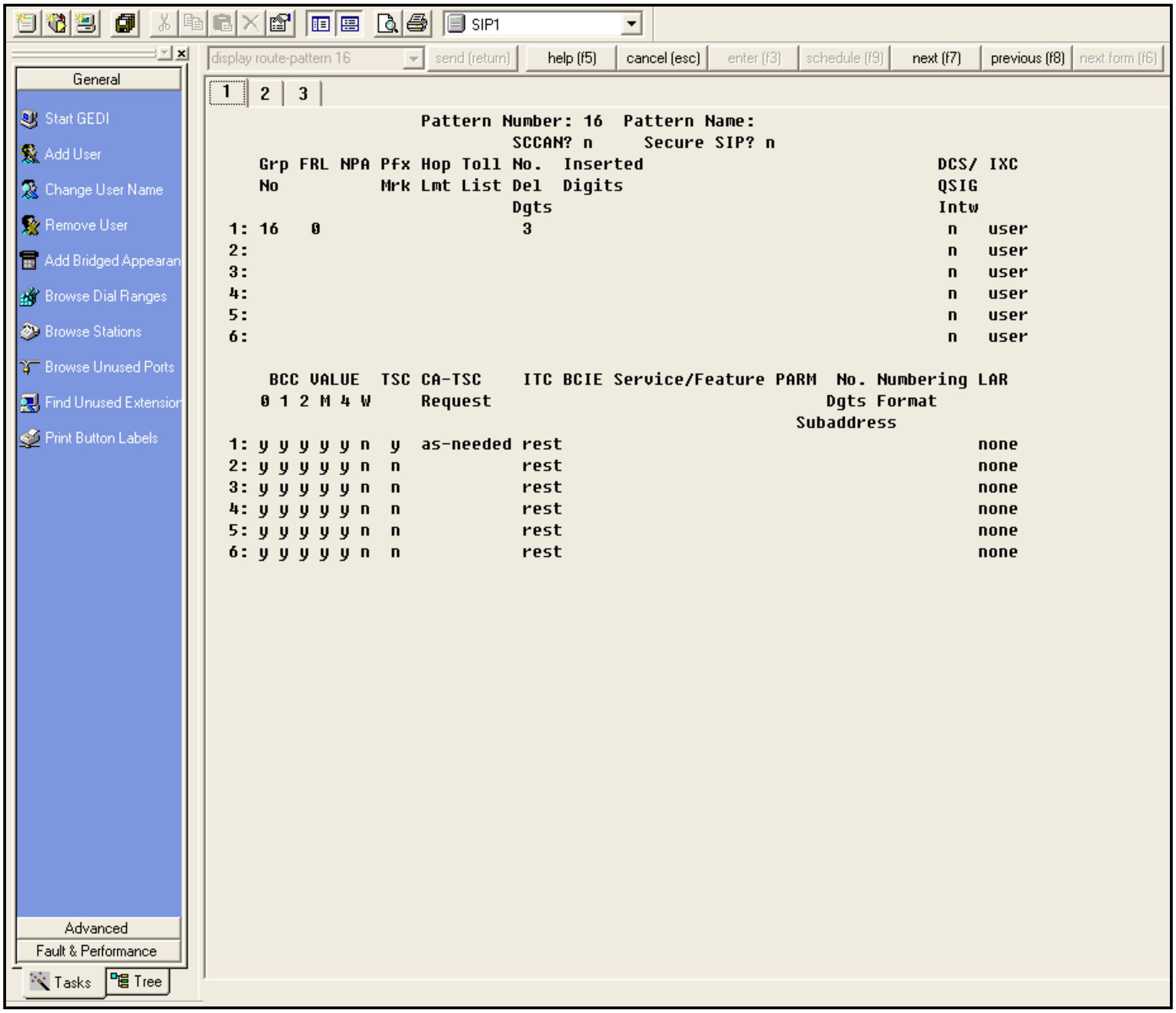


AAR DIGIT ANALYSIS TABLE

Percent Full: 2

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
216	7	7	16	aar		n
217	7	7	17	aar	3	n
221	7	7	11	aar	4	n
222	7	7	21	aar		n
224	7	7	99	aar		n
225	4	4	13	aar		n
226	7	7	13	aar		n
227	7	7	21	aar		n
228	7	7	44	aar		n
229	7	7	7	aar		n
3	7	7	999	aar		n
5	7	7	999	aar		n
5050	7	7	13	aar		n
5554050	7	7	11	aar		n
7	7	7	999	aar		n

Route Pattern



Cisco IOS configuration

Cisco 2432 voice gateway configuration

```
E1-NET5#sh ver
Cisco IOS Software, 2400 Software (C2430-IK9O3S-M), Version 12.4(15)T3, RELEASE
SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2008 by Cisco Systems, Inc.
Compiled Thu 24-Jan-08 14:04 by prod_rel_team

ROM: System Bootstrap, Version 12.3(7r)T2, RELEASE SOFTWARE (fc1)

E1-NET5 uptime is 1 day, 23 hours, 10 minutes
System returned to ROM by reload
System image file is "flash:c2430-ik9o3s-mz.124-15.T3.bin"


This product contains cryptographic features and is subject to United
States and local country laws governing import, export, transfer and
use. Delivery of Cisco cryptographic products does not imply
third-party authority to import, export, distribute or use encryption.
Importers, exporters, distributors and users are responsible for
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agree to comply with applicable laws and regulations. If you are unable
to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:
http://www.cisco.com/wwl/export/crypto/tool/stqrg.html

If you require further assistance please contact us by sending email to
export@cisco.com.

Cisco IAD2432 (R527x) processor (revision 4.1) with 119808K/11264K bytes of memo
ry.
Processor board ID FHK1013F23V
R527x CPU at 225MHz, Implementation 40, Rev 3.1
1 On-Board Twenty-Four FXS Analog Voice Module V1.3
2 FastEthernet interfaces
11 Serial interfaces
2 Channelized E1/PRI ports
1 Virtual Private Network (VPN) Module
DRAM configuration is 64 bits wide with parity disabled.
63K bytes of non-volatile configuration memory.
System fpga version is 250027
System readonly fpga version is 250025
Option for system fpga is 'system'.
62720K bytes of ATA System CompactFlash (Read/Write)
62720K bytes of ATA Slot0 CompactFlash (Read/Write)

Configuration register is 0x2102

==========================================

E1-NET5#sh run
Building configuration...

Current configuration : 2405 bytes
!
```

```
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname E1-NET5
!
boot-start-marker
boot system flash:c2430-ik9o3s-mz.124-15.T3.bin
boot-end-marker
!
card type e1 1[1]
logging buffered 1000000
!
no aaa new-model
network-clock-participate E1 1/0
network-clock-participate E1 1/1
!
!
no ip domain lookup
!
!
ip auth-proxy max-nodata-conns 3
ip admission max-nodata-conns 3
!
multilink bundle-name authenticated
isdn switch-type primary-net5
isdn gateway-max-interworking
!
voice-card 0
!
!
voice rtp send-recv
!
voice service voip
 h323
!
!
voice translation-rule 1[2]
 rule 1 /23/ /20\1/
!
!
voice translation-profile rule1[3]
 translate called 1
!
!
archive
 log config
  hidekeys
!
!
controller E1 1/0
 pri-group timeslots 1-10,16
!
```

[1] This sets or changes the card type (eg E1/T1)
[2] This individual rule translate the number that starts with “23” and replaces the first two digits with “20”, then copy the rest of the dialed digits.
[3] The voice translation rules are applied to voice translation profile, which can then applied to dial peers.

```
controller E1 1/1
!
!
interface FastEthernet0/0
 no ip address
 shutdown
 duplex half
 speed 100
!
interface FastEthernet0/1
 ip address 172.20.174.40 255.255.255.0
 duplex auto
 speed auto
!
interface Serial1/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5[4]
 isdn timer T310 10000
 isdn incoming-voice voice
 isdn supp-service name calling
 isdn send-alerting
 isdn bchan-number-order ascending
 no cdp enable
!
ip default-gateway 172.20.174.1
ip http server
no ip http secure-server
!
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 172.20.174.1
!
!
control-plane
!
!
!
voice-port 1/0:15
!
!
dial-peer voice 2000 pots
 destination-pattern 20..
 direct-inward-dial
 port 1/0:15
 forward-digits all
!
!
!
dial-peer voice 4100 voip
 translation-profile incoming rule1[5]
 destination-pattern 41..
 rtp payload-type nte 127
 session target ipv4:172.20.212.253
 incoming called-number 23..
```

[4] Specify E1-NET5 interface.
[5] This specifies a digit string that can be matched by an incoming call to associate the call with a dial peer.

```
 dtmf-relay rtp-nte[6]
 codec g729br8[7]
!
!
gateway
 timer receive-rtp 1200
!
sip-ua
 no remote-party-id
!
!
line con 0
 exec-timeout 0 0
 password cisco
 login
line aux 0
line vty 0 4
 exec-timeout 0 0
 password cisco
 login
!
end
```

[6] Specify DTMF parameter here. This command is for testing DTMF RFC2833; removing this command will do DTMF in-band. NOTE: Insert "dtmf-relay h245-alphnumeric h245-signal", and remove "dtmf-relay rtp-nte" to test DTMF for H245/Out-of-band signaling when using H323 trunk.

[7] Specify CODEC here when testing CODEC. Also change CODEC settings at the PBX end to match the specified codec at the IOS Media Gateway

Acronyms

Acronym	Definitions
Cisco IOS	Cisco Internetwork Operating System
PSTN	Public switched Telephone Network

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Test results

	Test Case	Description	Test Result
1	Phone A calls Phone C. Note: If IP PBX supports more than one codec (e.g. G729, G723..etc) test each supported codec and make separate results for each codec under "Test Results"	Phone A hears ring-back Ensure voice path, in both directions, is achieved when Phone C is answered Verify Calling Name and number are presented to called phone Verify call is disconnected at both ends when phone is hung-up	**G.711alaw: PASS** **G.711ulaw: PASS** **G.722-64K: FAIL** **G.723-5.3K: PASS** **G.723-6.3K: PASS** **G.726-32K: FAIL** **G.729: PASS** **G.729B: PASS**
2	Phone D calls Phone B. Note: If IP PBX supports more than one codec (e.g. G729, G723..etc) test each supported codec and make separate results for each codec under "Test Results"	Phone D hears ring-back Ensure voice path, in both directions, is achieved when Phone B is answered Verify Calling Name and number are presented to called phone Verify call is disconnected at both ends when phone is hung-up	**G.711alaw: PASS** **G.711ulaw: PASS** **G.722-64K: FAIL** **G.723-5.3K: PASS** **G.723-6.3K: PASS** **G.726-32K: FAIL** **G.729: PASS** **G.729B: PASS**
3	Phone A calls Phone C, Phone A hangs-up before Phone C answers	Ensure call is released and Phone C stops ringing	**PASS**
4	Phone C calls Phone A, Phone C hangs-up before Phone A answers	Ensure call is released and Phone A stops ringing	**PASS**
5	Call Transfer – Blind Phone D calls Phone B, Phone B transfers to Phone C before phone C answers the call.	Ensure call is transferred to Phone C and Calling number/name are updated correctly.	**PASS – Calling name/number are not updated correctly. Phone C display Phone B name and number.**
6	Call Transfer – Blind Phone D calls Phone B, Phone B transfers to Phone A before phone A answers the call.	Ensure call is transferred to Phone A and Calling number/name are updated correctly.	**PASS – Calling name/number are updated correctly.**
7	Call Transfer – Blind Phone A calls Phone B, Phone B transfers to Phone C before phone C answers the call.	Ensure call is transferred to Phone C and Calling number/name are updated correctly.	**PASS - Calling name/number are not updated correctly. Phone C display Phone B name and number.**
8	Call Transfer – Supervised Phone D calls Phone B, Phone B transfers to Phone C after phone C answers the call.	Ensure call is transferred to Phone C and Calling number/name are updated correctly.	**PASS - Calling name/number are not updated correctly. Phone C display Phone B name and number.**
9	Call Transfer – Supervised Phone D calls Phone B; Phone B transfers to Phone A after phone A answers the call.	Ensure call is transferred to Phone A and Calling number/name are updated correctly.	**PASS – Calling name/number are updated correctly.**
10	Call Transfer – Supervised Phone A calls Phone B, Phone B transfers to Phone C after	Ensure call is transferred to Phone C and Calling number/name are updated correctly.	**PASS – Calling name/name are not updated correctly. Phone C**

	Test Case	Description	Test Result
	phone C answers the call.		**display Phone B name and number.**
11	Call Conference – Using phone conference button. Phone C calls phone A, phone A calls phone D. When phone D answers, activate "conference" on phone A	Ensure all three phones (Phone A, C and D) are in a conference call. Voice path is established all three ways at each phone.	**PASS**
12	Call Conference – Phone D leaves call	Ensure Phones C and A remain on call	**PASS**
13	Call Conference – Phone C leaves call	Ensure Phone A and D remain on call	**PASS**
14	Call Conference – Phone A leaves call	Ensure Phone C and D remain on call	**PASS**
15	Call Conference – Using phone conference button Phone C calls phone A, phone A calls phone B. When phone B answers, activate "conference" on phone A	Ensure all three phones (Phone A, B and C) are in a conference call. Voice path is established all three ways at each phone.	**PASS**
16	Call Conference – Phone B leaves call	Ensure Phones C and A remain on call	**PASS**
17	Call Conference – Phone C leaves call	Ensure Phone A and B remain on call	**PASS**
18	Call Conference – Phone A leaves call	Ensure Phone B and C remain on call	**PASS**
19	Call on Hold Phone A calls Phone C, press the "Hold" button on phone A.	Ensure established call between phone A and phone C is not dropped during the time the call is placed on hold, and call resumes normally after removing call on hold.	**PASS**
20	Caller ID restricted Restrict Calling number and name on phone A. Place a call from phone A to phone C, answer phone C	Ensure the display on phone C does not display phone A's number. Must be restricted.	**PASS**[8]
21	Caller ID restricted Restrict Calling number and name on phone C. Place a call from phone C to phone A, answer phone A	Ensure the display on phone A does not display phone C's number. Must be restricted	**PASS**
22	Call Forward – All	Ensure the call is forwarded	**PASS – Calling name/number are**

[8] Change Avaya station display restricted to **blank or yes**, and change the trunk group number to restrict.

	Test Case	Description	Test Result
	Place phone A to forward all calls to phone D. Place a call from phone C to phone A.	immediately to phone D and the call is established successfully between phone C and phone D. Calling name/number shall be updated correctly	**updated correctly.**
23	Call Forward – All Place phone A to forward all calls to phone B. Place a call from phone D to phone A.	Ensure the call is forwarded immediately to phone B and the call is established successfully between phone D and phone B. Calling name/number shall be updated correctly	**PASS – Calling name is updated correctly (No Calling number being display).**
24	Call Forward – All Place phone A to forward all calls to phone D. Place a call from phone B to phone A.	Ensure the call is forwarded immediately to phone D and the call is established successfully between phone B and phone D. Calling name/number shall be updated correctly	**PASS – Calling name/number are updated correctly.**
25	Call Forward – No Reply Place phone A to forward calls on no reply to phone D. Place a call from phone C to phone A.	Ensure the call is forwarded to phone D, after phone A rings a predetermined number of times with no answer. The call shall be established successfully between phone C and phone D. Calling name/number shall be updated correctly	**PASS – Calling name/number are updated correctly.**
26	Call Forward – No Reply Place phone A to forward calls on no reply to phone B. Place a call from phone D to phone A.	Ensure the call is forwarded to phone B, after phone A rings a predetermined number of times with no answer. The call shall be established successfully between phone D and phone B. Calling name/number shall be updated correctly	**PASS – Calling name is updated correctly. (No Calling number being display).**
27	Call Forward – No Reply Place phone B to forward calls on no reply to phone D. Place a call from phone A to phone B.	Ensure the call is forwarded to phone D, after phone B rings a predetermined number of times with no answer. The call shall be established successfully between phone A and phone D. Calling name/number shall be updated correctly	**PASS – Calling name/number are updated correctly.**
28	Call Forward – Busy Place phone A to forward calls on busy to phone D. Place a call from phone B to phone A, then place a call from phone C to phone A.	Ensure the second call is forwarded to phone D, on busy. The call shall be established successfully between phone C and phone D. Calling name/number and connected name/number shall be updated correctly	**PASS – Calling name/number are updated correctly.**
29	Call Forward – Busy Place phone A to forward calls on busy to phone B. Place a call from phone D to phone A, then place a call from phone C to phone A.	Ensure the second call is forwarded to phone B, on busy. The call shall be established successfully between phone C and phone B. Calling name/number shall be updated correctly	**PASS – Calling name displayed correctly on Phone B (No Calling number.)**
30	Call Forward – Busy Place phone B to forward calls on busy to phone C. Place a call from phone D to phone B, then place a call from phone A to phone B.	Ensure the second call is forwarded to phone D, on busy. The call shall be established successfully between phone A and phone C. Calling name/number shall be updated correctly	**PASS – Calling name/number are updated correctly.**

	Test Case	Description	Test Result
31	DTMF Phone A calls Phone C. Phone C calls Phone A. RFC2833 and/or H245-(Alphanumeric or Signal) to G711 and vice-versa	Run independent tests from both ends to ensure DTMF tones arrive at both ends correctly.	**In-band: PASS** **H245/Out-of-band signaling: PASS** **RFC2833: PASS**
32	Digit translation – The voice gateway can manipulate the digits of the called 10-digit number sent by PSTN and 4 digit number sent by PBX	Ensure the voice gateway is capable of manipulating the called and calling digits to match configured dial-peers and route calls appropriately	**PASS**

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