Application Note

Avaya S8500 Version 4.0 to Cisco IOS Voice Gateway using SIP with T1-PRI to PSTN

February 20, 2008

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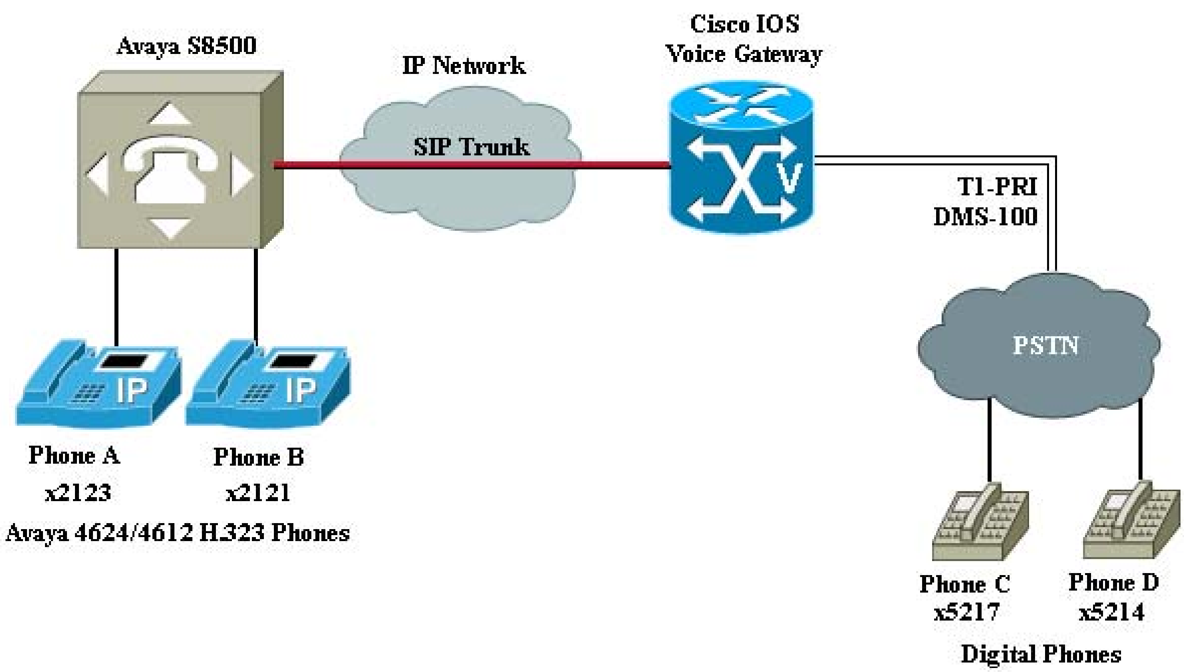
Introduction

- This Application note provides basic call interoperability and documented steps and configurations necessary for SIP integration between Avaya S8500 to Cisco IOS Voice Gateway providing PSTN (T1-PRI) connectivity.
- The SIP protocol is used between Cisco IOS Voice gateway and Avaya S8500. The connection between Cisco IOS gateway and PSTN uses T1-PRI (DMS-100 switch type).
- Features tested include Basic call, Call Transfer supervised, Call Transfer blind, Call Forward (Unconditional, Busy and No Answer), Three-way Conference, DTMF tones (in-band and relay via RFC2833), Digit Translation, and Call Hold.
- The Cisco IOS Voice Gateway offers the advantage of providing connectivity between Avaya S8500 and PSTN by offering SIP to T1-PRI 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 (10/100baseT) and connected to the PSTN via T1-PRI.
- This Application Notes uses the Cisco 3845 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 IAD2400 Series 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.726-32K 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.
- The Avaya H323 phones did not work when the trunk codec was set to G.723-5.3K or G.723-6.3K.
- For inband DTMF, the Avaya ”DTMF over IP” setting in the signalling group had to be set to ”in-band”, not ”in-band-g711”.
- For basic calls with RFC2833 DTMF relay, it was noticed that the PBX phone (e.g., ’A’) had to send DTMF to the PSTN digital phone (e.g., ’C’) before it could play tones originating from the PSTN digital phone. This was due to the H.323/SIP interworking on the PBX itself, and was also noticed on calls involving only Avaya SIP and H.323 phones.
- On a call that originates from a PSTN station to a PBX station and that is then forwarded unconditionally or on busy (e.g., phone C calls phone A, and phone A forwards to phone D), the originating PSTN user does not get ringback. Cisco IOS Voice Gateway does not support SIP 181 message (call being forwarded) which causes SDP establishment failure during a call forward between PSTN phone and Avaya stations.

System Components

Hardware requirements

Cisco equipment

- Cisco 3845 (Cisco 3800 family routers)

Avaya equipment

- Avaya S8500
- Avaya SIP Enablement Server (SES)
- (2) Avaya 4624/4612 IP phones
- TN2312BP IPSI
- TN799DP C-LAN
- TN2302AP IP Media Processor

Software Requirements

- Cisco IOS Voice Gateway: Cisco IOS Release – Cisco 3845 Version 12.4(15)T3: c3845-ipvoice-mz.124-15.T3.bin
- Avaya Communications Manager Release 4.0
- Avaya SES software version 4.0
- Avaya 4612/4624 (H.323 phones) load: def24rl_8_3.bin

Features

Features Supported

- Basic call with G.729, G.729B, G.711ulaw and G.711Alaw
- Call Transfer Blind and Call Transfer Supervised
- Call Conference
- Call on Hold
- Call Forward All (See Limitations Section)
- Call Forward No Reply
- Call Forward Busy (See Limitations Section)
- Digit Translation
- DTMF in-band
- DTMF - RFC2833 (See Limitations Section)

Configuration

Avaya S8500 Configuration

Figure 2. **Signaling group (In band)**

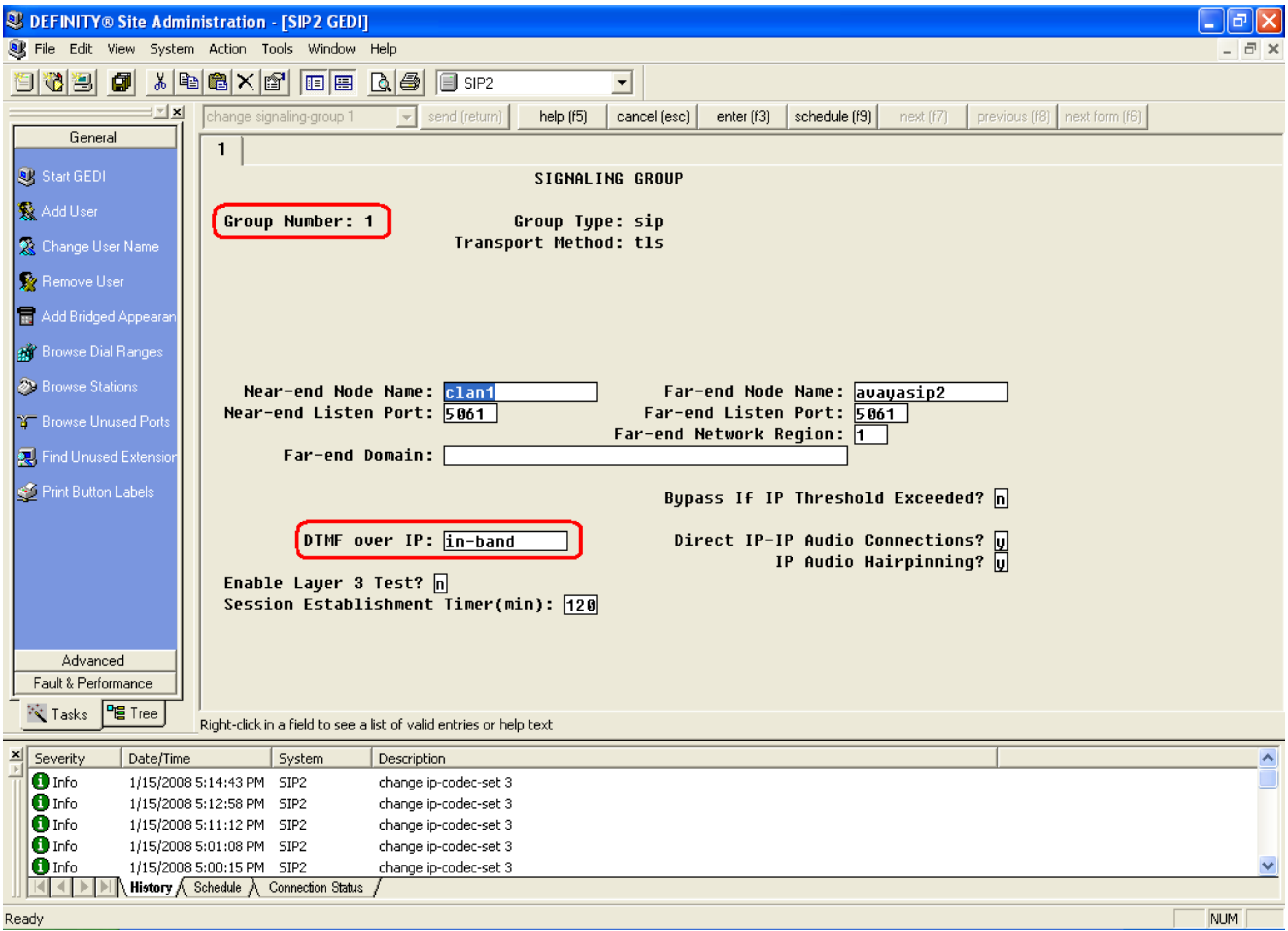


Figure 3. **Signaling group (RFC 2833)**

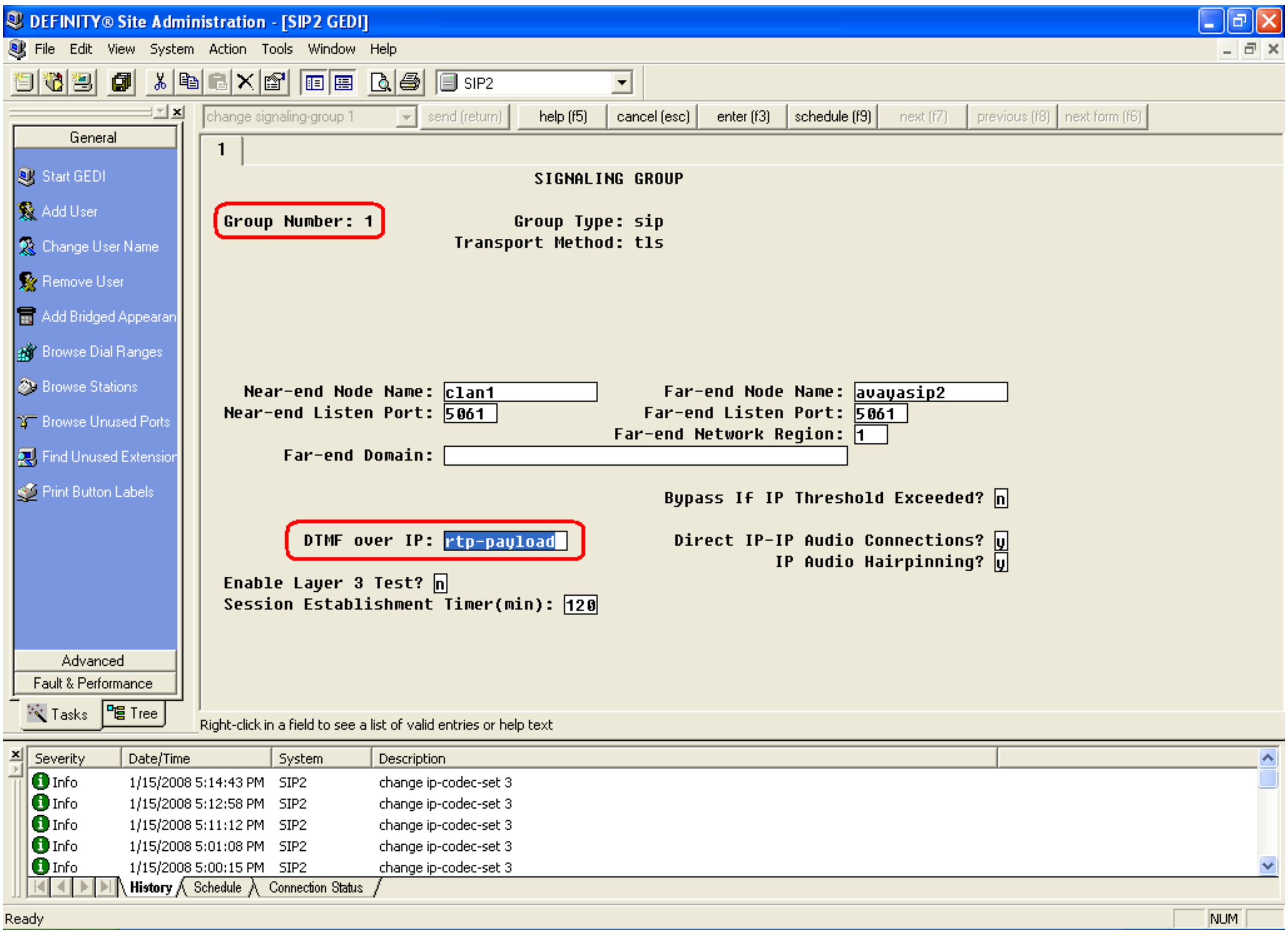


Figure 4. **Trunk group – p1 of 3**

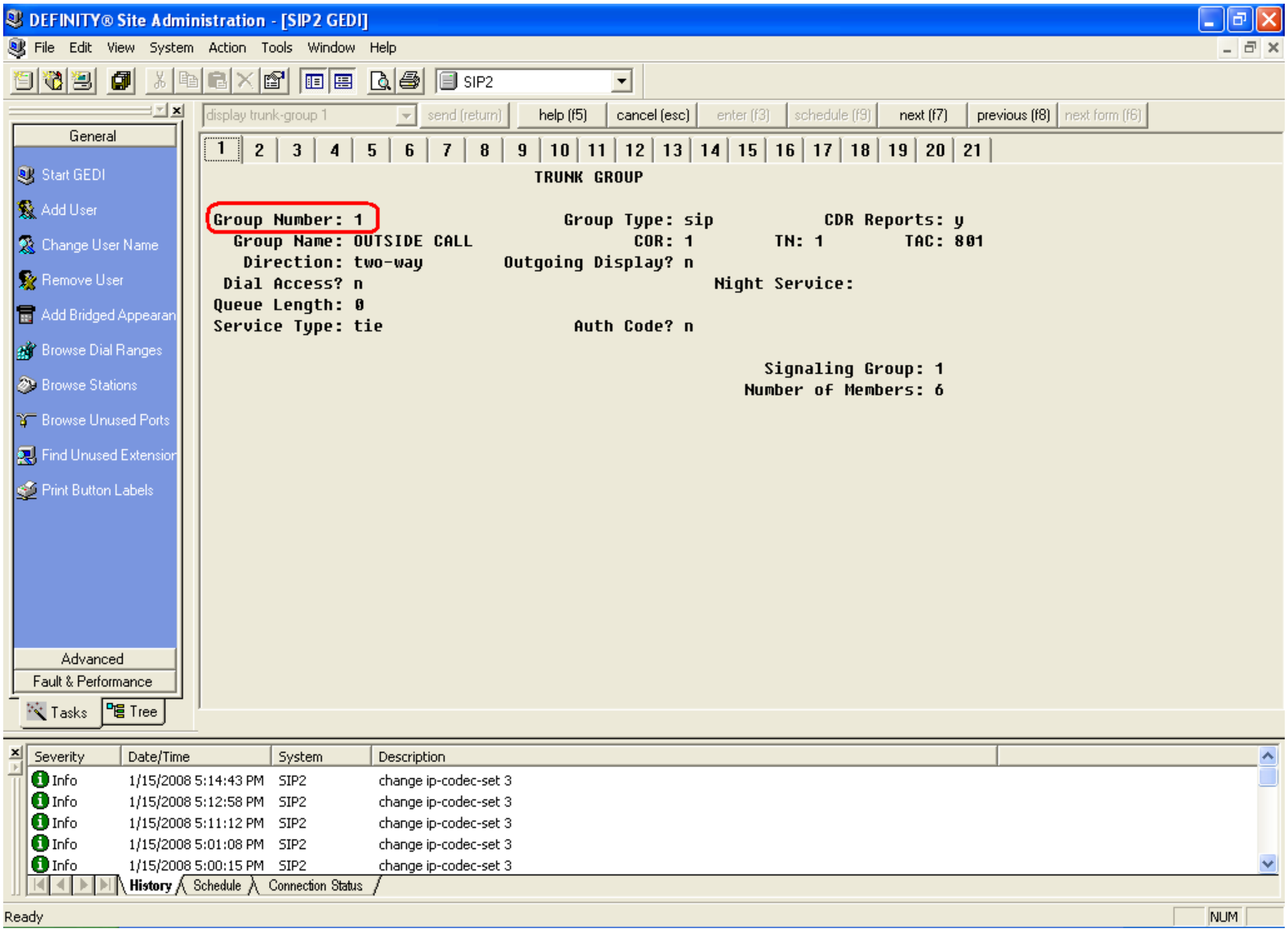


Figure 5. **Trunk group – p2 of 3**

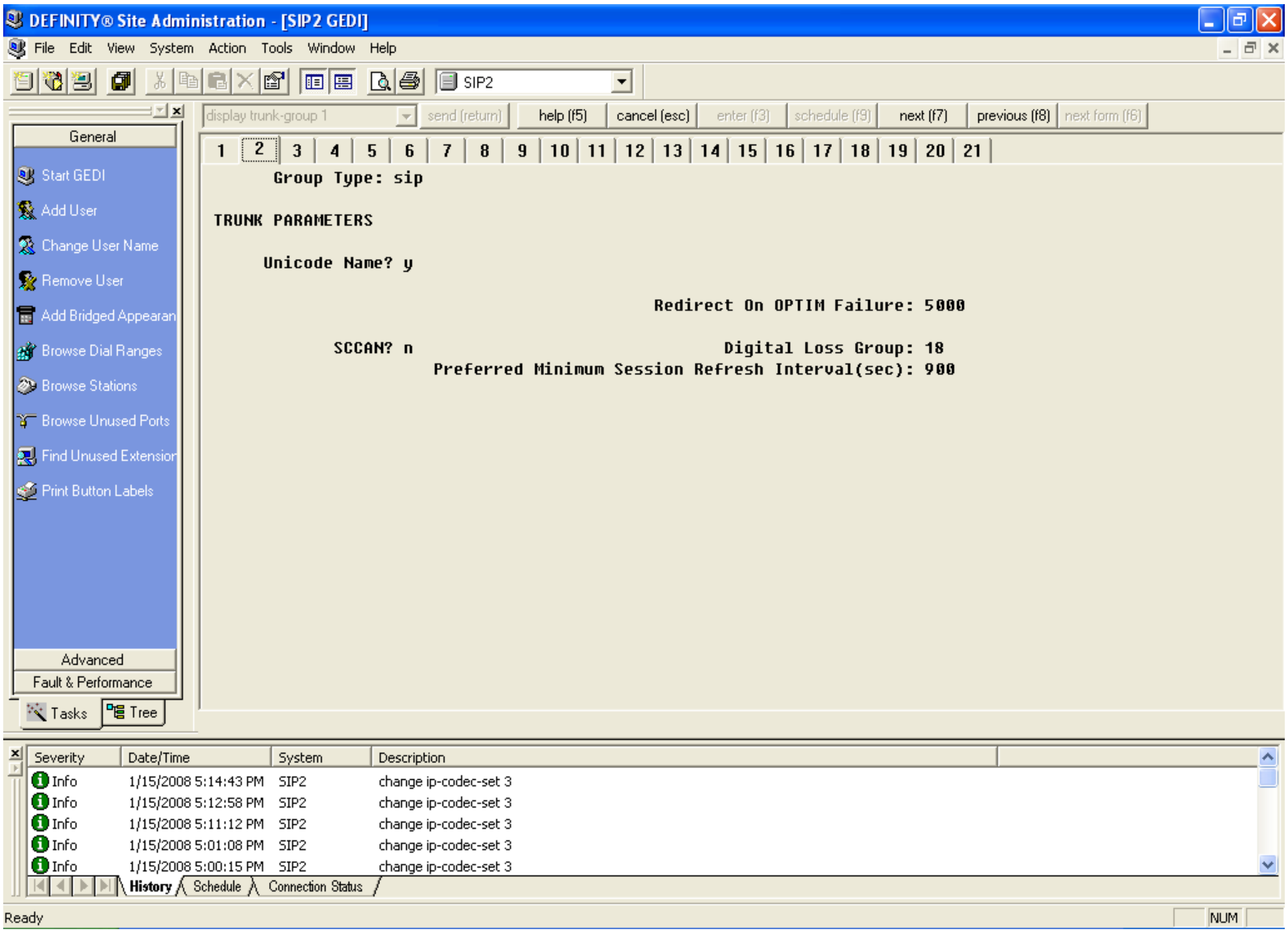


Figure 6. Trunk group – p3 of 3

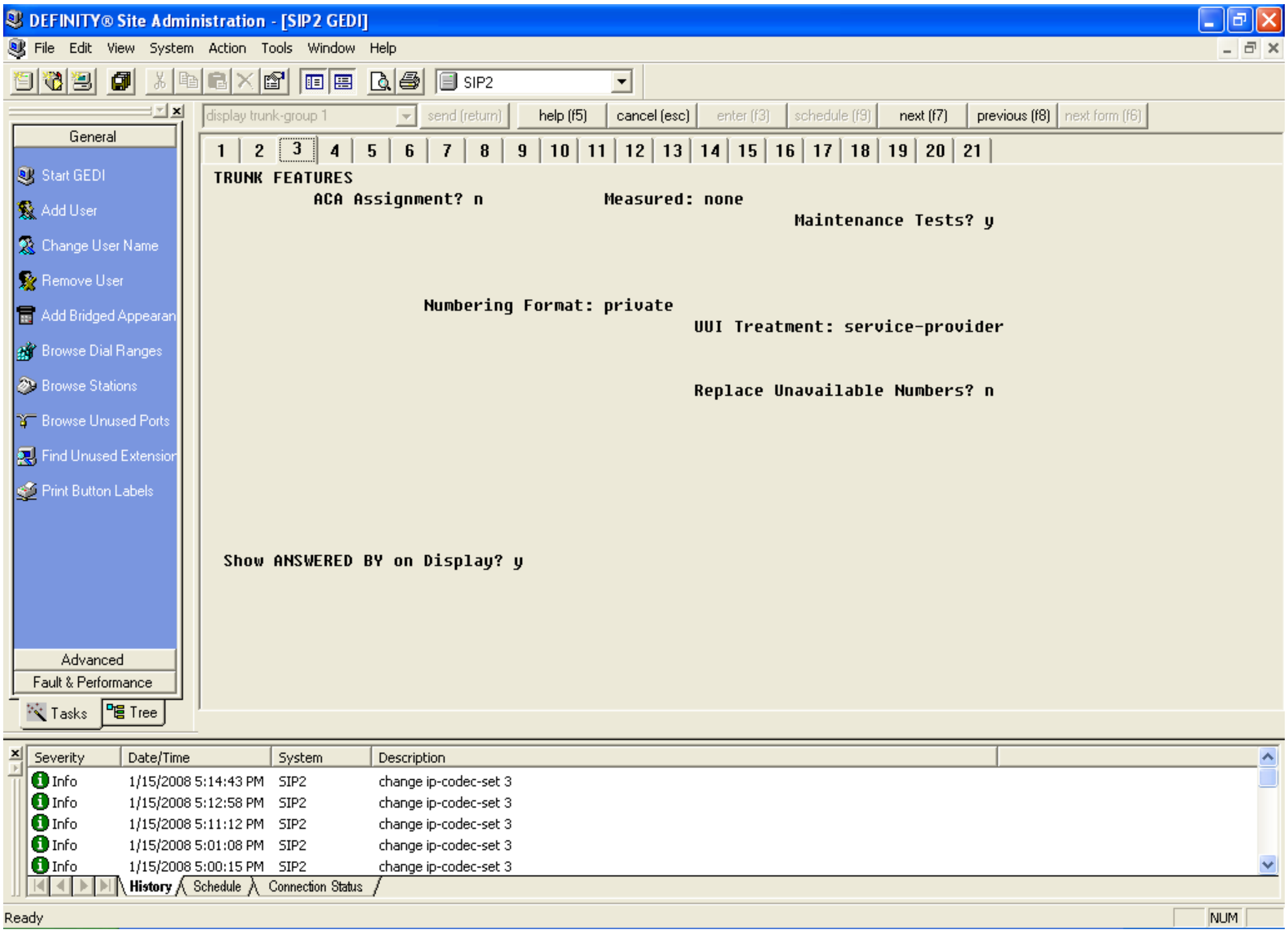


Figure 7. Node-names IP

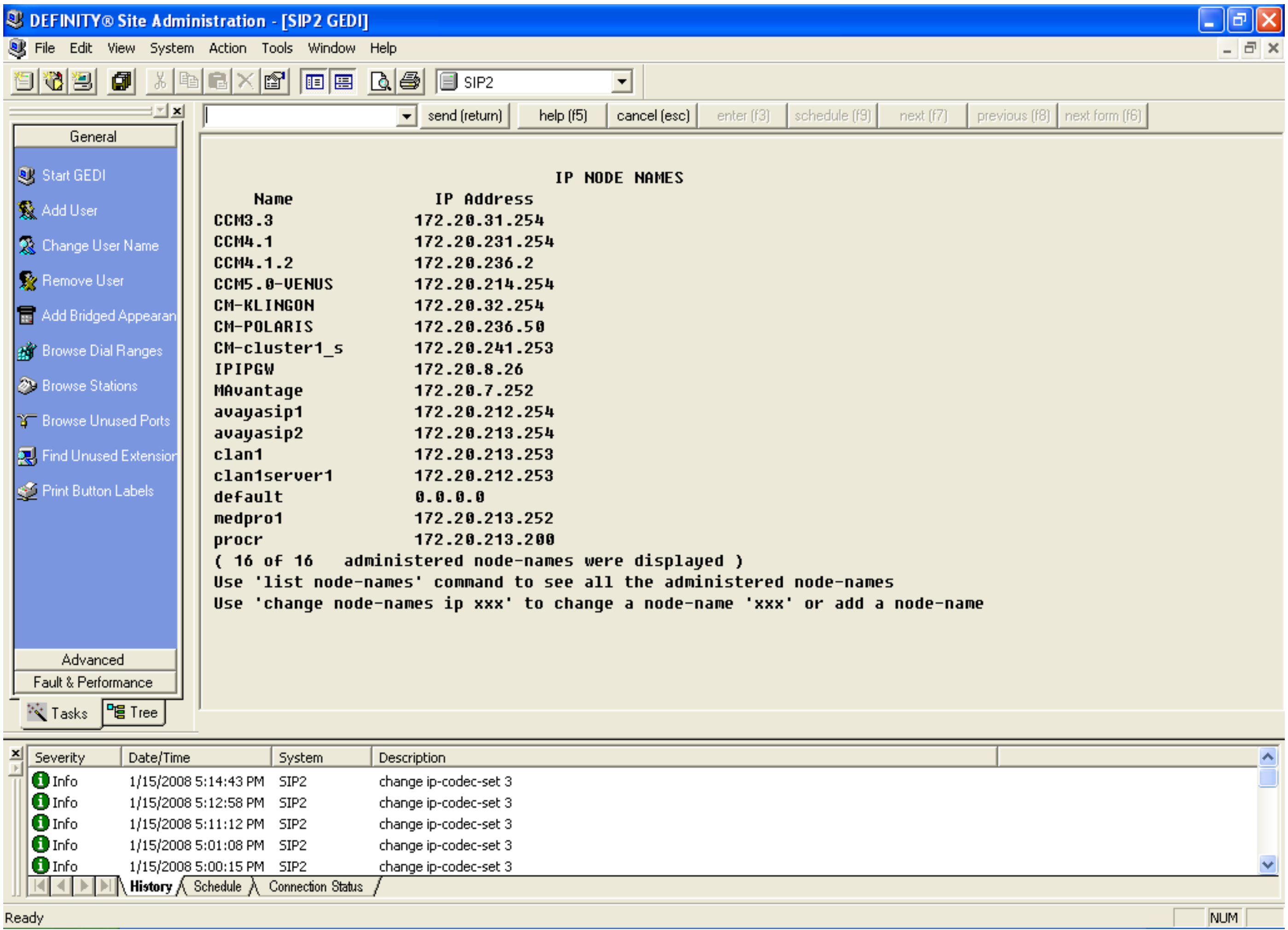


Figure 8. **IP Network Region**

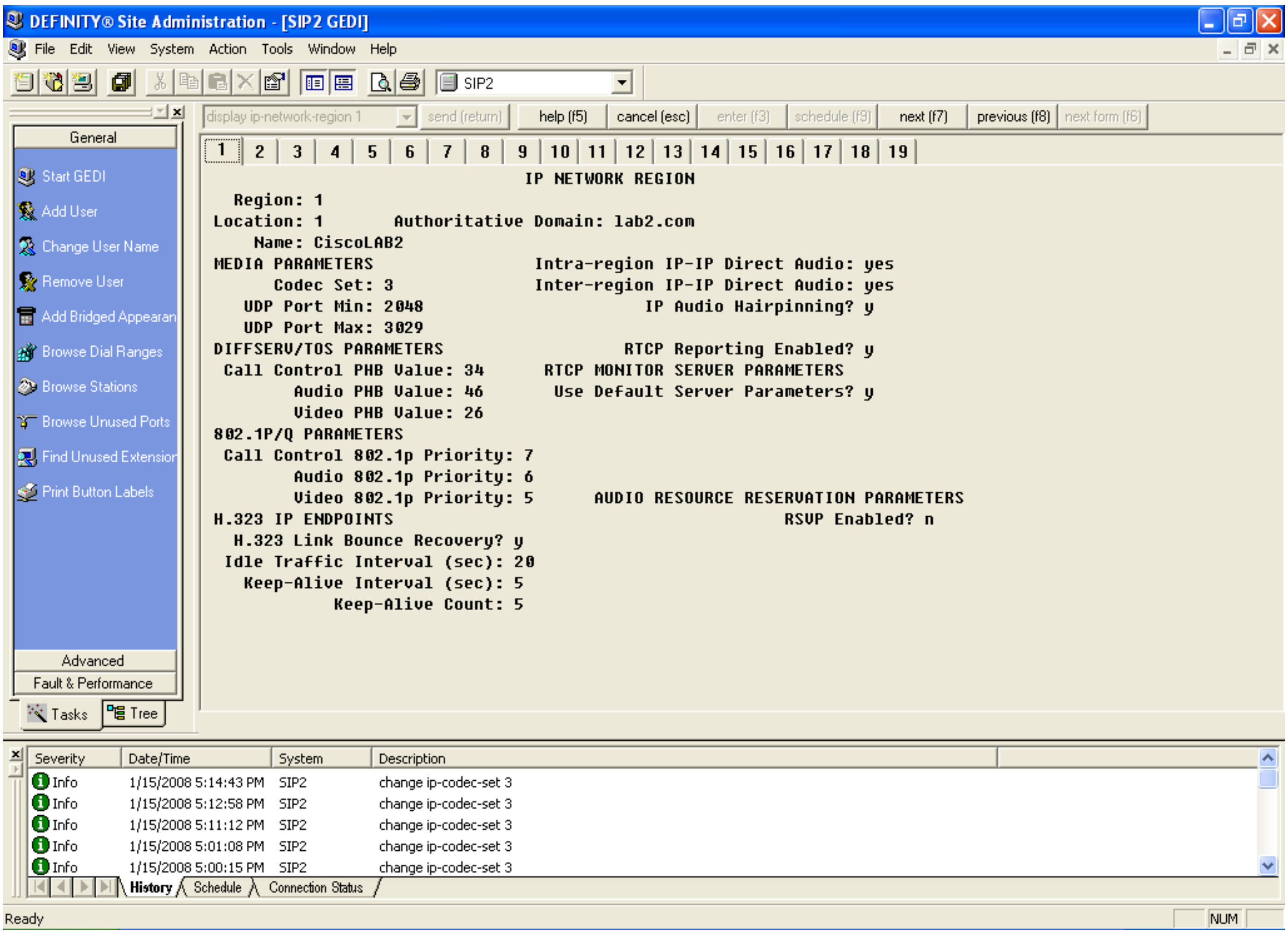


Figure 9. **IP codec set (showing G.729, annex B).**

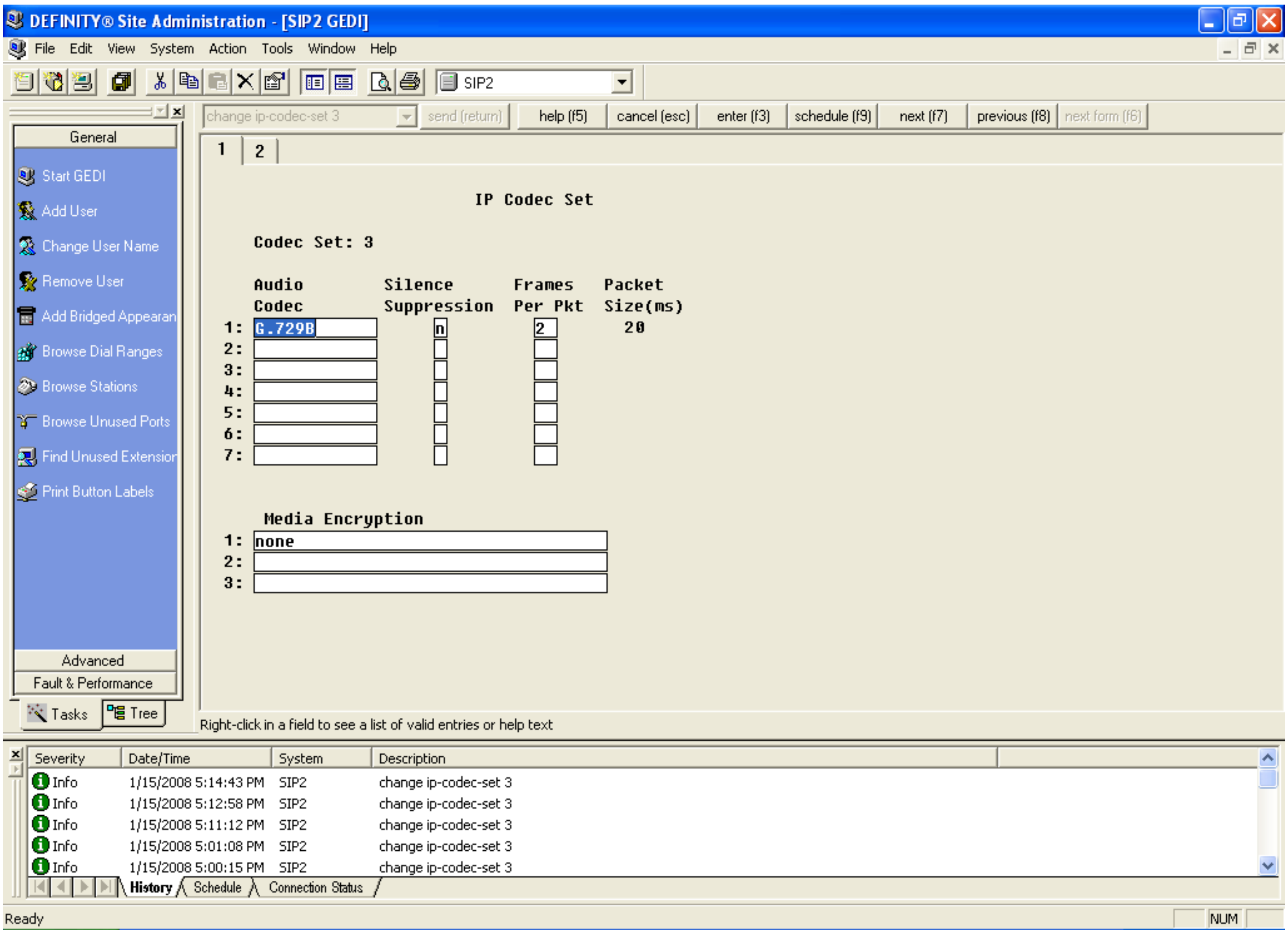


Figure 10. **Uniform dialing plan**

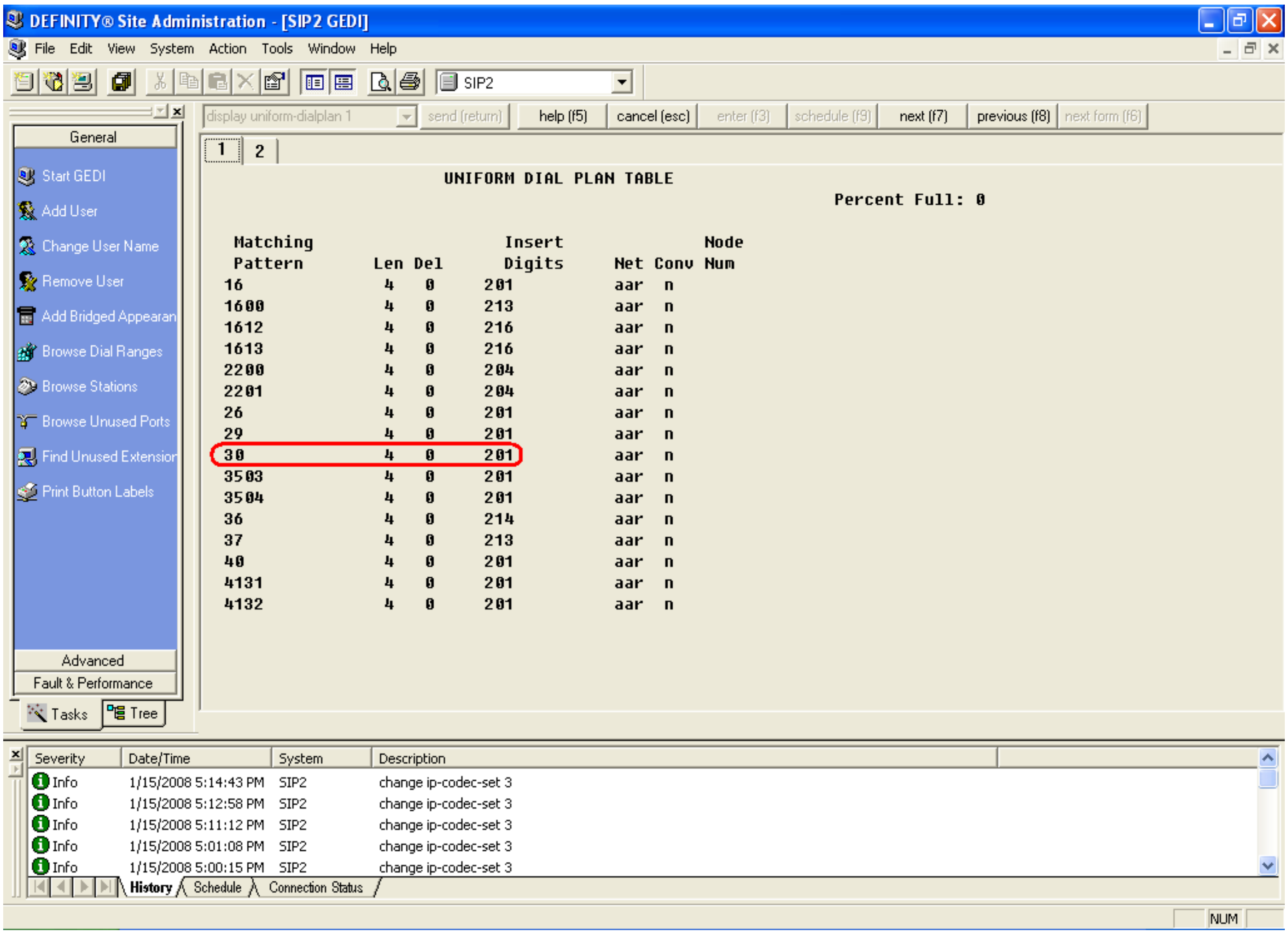


Figure 11. **AAR analysis**

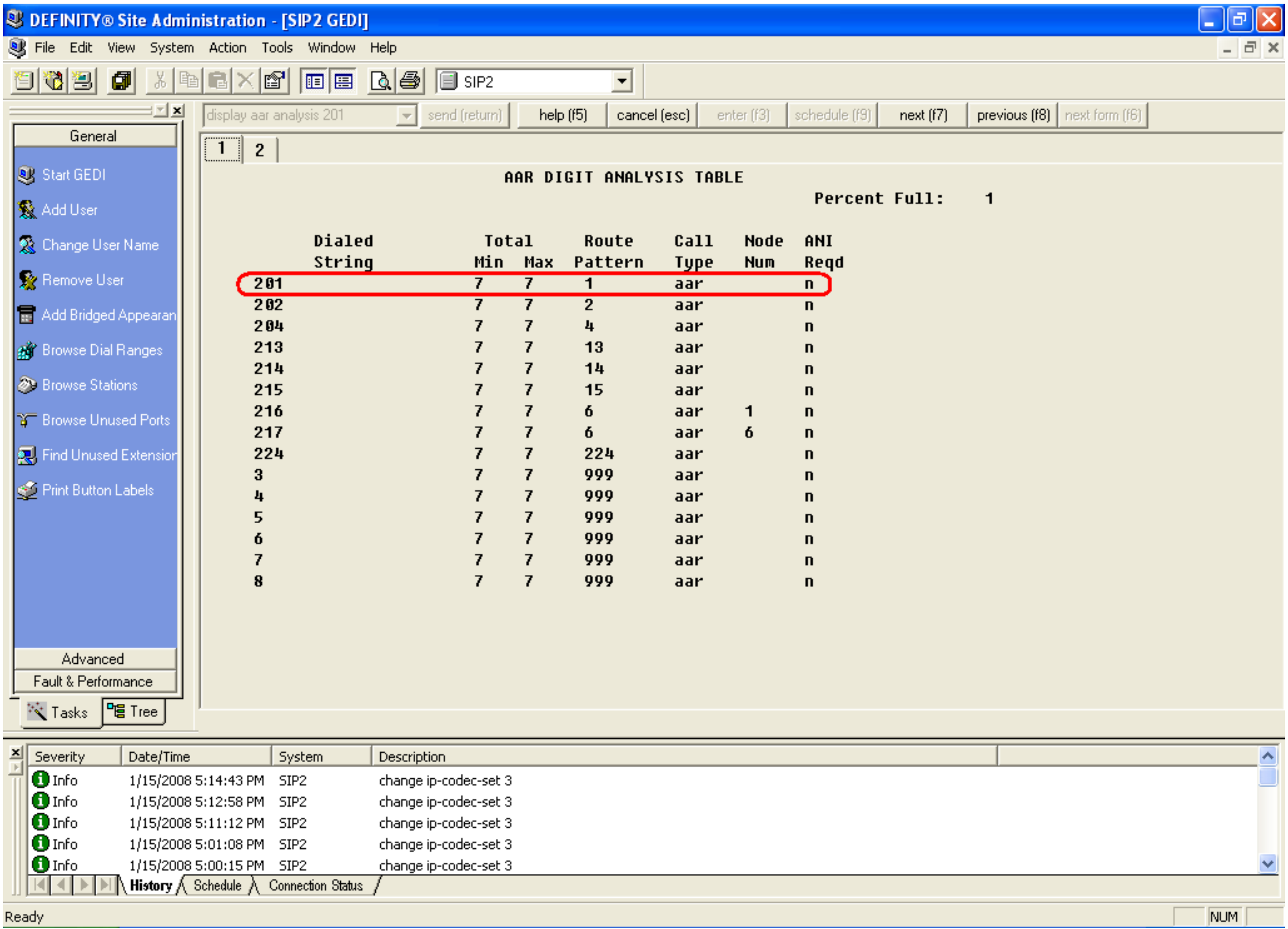
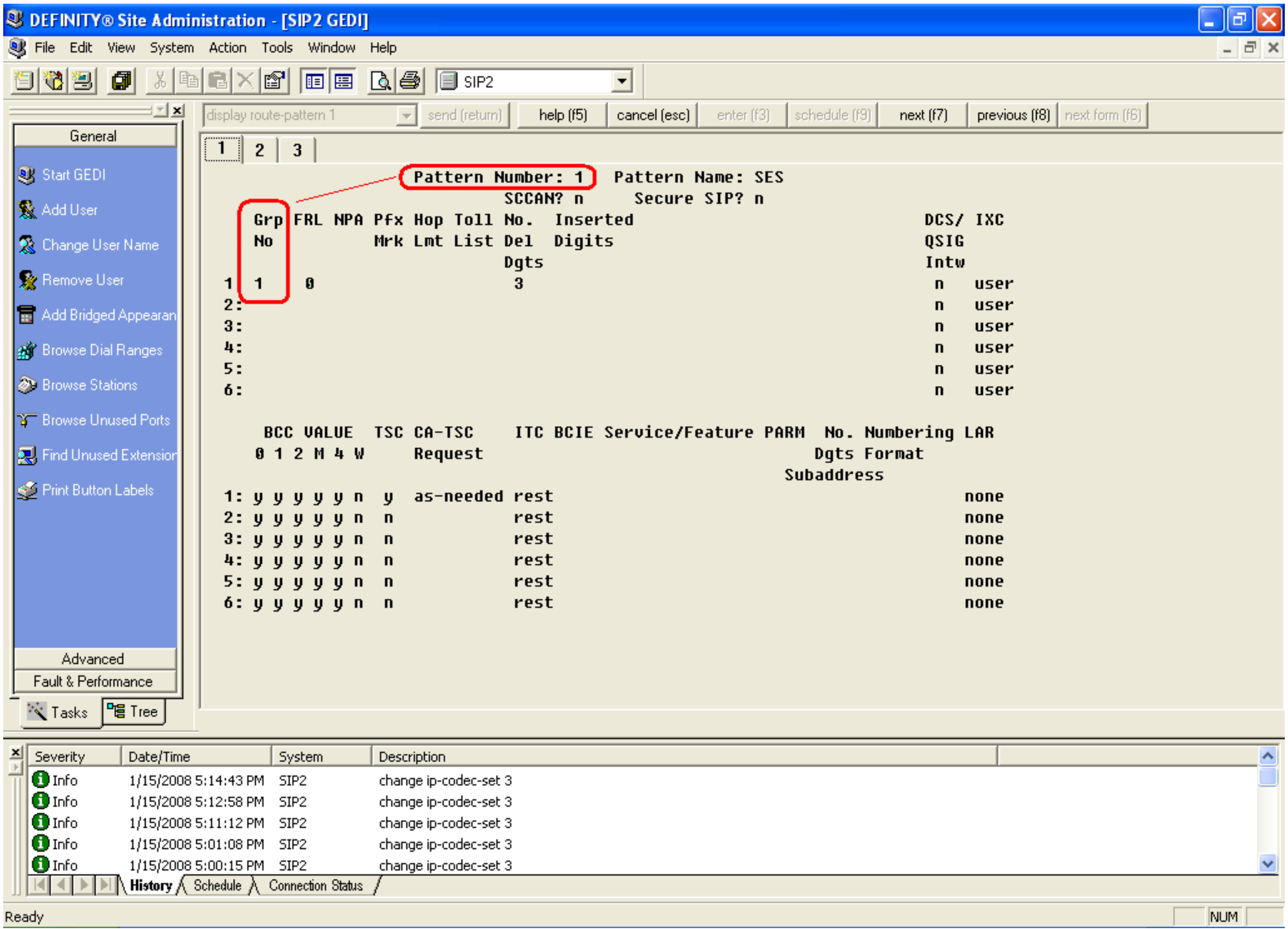


Figure 12. **Route Pattern**



Avaya SIP Enablement Server (SES) Configuration

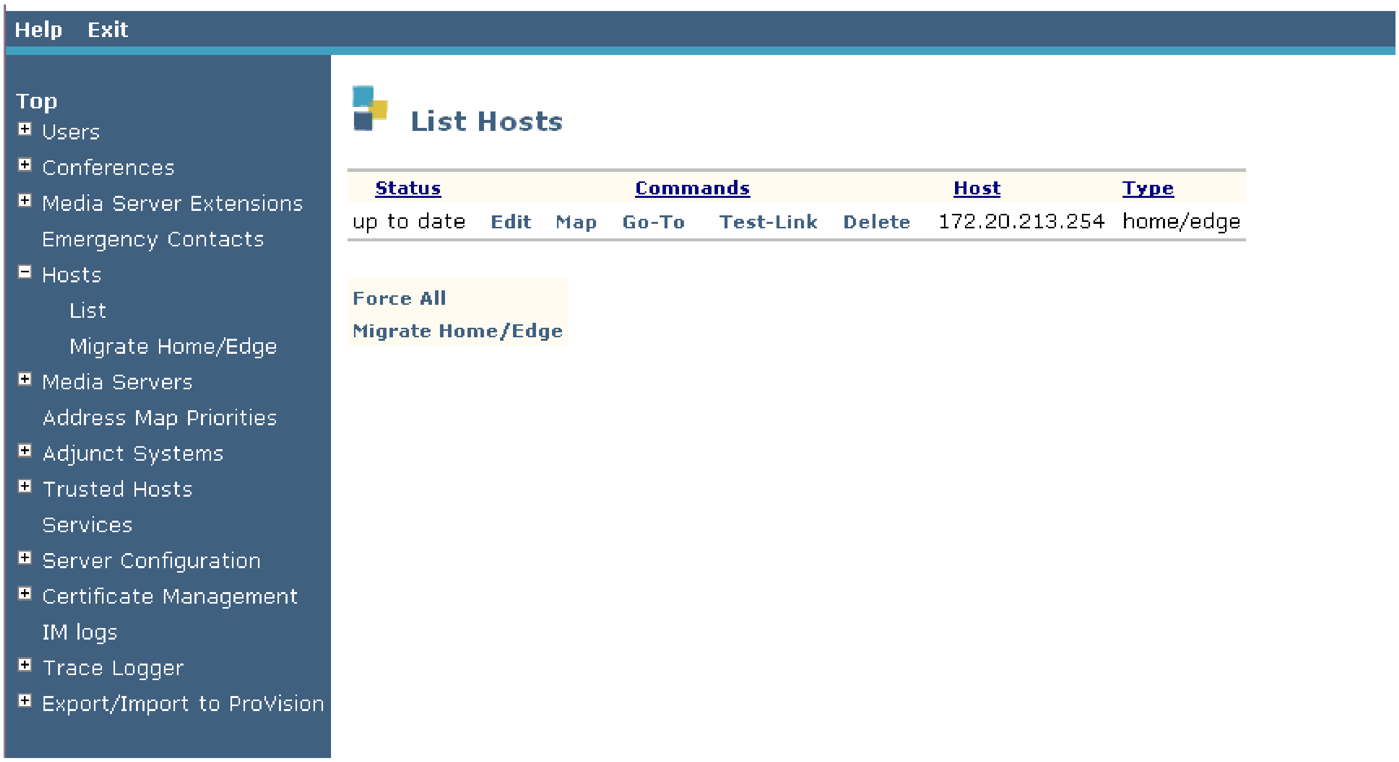


Figure 13. SES Server

Figure 14. SES Trusted Hosts

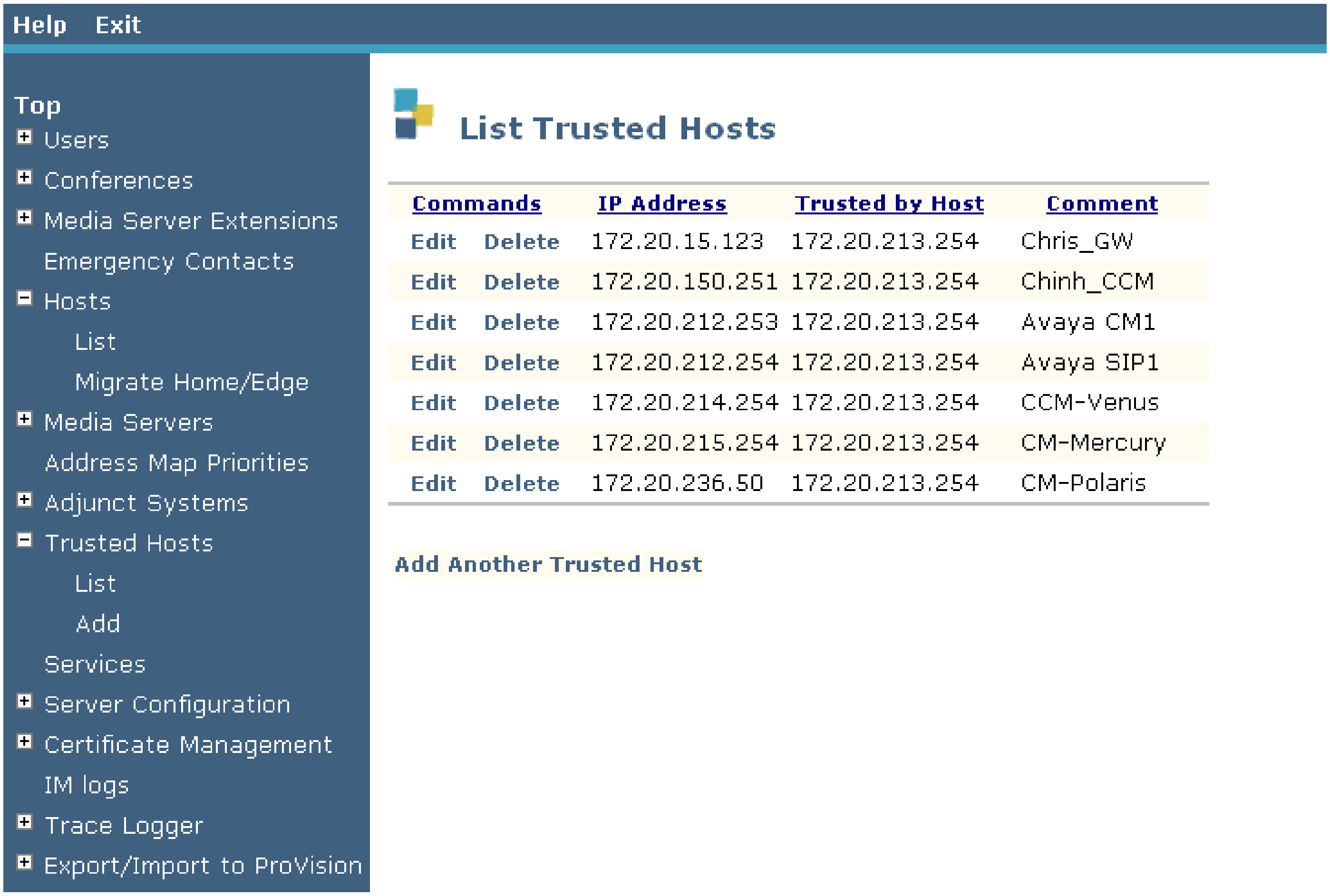


Figure 15. SES DN Map

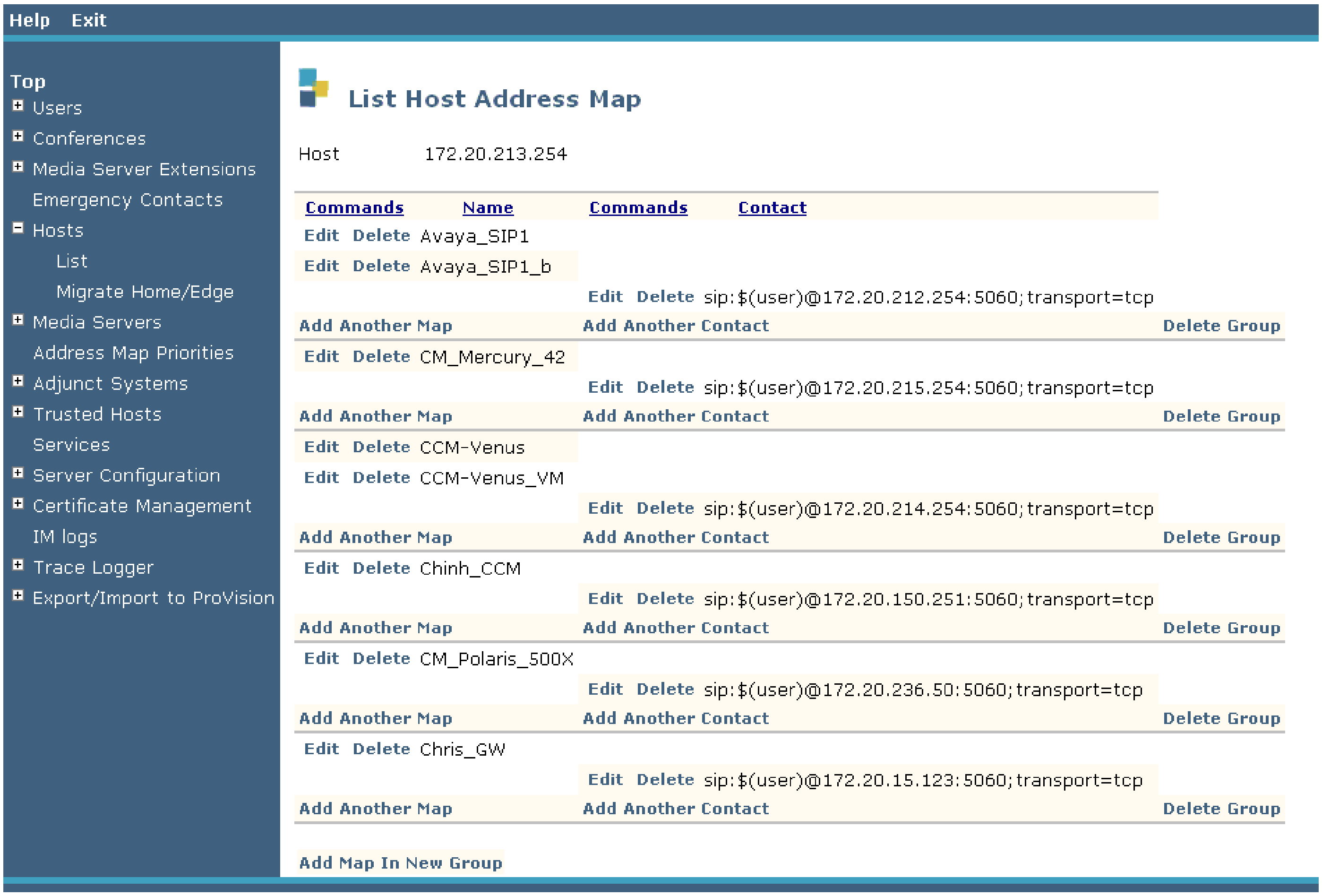
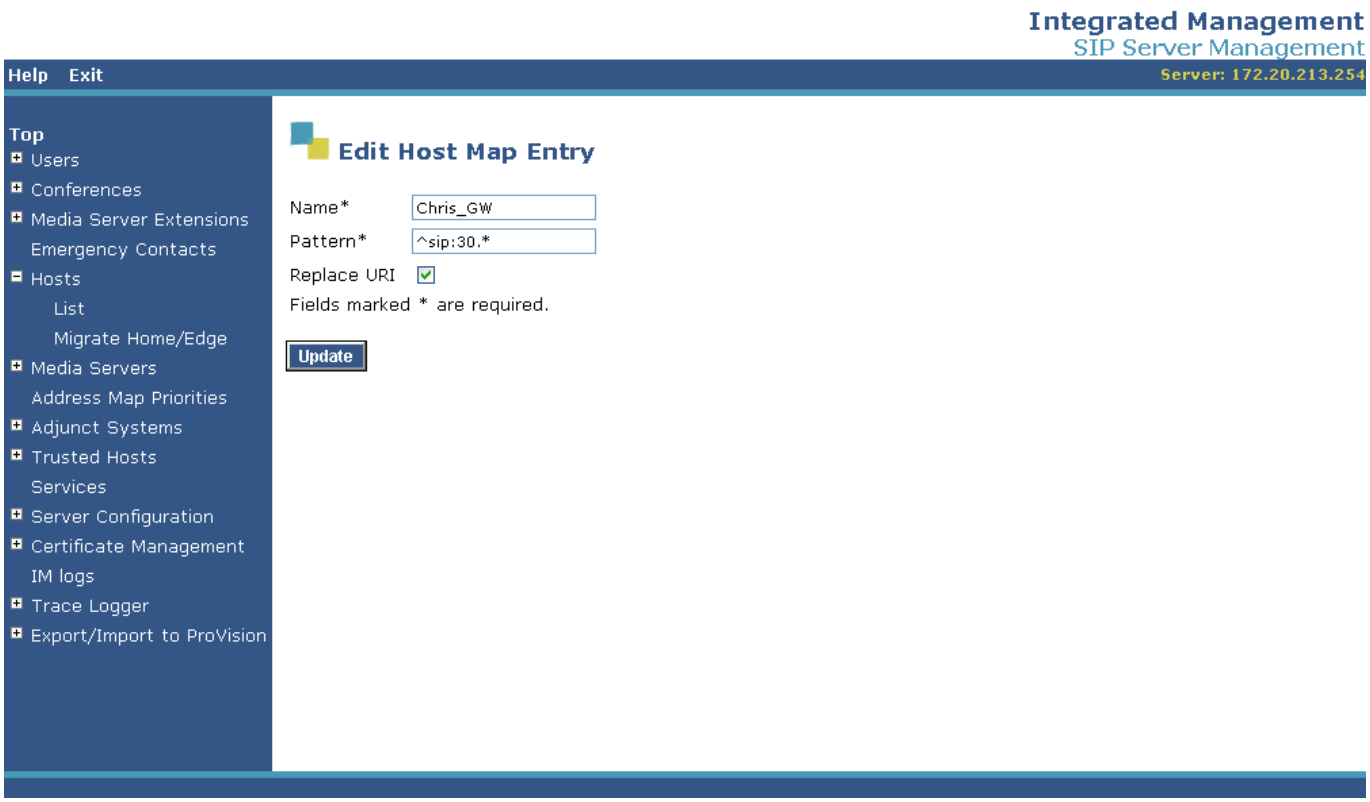


Figure 16. **SES 30XX Routing Entry**



Cisco IOS Voice Gateway (c3845) configuration

show version

Cisco IOS Software, 3800 Software (C3845-IPVOICE-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 Fri 25-Jan-08 00:11 by prod_rel_team

ROM: System Bootstrap, Version 12.3(11r)T1, RELEASE SOFTWARE (fc1)

3845_West uptime is 2 weeks, 43 minutes
System returned to ROM by reload at 18:17:14 UTC Wed Feb 6 2008
System image file is "flash:c3845-ipvoice-mz.124-15.T3.bin"

Cisco 3845 (revision 1.0) with 224256K/37888K bytes of memory.
Processor board ID FHK0847F136
2 Gigabit Ethernet interfaces
24 Serial interfaces
2 Channelized E1/PRI ports
2 Channelized T1/PRI ports
2 Voice FXS interfaces
DRAM configuration is 64 bits wide with parity enabled.
479K bytes of NVRAM.
62592K bytes of ATA System CompactFlash (Read/Write)

Configuration register is 0x2102

show run
Building configuration...

Current configuration : 2936 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname 3845_West
!
boot-start-marker
boot-end-marker
!
card type t1 0 0
logging buffered 1000000
no logging console
enable secret 5 1MFhi$AqqpDsFeO4Sb/IkzkrcmO/
!
!
no aaa new-model
network-clock-participate wic 0
network-clock-select 1 T1 0/0/0
voice-card 0
no dspfarm

```
!
ip cef
!
no ip domain lookup
!
!
isdn switch-type primary-dms100
isdn gateway-max-interworking
!
!
voice call carrier capacity active
!
voice service pots
!
voice service voip
!
!
!
voice translation-rule 1 [1]
 rule 1 /30/ /52\1/
!
voice translation-rule 2
 rule 1 /11/ /21\1/
!
!
voice translation-profile PSTN-to-SIP
 translate called 2
!
voice translation-profile SIP-to-PSTN [2]
 translate called 1
!
!
controller T1 0/0/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
controller T1 0/0/1
 framing esf
 linecode b8zs
!
ip tcp synwait-time 13
!
!
interface GigabitEthernet0/0
 description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
 ip address 10.10.10.1 255.255.255.248
 shutdown
 duplex auto
 speed auto
 media-type rj45
 no keepalive
!
```

[1] The above translation rule replaces the number “30xx” with “52xx”

[2] This defines the translation profile for the called number. In this particular example, the called number “30xx” is converted to “52xx”.

```
interface GigabitEthernet0/1
 ip address 172.20.15.123 255.255.255.0
 duplex auto
 speed auto
 media-type rj45
no keepalive
!
!
interface Serial0/0/0:23
 no ip address
 encapsulation hdlc
 isdn switch-type primary-dms100
 isdn protocol-emulate network
 isdn incoming-voice voice
 isdn channel-id invert extend-bit
 no cdp enable
!
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 172.20.15.1
!
ip http server
ip http authentication local
!
!
control-plane
!
voice-port 0/0/0:23
!
voice-port 0/1/0
!
voice-port 0/1/1
!
dial-peer voice 4000 pots
 translation-profile incoming PSTN-to-SIP
 destination-pattern 5...
 incoming called-number 1...
 direct-inward-dial
 port 0/0/0:23
forward-digits all
!
dial-peer voice 519 voip
 translation-profile incoming SIP-to-PSTN [3]
 destination-pattern 2...
< rtp payload-type nte 127>[4]
 session protocol sipv2
 session target ipv4:172.20.213.254
 incoming called-number 3... [5]
 dtmf-relay rtp-nte [6]
 codec g711ulaw [7]
```

[3] This defines the call number translation profile for incoming calls. Here, a called number "30xx" is converted to "40xx" for an incoming call assocated with this dial peer (i.e., via this SIP trunk).

[4] This command is not necessary when using Avaya Communications Manager 4.0, and was not used for the creation of this application note. It is only needed with prior releases of Avaya Communications Manager.

[5] This is to specify a digit string that can be matched by an incoming call to associate the call with a dial peer. For this example, user dials 3xxx will associate to this dial-peer.

[6] This is for Dual-Tone Multifrequency (DTMF) tones to be sent RFC2833. It is omitted for inband DTMF.

[7] Also changed to G.711A, G.729, and G.729B during testing.

```
!
banner login ^C
!
line con 0
 exec-timeout 600 0
 password cisco
 login
line aux 0
line vty 0 4
 exec-timeout 600 0
privilege level 15
 password cisco
 login
 transport input telnet
line vty 5 15
 privilege level 15
 login local
 transport input telnet
!
scheduler allocate 20000 1000
!
end
```

Acronyms

Acronym	Definitions
codec	compressor/decompressor
PBX	Private Branch Exchange
PSTN	Public Switched Telephone Network
IOS	Internetworking Operating System

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