Application Note

Microsoft Office Communication Server 2007 version RTM to Cisco IOS Voice Gateway using SIP with E1 ISDN

October 3, 2007 Revision 4

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Introduction

This Application note provides basic call interoperability and documented steps and configurations necessary for SIP integration between Microsoft (MSFT) Office Communications Server (OCS) 2007 version Release To Manufacturing (RTM) and MSFT Mediation Server to Cisco ISR Voice Gateway providing PSTN E1 connectivity.

The SIP Protocol is used between Cisco ISR Voice gateway and MSFT Mediation Server. The connection between Cisco ISR 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 Microsoft Office Communicator (MOC) and PSTN

The Cisco ISR Voice Gateway offers the advantage of providing connectivity between Microsoft Office Communication Server 2007 and PSTN by offering SIP to ISDN inter-working functionality.

The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability with the Cisco ISR Voice Gateway connected to the OCS/Mediation server via SIP (10/100baseT) and connected to the PSTN via E1 PRI ISDN.

This Application Note uses the C3825 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. Here is a list of Cisco Products capable of voice gateway functionality: Care must be taken when selecting a voice gateway platform depending of the capacity required for the intended deployments

Cisco 1861 Integrated Services Router
Cisco 2800 Series Integrated Services Routers
Cisco 3600 Series Routers
Cisco 3800 Series Integrated Services Routers
Cisco AS5350XM Universal Gateway
Cisco AS5400XM Universal Gateway

Network Topology

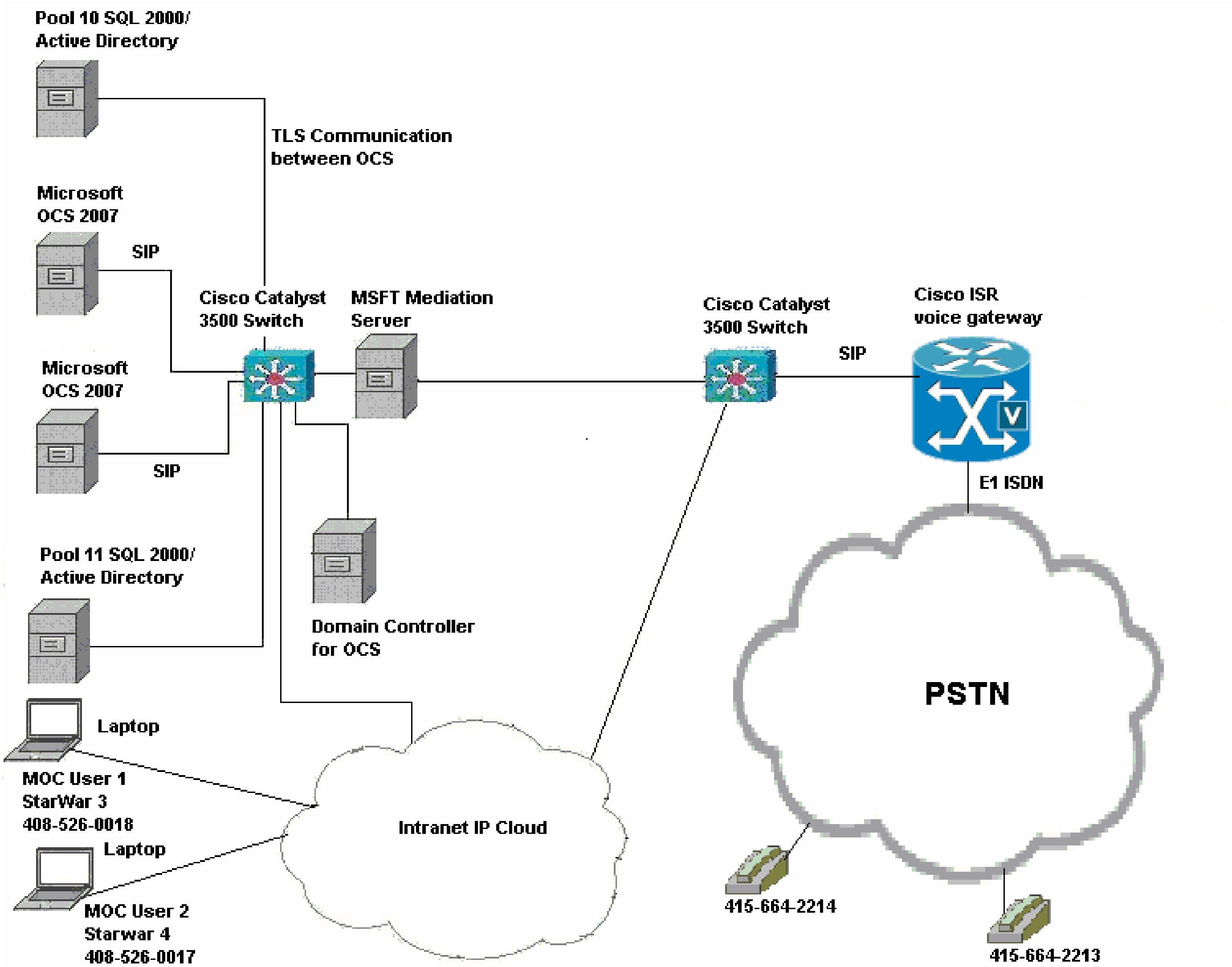


Figure 1. Network Topology

Limitations

Call transfer supervision is not supported by OCS 2007 RTM.

In the event of OCS MOC calling PSTN user: Caller ID restrict feature is not supported by OCS 2007 RTM. The OCS does not have the caller ID restriction feature. However, the PSTN outbound dial-peer on Cisco's ISR gateway can be configured to restrict caller ID as a work-around. For details, refer to the IOS dial-peer configuration.

In the event of PSTN user calling OCS MOC Client: Caller ID restriction using SIP Remote-Party-Identifier (RPID) is not honored by OCS. The work around is to strip the calling number from the SIP Invite at the ISR gateway when the incoming ISDN set up message contains calling number Information Element (IE) set to restricted. For details, refer to the IOS dial-peer configuration.

Call forward busy is not supported by OCS 2007 RTM.

System Components

Hardware requirements

Cisco Hardware

Cisco 3825 ISR Voice Gateway
Cisco Cat 3550 Power Ethernet switch.

Microsoft Hardware

OCS 2007 Enterprise Edition - MCS 7825H - Windows Server 2003 Enterprise, R2 with Service Pack 1 (SP1)
Microsoft Mediation Server
Windows Active Directory Node also serves as DNS for OCS - MCS 7825H - Windows Server 2003 Enterprise R2, with SP 1
Windows SQL - MCS 7825H - SQL Server 2000 Enterprise Edition with Window 2003 Server SP1
Laptop for Microsoft Office Communicator (OCS 2007 end-point)

Software Requirements

IOS Software releases: C3825adventerprisek9_ivs-mz.124-11.T1
Microsoft Software: Microsoft Office Communications Server 2007 version RTM

Features

Features Supported

SIP call establishment with TCP

Codec G.711 Ulaw and Alaw

Calling number

Call Transfer blind

Call Conference

Call on-hold

Call Forward No Reply

Call Forward all

DTMF tones using RFC2833

Digit translation – The voice gateway can modify the digits of the called 10-digit number sent by Microsoft Mediation Server.

Features Not Supported

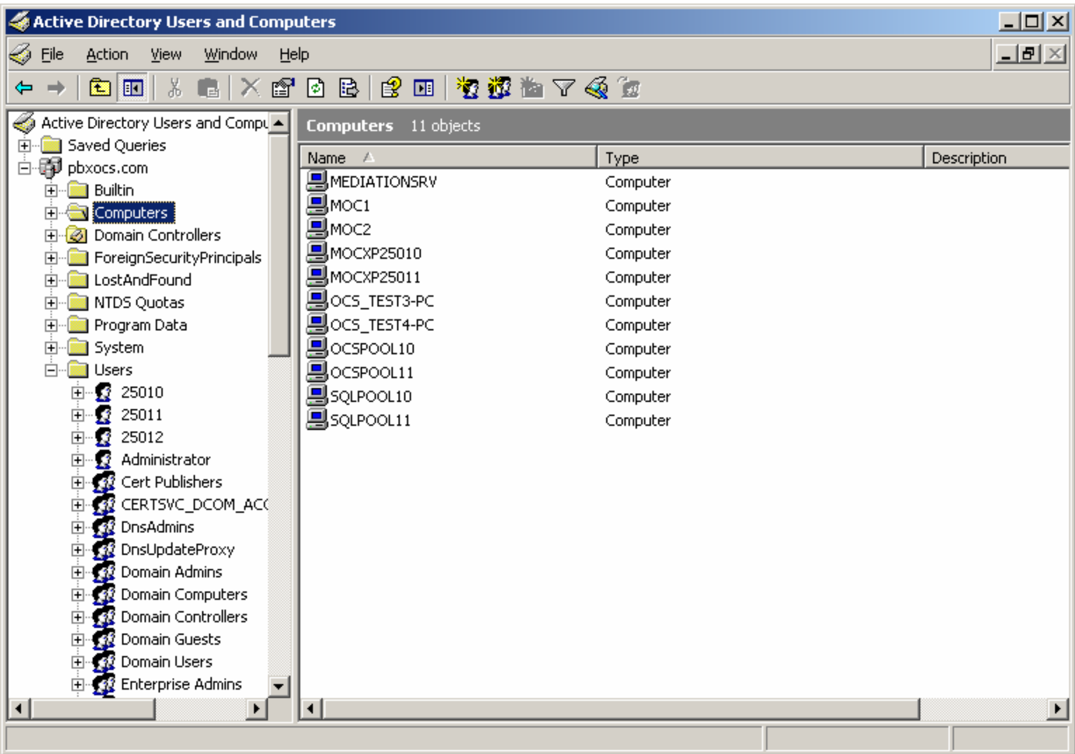
Call Forward Busy

Call Transfer supervised initiated by MOC

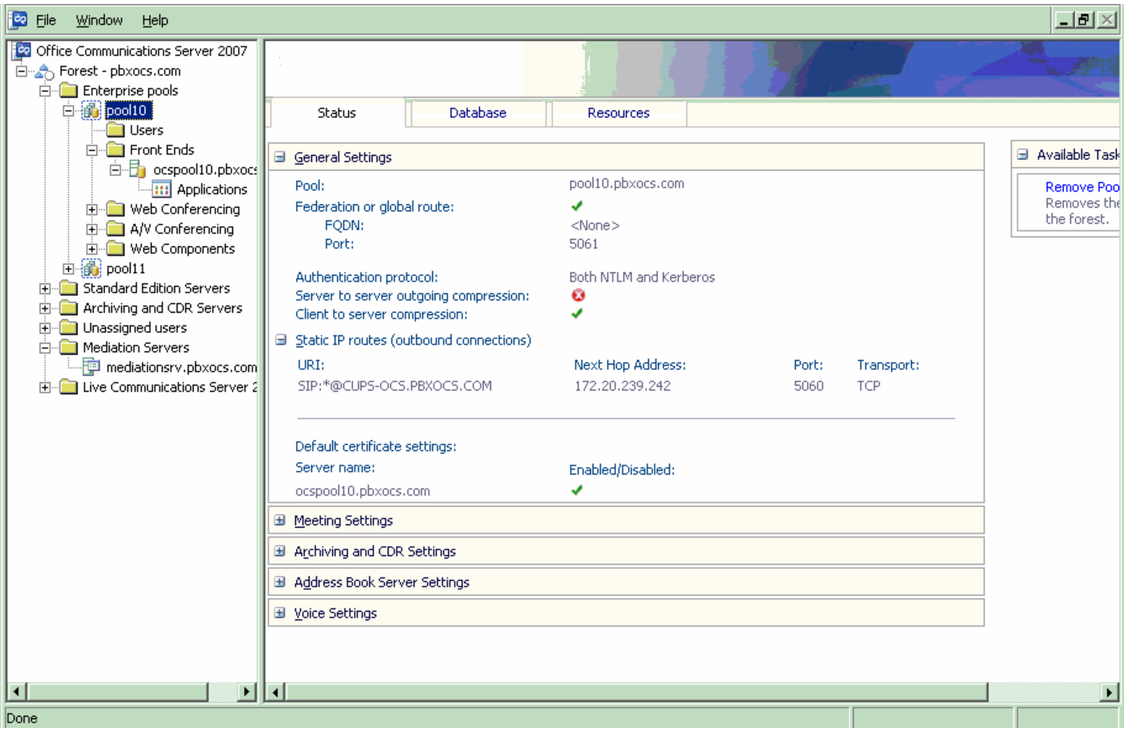
System Configuration

Configuring Microsoft OCS 2007 Version RTM

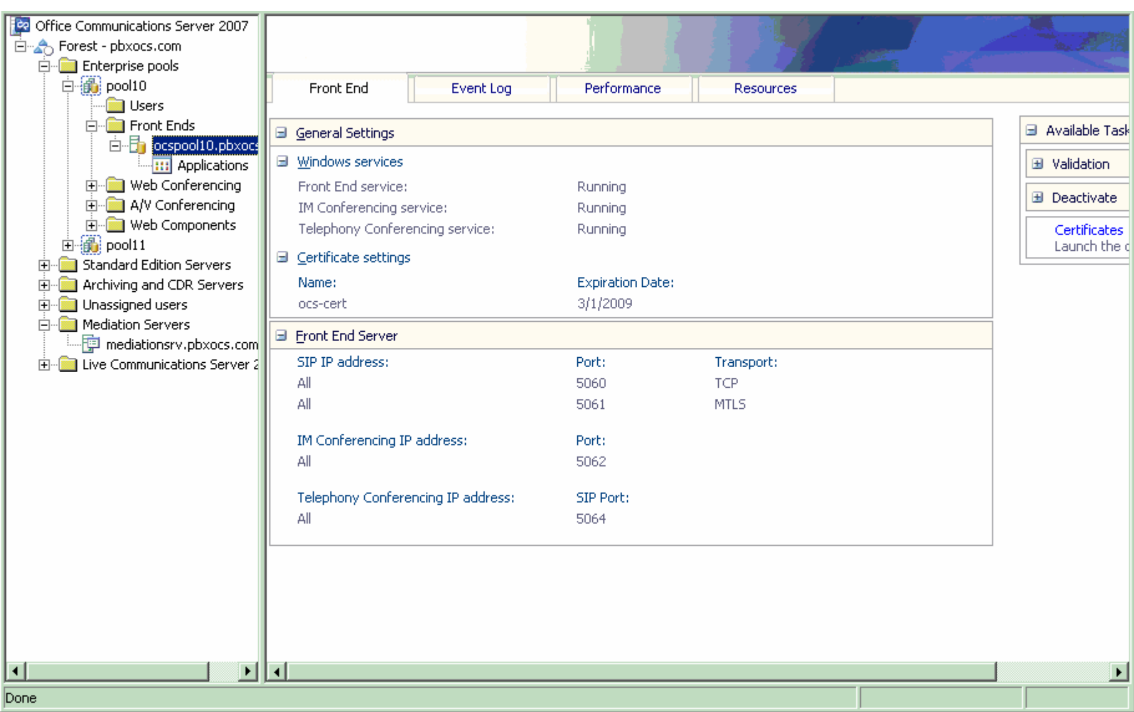
Domain Name Server Configuration



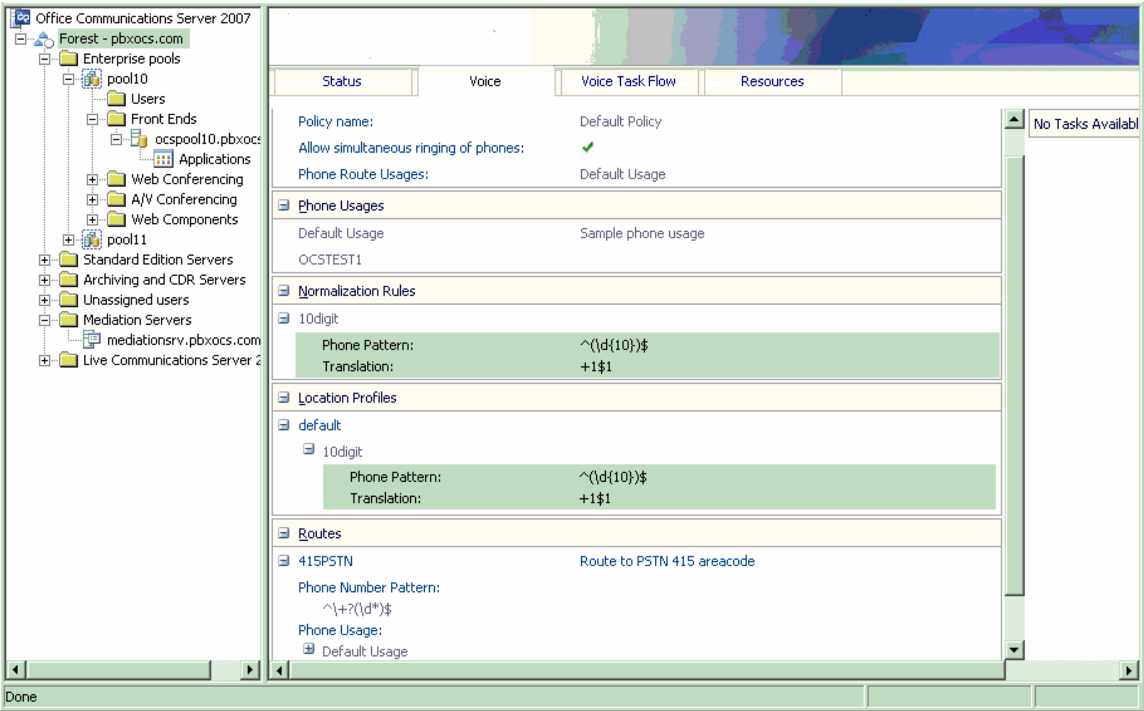
Pool General Settings

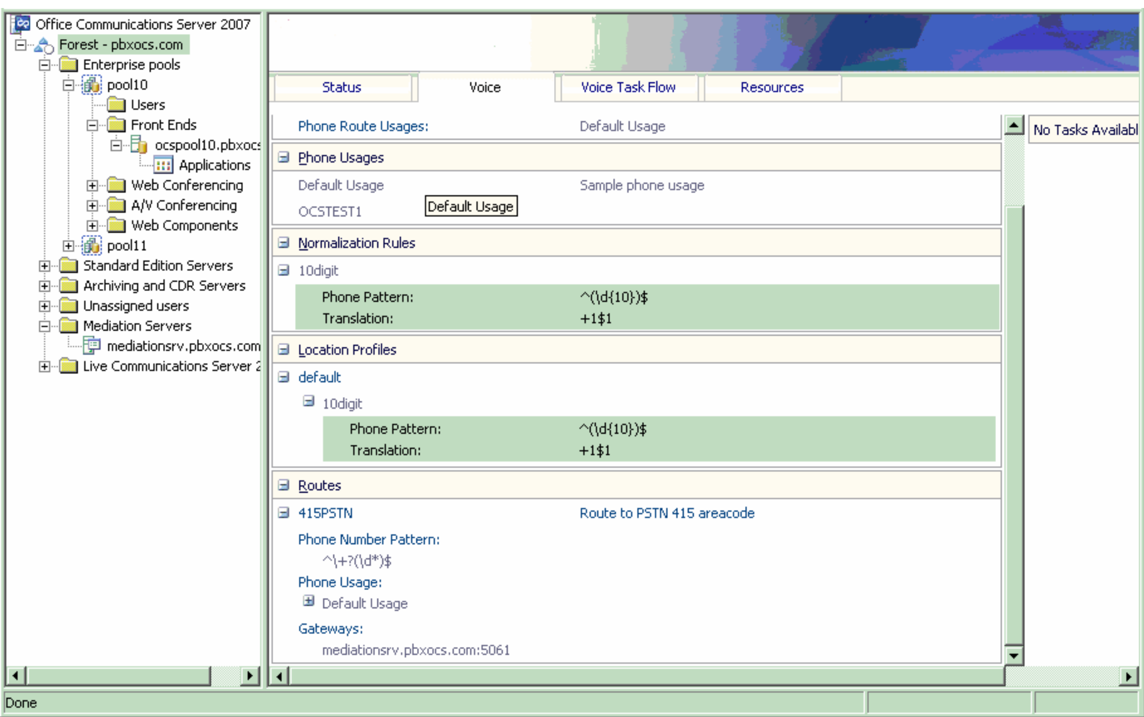


Pool Front-end Configuration

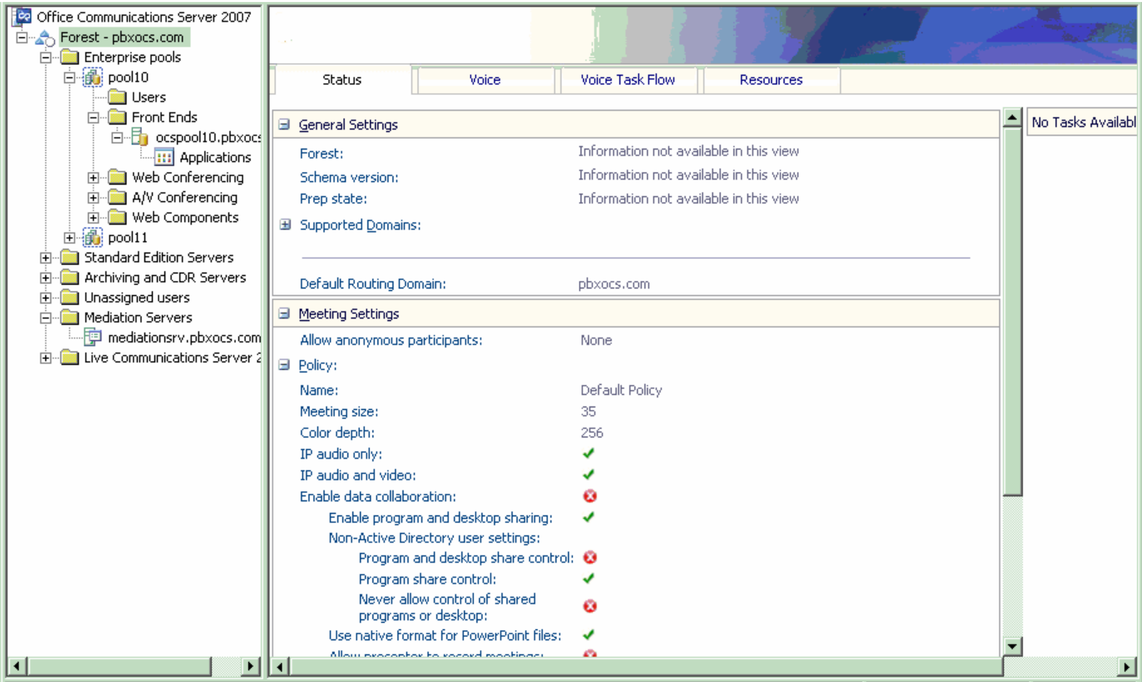


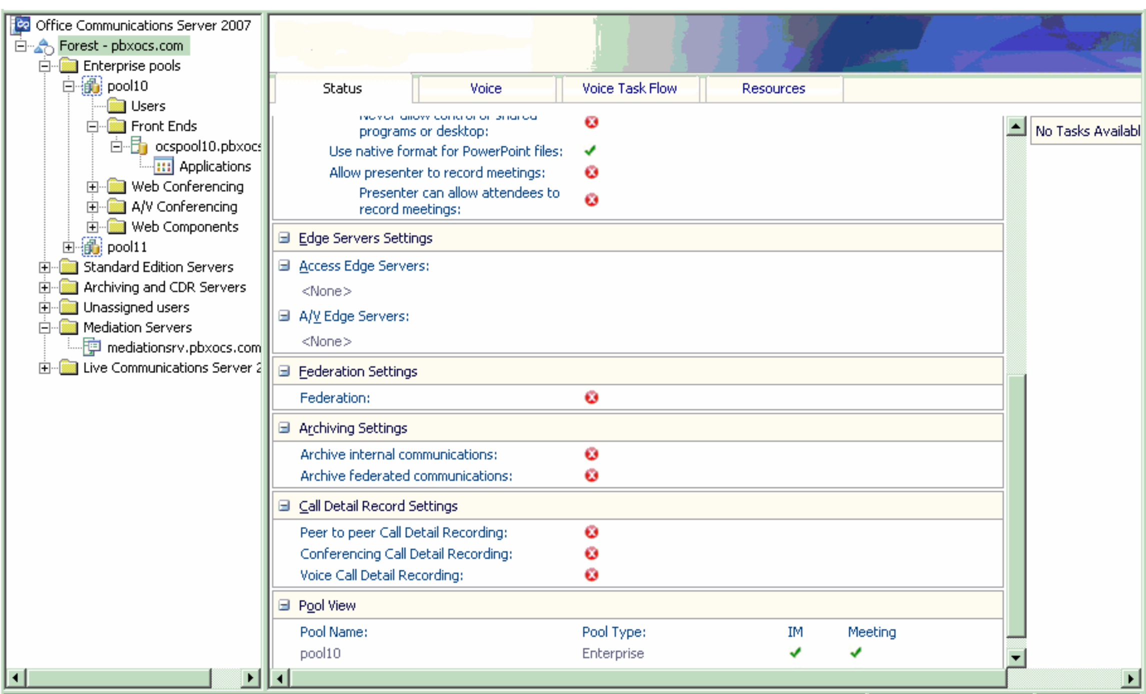
Forest Voice



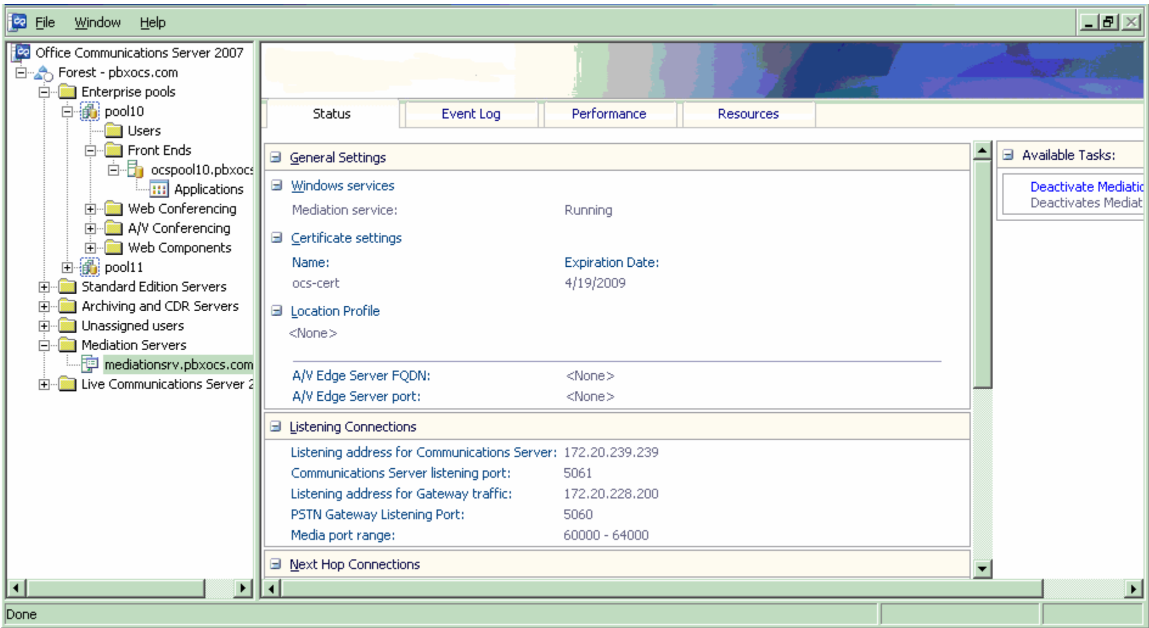


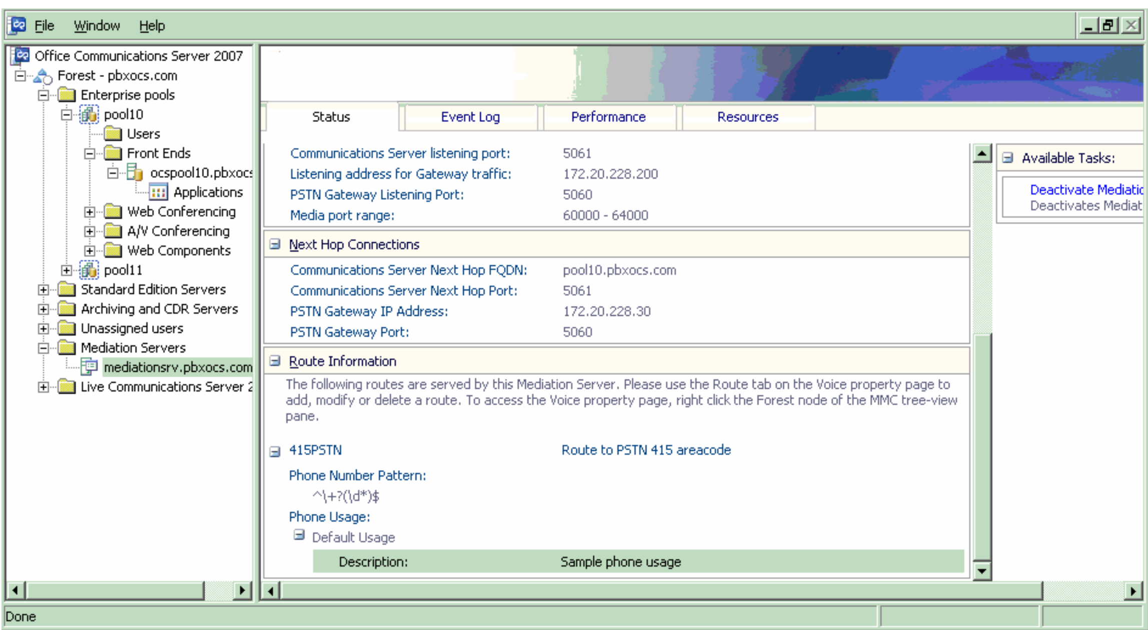
Forest Status



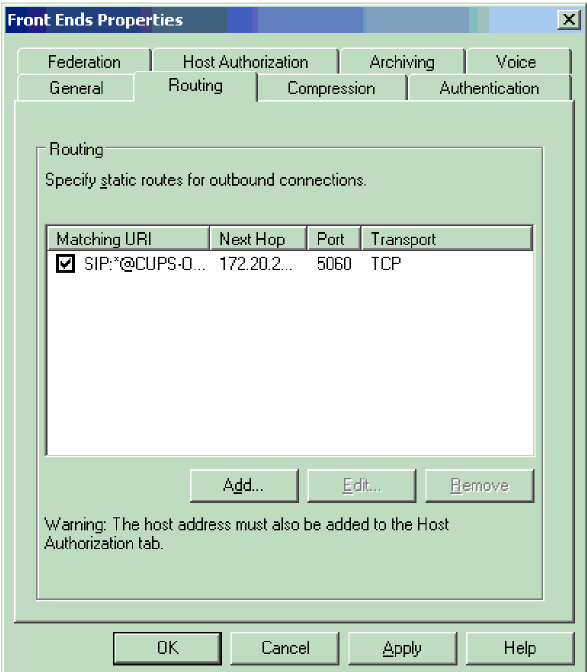


Mediation Sever

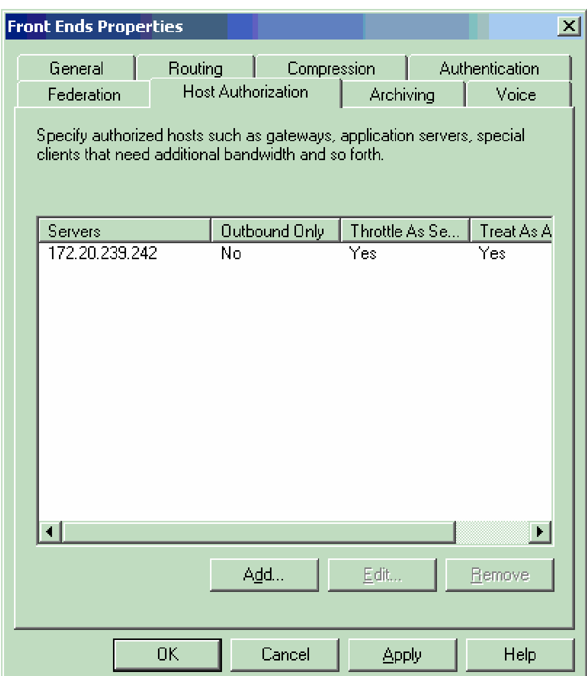




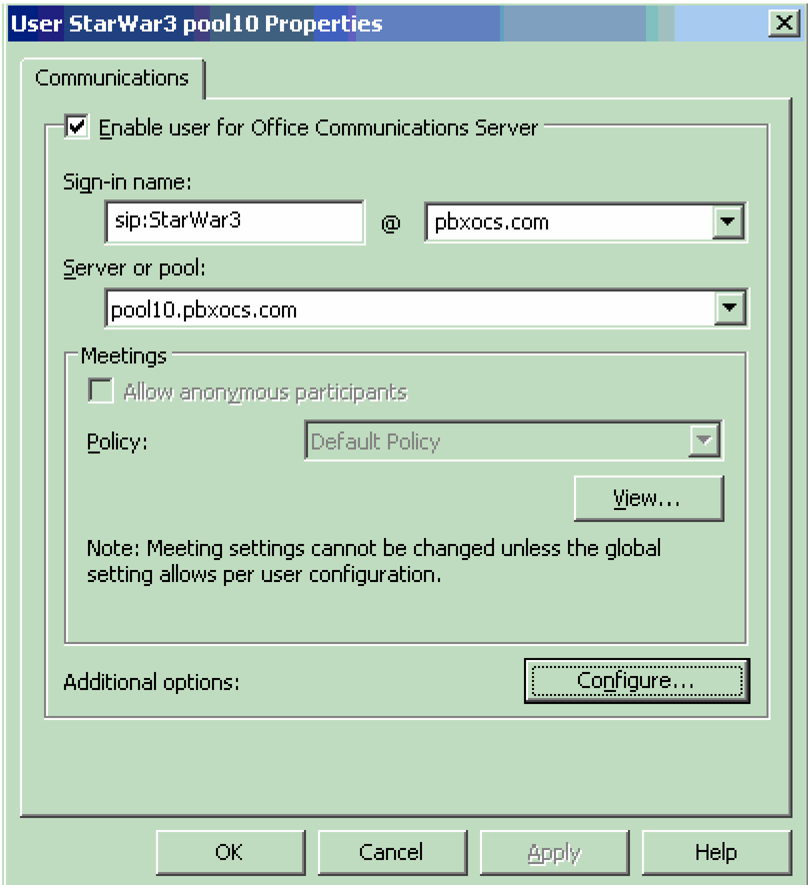
Static Route

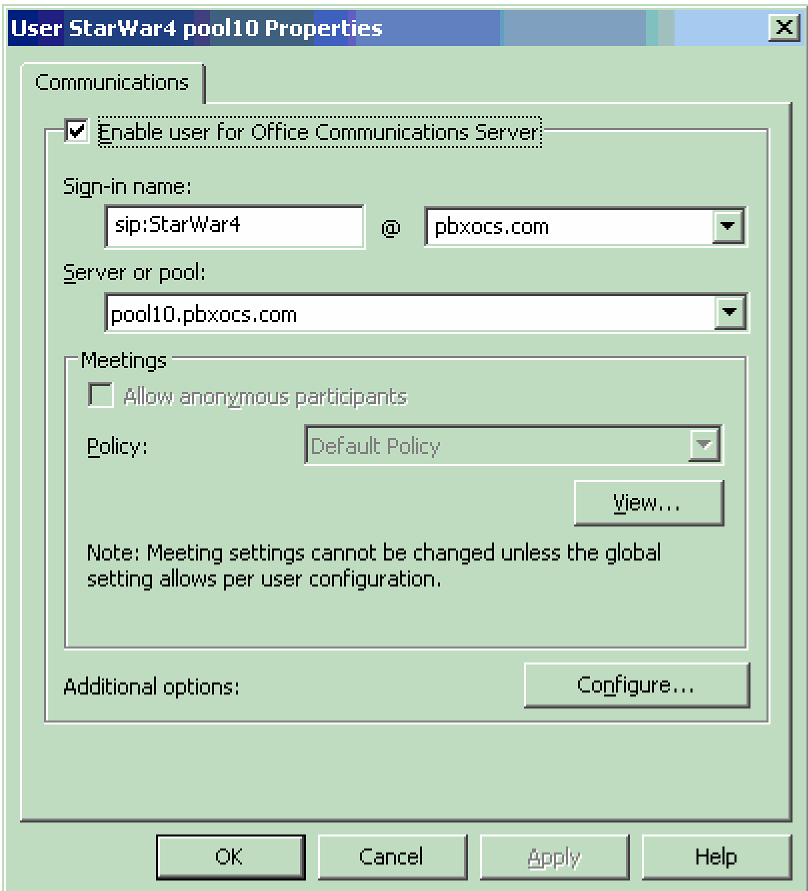


Authorized Host

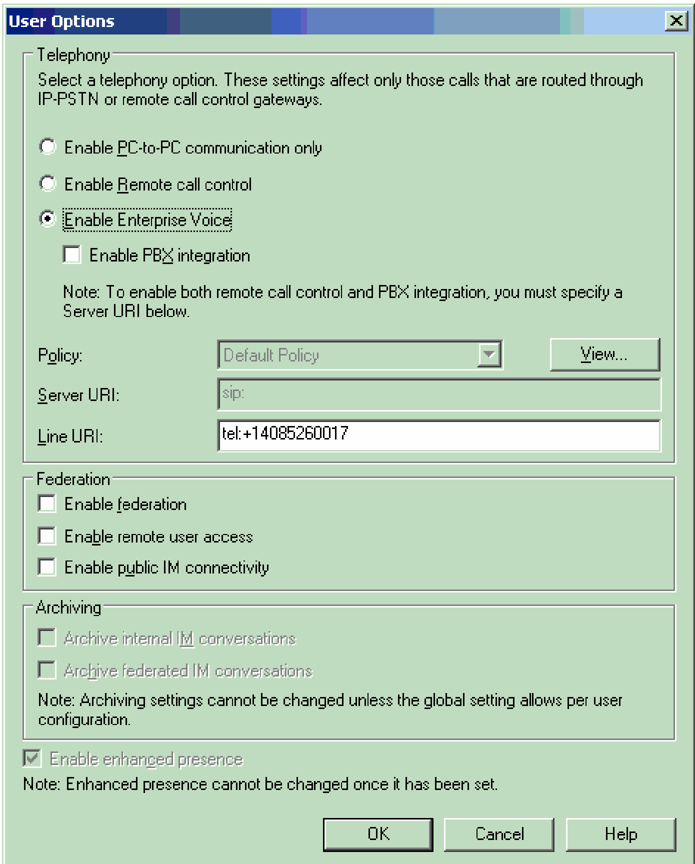


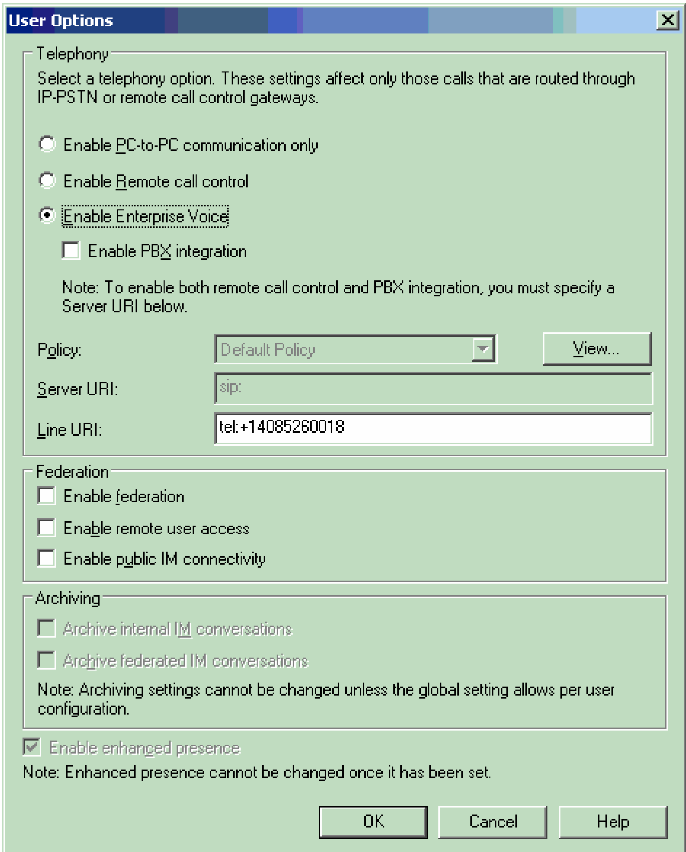
User Configuration




User StarWar4 pool10 Properties
Communications
Enable user for Office Communications Server
Sign-in name:
sip:StarWar4
@
pbxocs.com
Server or pool:
pool10.pbxocs.com
Meetings
Allow anonymous participants
Policy:
Default Policy
View...
Note: Meeting settings cannot be changed unless the global setting allows per user configuration.
Additional options:
Configure...
OK
Cancel
Apply
Help

User option


User Options
Telephony
Select a telephony option. These settings affect only those calls that are routed through IP-PSTN or remote call control gateways.
Enable PC-to-PC communication only
Enable Remote call control
Enable Enterprise Voice
Enable PBX integration
Note: To enable both remote call control and PBX integration, you must specify a Server URI below.
Policy:
Default Policy
View...
Server URI:
sip:
Line URI:
tel:+14085260017
Federation
Enable federation
Enable remote user access
Enable public IM connectivity
Archiving
Archive internal IM conversations
Archive federated IM conversations
Note: Archiving settings cannot be changed unless the global setting allows per user configuration.
Enable enhanced presence
Note: Enhanced presence cannot be changed once it has been set.
OK
Cancel
Help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Default Policy
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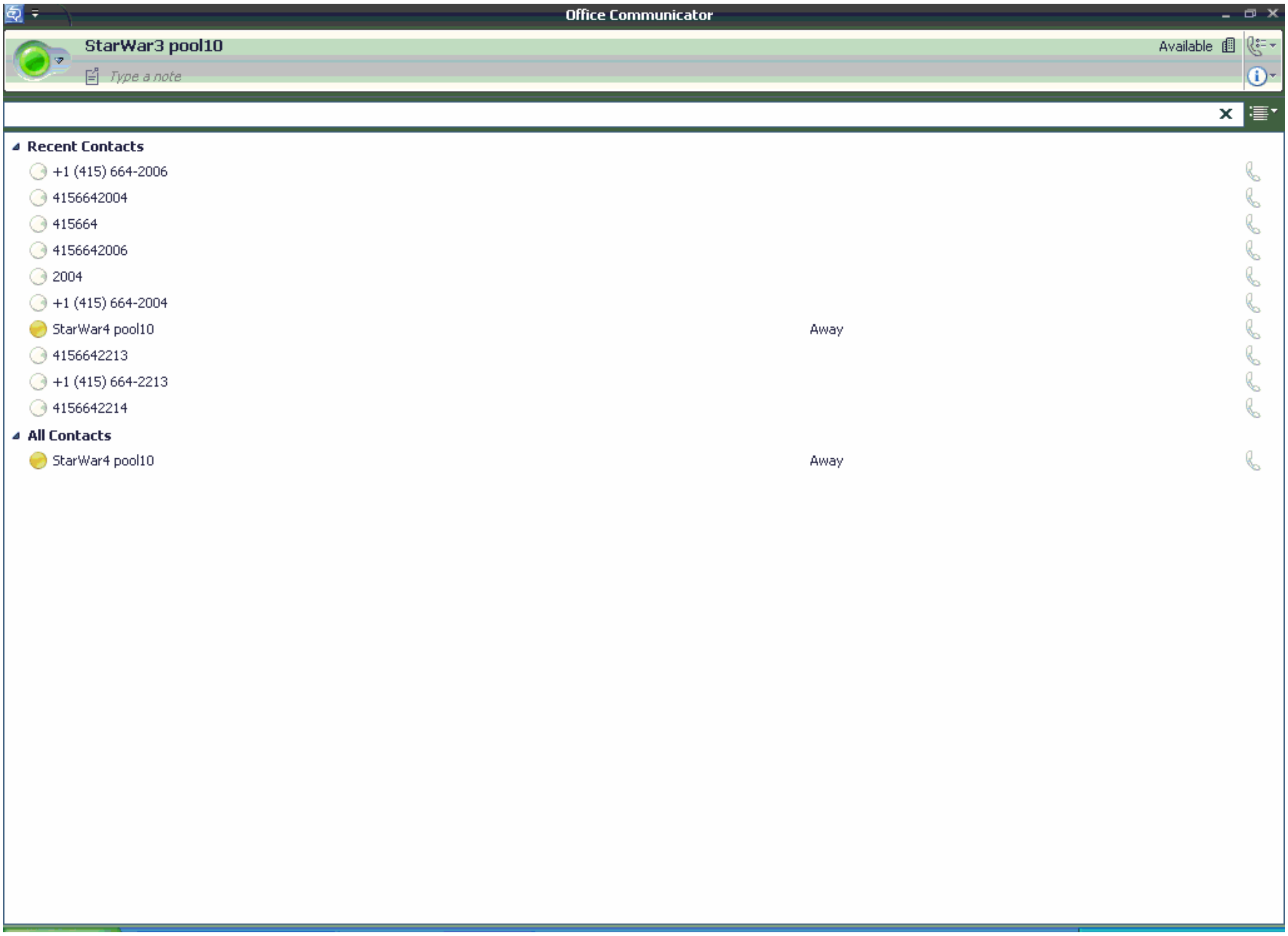
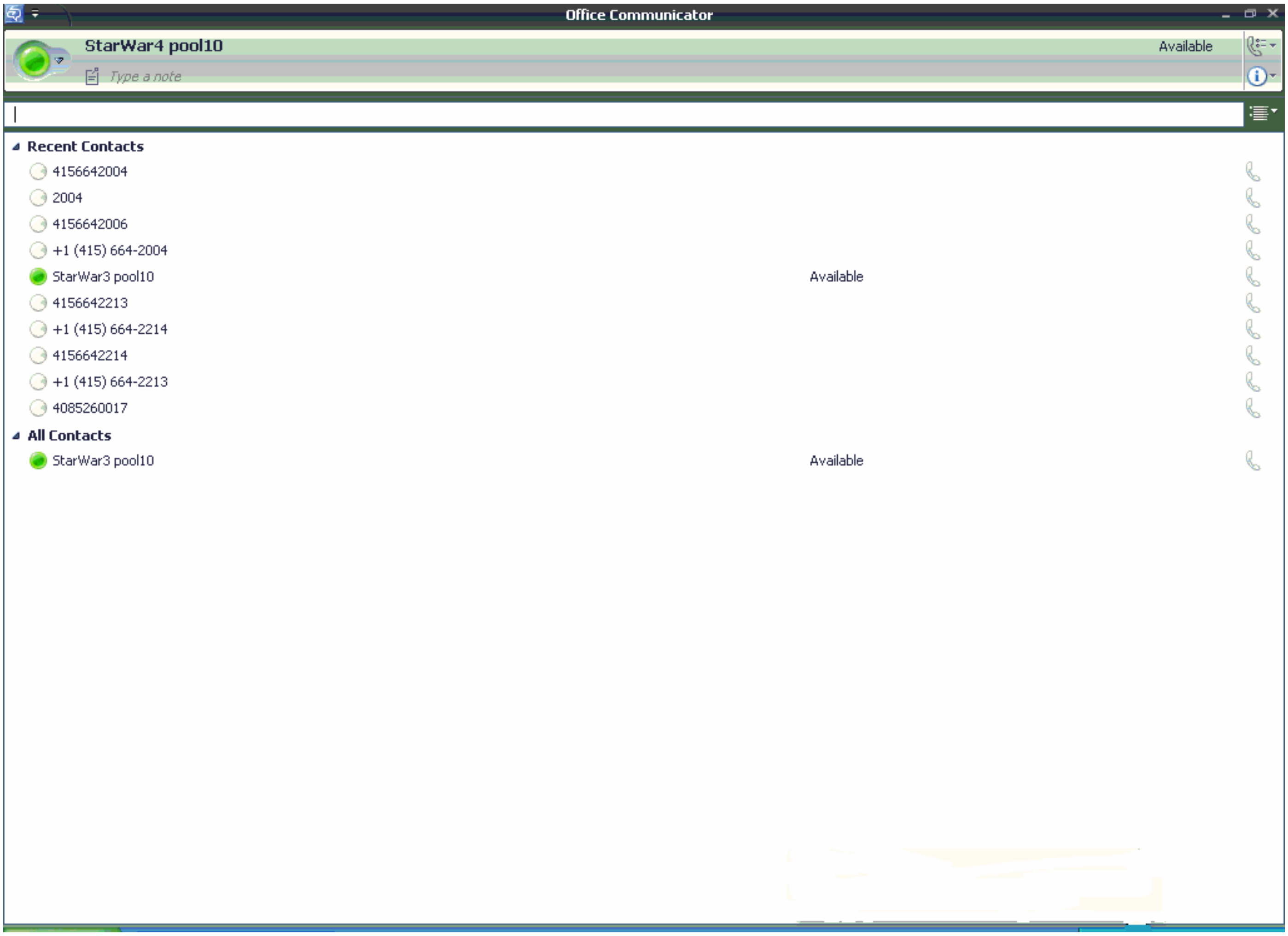
Server URI:
sip:
Line URI:
tel:+14085260018
Federation
Enable federation
Enable remote user access
Enable public IM connectivity
Archiving
Archive internal IM conversations
Archive federated IM conversations
Note: Archiving settings cannot be changed unless the global setting allows per user configuration.
Enable enhanced presence
Note: Enhanced presence cannot be changed once it has been set.
OK
Cancel
Help

Microsoft Office Communicator (MOC) Configuration




Office Communicator
StarWar4 pool10
Available
Type a note
Recent Contacts
4156642004
2004
4156642006
+1 (415) 664-2004
StarWar3 pool10
Available
4156642213
+1 (415) 664-2214
4156642214
+1 (415) 664-2213
4085260017
All Contacts
StarWar3 pool10
Available

Configuring the Cisco 3825

Router#sh ver

Cisco IOS Software, 3800 Software (C3845-IPVOICE-M), Version 12.4(11)T, RELEASE

SOFTWARE (fc2)

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

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Compiled Sat 18-Nov-06 23:46 by prod_rel_team

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

Router uptime is 4 weeks, 6 days, 21 hours, 13 minutes

System returned to ROM by reload at 21:43:20 UTC Tue May 29 2007

System image file is "flash:c3845-ipvoice-mz.124-11.T.bin"

Cisco 3845 (revision 1.0) with 225280K/36864K bytes of memory.

Processor board ID FHK0847F0W7

2 Gigabit Ethernet interfaces

24 Serial interfaces

1 terminal line

2 Channelized T1/PRI ports

1 cisco service engine(s)

DRAM configuration is 64 bits wide with parity enabled.

479K bytes of NVRAM.

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

Configuration register is 0x2102

Router#sh run

```
Router#sh run
Building configuration...

Current configuration : 2236 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
!
boot-start-marker
boot-end-marker
!
```

```
card type e1 0 0
logging buffered 1000000
no logging console
enable password cisco
!
no aaa new-model
network-clock-participate wic 0
network-clock-select 1 E1 0/0/0
ip cef
!
!
!
!
multilink bundle-name authenticated
!
isdn switch-type primary-net5[1]
voice-card 0
 no dspfarm
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
voice translation-rule 1
 rule 1 /5/ /+14085260\1/
 rule 2 // /415664\1/
!
voice translation-rule 2[2]
 rule 1 /^\+14156642/ /2\1/
 rule 2 /^\+14085260/ /5\1/
!
!
voice translation-profile pots
 translate calling 1
 translate called 1
!
voice translation-profile voip
 translate calling 2
 translate called 2
!
!
```

[1] PSTN interface type

[2] The voice gateway manipulates the called and calling digits to match configured dial-peers and to route calls appropriately. For example: Digit manipulation rule 1 of voice translation rule 2 instructs ISR gateway that when it receives +14156642xxx ISR gateway is to strip +14156642,and add digit 2 as leading number to the remaining digits xxx (xxx in this case are either 213 or 214) and send them to the appropriate dial-peer.

```
!
!
!
!
controller E1 0/0/0
 pri-group timeslots 1-10,16
!
controller E1 0/0/1
!
!
!
!
interface GigabitEthernet0/0
 ip address 172.20.192.103 255.255.255.0
 shutdown
 duplex auto
 speed auto
 media-type rj45
 no keepalive
!
interface GigabitEthernet0/1
 ip address 172.20.228.30 255.255.255.0
 duplex auto
 speed auto
 media-type rj45
 no keepalive
!
interface Serial0/0/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5[3]
 isdn protocol-emulate network
 isdn incoming-voice voice
 isdn supp-service name calling
 no cdp enable
!
interface Service-Engine1/0
 no ip address
 shutdown
!
ip route 0.0.0.0 0.0.0.0 172.20.192.1
ip route 172.20.0.0 255.255.0.0 172.20.228.1
!
!
ip http server
!
!
!
!
control-plane
!
!
!
voice-port 0/0/0:15
!
!
```

[3] ISDN E1 interface

```
!
!
!
dial-peer voice 408 voip[4]
 translation-profile incoming voip
 destination-pattern +140852600..
 session protocol sipv2
 session target ipv4:172.20.228.200
 session transport tcp
 incoming called-number +1415664....
 codec g711ulaw
 clid strip pi-restrict
!
dial-peer voice 2200 pots[5]
 translation-profile incoming pots
 destination-pattern 22..
 incoming called-number 5...
 direct-inward-dial
 port 0/0/0:15
 forward-digits all
 clid restrict[6]
!
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line 66
 no activation-character
 no exec
 transport preferred none
 transport input all
 transport output pad telnet rlogin lapb-ta mop udptn v120
line vty 0 4
 password cisco
 login
!
scheduler allocate 20000 1000
!
end
```

[4] Dial-peer toward OCS
[5] Dial-peer toward PSTN
[6] If this command is set, the MOC client caller ID toward PSTN will be restricted. To allow caller ID, remove this command from the dial-peer.

Acronyms

Acronym	Definitions
OCS	Office Communication Server
Cisco IOS	Cisco Internetwork Operating System
SIP	Session Initiation Protocol
RTP	Real-Time Protocol
MOC	Microsoft Office Communicator
MSFT	Microsoft
MS	Mediation Server
SP	Service Pack
ISR	Integrated Services Router

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