



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

October 3, 2007 Revision 4

Table of Contents

Introduction	2
Network Topology.....	3
Limitations.....	4
System Components	4
Hardware requirements	4
Software Requirements	4
Features	4
Features Supported.....	4
Features Not Supported.....	5
System Configuration	6
Configuring Microsoft OCS 2007 Version RTM.....	6
Configuring the Cisco 3825.....	18
Acronyms	22



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

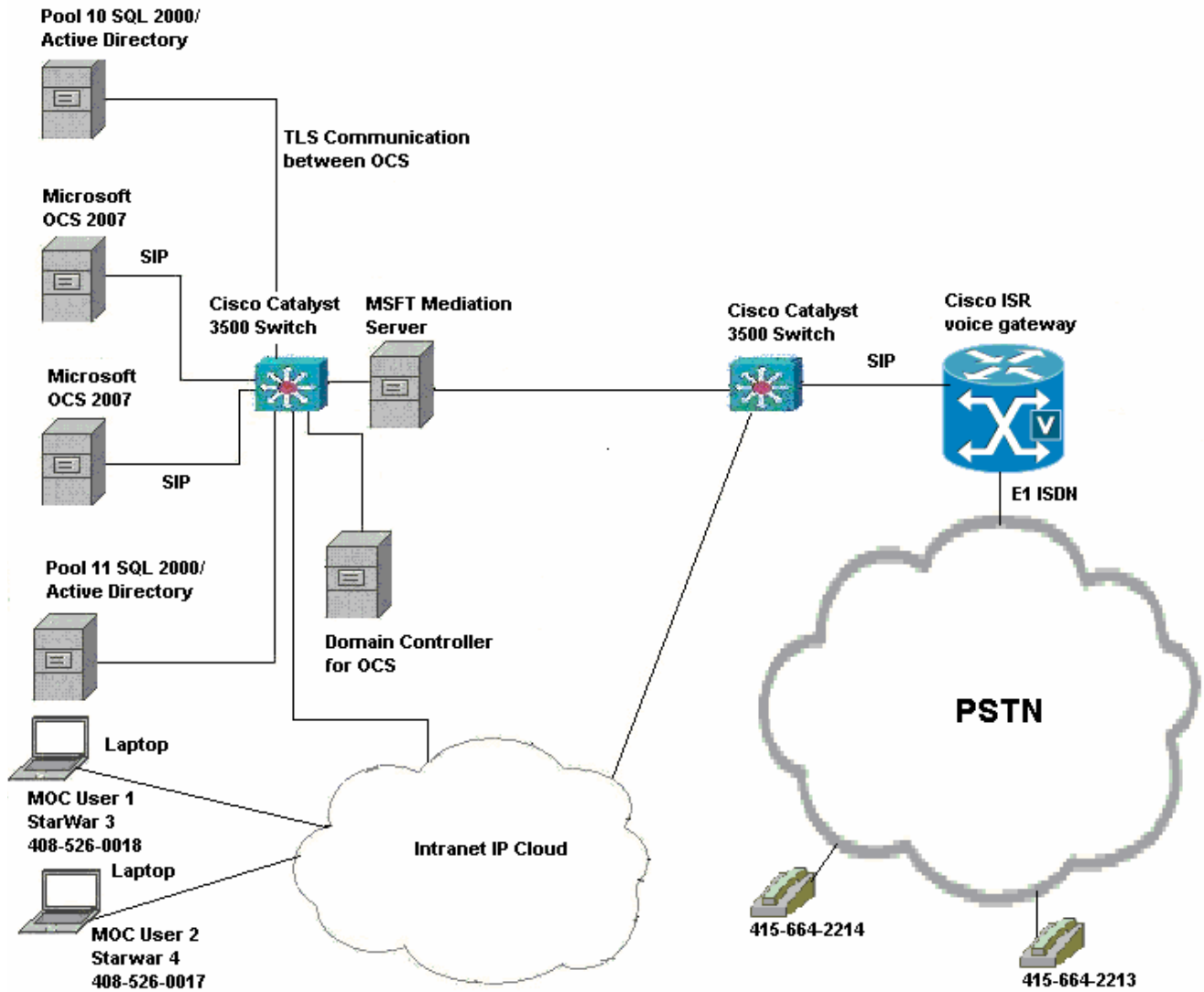


Figure 1. Network Topology



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

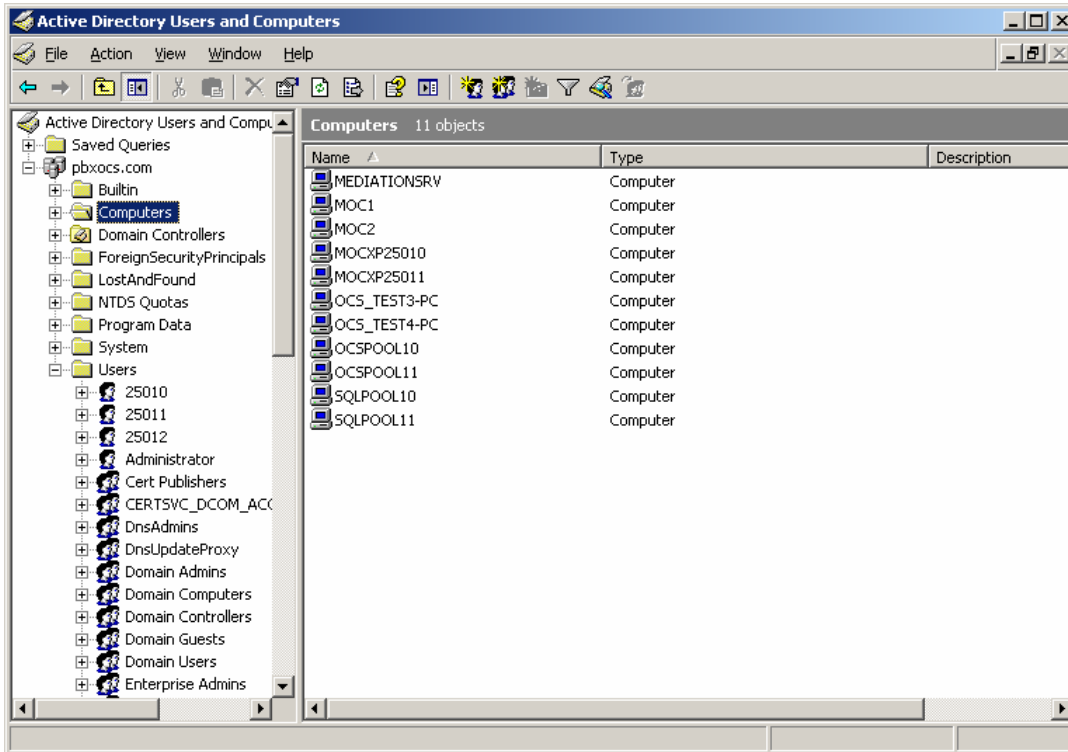
Call Transfer supervised initiated by MOC



System Configuration

Configuring Microsoft OCS 2007 Version RTM

Domain Name Server Configuration





Pool General Settings

The screenshot shows the 'Office Communications Server 2007' interface. The left-hand tree view is expanded to 'Enterprise pools' > 'pool10' > 'Users'. The main pane displays the 'General Settings' for the selected pool. The 'Status' tab is active, showing various configuration parameters.

Parameter	Value	Status
Pool:	pool10.pbxocs.com	
Federation or global route:	<None>	✓
FQDN:	<None>	
Port:	5061	
Authentication protocol:	Both NTLM and Kerberos	
Server to server outgoing compression:		✗
Client to server compression:		✓

Static IP routes (outbound connections)

URI	Next Hop Address	Port	Transport
SIP:*@CUPS-OC5.PBXOCS.COM	172.20.239.242	5060	TCP

Default certificate settings:

Parameter	Value	Status
Server name:	ocs.pool10.pbxocs.com	Enabled/Disabled: ✓

Other settings tabs visible: Meeting Settings, Archiving and CDR Settings, Address Book Server Settings, Voice Settings.

Pool Front-end Configuration

The screenshot shows the 'Office Communications Server 2007' interface. The left-hand tree view is expanded to 'Enterprise pools' > 'pool10' > 'Front Ends' > 'ocs.pool10.pbxocs.com'. The main pane displays the 'Front End' configuration for the selected pool.

Service	Status
Front End service:	Running
IM Conferencing service:	Running
Telephony Conferencing service:	Running

Certificate settings

Parameter	Value
Name:	ocs-cert
Expiration Date:	3/1/2009

Front End Server

Service	IP address	Port	Transport
SIP IP address:	All	5060	TCP
	All	5061	MTLS
IM Conferencing IP address:	All	5062	
Telephony Conferencing IP address:	All	SIP Port: 5064	



Forest Voice

Office Communications Server 2007
Forest - pbxocs.com

- Enterprise pools
 - pool10
 - Users
 - Front Ends
 - ocspool10.pbxocs.com
 - Applications
 - Web Conferencing
 - A/V Conferencing
 - Web Components
 - pool11
 - Standard Edition Servers
 - Archiving and CDR Servers
 - Unassigned users
 - Mediation Servers
 - mediationsrv.pbxocs.com
 - Live Communications Server 2

Status	Voice	Voice Task Flow	Resources
Policy name:	Default Policy		No Tasks Available
Allow simultaneous ringing of phones:	✓		
Phone Route Usages:	Default Usage		
Phone Usages			
Default Usage:	Sample phone usage		
OCSTEST1			
Normalization Rules			
10digit			
Phone Pattern:	^\{d{10}\}\$		
Translation:	+1\$1		
Location Profiles			
default			
10digit			
Phone Pattern:	^\{d{10}\}\$		
Translation:	+1\$1		
Routes			
415PSTN Route to PSTN 415 areacode			
Phone Number Pattern:	^\{+?(d*)\}\$		
Phone Usage:	Default Usage		

Office Communications Server 2007
Forest - pbxocs.com

- Enterprise pools
 - pool10
 - Users
 - Front Ends
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Location Profiles			
default			
10digit			
Phone Pattern:	^\{d{10}\}\$		
Translation:	+1\$1		
Routes			
415PSTN Route to PSTN 415 areacode			
Phone Number Pattern:	^\{+?(d*)\}\$		
Phone Usage:	Default Usage		
Gateways:			
mediationsrv.pbxocs.com:5061			



Forest Status

Office Communications Server 2007

Forest - pbxocs.com

- Enterprise pools
 - pool10
 - Users
 - Front Ends
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 - Applications
 - Web Conferencing
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 - Standard Edition Servers
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 - Mediation Servers
 - mediationsrv.pbxocs.com
 - Live Communications Server 2

Tabbed interface: Status | Voice | Voice Task Flow | Resources

General Settings

- Forest: Information not available in this view
- Schema version: Information not available in this view
- Prep state: Information not available in this view
- Supported Domains:
- Default Routing Domain: pbxocs.com

Meeting Settings

- Allow anonymous participants: None
- Policy:
 - Name: Default Policy
 - Meeting size: 35
 - Color depth: 256
 - IP audio only: ✓
 - IP audio and video: ✓
 - Enable data collaboration: ✗
 - Enable program and desktop sharing: ✓
 - Non-Active Directory user settings:
 - Program and desktop share control: ✗
 - Program share control: ✓
 - Never allow control of shared programs or desktop: ✗
 - Use native format for PowerPoint files: ✓
 - Allow presenter to record meetings: ✗

Office Communications Server 2007

Forest - pbxocs.com

- Enterprise pools
 - pool10
 - Users
 - Front Ends
 - ocspool10.pbxocs.com
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 - Standard Edition Servers
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 - mediationsrv.pbxocs.com
 - Live Communications Server 2

Tabbed interface: Status | Voice | Voice Task Flow | Resources

Meeting Settings

- Enable program and desktop sharing: ✓
- Non-Active Directory user settings:
 - Program and desktop share control: ✗
 - Program share control: ✓
 - Never allow control of shared programs or desktop: ✗
- Use native format for PowerPoint files: ✓
- Allow presenter to record meetings: ✗
- Presenter can allow attendees to record meetings: ✗

Edge Servers Settings

- Access Edge Servers: <None>
- A/V Edge Servers: <None>

Federation Settings

- Federation: ✗

Archiving Settings

- Archive internal communications: ✗
- Archive federated communications: ✗

Call Detail Record Settings

- Peer to peer Call Detail Recording: ✗
- Conferencing Call Detail Recording: ✗
- Voice Call Detail Recording: ✗

Pool View

Pool Name:	Pool Type:	IM	Meeting
pool10	Enterprise	✓	✓



Mediation Sever

The screenshot shows the Cisco Mediation Server configuration console. The left pane displays a tree view of the configuration hierarchy, with 'mediationsrv.pbxcos.com' selected. The main pane shows the 'General Settings' tab, which includes the following sections:

- Windows services:** Mediation service: Running
- Certificate settings:** Name: ocs-cert, Expiration Date: 4/19/2009
- Location Profile:** <None>
- Listening Connections:**
 - Listening address for Communications Server: 172.20.239.239
 - Communications Server listening port: 5061
 - Listening address for Gateway traffic: 172.20.228.200
 - PSTN Gateway Listening Port: 5060
 - Media port range: 60000 - 64000
- Next Hop Connections:** (Empty)

An 'Available Tasks' pane on the right contains a button labeled 'Deactivate Mediation Service'.

The screenshot shows the Cisco Mediation Server configuration console with the 'Next Hop Connections' and 'Route Information' sections expanded. The 'Next Hop Connections' section contains the following data:

Communications Server Next Hop FQDN:	pool10.pbxcos.com
Communications Server Next Hop Port:	5061
PSTN Gateway IP Address:	172.20.228.30
PSTN Gateway Port:	5060

The 'Route Information' section displays a route for PSTN 415 area code:

- 415PSTN:** Route to PSTN 415 areacode
- Phone Number Pattern:** ^\+?(\d*)\$
- Phone Usage:** Default Usage
- Description:** Sample phone usage

The 'Available Tasks' pane on the right remains visible with the 'Deactivate Mediation Service' button.



Static Route

Front Ends Properties

Federation | Host Authorization | Archiving | Voice
General | Routing | Compression | Authentication

Routing

Specify static routes for outbound connections.

Matching URI	Next Hop	Port	Transport
<input checked="" type="checkbox"/> SIP:*@CUPS-0...	172.20.2...	5060	TCP

Add... Edit... Remove

Warning: The host address must also be added to the Host Authorization tab.

OK Cancel Apply Help

Authorized Host

Front Ends Properties

General | Routing | Compression | Authentication
Federation | Host Authorization | Archiving | Voice

Specify authorized hosts such as gateways, application servers, special clients that need additional bandwidth and so forth.

Servers	Outbound Only	Throttle As Se...	Treat As A
172.20.239.242	No	Yes	Yes

Add... Edit... Remove

OK Cancel Apply Help



User Configuration

User StarWar3 pool10 Properties [X]

Communications

Enable user for Office Communications Server

Sign-in name:
sip:StarWar3 @ pbxocs.com

Server or pool:
pool10.pbxocs.com

Meetings

Allow anonymous participants

Policy: Default Policy [v]
[View...]

Note: Meeting settings cannot be changed unless the global setting allows per user configuration.

Additional options: [Configure...]

OK Cancel Apply Help



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 [X]

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:

Server URI:

Line URI:

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.



User Options [X]

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 Enable federation
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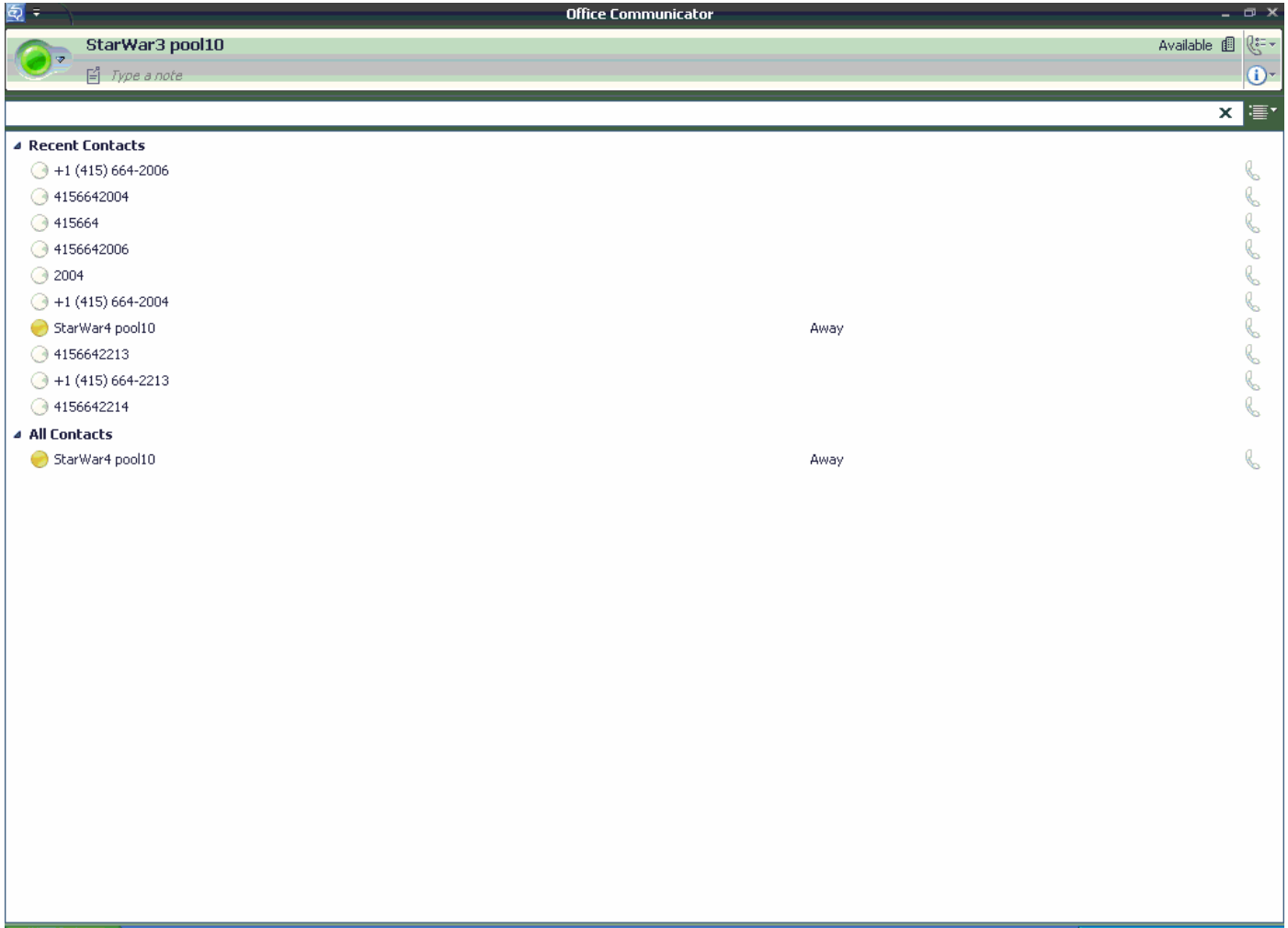
Archiving
 Archive internal IM conversations
 Archive federated IM conversations

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.



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>

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

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-net51
voice-card 0
no dspfarm
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
voice translation-rule 1
rule 1 /5/ /+14085260\1/
rule 2 // /415664\1/
!
voice translation-rule 22
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-net53  
  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 voip4  
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 pots5  
translation-profile incoming pots  
destination-pattern 22..  
incoming called-number 5...  
direct-inward-dial  
port 0/0/0:15  
forward-digits all  
clid restrict6  
!  
!  
!  
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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