



# Microsoft Office Communication Server 2007 version RTM to Cisco IOS Voice Gateway using SIP with T1 ISDN

October 9, 2007 Revision 5

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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 connectivity.

The SIP Protocol is used between Cisco ISR Voice gateway and MSFT Mediation Server. The connection between Cisco ISR gateway and PSTN uses T1-PRI with switch-type DMS-100 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 PRI ISDN.

This Application Notes 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

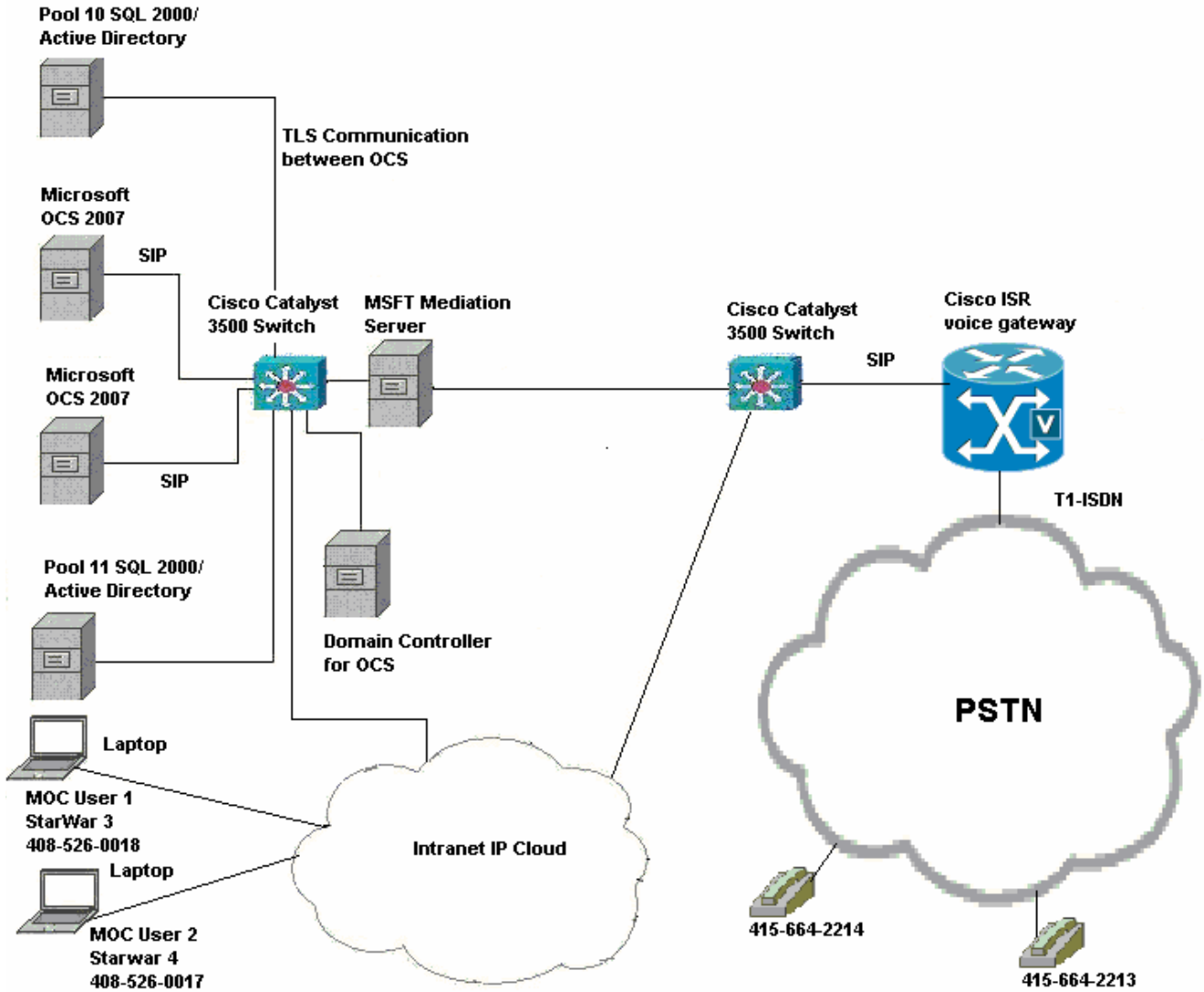


Figure 1. Network Topology



## 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

- Cisco 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

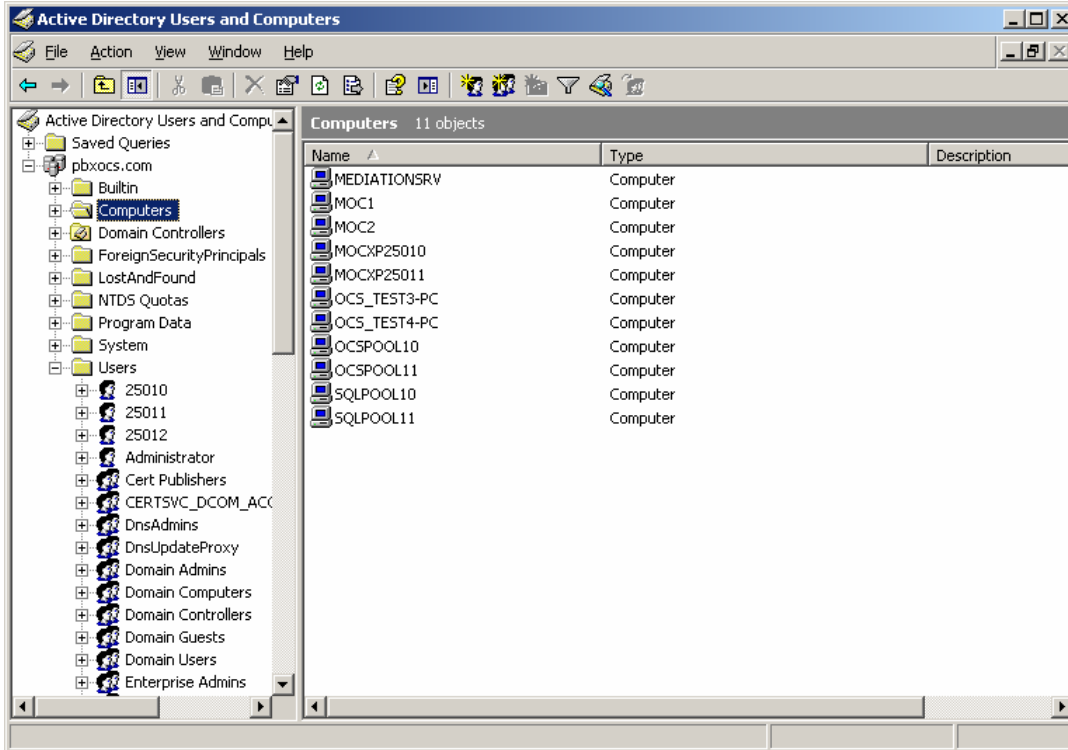
Call Transfer supervised initiated by MOC



## Configuration

### Configuring Microsoft OCS 2007 Version RTM

#### Domain Name Server Configuration





## Pool General Settings

The screenshot shows the 'Office Communications Server 2007' console. The left-hand tree view is expanded to 'Enterprise pools' > 'pool10'. The main pane displays the 'General Settings' tab for 'pool10.pbxocs.com'. The status bar at the top indicates 'Status', 'Database', and 'Resources'.

Static IP routes (outbound connections)			
URI	Next Hop Address	Port	Transport
SIP:*@CUPS-OCS.PBXOCS.COM	172.20.239.242	5060	TCP

Default certificate settings:

Server name:	Enabled/Disabled:
ocspool10.pbxocs.com	✓

Available Task: Remove Pool (Removes the forest...)

## Pool Front-end configuration

The screenshot shows the 'Office Communications Server 2007' console. The left-hand tree view is expanded to 'Enterprise pools' > 'pool10' > 'Front Ends' > 'ocspool10.pbxocs.com'. The main pane displays the 'Front End' tab for 'ocspool10.pbxocs.com'. The status bar at the top indicates 'Front End', 'Event Log', 'Performance', and 'Resources'.

Certificate settings	
Name:	Expiration Date:
ocs-cert	3/1/2009

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

Available Task: Validation, Deactivate, Certificates (Launch the c...)



Forest Voice

This screenshot shows the configuration for a 10digit route profile. The left pane displays a tree view of the system configuration, including Enterprise pools, Front Ends, Applications, and various servers. The main pane shows the configuration for the 10digit profile under Normalization Rules and Location Profiles.

Section	Item	Value
Policy	Policy name:	Default Policy
Policy	Allow simultaneous ringing of phones:	✓
Policy	Phone Route Usages:	Default Usage
Phone Usages	Default Usage	Sample phone usage
Phone Usages	OCSTEST1	
Normalization Rules	10digit	
Normalization Rules	Phone Pattern:	^\(d{10})\$
Normalization Rules	Translation:	+1\$1
Location Profiles	default	
Location Profiles	10digit	
Location Profiles	Phone Pattern:	^\(d{10})\$
Location Profiles	Translation:	+1\$1
Routes	415PSTN	Route to PSTN 415 areacode
Routes	Phone Number Pattern:	^\+?(d*)\$
Routes	Phone Usage:	Default Usage

This screenshot shows the configuration for a 415PSTN route profile. The left pane displays the same tree view as the first screenshot. The main pane shows the configuration for the 415PSTN profile under Routes.

Section	Item	Value
Phone Route Usages	Default Usage	Sample phone usage
Phone Usages	OCSTEST1	Default Usage
Normalization Rules	10digit	
Normalization Rules	Phone Pattern:	^\(d{10})\$
Normalization Rules	Translation:	+1\$1
Location Profiles	default	
Location Profiles	10digit	
Location Profiles	Phone Pattern:	^\(d{10})\$
Location Profiles	Translation:	+1\$1
Routes	415PSTN	Route to PSTN 415 areacode
Routes	Phone Number Pattern:	^\+?(d*)\$
Routes	Phone Usage:	Default Usage
Routes	Gateways:	mediationsrv.pbxocs.com:5061





## Forest Status

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 2005

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: ✗

No Tasks Available

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 2005

Status | Voice | Voice Task Flow | Resources

Enable program and desktop sharing: ✗  
 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	✓	✓

No Tasks Available



## Mediation Sever

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

- Status:** Running
- Windows services:** Mediation service: Running
- Certificate settings:**
  - Name: ocs-cert
  - Expiration Date: 4/19/2009
- Location Profile:** <None>
- A/V Edge Server FQDN:** <None>
- A/V Edge Server port:** <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 shows a button for 'Deactivate Mediation Service'.

The screenshot shows the Cisco Mediation Server configuration console with the 'Next Hop Connections' and 'Route Information' tabs selected. The configuration details are as follows:

- Next Hop Connections:**
  - 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
- Route Information:**

The following routes are served by this Mediation Server. Please use the Route tab on the Voice property page to add, modify or delete a route. To access the Voice property page, right click the Forest node of the MMC tree-view pane.

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

An 'Available Tasks' pane on the right shows a button for 'Deactivate Mediation Service'.



### 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

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

[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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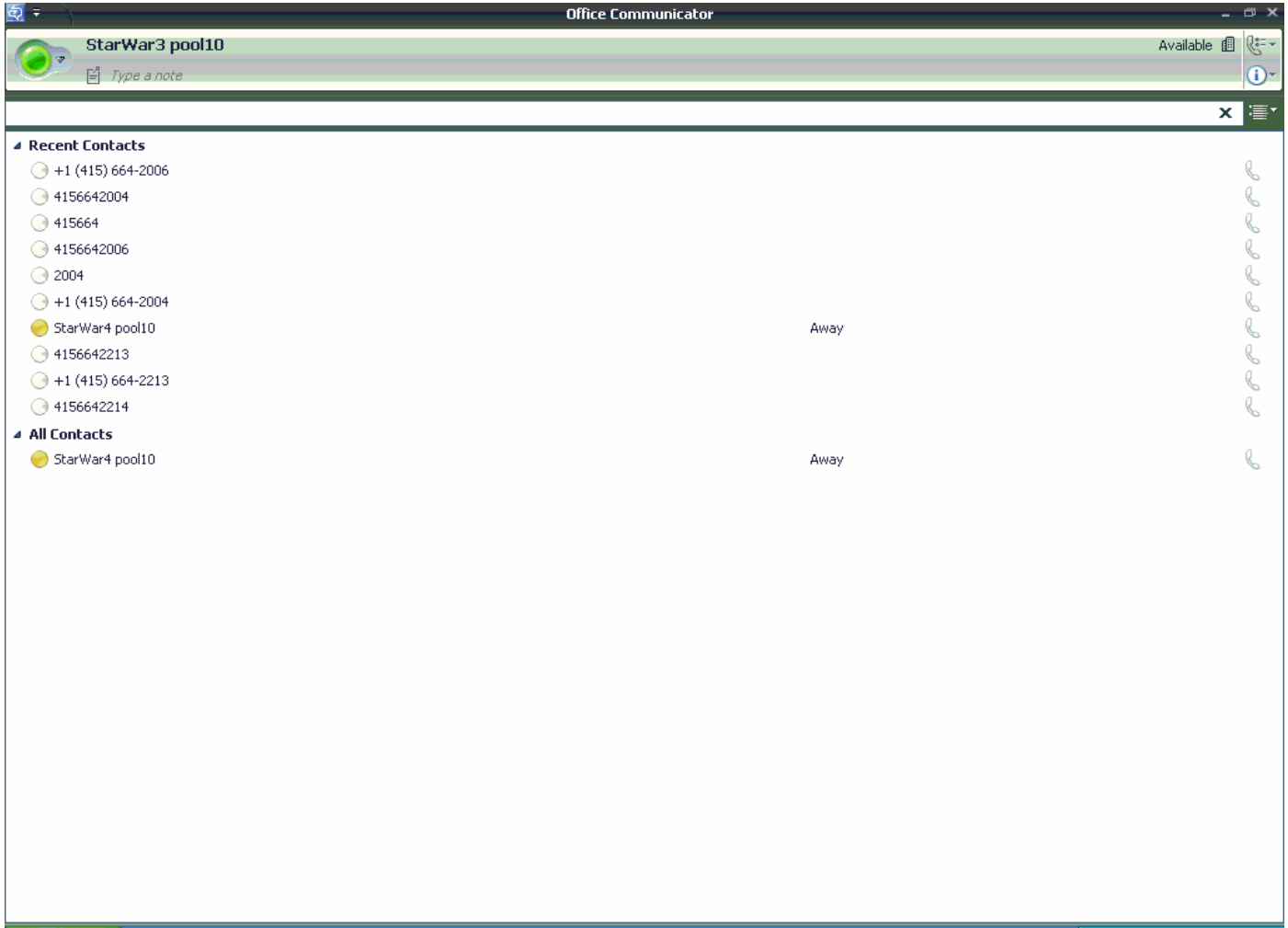
**Archiving**  
 Archive internal IM conversations  
 Archive federated IM conversations

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**

Building configuration...

Current configuration : 2327 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 t1 0 0
logging buffered 1000000
no logging console
enable password cisco
!
no aaa new-model
network-clock-participate wic 0
ip cef
!
!
multilink bundle-name authenticated
!
isdn switch-type primary-dms1001
isdn gateway-max-interworking
voice-card 0
no dspfarm
!
!
voice translation-rule 1
rule 1 /5/ /+14085260\1/
rule 2 /314771/ /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
!
!
controller T1 0/0/0
framing esf
linecode b8zs
pri-group timeslots 1-24
!
controller T1 0/0/1
framing esf
linecode b8zs
!
interface GigabitEthernet0/0
ip address 172.20.192.103 255.255.255.0
shutdown
duplex auto
speed auto
```

---

<sup>1</sup> PSTN interface type.

<sup>2</sup> 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 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.



```
media-type rj45
no keepalive
!
interface GigabitEthernet0/1
Description Interface to Mediation server
ip address 172.20.228.30 255.255.255.0
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 supp-service name calling ie 40 cs 0
isdn channel-id invert extend-bit
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 0/0/0:23
!
!
dial-peer voice 408 voip 3
translation-profile incoming voip
destination-pattern 40852600..
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 4
translation-profile incoming pots
destination-pattern 22..
incoming called-number 5...
direct-inward-dial
port 0/0/0:23
```

---

<sup>3</sup> Sip Signaling Dial-peer

<sup>4</sup> PSTN Dial-peer



```
forward-digits all
clid restrict5
!
!
!
line con 0
stopbits 1
line aux 0
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
```

---

<sup>5</sup> 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

<b>Acronym</b>	<b>Definitions</b>
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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