



# 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

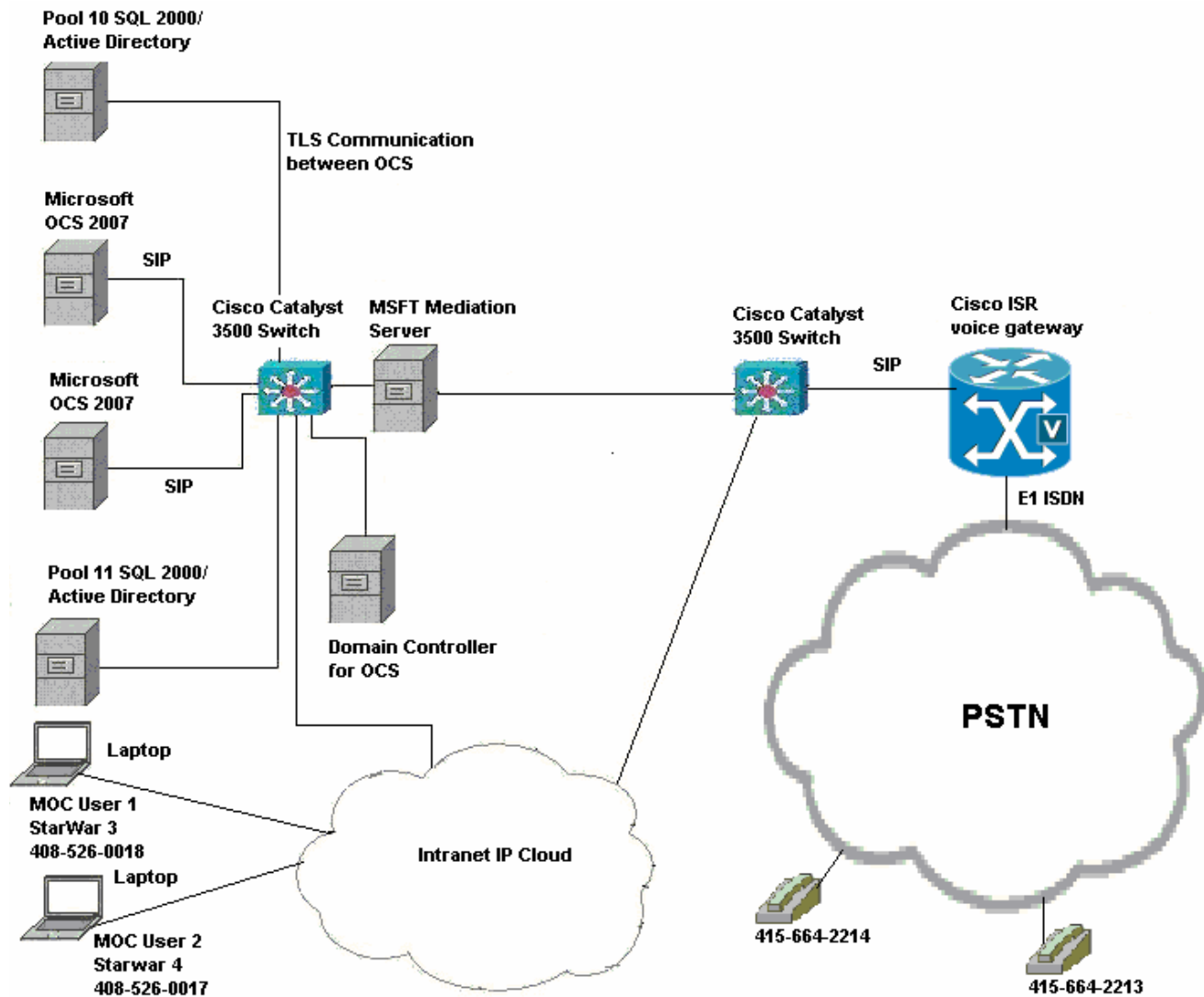


Figure 1. Network Topology



## 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

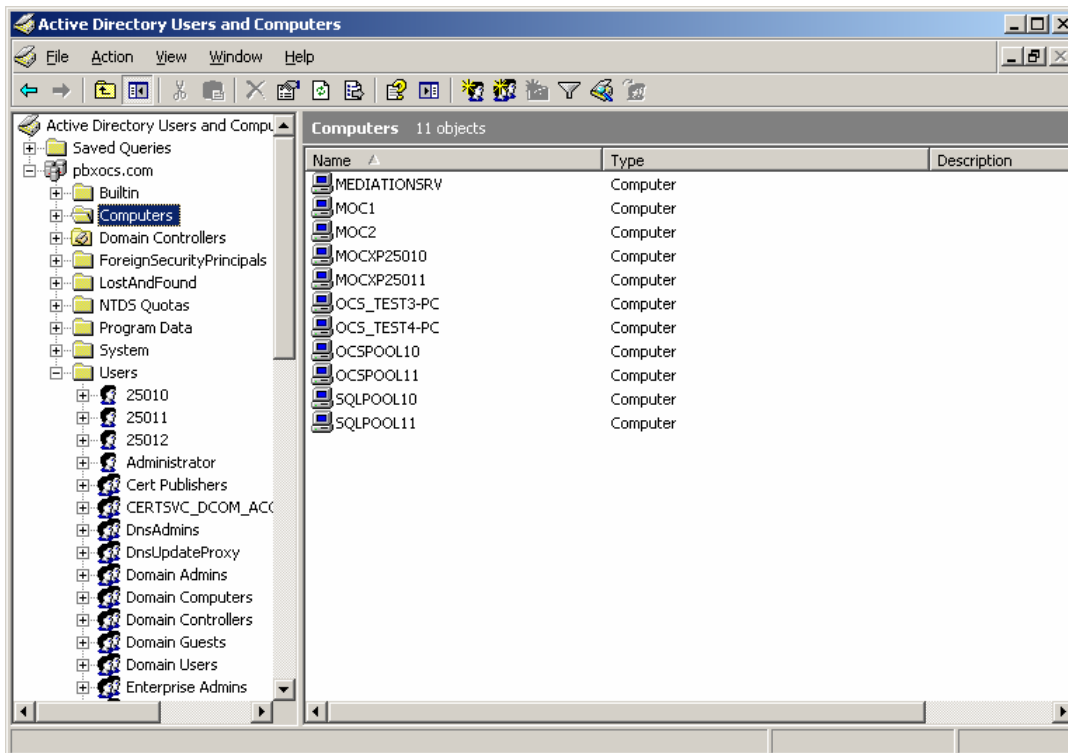
Call Transfer supervised initiated by MOC



## System Configuration

### Configuring Microsoft OCS 2007 Version RTM

#### Domain Name Server Configuration





## Pool General Settings

The screenshot shows the Office Communications Server 2007 configuration console. The left pane displays a tree view of the configuration hierarchy, with 'pool10' selected under 'Enterprise pools'. The right pane shows the 'General Settings' tab for 'pool10.pbxocs.com'. The settings are as follows:

Setting	Value
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:	SIP:*@CUPS-OC5.PBXOCS.COM
Next Hop Address:	172.20.239.242
Port:	5060
Transport:	TCP
Default certificate settings:	
Server name:	ocspool10.pbxocs.com
Enabled/Disabled:	✓

Below the General Settings, there are expandable sections for Meeting Settings, Archiving and CDR Settings, Address Book Server Settings, and Voice Settings. An 'Available Task' pane on the right shows a 'Remove Pool' task that removes the forest.

## Pool Front-end Configuration

The screenshot shows the Office Communications Server 2007 configuration console with 'pool10' selected. The right pane shows the 'Front End' tab for 'ocspool10.pbxocs.com'. The settings are as follows:

Setting	Value
Front End service:	Running
IM Conferencing service:	Running
Telephony Conferencing service:	Running

Below the Front End services, there are sections for Certificate settings and Front End Server configuration.

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

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

An 'Available Task' pane on the right shows tasks for Validation, Deactivate, and Certificates.



## Forest Voice

Office Communications Server 2007

Forest - pbxocs.com

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

Status Voice Voice Task Flow Resources

Policy name: Default Policy

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:  $^(\+7(\d{10})\$$

Phone Usage: Default Usage

Done

Office Communications Server 2007

Forest - pbxocs.com

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

Status Voice Voice Task Flow Resources

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Phone Usages

Default Usage Sample phone usage

OCSTEST1 **Default Usage**

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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:  $^(\+7(\d{10})\$$

Phone Usage: Default Usage

Gateways: mediationsrv.pbxc.com:5061

Done





## 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

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: pool10

Pool Type: Enterprise

IM: ✓

Meeting: ✓

No Tasks Available



## Mediation Sever

The screenshot shows the Cisco Mediation Server MMC console. The left pane displays a tree view of the configuration hierarchy, with 'mediationsrv.pbxocs.com' selected. The right pane shows the 'General Settings' tab, which includes sections for Windows services, Certificate settings, Location Profile, Listening Connections, and Next Hop Connections. The 'Available Tasks' pane on the right shows 'Deactivate Mediation Services'.

Section	Property	Value
Windows services	Mediation service:	Running
	Mediation service:	Running
Certificate settings	Name:	ocs-cert
	Expiration Date:	4/19/2009
Location Profile	Name:	<None>
	Location Profile:	<None>
Listening Connections	A/V Edge Server FQDN:	<None>
	A/V Edge Server port:	<None>
	Listening address for Communications Server:	172.20.239.239
	Communications Server listening port:	5061
	Listening address for Gateway traffic:	172.20.228.200
Next Hop Connections	PSTN Gateway Listening Port:	5060
	Media port range:	60000 - 64000

The screenshot shows the Cisco Mediation Server MMC console with the 'Next Hop Connections' and 'Route Information' tabs selected. The 'Next Hop Connections' tab shows the Communications Server Next Hop FQDN, Communications Server Next Hop Port, PSTN Gateway IP Address, and PSTN Gateway Port. The 'Route Information' tab shows the 415PSTN route to PSTN 415 areacode, including the Phone Number Pattern, Phone Usage, and Default Usage.

Section	Property	Value
Next Hop Connections	Communications Server Next Hop FQDN:	pool10.pbxocs.com
	Communications Server Next Hop Port:	5061
	PSTN Gateway IP Address:	172.20.228.30
	PSTN Gateway Port:	5060
Route Information	415PSTN	Route to PSTN 415 areacode
	Phone Number Pattern:	^+?(\\d*)\$
	Phone Usage:	Default Usage



## Static Route

**Front Ends Properties** [X]

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-O...	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** [X]

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

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



**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

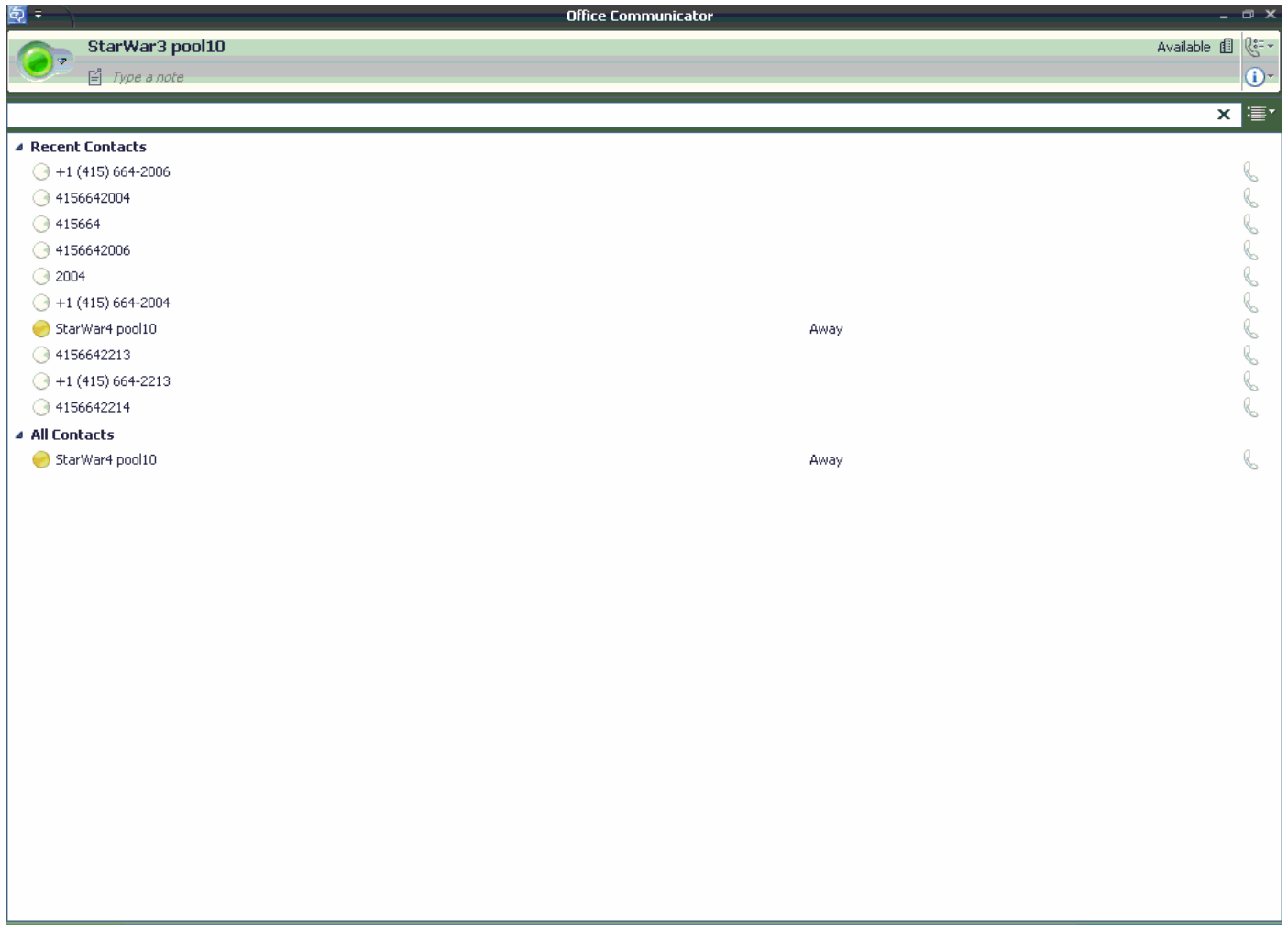
**Archiving**  
☐ Archive internal IM conversations  
☐ Archive federated IM conversations

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

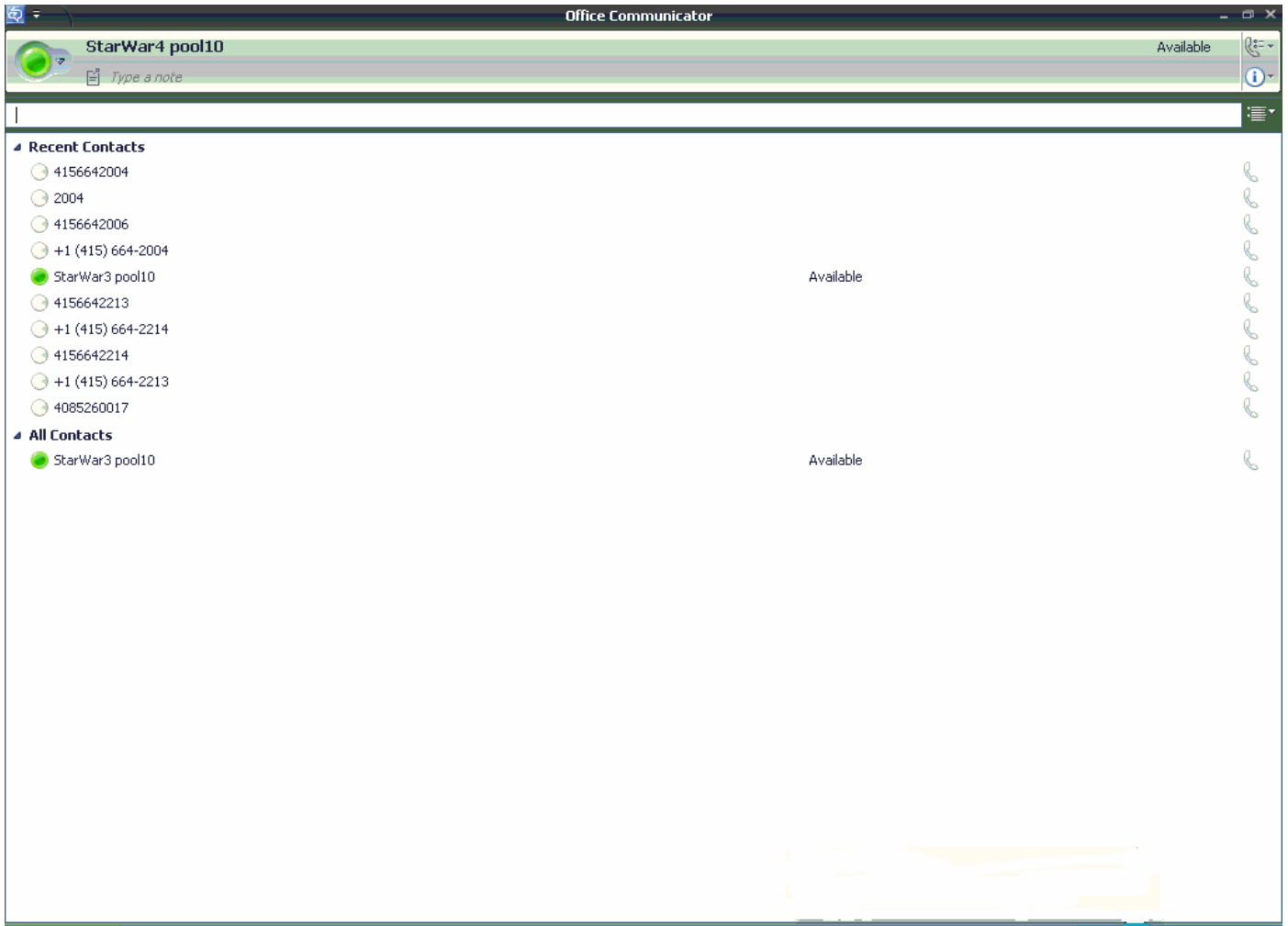
☒ Enable enhanced presence  
Note: Enhanced presence cannot be changed once it has been set.



## Microsoft Office Communicator (MOC) Configuration









## 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
!
!
```

---

<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 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  
!  
!
```

---

<sup>3</sup> 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
```

---

<sup>4</sup> Dial-peer toward OCS

<sup>5</sup> Dial-peer toward PSTN

<sup>6</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

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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