

## DialPlan for Site B

« Dial Plan for HQ Site (using single Route Pattern) (/workbook/view/ccie-collaboration/task/dial-plan-for-hq-site-using-single-route-pattern-MzAzNQ%3D%3D) | PLUS (+) Dialing For Site B (/workbook/view/ccie-collaboration/task/plus-dialing-for-site-b-MzAzNw%3D%3D) »

Last updated: November 13, 2017

### The following call routing policies are in effect at Site B's PSTN Service Provider

1. Site B PSTN service provider mandates proper use of both ISDN "called party number" and "called party number type" (Subscriber, National and International)
2. Called party number types (local, National, International) must be set in ISDN setup "called party number type" field for different types of calls (Local, National and international)
3. Do not send leading digits in the ISDN called party number string, such as **1** for national and **011** for international to signal type of calls.

## Tasks

- Configure Site B CUCM to satisfy the following dial plan requirements.
- Access code for all PSTN call is 9
- Configure local route group (R2) for Site B local Voice gateway.
- Use Route-List RL-R2 as the naming convention.
- Calls traversing between Site B Phone and R2 PRI trunk should always use G711 codec.
- Ensure that all calls will be sending Display Name and Number over PSTN.
  - Telco delivers 10 digit direct inward dialing called numbers in inbound calls, 972603XXXX, where XXXX is any Site B internal 4-digits extension.
  - Verify Site B gateway configuration by placing an inbound call from line 2 on the PSTN phone to 603XXXX, where XXXX is any Site B internal 4-digit extension. Make sure your MOH works. Do not use a Transcoder.
  - Modify the previous settings to use g711ulaw codec. Use a Transcoder for MOH to work.
  - Configure local route group for Site B's local Voice gateway : R2
  - For outbound calls, ensure 911 calls from any Site B IP phone terminate media on R2. For 911 calls, send 9726033XXX, where XXXX is the 4-digit extension as the calling number. 9911 is not required.

Calling Party Type = **National**

Calling Party Plan = **ISDN**

Called Party Type = **Unknown**

Called Party Plan = **ISDN**

- All Site B IP phones should be able to call local 7 digits PSTN numbers by dialing the access code **9** followed by 7 additional digits. The First digit after the access code could be any digit from 2 to 9, the remaining digit could be 0 (zero) to 9. This type of call should always use the local gateway R2 as the primary route. 7-digit calling number (603XXXX where xxxx is internal extension) and calling name should be sent to PSTN.

- All Site B IP phones should be able to place a national long distance PSTN calls by dialing the access code **9** followed by **1**, followed by 3 digit area code and lastly 7 digit subscriber number. The First digit of the area code and subscriber number could be any digit from 2 to 9, the remaining area code and subscriber number digit could be 0 (zero) to 9. This type of call should always use the local gateway R2 as the primary route. 10-digit calling number (972603XXXX where xxxx is internal extension) and calling name should be sent to PSTN.
- All Site B IP phones should also be able to place international calls by dialing the access code **9** followed by **011**, followed by variable length digit and ending with # sign. This type of call should always use local gateway R2. +1972603XXXX where XXXX is internal extension should be sent as calling number. Calling number should be sent to PSTN.

All the mentioned calls must be serviced by single route pattern **+**! on the Site B Publisher CallManager. H323 Gateway R2 should be SRST compliant. However SRST configuration task is not part of this section. Also make sure for all calls the call is routed within 5 seconds.

**Note:**

The exception to this rule is SIP trunk calls.

**Configuration:** [Click to collapse](#)

**R2 – Site B**

```
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g711alaw
  codec preference 3 g729br8
  codec preference 4 g729r8
!
dial-peer voice 1 pots
  incoming called-number .T
  direct-inward-dial
!
dial-peer voice 3100 voip
  destination-pattern ^9726033...$
  session target ipv4:11.100.65.11
  voice-class codec 1
```

**Site B – Publisher**

Device > Gateway

**Call Routing Information - Inbound Calls**

Significant Digits\*

System > Region Information > Audio Codec Preference List

CONTENTS

### Audio Codec Preference List Configuration

Save Delete Copy Add New

#### Status

Status: Ready

#### Audio Codec Preference List Information

Name\* NEW AUDIO List-H323Gateway

Description\* Low Loss Codec List

- Codecs in List\*
- G.711 U-Law 64k
  - G.711 A-Law 64k
  - G.711 U-Law 56k
  - G.711 A-Law 56k
  - ILBC 16k
  - AMR-WB (7k-24k)
  - AMR (5k-13k)
  - MP4A-LATM 128k
  - AAC-LD (MP4A Generic)
  - MP4A-LATM 64k
  - MP4A-LATM 56k
  - L16 256k
  - MP4A-LATM 48k
  - G.722 64k
  - ISAC 32k
  - MP4A-LATM 32k
  - G.722.1 32k
  - G.722 56k
  - G.722.1 24k
  - G.722 48k
  - MP4A-LATM 24k
  - G.728 16k
  - GSM Enhanced Full Rate 13k
  - GSM Full Rate 13k
  - G.729b 8k
  - G.729ab 8k
  - G.729 8k
  - G.729a 8k
  - GSM Half Rate 6k
  - G.723.1 7k

Save Delete Copy Add New

### System > Region Information > Region

Region Configuration

Region Information

Name\* RegionH323Gateway

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
RegionH323Gateway	NEW AUDIO List-H323Gateway	64 kbps (G.722, G.711)	384
Region for Phones	NEW AUDIO List-H323Gateway	64 kbps (G.722, G.711)	384

NOTE: Regions not displayed

Use System Default

Modify Relationship to other Regions

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
Default	Keep Current Setting	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting
Region for Phones			<input type="radio"/> Use System Default
Region for E2-CUBE			<input type="radio"/> None
RegionH323Gateway			<input type="radio"/> kbps

Save Delete Reset Apply Config Add New

### System > Device Pool

### Device Pool Configuration

Save Delete Copy Reset Apply Config Add New

#### Device Pool Settings

Device Pool Name\* DP-H323Gateway

Cisco Unified Communications Manager Group\* Default

Calling Search Space for Auto-registration < None >

Adjunct CSS < None >

Reverted Call Focus Priority Default

Intercompany Media Services Enrolled Group < None >

#### Roaming Sensitive Settings

Date/Time Group\* CMLocal

Region\* RegionH323Gateway

System > Device Gateway

Assign the H323 Gateway the above Device Pool created. Apply & Reset.

## Verification

Place the call from the PSTN phone line 2 to 603XXXX. Verify the codec.  
Press hold on Site B Phone to check if MOH works.

### > R2 – Site B Gateway

CONTENTS

```
dial-peer voice 3100 voip
no voice-class codec 1
codec g711ulaw
```

**Now your MOH will not work and for the same you would require a Transcoder**

```
voice-card 0
dspfarm
dsp services dspfarm
!
dspfarm profile 1 transcode
codec g729abr8
codec g729ar8
codec g711alaw
codec g711ulaw
codec g729br8
codec g729r8
maximum sessions 2
associate application SCCP
no shutdown
!
sccp local Loopback0
sccp ccm 11.100.65.11 identifier 1 priority 1 version 7.0+
sccp ccm group 1
bind interface Loopback0
associate ccm 1 priority 1
associate profile 1 register SBXcode
!
sccp
```

### Site B – Publisher

Media Resources > Transcoder

### Transcoder Configuration

---

#### Transcoder Information

**Transcoder:** SBXcode (SBXcode)  
 Registration Registered with Cisco Unified Communications Manager 11.100.65.11  
 IP Address 11.11.11.2  
 IPv6 Address 0000:0000:0000:0000:0000:0000:0000:0000

---

#### IOS Transcoder Info

Transcoder Type\* Cisco IOS Enhanced Media Termination Point  
 Description SBXcode  
 Device Name\* SBXcode  
 Device Pool\* DP-H323Gateway [View Details](#)  
 Common Device Configuration < None > [View Details](#)  
 Special Load Information  Leave blank to use default  
 Trusted Relay Point

## Verification

Place a call from PSTN Phone Line 2 to 603XXXX and place the call on hold from the Site B Phone 1

### R2 – Site B Gateway

show sccp connections

```

R2#show sccp connections

Total number of active session(s) 0, and connection(s) 0

R2#show sccp connections
sess_id   conn_id   stype mode      codec   sport rport ripaddr conn_id_tx
-----
16778220  16777333  xcode sendonly g711u   16562 16558 11.11.11.2
16778220  16777332  xcode recvonly g729    16560 0      0.0.0.0
  
```

show call active voice brief

```

Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 1
Call agent controlled call-legs: 0
SCCP call-legs: 2
Multicast call-legs: 0
Total call-legs: 4
11F3 : 61 169387550ms.1 (18:28:32.559 CST Tue Jan 5 2016) +1810 pid:1 Answer 6043000 active
dur 00:00:47 tx:2375/399000 rx:1523/241770 dscp:0 media:0 audio tos:0x0 video tos:0x0
Tele 0/1/0:23 (61) [0/1/0.1] tx:14660/10680/0ms g711ulaw noise:-83 acom:51 i/0/-57/-42 dBm

11F3 : 62 169387560ms.1 (18:28:32.569 CST Tue Jan 5 2016) +1800 pid:3100 Originate 9726033001 active
dur 00:00:47 tx:979/156083 rx:2375/380000 dscp:0 media:0 audio tos:0xB8 video tos:0x0
IP 0.0.0.0:0 SRTP: off rtt:0ms pl:10020/70ms lost:0/2/0 delay:55/55/75ms g711ulaw TextRelay: off Transcoded: No
media inactive detected:n media contrl rcvrd:n/a timestamp:n/a
long duration call detected:n long duration call duration:n/a timestamp:n/a

0 : 63 169421920ms.1 (18:29:06.929 CST Tue Jan 5 2016) +0 pid:0 Originate connecting
dur 00:00:15 tx:0/0 rx:764/15280 dscp:0 media:0 audio tos:0x0 video tos:0x0
IP 0.0.0.0:0 SRTP: off rtt:0ms pl:0/0ms lost:0/0/0 delay:0/0/0ms g729r8 TextRelay: off Transcoded: No
media inactive detected:n media contrl rcvrd:n/a timestamp:n/a
long duration call detected:n long duration call duration:n/a timestamp:n/a

0 : 65 169421930ms.1 (18:29:06.933 CST Tue Jan 5 2016) +0 pid:0 Originate connecting
dur 00:00:15 tx:793/126890 rx:0/0 dscp:0 media:0 audio tos:0xB8 video tos:0x0
IP 11.11.11.2:16558 SRTP: off rtt:0ms pl:0/0ms lost:0/0/0 delay:0/0/0ms g711ulaw TextRelay: off Transcoded: No
media inactive detected:n media contrl rcvrd:n/a timestamp:n/a
long duration call detected:n long duration call duration:n/a timestamp:n/a

Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 1
Call agent controlled call-legs: 0
SCCP call-legs: 2
Multicast call-legs: 0
Total call-legs: 4

R2#
  
```

### Site B – Publisher

Call Routing > Class of Control > Partitions

<input type="checkbox"/>	<a href="#">PT-Called-Xform</a>	PT-Called-Xform
<input type="checkbox"/>	<a href="#">PT-SingleRP</a>	PT-SingleRP

Call Routing > Class of Control > Calling Search Space

### Calling Search Space Configuration

Save Delete Copy Add New

**Status**  
 Status: Ready

**Calling Search Space Information**  
Name\*   
Description

**Route Partitions for this Calling Search Space**

Available Partitions\*\*

▼ ▲

Selected Partitions

▼  
▲

### Calling Search Space Configuration

Save Delete Copy Add New

**Status**  
 Status: Ready

**Calling Search Space Information**  
Name\*   
Description

**Route Partitions for this Calling Search Space**

Available Partitions\*\*

▼ ▲

Selected Partitions

▼  
▲

Call Routing > Translation Pattern

**Translation Pattern Configuration**

Save Delete Copy Add New

---

**Status**  
 Status: Ready

---

**Pattern Definition**

Translation Pattern:

Partition:

Description:

Numbering Plan:

Route Filter:

MLPP Precedence\*:

Resource Priority Namespace Network Domain:

Route Class\*:

Calling Search Space:

External Call Control Profile:

Route Option:  
 Route this pattern  
 Block this pattern

Provide Outside Dial Tone  
 Urgent Priority

---

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask:

Prefix Digits (Outgoing Calls):

Calling Line ID Presentation\*:

Calling Name Presentation\*:

Calling Party Number Type\*:

Calling Party Numbering Plan\*:

---

**Connected Party Transformations**

Connected Line ID Presentation\*:

Connected Name Presentation\*:

---

**Called Party Transformations**

Discard Digits:

Called Party Transform Mask:

Prefix Digits (Outgoing Calls):

Called Party Number Type\*:

Called Party Numbering Plan\*:

Save Delete Copy Add New

**Translation Pattern Configuration**

Save Delete Copy Add New

---

**Status**  
 Status: Ready

---

**Pattern Definition**

Translation Pattern:

Partition:

Description:

Numbering Plan:

Route Filter:

MLPP Precedence\*:

Resource Priority Namespace Network Domain:

Route Class\*:

Calling Search Space:

External Call Control Profile:

Route Option:  
 Route this pattern  
 Block this pattern

Provide Outside Dial Tone  
 Urgent Priority

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\*

Calling Name Presentation\*

Calling Party Number Type\*

Calling Party Numbering Plan\*

**Connected Party Transformations**

Connected Line ID Presentation\*

Connected Name Presentation\*

**Called Party Transformations**

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type\*

Called Party Numbering Plan\*

**Translation Pattern Configuration**

**Status**

Status: Ready

**Pattern Definition**

Translation Pattern

Partition

Description

Numbering Plan

Route Filter

MLPP Precedence\*

Resource Priority Namespace Network Domain

Route Class\*

Calling Search Space

External Call Control Profile

Route Option   
 Route this pattern   
 Block this pattern

Provide Outside Dial Tone

Urgent Priority

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\*

Calling Name Presentation\*

Calling Party Number Type\*

Calling Party Numbering Plan\*

**Connected Party Transformations**

Connected Line ID Presentation\*

Connected Name Presentation\*

**Called Party Transformations**

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type\*

Called Party Numbering Plan\*

**Translation Pattern Configuration**

**Status**

Status: Ready

**Pattern Definition**

Translation Pattern

Partition

Description

Numbering Plan

Route Filter

MLPP Precedence\*

Resource Priority Namespace Network Domain

Route Class\*

Calling Search Space

External Call Control Profile

Route Option

Route this pattern

Block this pattern

Provide Outside Dial Tone

Urgent Priority

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\*

Calling Name Presentation\*

Calling Party Number Type\*

Calling Party Numbering Plan\*

**Connected Party Transformations**

Connected Line ID Presentation\*

Connected Name Presentation\*

**Called Party Transformations**

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type\*

Called Party Numbering Plan\*

**Translation Pattern Configuration**

**Status**

Status: Ready

**Pattern Definition**

Translation Pattern

Partition

Description

Numbering Plan

Route Filter

MLPP Precedence\*

Resource Priority Namespace Network Domain

Route Class\*

Calling Search Space

External Call Control Profile

Route Option  
 Route this pattern  
 Block this pattern

Provide Outside Dial Tone

Urgent Priority

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\*

Calling Name Presentation\*

Calling Party Number Type\*

Calling Party Numbering Plan\*

---

**Connected Party Transformations**

Connected Line ID Presentation\*

Connected Name Presentation\*

---

**Called Party Transformations**

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type\*

Called Party Numbering Plan\*

—     —

**Find and List Translation Patterns**

**Status**

5 records found

**Translation patterns (1 - 5 of 5)**

Find Translation patterns where  begins with

<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	Translation Pattern ^	Partition
<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	90111	
<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	90111#	
<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	911	
<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	91[2-9]x[2-9]xxxxxxx	
<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	9[2-9]xxxxxxx	

Call Routing > Transformation > Transformation Pattern > Called Party Transformation Pattern

**Called Party Transformation Pattern Configuration**

**Status**

Status: Ready

---

**Pattern Definition**

Pattern\*

Partition

Description

Numbering Plan

Route Filter

Urgent Priority

---

**Called Party Transformations**

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type\*

Called Party Numbering Plan\*

—     —

### Route Group Configuration

Save 
 Delete 
 Add New

---

Save

Add successful

---

#### Route Group Information

Route Group Name\*

Distribution Algorithm\*

---

#### Route Group Member Information

##### Find Devices to Add to Route Group

Device Name contains

Available Devices\*\*

11.11.11.2
------------

Port(s)

---

#### Current Route Group Members

Selected Devices (ordered by priority)\*

Call Routing > Route/Hunt > Route List

### Route List Configuration

Save 
 Delete 
 Copy 
 Reset 
 Apply Config 
 Add New

---

Update successful

---

#### Route List Information

Registration Registered with Cisco Unified Communications Manager 11.100.65.11

IP Address 11.100.65.11

Device is trusted

Name\*

Description

Cisco Unified Communications Manager Group\*

Enable this Route List (change effective on Save; no reset required)

Run On All Active Unified CM Nodes

---

#### Route List Member Information

Selected Groups\*\*

System > Device Pool

### Device Pool Configuration

Save 
 Delete 
 Copy 
 Reset 
 Apply Config 
 Add New

---

Status: Ready

---

#### Device Pool Information

Device Pool: DP-SiteB-Phones (5 members\*\*)

---

#### Device Pool Settings

Device Pool Name\*

Cisco Unified Communications Manager Group\*

Calling Search Space for Auto-registration

Adjunct CSS

Reverted Call Focus Priority

Local Route Group

Call Routing > Route/Hunt > Route Pattern

### Route Pattern Configuration

Save 
 Delete 
 Copy 
 Add New

---

**Status**  
 Status: Ready

---

**Pattern Definition**

Route Pattern\*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence\*

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

Route Class\*

Gateway/Route List\*  [\(Edit\)](#)

Route Option  
 Route this pattern  
 Block this pattern

Call Classification\*

Allow Device Override  Provide Outside Dial Tone  Allow Overlap Sending  Urgent Priority

Require Forced Authorization Code

Authorization Level\*

System > Device Pool

### Device Pool Configuration

Save 
 Delete 
 Copy 
 Reset 
 Apply Config 
 Add New

---

**Device Pool Settings**

Device Pool Name\*

Cisco Unified Communications Manager Group\*

Calling Search Space for Auto-registration

Adjunct CSS

Reverted Call Focus Priority

Intercompany Media Services Enrolled Group

---

**Roaming Sensitive Settings**

Date/Time Group\*

Region\*

Media Resource Group List

Location

Network Locale

SRST Reference\*

Connection Monitor Duration\*\*\*

Single Button Barge\*

Join Across Lines\*

Physical Location

Device Mobility Group

Local Route Group

---

**Device Mobility Related Information\*\*\*\***

Device Mobility Calling Search Space

AAR Calling Search Space

AAR Group

Calling Party Transformation CSS

Called Party Transformation CSS

R2 – Site B

```
voice translation-rule 911
  rule 1 /^3...$/ /972603&/ type any national plan any isdn
!
voice translation-rule 912
  rule 1 // // type any unknown plan any isdn
!
voice translation-rule 9011
  rule 1 /^3...$/ /+1972603&/ type any international plan any isdn
!
voice translation-rule 9012
  rule 1 // // type any international plan any isdn
!
voice translation-rule 9303
  rule 1 /^3...$/ /603&/ type any subscriber plan any isdn
!
voice translation-rule 9304
  rule 1 // // type any subscriber plan any isdn
!
voice translation-rule 91408
  rule 1 /^3...$/ /972603&/ type any national plan any isdn
!
voice translation-rule 91409
  rule 1 // // type any national plan any isdn
!
voice translation-profile EMR
  translate calling 911
  translate called 912
!
voice translation-profile INT
  translate calling 9011
  translate called 9012
!
voice translation-profile LOCAL
  translate calling 9303
  translate called 9304
!
voice translation-profile NATIONAL
  translate calling 91408
  translate called 91409
!
dial-peer voice 10000 voip
  incoming called-number .T
  codec g711ulaw
!
dial-peer voice 911 pots
  translation-profile outgoing EMR
  destination-pattern ^911$
  no digit-strip
  port 0/1/0:23
!
dial-peer voice 9303 pots
  translation-profile outgoing LOCAL
  destination-pattern 9[2-9].....$
```

```

port 0/1/0:23
forward-digits 7
!
dial-peer voice 91408 pots
translation-profile outgoing NATIONAL
destination-pattern 91[2-9]..[2-9].....$
port 0/1/0:23
forward-digits 10
!
dial-peer voice 9011 pots
translation-profile outgoing INT
destination-pattern 9011T
port 0/1/0:23

```

System > Service Parameters

### Service Parameter Configuration

Save
 Set to Default
 Advanced

---

**Status**

Status: Ready

---

**Select Server and Service**

Server\* 11.100.65.11 (Active)

Service\* Cisco CallManager (Active)

---

T302 Timer\* 5000

Device > Gateway

**Call Routing Information - Outbound Calls**

Calling Party Selection*	<span style="border: 1px solid #ccc; padding: 2px;">Originator</span>
Calling Party Presentation*	<span style="border: 1px solid #ccc; padding: 2px;">Default</span>
Called party IE number type unknown*	<span style="border: 1px solid #ccc; padding: 2px;">Cisco CallManager</span>
Calling party IE number type unknown*	<span style="border: 1px solid #ccc; padding: 2px;">Cisco CallManager</span>
Called Numbering Plan*	<span style="border: 1px solid #ccc; padding: 2px;">Cisco CallManager</span>
Calling Numbering Plan*	<span style="border: 1px solid #ccc; padding: 2px;">Cisco CallManager</span>
Caller ID DN	<span style="border: 1px solid #ccc; padding: 2px;"></span>

Display IE Delivery

Redirecting Number IE Delivery - Outbound

R2 – Gateway

```

Interface serial 0/1/0:23
isdn outgoing display-ie
isdn outgoing ie redirecting number

```

## Verification

R2 - Site B Gateway

```

debug isdn q931

```

Dial 911  
Dial 96043000  
Dial 914085032000  
Dial 9011442050075000 with # and without #

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