

# Native Call Queueing Enhancement in CUCM 11.5

## Contents

[Introduction](#)

[Components Used](#)

[Background Information](#)

[Feature Overview](#)

[Configuration](#)

[H.225 Trunk \(Gatekeeper Controlled\)](#)

[Inter-Cluster Trunk \(Non-Gatekeeper Controlled\)](#)

[Inter-Cluster Trunk \(Gatekeeper Controlled\)](#)

[H.323 Gateway](#)

[SIP Profile](#)

[MGCP \(E1 PRI, T1 PRI, T1 CAS, and BRI\)](#)

[Log Analysis](#)

[Troubleshoot](#)

## Introduction

Cisco Unified Communications Manager(CUCM) provides Call Queueing to place callers in a queue until hunt members are available to answer them. An administrator can set the default so callers receive an initial greeting announcement before the call is extended to an agent or the default can be changed so the initial announcement plays only after the caller is put in the queue followed by Music On Hold or Tone On Hold. If the caller remains in queue for a specified period of time, a secondary announcement is played at a configured interval until the call can be answered or until the maximum wait timer expires.

## Components Used

- Cisco Unified Communication Manager Version 11.5.1
- Cisco IP Phone Version 8.6.6.0

## Background Information

This section describes the basic function of native call queuing prior to the enhancement in CUCM 11.5

When a call comes in and reaches the hunt pilot, these functions are provided:

- A caller can be connected to an initial customizable greeting announcement before proceed.
- If one or more line members are logged in to the hunt pilot and are in an idle state, and if no calls are

queued, the call is extended to the line member that has been idle for the longest period of time.

- If no line members answer a call, that caller is not placed in queue. The call is routed to a new destination or disconnected, based on the setup. When no hunt members answer, are logged in, or registered.
- If a line member does not answer a queue-enabled call, that line member is logged off the hunt group only if the option Automatically Logout Hunt Member on No Answer is selected in the Line Group setup window.
- Calls are placed in queue only if all members are busy.
- A caller who is connected in queue can hear Music On Hold and a repeating (customizable) periodic announcement.
- After a line member becomes idle, the caller with the longest wait time across multiple hunt groups is extended to the idle line member. If the idle line member does not answer the call, the caller is returned to the previous position in the queue.
- If a queued call exceeds its maximum wait time or the maximum number of callers allowed in queue is exceeded, the call can be routed to an alternate number or it can be disconnected, depending on how the hunt pilot is configured. The alternate number can be one of the following: A hunt pilot DN with queuing either enabled or disabled A voicemail DN A line DN A shared DN
- Line members can display the queue status of their queue-enabled hunt pilots. The queue status display provides the following types of information: Hunt pilot pattern Number of queued callers on each hunt pilot Longest waiting time

Call queuing works in conjunction with existing hunt pilots, but there are no changes in the behavior of the hunting operation for either queuing or nonqueuing hunt pilots. Hunt pilots that have call queuing enabled provide the following features:

- Queuing-enabled hunt pilot calls can only be received by line members one call at a time. Two queuing-enabled hunt pilot calls cannot be offered to a line member. A line member can receive calls directly to the DN or from non-queuing hunt pilots.
- Line members who do not answer calls that are routed by hunt pilots are automatically logged out. A line member is automatically logged out of a device if the line member receives a queuing-enabled hunt pilot call and does not answer the call before timeout occurs. In the case of a shared-line deployment, all devices configured with the same shared line are logged out. You can configure this behavior from the Line Group setting window by selecting Automatically Logout Hunt Member on No Answer. Line

members are logged out only if this check box is checked.

With the working of call queuing as described there were many instances where the end user would hear dead air or silence during the initial announcement, thus causing the user to think that the call was not successful. This situation would arise when one end could not be able to support early media in the call.

## Feature Overview

Starting with Cisco Unified Communications Manager Release 11.5, you can configure the inbound calls to change to the connected call state before playing the queuing announcement, while the call is extended to a hunt member in the queuing-enabled hunt pilot.

The new **Connect Inbound Call before Playing Queuing Announcement** check box is added to the following trunk and gateway configuration windows :

- H.225 Trunk (Gatekeeper Controlled)
- Inter-Cluster Trunk (Non- Gatekeeper Controlled)
- Inter Cluster Trunk(Gatekeeper Controlled)
- H.323 Gateway(Gateway Type)
- SIP Profile (Trunk Specific Configuration)
- MGCP (E1 PRI, T1 PRI, T1 CAS, and BRI)

Once the user checks this box, CUCM will send 200OK after the 100Trying in case of SIP and in case of H323/MGCP CUCM will send a Connect in the Hunt Pilot call flow. This will ensure that the user can hear the initial announcement instead of silence or dead air in case the other end is not able to support Early Media.

## Configuration

Below are the configuration snapshots with the newly added parameter on the CUCM

### H.225 Trunk (Gatekeeper Controlled)

## Trunk Configuration

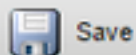


Save

Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input checked="" type="checkbox"/> Wait for Far End H.245 Terminal Capability Set	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, IPSec needs to be configured in th	
<input type="checkbox"/> H.235 Pass Through Allowed	
Use Trusted Relay Point*	Default
<input type="checkbox"/> PSTN Access	
<input checked="" type="checkbox"/> Connect Inbound Call before Playing Queuing Announcement	

### Inter-Cluster Trunk (Non-Gatekeeper Controlled)

## Trunk Configuration



Save

Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, IPsec needs to be configured	
<input type="checkbox"/> H.235 Pass Through Allowed	
<input type="checkbox"/> Enable SAF	
Use Trusted Relay Point*	Default
<input type="checkbox"/> PSTN Access	
<input checked="" type="checkbox"/> Connect Inbound Call before Playing Queuing Announcement	
<input type="checkbox"/> Run On All Active Unified CM Nodes	

### Inter-Cluster Trunk (Gatekeeper Controlled)

## Trunk Configuration



Save

Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, IPsec needs to be configured	
<input type="checkbox"/> H.235 Pass Through Allowed	
Use Trusted Relay Point*	Default
<input type="checkbox"/> PSTN Access	
<input type="checkbox"/> Connect Inbound Call before Playing Queuing Announcement	

## H.323 Gateway

### Gateway Configuration

Save

Queue Name:

ASN.1 ROSE OID Encoding\*:

Use Trusted Relay Point\*:

Signaling Port\*:

Media Termination Point Required

Retry Video Call As Audio

Wait for Far End H.245 Terminal Capability Set

Path Replacement Support

Transmit UTF-8 for Calling Party Name

SRTP Allowed - When this flag is checked, IPsec needs to be config

H.235 Pass Through Allowed

PSTN Access

Connect Inbound Call before Playing Queuing Announcement

## SIP Profile

### SIP Profile Configuration

Save

Calling Line Identification Presentation\*:

Session Refresh Method\*:

Early Offer support for voice and video calls\*:

Enable ANAT

Deliver Conference Bridge Identifier

Allow Passthrough of Configured Line Device Caller Information

Reject Anonymous Incoming Calls


Reject Anonymous Outgoing Calls

Send ILS Learned Destination Route String

Connect Inbound Call before Playing Queuing Announcement

## MGCP (E1 PRI, T1 PRI, T1 CAS, and BRI)

## Gateway Configuration

 Save

Confidential Access Level	< None >
<input type="checkbox"/> Handle DTMF Precedence Signals	
<input type="checkbox"/> Encode Voice Route Class	
Load Information	
Port Selection Order*	Top Down
Digit Sending*	DTMF
Network Locale	United States
SMDI Base Port*	0
Use Trusted Relay Point*	Default
Route Class Signaling Enabled*	Off
<input type="checkbox"/> V150 (subset)	
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
<input type="checkbox"/> PSTN Access	
<input type="checkbox"/> Connect Inbound Call before Playing Queuing Announcement	

## Log Analysis

The below section focuses on the differences seen in the trace files when the "**Connect Inbound Call before Playing Queuing Announcement**" is checked and unchecked.

### SIP Normal Call Flow

#### Incoming Invite to the CUCM

```
00455394.002 |18:33:30.036 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from
10.127.227.7 on port 55522 index 16 with 1182 bytes:
[14599,NET]
INVITE sip:0000@10.106.111.105:5060 SIP/2.0
Via: SIP/2.0/TCP 10.127.227.7:5060;branch=z9hG4bK4e222dea4e0
From: <sip:888819@10.127.227.7>;tag=107999~6c65cba7-94a0-4069-84c7-4774aecf0647-33198813
To: <sip:0000@10.106.111.105>
.
.
//Truncated Output
```

#### 100 Trying Sent

```
00455398.001 |18:33:30.037 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to
10.127.227.7 on port 55522 index 16
[14600,NET]
SIP/2.0 100 Trying
Via: SIP/2.0/TCP 10.127.227.7:5060;branch=z9hG4bK4e222dea4e0
From: <sip:888819@10.127.227.7>;tag=107999~6c65cba7-94a0-4069-84c7-4774aecf0647-33198813
```



To: <sip:0000@10.106.111.105>

.

//Truncated Output

#### Digit Analysis takes place

00455415.007 |18:33:30.038 |AppInfo |Digit analysis: match(pi="2", fqcn="",  
cn="888819", plv="5", pss="", TodFilteredPss="", dd="0000", dac="0")  
00455415.008 |18:33:30.038 |AppInfo |Digit analysis: analysis results  
00455415.009 |18:33:30.038 |AppInfo ||PretransformCallingPartyNumber=888819  
|CallingPartyNumber=888819  
|DialingPartition=  
|DialingPattern=0000  
|FullyQualifiedCalledPartyNumber=0000

#### Allocate Annunciater for the Initial Announcement

00455426.001 |18:33:30.039 |AppInfo |QueueControlCdr(17) - get\_call\_info\_SsCallInfoRes,  
huntPilotQueueProfile.alwaysplayinitialannouncement=1  
00455432.001 |18:33:30.039 |AppInfo |MediaResourceCdpc(22)::waiting\_MrmAllocateAnnResourceReq -  
CI = 21438416

#### Media Negotiation takes place for initial announcement

00455454.001 |18:33:30.041 |AppInfo |ARBTRY-ConnectionManager-  
wait\_MediaConnectRequest(21438414,21438416)  
00455478.001 |18:33:30.041 |AppInfo |ARBTRY-ConnectionManager-  
wait\_MediaConnectReply(21438414,21438416)

#### 183 Session Progress sent for early media with SDP a=sendonly

00455494.001 |18:33:30.143 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to  
10.127.227.7 on port 55522 index 16  
[14601,NET]  
SIP/2.0 183 Session Progress  
Via: SIP/2.0/TCP 10.127.227.7:5060;branch=z9hG4bK4e222dea4e0  
From: <sip:888819@10.127.227.7>;tag=107999~6c65cba7-94a0-4069-84c7-4774aecf0647-33198813  
To: <sip:0000@10.106.111.105>;tag=4705~8b68bd5c-f78f-44c5-b1ce-8ea93a8efbb6-21438414

.

//Truncated Output

.

v=0  
o=CiscoSystemsCCM-SIP 4705 1 IN IP4 10.106.111.105  
s=SIP Call  
c=IN IP4 10.106.111.105  
t=0 0  
m=audio 4000 RTP/AVP 0 8 18  
a=X-cisco-media:umoh+ConnSendOnly  
a=rtpmap:0 PCMU/8000  
a=rtpmap:8 PCMA/8000  
a=rtpmap:18 G729/8000  
a=fmtp:18 annexb=no  
a=sendonly

#### SIP Call Flow with "Connect Inbound Call before Playing Queuing Announcement" checked

#### Incoming Invite to the CUCM

00452822.002 |18:22:22.842 |AppInfo |SIPTcp - wait\_SdlReadRsp: Incoming SIP TCP message from  
10.127.227.7 on port 56658 index 14 with 1182 bytes:  
[14494,NET]  
INVITE sip:0000@10.106.111.105:5060 SIP/2.0  
Via: SIP/2.0/TCP 10.127.227.7:5060;branch=z9hG4bK4d2425c95ba

From: <sip:888819@10.127.227.7>;tag=107977~6c65cba7-94a0-4069-84c7-4774aecf0647-33198808  
To: <sip:0000@10.106.111.105>

.  
.  
//Truncated Output

#### 100 Trying sent

00452826.001 |18:22:22.843 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to  
10.127.227.7 on port 56658 index 14  
[14495,NET]

SIP/2.0 100 Trying

Via: SIP/2.0/TCP 10.127.227.7:5060;branch=z9hG4bK4d2425c95ba

From: <sip:888819@10.127.227.7>;tag=107977~6c65cba7-94a0-4069-84c7-4774aecf0647-33198808

To: <sip:0000@10.106.111.105>

.  
.  
//Truncated Output

#### Digit Analysis takes place

00452843.007 |18:22:22.844 |AppInfo |Digit analysis: match(pi="2", fqcn="",  
cn="888819",plv="5", pss="", TodFilteredPss="", dd="0000",dac="0")

00452843.008 |18:22:22.844 |AppInfo |Digit analysis: analysis results

00452843.009 |18:22:22.844 |AppInfo ||PretransformCallingPartyNumber=888819

|CallingPartyNumber=888819

|DialingPartition=

|DialingPattern=0000

|FullyQualifiedCalledPartyNumber=0000

#### Annunciater allocated for Initial announcement

00452854.001 |18:22:22.845 |AppInfo |QueueControlCdr(15) - get\_call\_info\_SsCallInfoRes,  
huntPilotQueueProfile.alwaysplayinitialannouncement=1

00452860.001 |18:22:22.845 |AppInfo |MediaResourceCdpc(19)::waiting\_MrmAllocateAnnResourceReq -  
CI = 21438406

#### Media Negotiation for the initial announcement

00452882.001 |18:22:22.846 |AppInfo |ARBTRY-ConnectionManager-  
wait\_MediaConnectRequest(21438404,21438406)

00452906.001 |18:22:22.847 |AppInfo |ARBTRY-ConnectionManager-  
wait\_MediaConnectReply(21438404,21438406)

#### 200 OK with SDP a=sendonly sent instead of 183 session progress thus connecting the call rather than an early media.

00452928.001 |18:22:22.848 |AppInfo |SIPTcp - wait\_SdlSPISignal: Outgoing SIP TCP message to  
10.127.227.7 on port 56658 index 14  
[14496,NET]

SIP/2.0 200 OK

Via: SIP/2.0/TCP 10.127.227.7:5060;branch=z9hG4bK4d2425c95ba

From: <sip:888819@10.127.227.7>;tag=107977~6c65cba7-94a0-4069-84c7-4774aecf0647-33198808

To: <sip:0000@10.106.111.105>;tag=4690~8b68bd5c-f78f-44c5-b1ce-8ea93a8efbb6-21438404

.  
.  
//Truncated Output

.  
.  
v=0

o=CiscoSystemsCCM-SIP 4690 1 IN IP4 10.106.111.105

s=SIP Call

c=IN IP4 10.106.111.105

t=0 0

m=audio 4000 RTP/AVP 0 8 18

a=X-cisco-media:umoh+ConnSendOnly

a=rtpmap:0 PCMU/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:18 G729/8000

a=fmtp:18 annexb=no  
a=sendonly

## H323 Normal Call Flow

### **Incoming H323 Setup Message**

```
00091345.011 |09:03:06.341 |AppInfo |SPROCRas - {  
  h323-uu-pdu  
  {  
    h323-message-body setup :  
    {  
      protocolIdentifier { 0 0 8 2250 0 5 },  
      sourceAddress  
      {  
        dialedDigits : "999919",  
        h323-ID : {"999919", {0, 0, 0, 0}, ...}  
      }  
    }  
  }  
.  
.  
//Truncated Output
```

### **Digit Analysis takes place**

```
00091367.006 |09:03:06.384 |AppInfo |Digit analysis: match(pi="2", fqcn="",  
cn="999919", plv="5", pss="", TodFilteredPss="", dd="0000", dac="0")  
00091367.007 |09:03:06.384 |AppInfo |Digit analysis: analysis results  
00091367.008 |09:03:06.384 |AppInfo ||PretransformCallingPartyNumber=999919  
|CallingPartyNumber=999919  
|DialingPartition=  
|DialingPattern=0000
```

### **Annunciator Allocated for initial announcement**

```
00091378.001 |09:03:06.388 |AppInfo |QueueControlCdrc(1) - get_call_info_SsCallInfoRes,  
huntPilotQueueProfile.alwaysplayinitialannouncement=1  
00091384.001 |09:03:06.388 |AppInfo |MediaResourceCdpc(1)::waiting_MrmAllocateAnnResourceReq -  
CI = 25333775
```

### **Call Proceeding Message sent**

```
00091386.005 |09:03:06.389 |AppInfo |{  
  h323-uu-pdu  
  {  
    h323-message-body callProceeding :  
    {  
      protocolIdentifier { 0 0 8 2250 0 5 },  
    }  
  }  
.  
.  
//Truncated Output
```

### **Media Negotiation takes place for the initial announcement**

```
00091407.001 |09:03:06.392 |AppInfo |ARBTRY-ConnectionManager-  
wait_MediaConnectRequest(25333773,25333775)  
  
00091447.001 |09:03:06.411 |AppInfo |ARBTRY-ConnectionManager-  
wait_MediaConnectReply(25333773,25333775)
```

### **H323 Progress message sent for early media, which is followed by the H245 messages for media negotiation**

```
00091456.005 |09:03:06.411 |AppInfo |SPROCRas - {  
  h323-uu-pdu  
  {  
    h323-message-body progress :  
    {  
      protocolIdentifier { 0 0 8 2250 0 5 },  
    }  
  }  
.  
.
```

.  
//Truncated Output

**H323 Call flow with the "Connect Inbound Call before Playing Queuing Announcement" checked**

**Incoming setup message to the CUCM**

```
00092572.010 |09:07:25.234 |AppInfo |SPROCRas - {
  h323-uu-pdu
  {
    h323-message-body setup :
    {
      protocolIdentifier { 0 0 8 2250 0 5 },
      sourceAddress
      {
        dialedDigits : "999919",
        h323-ID : {"999919", {0, 0, 0, 0}, ...}
      },
    },
  },
}
```

.  
//Truncated Output

**Digit Analysis takes place**

```
00092594.006 |09:07:25.236 |AppInfo |Digit analysis: match(pi="2", fqcn="",
cn="999919",plv="5", pss="", TodFilteredPss="", dd="0000",dac="0")
00092594.007 |09:07:25.236 |AppInfo |Digit analysis: analysis results
00092594.008 |09:07:25.236 |AppInfo ||PretransformCallingPartyNumber=999919
|CallingPartyNumber=999919
|DialingPartition=
|DialingPattern=0000
```

**Annunciator is invoked for initial announcement**

```
00092605.001 |09:07:25.236 |AppInfo |QueueControlCdr(2) - get_call_info_SsCallInfoRes,
huntPilotQueueProfile.alwaysplayinitialannouncement=1
00092611.001 |09:07:25.237 |AppInfo |MediaResourceCdpc(2)::waiting_MrmAllocateAnnResourceReq -
CI = 25333779
```

**H323 Proceeding message sent out**

```
00092612.005 |09:07:25.237 |AppInfo |{
  h323-uu-pdu
  {
    h323-message-body callProceeding :
    {
      protocolIdentifier { 0 0 8 2250 0 5 },
    },
  },
}
```

.  
//Truncated Output

**Media negotiation takes place**

```
00092634.001 |09:07:25.238 |AppInfo |ARBTRY-ConnectionManager-
wait_MediaConnectRequest(25333777,25333779)
00092674.001 |09:07:25.240 |AppInfo |ARBTRY-ConnectionManager-
wait_MediaConnectReply(25333777,25333779)
```

**Connect message is sent out instead of H323 Progress message placing the call in connected state rather than early media. The H245 messages will be exchanged post this message.**

```
00092686.006 |09:07:25.240 |AppInfo |SPROCRas - {
  h323-uu-pdu
  {
    h323-message-body connect :
    {
      protocolIdentifier { 0 0 8 2250 0 5 },
      h245Address ipAddress :
    },
  },
}
```

```
{  
  ip '0A6A6F69'H,  
  port 34408  
},
```

```
.  
.
```

*//Truncated Output*

## Troubleshoot

There is currently no specific troubleshooting information available for this configuration.