Call Recording Examples for Network-Based and Phone-Based Recording

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Call Recording Examples for Network-Based and Phone-Based Recording

Recording Overview

This document contains use case examples and call flows for Cisco Unified Communications Manager's Call Recording feature. Cisco Unified Communications Manager offers IP phone-based or network-based recording:

- In IP phone based recording, recording media is sourced from the phone. The phone forks two media streams to the recording server.
- In network-based recording, recording media can be sourced from either the phone or the gateway. When you implement network-based recording, the gateway in your network must connect to Cisco Unified Communications Manager over a SIP trunk.

Configuration

For more details on Call Recording, including configuration steps, see the "Recording" chapter of the Feature Configuration Guide for Cisco Unified Communications Manager for release 10.5(2) or greater at https://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-installation-and-configuration-guides-list.html.

Note  With Release 10.0(1), see the "Monitoring and Recording" chapter of the Features and Services Guide for Cisco Unified Communications Manager.

Network-Based Recording Call Flow Examples

This section contains network-based recording call flow examples.

IP Phone to IP Phone - Both Enabled For Recording

In this use case, Gateway is preferred, but the phone is selected.

- 1–3. An IP phone calls another IP phone
• 4–5. Two recording sessions are started automatically from both IP phones. The recording media source for both phones is Gateway Preferred. Since the gateway is not in the call flow Cisco Unified Communications Manager selects the IP Phones to fork the media to the recorder.

**Figure 1: IP Phone to IP Phone Both Enabled for Recording**

External Call to IP phone – Selective Recording

In this use case, Gateway is preferred and selected.

• 1–3. An external call is answered by a user with a Cisco IP phone.

• 4–5. The Cisco IP phone presses the Record softkey thereby starting a new recording session. The gateway is configured as the preferred recording media source and the gateway is valid for the call flow. Cisco Unified Communications Manager selects the gateway to fork the media to the recorder.
6. The Cisco IP phone displays "Recording..." and the softkey label changes to "StopRec".

**Figure 2: External Call to IP phone – Selective Recording**

**Mid-Call - Gateway Recording Session Stops When Call is Held**

This use case demonstrates the Hold function on the IP phone.

- 1–3. A call is in progress and the gateway is recording the media. User 1 places the call on hold. Cisco Unified Communications Manager plays Music-on-Hold to the caller.
- 4. Cisco Unified Communications Manager instructs the gateway to stop forking media to the recorder.

**Figure 3: Mid-Call – Gateway Recording Session Stops When Call Is Held**
Mobility - External Call to Mobile Jabber Client

In this use case, the gateway is preferred and selected.

- 1–3. An external call is answered on a Mobile phone configured as a Dual-mode device
- 4. The recording session starts automatically for the dual mobile phone. The gateway is configured as the preferred recording source and is valid for the call flow. Cisco Unified Communications Manager selects the gateway to fork the media to the recorder.

Figure 4: Mobility – External Call to Mobile Jabber Client

Mobility - External Call to Remote Destination Profile

In this use case, gateway is preferred and selected.

- 1–4. An external call is answered on a mobile phone using a remote destination configured on a Remote Destination Profile.
• 5–6. The mobile presses *86 to use DTMF to start the recording session. Cisco Unified Communications Manager selects the ingress gateway to fork the media to the recorder.

**Figure 5: Mobility – External Call to Remote Destination Profile**

**Extend and Connect - External Call to CTI Remote Device**
In this use case, gateway is preferred and selected.

• 1–4. An external call is answered on an Extend & Connect remote destination that is configured as a CTI Remote Device.
5. The recording session starts from the CTI Remote Device. The gateway is configured as the preferred recording media source and is valid for the call flow. The ingress gateway is the first gateway in the call flow so Cisco Unified Communications Manager selects the ingress gateway to fork recording media to the recorder.

**Figure 6: Extend & Connect – External Call to CTI Remote Device**

**Inter-Cluster Recording - External Call From Cluster 1 to IP Phone on Cluster**

In this use case, the gateway is selected. The Recording Profile specifies a central recorder.

1–4. An external call is received on a gateway in Cluster 1 and is answered by an IP phone in Cluster 2.
5. Recording starts automatically for the IP Phone in Cluster 2; the Phone is selected to fork the media. The recording profile assigned to the IP Phone is configured to use a central recorder.

Inter-Cluster Recording – External Call From Cluster 1 to IP Phone on Cluster 2

This use case demonstrates a mid-call Hold in an inter-cluster call.

- 1–5. An external call is underway to User 1 in Cluster 2. User 1 places the call on hold. The media stream is dropped.
- 6. Cisco Unified Communications Manager in Cluster 2 plays Music On Hold to the external caller.
7–8. Cisco Unified Communications Manager in Cluster 2 requests that Cluster 1 stop the recording session. Cluster 1 instructs the gateway to stop the session.

Figure 8: Inter-Cluster Recording – External Call From Cluster 1 Is HELD by User on Cluster 2

IP Phone-Based Recording Call Flow Examples
This section contains network-based recording call flow examples.

Automatic Call Recording
In automatic call recording, after an agent call becomes active, two server calls get made to the built-in bridge (BIB) of the agent phone. The agent phone automatically answers. Two server calls then get redirected to the recorder.
The following figure illustrates automatic call recording.

**Figure 9: Automatic Call Recording**

In this example of an automatic call recording session, the following entities participate:

- The customer call originates from DN 1000 device A.
- The agent receives the call at DN 2000 device B.

During an automatic call recording session, the following steps take place:

1. A customer, party A (the far-end party) with DN 1000, calls into the call center.
2. The call routes to the agent, who is party B with DN 2000. The agent answers the call. The agent IP phone starts to exchange media streams with the customer.
3. Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager first makes a recording call to the built-in bridge (BIB) of the agent IP phone for the agent voice.
4. Cisco Unified Communications Manager makes the second recording call to the BIB of the agent IP phone for the customer voice.
5. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone starts to fork the agent voice stream to the recorder.
6. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone starts to fork the customer voice stream to the recorder.

In previous releases, the INVITE message contained only near-end information, but customer information was unknown. A CTI connection to Cisco Unified Communications Manager was required to obtain the customer information.

Be aware that the two INVITE messages for the two separate voice streams contain the same near-end and far-end call information. The only difference in the two From headers is the first x- parameter, which indicates whether the call is for the near-end voice stream or for the far-end voice stream. x-nearend indicates the near-end voice stream. x-farend indicates the far-end voice stream.

The INVITE message that is explained here (another INVITE message gets sent) contains the agent recording call.

Note the header information of the INVITE messages from step 5 and step 6. The SIP header enhancement feature adds the information in bold text to the INVITE message headers.

**Step 5 INVITE Message Header Information**

```plaintext
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;x-farendrefci=ci1;x-farenddevice=deviceA;x-farendaddr=1000>;tag=fromtag1
```
**Step 6 INVITE Message Header Information**

From: <sip:2000@ucm1;x-farend;x-refci=ci2;x-nearenddevice=deviceB;
x-farendrefci=ci1;x-farenddevice=deviceA;x-farendaddr=1000>;tag=fromtag2

In both message headers,

- **x-farendrefci** specifies the far-end (customer) call leg caller ID.
- **x-farenddevice** specifies the far-end device name.
- **x-farendaddr** specifies the far-end DN.

**Local Cluster Far-End Party Hold/Resume**

In this use case for automatic call recording, the far-end party that belongs to the local cluster places the call on hold and resumes the call from a different device. The following figure illustrates this use case.

*Figure 10: Far-End Party in Local Cluster Holds and Resumes Call From Different Device*

In this use case, the following entities participate:

- The customer call originates from DN 1000 device A.
- The agent receives the call at DN 2000 device B.
- After placing the call on hold, the customer resumes the call from DN 1000 device A’.

During an automatic call recording session that involves a customer (far-end party) in the local cluster that places the call on hold then resumes the call from a different device, the following steps take place:

1. Party A (far-end party = customer in local cluster) calls party B (near-end party = agent).
2. Party B answers the call.
Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager first makes a recording call to the built-in bridge (BIB) of the agent IP phone for the agent voice.

Cisco Unified Communications Manager makes the second recording call to the BIB of the agent IP phone for the customer voice.

The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone starts to fork the agent voice stream to the recorder.

The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone starts to fork the customer voice stream to the recorder.

Party A (far-end party = customer in local cluster) places the call on hold.

Party A (far-end party = customer in local cluster) resumes the held call from device A' (a different device with same DN). Upon call resumption, party B is now connected to the new far-end party A'. The far-end call information has changed.

The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone forks the agent voice stream to the recorder.

The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone forks the customer voice stream to the recorder.

Note the following particularities of call processing that apply in this use case:

- Insertion of MOH when the far-end party places the call on hold does not cause a change in the far-end party.
- When another device that shares the line resumes the call, a SIP UPDATE message gets sent to the recorder with the new far-end party device name.

The UPDATE message that is explained here contains the agent (near-end) recording call. The other UPDATE for the customer (far-end) recording call contains the same information for the x-farend.

Note the header information of the INVITE message from step 5 and the UPDATE message from step 9. The SIP header enhancement feature adds the information in bold text to the message headers.

**Step 5 INVITE Message Header Information**
```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;x-farendrefci=ci1;x-farenddevice=deviceA;x-farendaddr=1000>;tag=fromtag1
```

**Step 9 UPDATE Message Header Information**
```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;x-farendrefci=ci1;x-farenddevice=deviceA;x-farendaddr=1000>;tag=fromtag1
```
Far-End Party Transfers Call to Another Far-End Party in Local Cluster

In this use case for automatic call recording, the far-end party in the local cluster transfers the call to another far-end party in the same local cluster. The following figure illustrates this use case.

Figure 11: Far-End Party in Local Cluster Transfers Call to Another Far-End Party in Local Cluster

In this use case, the following entities participate:

- The customer call originates from DN 1000 device A in the local cluster.
- The agent receives the call at DN 2000 device B.
- The customer transfers the call to DN 1100 device C in the same local cluster.

During an automatic call recording session that involves a customer (far-end party) in the local cluster that places the call to the agent and then later transfers the call to another far-end party in the local cluster, the following steps take place:

1. Party A (far-end party = customer in local cluster) calls party B (near-end party = agent).
2. Party B answers the call.
3. Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager first makes a recording call to the built-in bridge (BIB) of the agent IP phone for the agent voice.
4. Cisco Unified Communications Manager makes the second recording call to the BIB of the agent IP phone for the customer voice.
5. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone starts to fork the agent voice stream to the recorder.
6. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone starts to fork the customer voice stream to the recorder.
7. Party A (far-end party = customer in local cluster) initiates a consultation transfer of the call to another party, party C at DN 1100, in the local cluster.
8. Party C answers the transferred call.
9. Party A completes the transfer.
10. Because party B is now connected to a new far-end party, party C, Cisco Unified Communications Manager sends two UPDATE messages to the recorder.
The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone forks the agent voice stream to the recorder.

The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone forks the customer voice stream to the recorder.

Note the following particularities of call processing that apply in this use case:

- After first press of the Transfer key, because no change in the far-end party information occurs, Cisco Unified Communications Manager does not update the recorder.
- After the second press of the Transfer key, Cisco Unified Communications Manager sends the SIP UPDATE message to the recorder with updated far-end party information.

Note the following particularities of call processing that apply in this use case:

1. Insertion of MOH when the far-end party places the call on hold does not cause a change in the far-end party.
2. When another device that shares the line resumes the call, a SIP UPDATE message gets sent to the recorder with the new far-end party device name.

Note the header information of the INVITE message from step 5 and the UPDATE message from step 10. The SIP header enhancement feature adds the information in bold text to the message headers.

**Step 5 INVITE Message Header Information**

```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB; x-farendrefci=ci1;x-farenddevice=deviceA;x-farendaddr=1000>;tag=fromtag1
```

**Step 10 UPDATE Message Header Information**

```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB; x-farendrefci=ci4;x-farenddevice=deviceC;x-farendaddr=1100>;tag=fromtag1
```

When you compare the INVITE message header in step 5 with the UPDATE message header in step 10, notice that the far-end values (farendrefci, farenddevice, and farendaddr) all change because of the transfer.
Near-End Party Transfers Call to Another Near-End Party in Local Cluster

In this use case for automatic call recording, the near-end party transfers a call to another near-end party in the local cluster. The following figure illustrates this use case.

Figure 12: Near-End Party Transfers Call to Another Near-End Party in Local Cluster

In this use case, the following entities participate:

- The customer call originates from DN 1000 device A.
- The agent receives the call at DN 2000 device B.
- The agent transfers the call to DN 2001 device D.

During an automatic call recording session where the agent on the call transfers the call to another party in the same local cluster, the following steps take place:

1. Party A (far-end party = customer in local cluster) calls party B (near-end party = agent).
2. Party B (near-end party = agent in local cluster) answers the call.
3. Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager first makes a recording call to the built-in bridge (BIB) of the agent IP phone for the agent voice.
4. Cisco Unified Communications Manager makes the second recording call to the BIB of the agent IP phone for the customer voice.
5. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone starts to fork the agent voice stream to the recorder.
6. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone starts to fork the customer voice stream to the recorder.
7. Party B initiates the consultation transfer. This action implicitly places the call on hold.
8. Cisco Unified Communications Manager terminates recording of the agent voice by sending a BYE message to the recorder through a SIP trunk.
Cisco Unified Communications Manager terminates recording of the customer voice by sending a BYE message to the recorder through a SIP trunk.

Party B calls party D (another far-end party = agent in local cluster).

Party D answers the call from party B.

Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager makes a recording call to the built-in bridge (BIB) of the party B IP phone for the agent voice.

Cisco Unified Communications Manager makes the second recording call to the BIB of the party B IP phone for the customer voice.

The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the party B (agent) voice through a SIP trunk. The agent IP phone starts to fork the agent voice stream to the recorder.

The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the party A (customer) voice through a SIP trunk. The agent IP phone starts to fork the customer voice stream to the recorder.

Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager makes a recording call to the built-in bridge (BIB) of the party D IP phone for the agent voice.

Cisco Unified Communications Manager makes the second recording call to the BIB of the party D IP phone for the customer voice.

The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the party D (agent) voice through a SIP trunk. The agent IP phone starts to fork the agent voice stream to the recorder.

The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the party A (customer) voice through a SIP trunk. The agent IP phone starts to fork the customer voice stream to the recorder.

Party B completes the transfer.

Cisco Unified Communications Manager terminates recording of the party B (agent) voice (the consultation call) by sending a BYE message to the recorder through a SIP trunk.

Cisco Unified Communications Manager terminates recording of the party A (customer) voice by sending a BYE message to the recorder through a SIP trunk.

Because party D is now connected to a new far-end party, party A, Cisco Unified Communications Manager sends two UPDATE messages to the recorder.

The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the party D (agent) voice through a SIP trunk. The agent IP phone forks the agent voice stream to the recorder.

The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the party A (customer) voice through a SIP trunk. The agent IP phone forks the customer voice stream to the recorder.

Note the following particularities of call processing that apply in this use case:

- When the near-end party B presses Transfer, the call is implicitly put on hold, and the recording session with party A terminates.
- When party B dials party D and party D answers, a new recording session starts for party D.
- When party B completes the transfer, party D and party A get connected and the recorder receives an update with information about the new far-end party A.
Generally, each time an agent puts a recording call on hold, the current recording session terminates. Each time the agent invokes a supplementary service, such as Transfer or hold, the call is implicitly put on hold. Each time the far-end information changes, Cisco Unified Communications Manager sends a SIP UPDATE message to the recorder.

Note the header information of the INVITE messages from step 5, step 14, step 18, and the UPDATE message from step 23. The SIP header enhancement feature adds the information in bold text to the message headers.

**Step 5 INVITE Message Header Information**
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;x-farendrefci=ci1;x-farenddevice=deviceA;x-farendaddr=1000>;tag=fromtag1

**Step 14 INVITE Message Header Information**
From: <sip:2000@ucm1;x-nearend;x-refci=ci3;x-nearenddevice=deviceB;x-farendrefci=ci4;x-farenddevice=deviceD;x-farendaddr=2001>;tag=fromtag2

**Step 18 INVITE Message Header Information**
From: <sip:2001@ucm1;x-nearend;x-refci=ci4;x-nearenddevice=deviceD;x-farendrefci=ci3;x-farenddevice=deviceB;x-farendaddr=2000>;tag=fromtag2

**Step 23 UPDATE Message Header Information**
From: <sip:2001@ucm1;x-nearend;x-refci=ci4;x-nearenddevice=deviceD;x-farendrefci=ci1;x-farenddevice=deviceA;x-farendaddr=1000>;tag=fromtag2

**Far-End Party Transfers Call to CFNA-Enabled Party**
In this use case for automatic call recording, the far-end party blind-transfers the call to a party that has Call Forward No Answer (CFNA) enabled. The following figure illustrates this use case.

*Figure 13: Far-End Party Transfers Call to CFNA-Enabled Party*

In this use case, the following entities participate:

- The customer call originates from DN 1000 device A.
- The agent receives the call at DN 2000 device B.
- The customer blind-transfers the call to DN 1100 device C.
- Device C does not answer but has CFNA enabled to forward to DN 1200 device D.
During an automatic call recording session where the far-end party (customer) on the call transfers the call to another far-end party in the same local cluster but the far-end party has CFNA enabled, so the call forwards to a third far-end party in the local cluster, the following steps take place:

1. Party A (far-end party = customer in local cluster) calls party B (near-end party = agent).
2. Party B (near-end party = agent in local cluster) answers the call.
3. Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager first makes a recording call to the built-in bridge (BIB) of the party B (agent) IP phone for the agent voice.
4. Cisco Unified Communications Manager makes the second recording call to the BIB of the party B (agent) IP phone for the customer voice.
5. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone starts to fork the agent voice stream to the recorder.
6. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone starts to fork the customer voice stream to the recorder.
7. Party A presses Transfer, dials DN 1100 device C, and presses Transfer again (performs a blind transfer).
8. Cisco Unified Communications Manager rings DN 1100 on device C, but this DN and device have CFNA configured: ringing times out, and Cisco Unified Communications Manager forwards the call to DN 1200 device D.
9. Far-end party D with DN 1200 on device D answers the call.
10. Because party B is now connected to a new far-end party, party D, Cisco Unified Communications Manager sends two UPDATE messages to the recorder.
11. The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the party B (agent) voice through a SIP trunk. The agent IP phone forks the agent voice stream to the recorder.
12. The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the party D (customer) voice through a SIP trunk. The agent IP phone forks the customer voice stream to the recorder.

Note the following particularities of call processing that apply in this use case:

- For local-cluster transfers, Cisco Unified Communications Manager updates the recorder only when a new far-end party answers.
- A SIP UPDATE message that contains updated far-end information gets sent to the recorder when party D answers.

Note the header information of the INVITE messages from step 5 and step 10. The SIP header enhancement feature adds the information in bold text to the INVITE and UPDATE message headers.

**Step 5 INVITE Message Header Information**
```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;
    x-farendrefci=ci1;x-farenddevice=deviceA;x-farendaddr=1000>;tag=fromtag1
```

**Step 10 UPDATE Message Header Information**
```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;
    x-farendrefci=ci5;x-farenddevice=deviceD;x-farendaddr=1200>;tag=fromtag1
```
**Far-End Party in Local Cluster Creates Conference**

In this use case for automatic call recording, the far-end party in a local cluster creates a conference. The following figure illustrates this use case.

*Figure 14: Far-End Party in Local Cluster Creates Conference*

In this use case, the following entities participate:

- The far-end customer call originates from DN 1000 device A.
- The near-end agent receives the call at DN 2000 device B.
- Party A creates a conference by conferencing in DN 1100 device C.

During an automatic call recording session where the far-end (customer) party in the local cluster creates a conference by bringing an additional far-end party into the call, the following steps take place:

1. Party A (far-end party = customer in local cluster) calls party B (near-end party = agent).
2. Party B (near-end party = agent in local cluster) answers the call.
3. Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager first makes a recording call to the built-in bridge (BIB) of the party B (agent) IP phone for the agent voice.
4. Cisco Unified Communications Manager makes the second recording call to the BIB of the party B (agent) IP phone for the customer voice.
5. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone starts to fork the party B (agent) voice stream to the recorder.
The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone starts to fork the party A (customer) voice stream to the recorder.

Party A initiates a conference by pressing Confn and dialing DN 1100.

Party C DN 1100 device C answers the call.

Party A completes the conference by pressing Confn again.

Because party B is now connected to a new far-end party, CFB_2, Cisco Unified Communications Manager sends two UPDATE messages to the recorder.

The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the party B (agent) voice through a SIP trunk. The agent IP phone forks the agent voice stream to the recorder.

The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for CFB_2 (conference bridge) through a SIP trunk. The agent IP phone forks the conference voice stream to the recorder.

Note the following particularities of call processing that apply in this use case:

- After the conference gets established, the far-end party is changed to the conference bridge (CFB).
- Cisco Unified Communications Manager sends a SIP UPDATE message to the recorder.

Note the header information of the INVITE messages from step 5 and step 10. The SIP header enhancement feature adds the information in bold text to the INVITE message headers.

### Step 5 INVITE Message Header Information
```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;
x-farendrefci=ci1;x-farenddevice=deviceA;x-farendaddr=1000>;tag=fromtag1
```

### Step 10 UPDATE Message Header Information
```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;
x-farendrefci=ci7;x-farenddevice=CFB_2;x-farendaddr=b001234567;isfocus>;tag=fromtag1
```

The UPDATE message in step 10 includes isfocus. This isfocus indicates that the near-end party is participating in a conference call. The UPDATE message also includes a b-number as the new far-end address. The b-number specifies the DN of the conference bridge (CFB).
**Near-End Party in Local Cluster Creates Conference**

In this use case for automatic call recording, the near-end party in a local cluster creates a conference. The following figure illustrates this use case.

*Figure 15: Near-End Party in Local Cluster Creates Conference*

In this use case, the following entities participate:

- The far-end customer call originates from DN 1000 device A.
- The near-end agent receives the call at DN 2000 device B.
- Party B creates a conference by conferencing in DN 1100 device C.

During an automatic call recording session where the near-end (agent) party in the local cluster creates a conference by bringing an additional far-end party into the call, the following steps take place:

1. Party A (far-end party = customer in local cluster) calls party B (near-end party = agent).
2. Party B (near-end party = agent in local cluster) answers the call.
3. Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager first makes a recording call to the built-in bridge (BIB) of the party B (agent) IP phone for the agent voice.
4. Cisco Unified Communications Manager makes the second recording call to the BIB of the party B (agent) IP phone for the customer voice.
5. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone starts to fork the party B (agent) voice stream to the recorder.
6 The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone starts to fork the party A (customer) voice stream to the recorder.

7 Party B, the near-end party, initiates a conference by pressing Confn. When the near-end party makes a consultation call for a conference participant, the near-end party call automatically gets put on hold.

8 Cisco Unified Communications Manager terminates recording of the party B (agent) voice (the consultation call) by sending a BYE message to the recorder through a SIP trunk.

9 Cisco Unified Communications Manager terminates recording of the party A (customer) voice by sending a BYE message to the recorder through a SIP trunk.

10 Near-end party B dials DN 1100 party C.

11 Party C answers the call.

12 Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager makes a recording call to the built-in bridge (BIB) of the party B IP phone for the near-end (agent) voice.

13 Cisco Unified Communications Manager makes the second recording call to the BIB of the party B IP phone for the far-end (customer) voice.

14 The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the party B (agent) voice through a SIP trunk. The agent IP phone starts to fork the near-end (agent) voice stream to the recorder.

15 The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the far-end (customer) voice through a SIP trunk. The agent IP phone starts to fork the customer voice stream to the recorder.

16 Party B completes the consultation conference by pressing Confn. All parties connect to the conference bridge (CFB_2).

17 Because party B is now connected to a new far-end party, CFB_2, Cisco Unified Communications Manager sends two UPDATE messages to the recorder.

18 The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the party B (agent) voice through a SIP trunk. The agent IP phone forks the agent voice stream to the recorder.

19 The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for CFB_2 (conference bridge) through a SIP trunk. The agent IP phone forks the conference voice stream to the recorder.

Note the following particularities of call processing that apply in this use case:

- Near-end party creates a conference; the first recording session gets torn down
- Connection of the consultation call re-establishes the recording session.
- Far-end party changes to CFB and Cisco Unified Communications Manager sends SIP UPDATE message to the recorder.

Note the header information of the INVITE messages from step 5 and step 14, and the UPDATE message from step 17. The SIP header enhancement feature adds the information in bold text to the INVITE and UPDATE message headers.

**Step 5 INVITE Message Header Information**

From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;x-farendrefci=ci1;x-farenddevice=deviceA;x-farendaddr=1000>;tag=fromtag1
Step 14 INVITE Message Header Information

From: <sip:2000@ucm1;x-nearend;x-refci=ci3;x-nearenddevice=deviceB;x-farendrefci=ci4;x-farenddevice=deviceC;x-farendaddr=1100;tag=fromtag1

Step 17 UPDATE Message Header Information

From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;x-farendrefci=ci7;x-farenddevice=CFB_2;x-farendaddr=B001234567;isfocus;tag=fromtag1

The UPDATE message in step 17 includes isfocus. This isfocus indicates that the near-end party is participating in a conference call. The UPDATE message also includes a b-number as the new far-end address. The b-number specifies the DN of the conference bridge (CFB).

Far-End Party in Remote Cluster Transfers Call to Another Party

In this use case for automatic call recording, the far-end party in a remote cluster transfers the call to another party in the remote cluster. The following figure illustrates this use case.

Figure 16: Far-End Party in Remote Cluster Transfers Call to Another Party in the Remote Cluster

In this use case, the following entities participate:

- The customer call originates from DN 3000 device D in cluster Cisco Unified CM2.
- The agent receives the call at DN 2000 device B in cluster Cisco Unified CM1.
- Agent D transfers the call to DN 3100 device E in cluster Cisco Unified CM2.

During an automatic call recording session where the far-end (agent) party in the remote cluster transfers the call to another party in the remote cluster, the following steps take place:

1. Party D (far-end party = customer in remote cluster) calls party B (near-end party = agent) in local cluster by dialing 82000.
2. The remote cluster (Cisco Unified CM2) sends an INVITE message to the local cluster (Cisco Unified CM1) through a SIP trunk. The message contains information about party D.
3. Party B (near-end party = agent in local cluster) answers the call.
4. Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager first makes a recording call to the built-in bridge (BIB) of the party B (agent) IP phone for the agent voice.
5. Cisco Unified Communications Manager makes the second recording call to the BIB of the party B (agent) IP phone for the customer voice.
The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone starts to fork the party B (agent) voice stream to the recorder.

The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone starts to fork the party D (customer) voice stream to the recorder.

Party D in the remote cluster initiates a transfer (presses Transfer) and dials DN 3100 device E, which is also in the remote cluster.

Party E answers the call.

Party D completes the transfer by pressing Transfer.

The remote cluster (Cisco Unified CM2) sends an INVITE message to the local cluster (Cisco Unified CM1) through a SIP trunk. The message contains information about party E.

Because party B is now connected to a new far-end party, party E, Cisco Unified Communications Manager sends two UPDATE messages to the recorder.

The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the party B (agent) voice through a SIP trunk. The agent IP phone forks the agent voice stream to the recorder.

The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the party E (customer) voice through a SIP trunk. The agent IP phone forks the customer voice stream to the recorder.

Note the following particularities of call processing that apply in this use case:

- Far-end party and transfer-to party are both in the remote (Cisco Unified CM2) cluster. The near-end party sees the far-end party via the SIP trunk that links the two clusters.
- When the transfer-to party answers, the recorder receives an UPDATE message that contains the far-end address.

Note the header information of the INVITE message from step 6 and the UPDATE message from step 12. The SIP header enhancement feature adds the information in bold text to the message headers.

**Step 6 INVITE Message Header Information**

```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;
x-farendrefci=ci1;x-farenddevice=SIPTrunkTocluster2;x-farendaddr=3000>;tag=fromtag1
```

**Step 12 UPDATE Message Header Information**

```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;
x-farendrefci=ci1;x-farenddevice=SIPTrunkTocluster2;x-farendaddr=3100>;tag=fromtag1
```
Far-End Party in Remote Cluster Blind-Transfers Call

In this use case for automatic call recording, the far-end party in the remote cluster blind-transfers the call to a remote party that does not answer and has Call Forward No Answer (CFNA) configured. The following figure illustrates this use case.

**Figure 17: Far-End Party in Remote Cluster Blind-Transfers Call To Remote-Cluster Party That Has CFNA Configured**

In this use case, the following entities participate:

- The customer call originates from DN 3000 device D in cluster Cisco Unified CM2.
- The agent receives the call at DN 2000 device B in cluster Cisco Unified CM1.
- Agent D blind-transfers the call to DN 3100 device E in cluster Cisco Unified CM2.
- Agent E does not answer and the call forwards to DN 3200 device F in cluster Cisco Unified CM2.

During an automatic call recording session where the far-end (agent) party in the remote cluster blind-transfers the call to another party in the remote cluster, but the second party does not answer and the call forwards to a third party in the remote cluster, the following steps take place:

1. Party D (far-end party = customer in remote cluster) calls party B (near-end party = agent) in local cluster by dialing 82000.
2. The remote cluster (Cisco Unified CM2) sends an INVITE message to the local cluster (Cisco Unified CM1) through a SIP trunk. The message contains information about party D.
3. Party B (near-end party = agent in local cluster) answers the call.
4. Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager first makes a recording call to the built-in bridge (BIB) of the party B (agent) IP phone for the agent voice.
5. Cisco Unified Communications Manager makes the second recording call to the BIB of the party B (agent) IP phone for the customer voice.
6. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone starts to fork the party B (agent) voice stream to the recorder.
7. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone starts to fork the party D (customer) voice stream to the recorder.
8. Party D in the remote cluster initiates a transfer (presses Transfer) and dials DN 3100 device E, which is also in the remote cluster.
9. Party E does not answer the call: ringing times out, so Cisco Unified Communications Manager sends the call to party F DN 3200 device F.

10. The remote cluster (Cisco Unified CM2) sends an UPDATE message to the local cluster (Cisco Unified CM1) through a SIP trunk. The message contains information about party E.

11. Because party B is now connecting to a new far-end party, party E, local Cisco Unified Communications Manager sends two UPDATE messages to the recorder.

12. The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the party B (agent) voice through a SIP trunk. The agent IP phone forks the agent voice stream to the recorder.

13. The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the party E (customer) voice through a SIP trunk. The agent IP phone forks the customer voice stream to the recorder.

14. Party F answers the forwarded call.

15. The remote cluster (Cisco Unified CM2) sends an UPDATE message to the local cluster (Cisco Unified CM1) through a SIP trunk. The message contains information about party F.

16. Because party B is now connected to a new far-end party, party F, Cisco Unified Communications Manager sends two UPDATE messages to the recorder.

17. The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the party B (agent) voice through a SIP trunk. The agent IP phone forks the agent voice stream to the recorder.

18. The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the party F (customer) voice through a SIP trunk. The agent IP phone forks the customer voice stream to the recorder.

Note the following particularities of call processing that apply in this use case:

- Far-end part D in remote cluster transfers call to party E in remote cluster; the remote Cisco Unified Communications Manager updates the recorder.
- Party E CFNA timer expires and Cisco Unified Communications Manager redirects call to party F; the remote Cisco Unified Communications Manager again updates the recorder.
- The call state of the local Cisco Unified Communications Manager remains call-active, so Cisco Unified Communications Manager updates the recorder for each forwarded remote device.

Note the header information of the INVITE messages from step 6, step 11, and step 15. The SIP header enhancement feature adds the information in bold text to the INVITE and UPDATE message headers.

**Step 6 INVITE Message Header Information**
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB; x-farendrefci=ci1;x-farenddevice=SIPTrunkTocluster2;x-farendaddr=3000>;tag=fromtag1

**Step 11 UPDATE Message Header Information**
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB; x-farendrefci=ci1;x-farenddevice=SIPTrunkTocluster2;x-farendaddr=3100>;tag=fromtag1

**Step 15 UPDATE Message Header Information**
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB; x-farendrefci=ci1;x-farenddevice=SIPTrunkTocluster2;x-farendaddr=3200>;tag=fromtag1
Far-End Party in Remote PBX Transfers Call to Phone in Local Cluster

In this use case for automatic call recording, the far-end party in a remote PBX transfers a call to a phone in the local cluster. The following figure illustrates this use case.

**Figure 18: Far-End Party in Remote PBX Transfers Call to Phone in Local Cluster**

In this use case, the following entities participate:

- The customer call originates from DN 3000 device D in PBX1.
- The agent receives the call at DN 2000 device B in cluster Cisco Unified CM1.
- Agent D transfers the call to DN 1000 device A in cluster Cisco Unified CM1.

During an automatic call recording session where the far-end (agent) party in a remote PBX transfers the call to another party in the local cluster, the following steps take place:

1. Party D (far-end party = customer in remote PBX) calls party B (near-end party = agent) in local cluster by dialing 82000.
2. The remote PBX sends an setup message to the local cluster (Cisco Unified CM1) through a PRI QSIG gateway. The message contains information about party D.
3. Party B (near-end party = agent in local cluster) answers the call.
4. Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager first makes a recording call to the built-in bridge (BIB) of the party B (agent) IP phone for the agent voice.
5. Cisco Unified Communications Manager makes the second recording call to the BIB of the party B (agent) IP phone for the customer voice.
6. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone starts to fork the party B (agent) voice stream to the recorder.
7. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone starts to fork the party D (customer) voice stream to the recorder.
8. Party D in the remote PBX initiates a consultation transfer (presses Transfer) and dials DN 81000 device A, which is in the local cluster.
9. The remote PBX sends an setup message to the local cluster (Cisco Unified CM1) through a PRI QSIG gateway. The message contains information about party D.
10 Party A answers the call from party D.
11 Party D presses Transfer to complete the transfer.
12 Remote PBX sends UPDATE.
13 Remote PBX sends UPDATE.
14 Because party B is now connecting to a new far-end party, party A, local Cisco Unified Communications Manager sends two UPDATE messages to the recorder.
15 The recorder receives and answers the recording call UPDATE message that is sent from local Cisco Unified Communications Manager for the party B (agent) voice through a SIP trunk. The agent IP phone forks the agent voice stream to the recorder.
16 The recorder receives and answers the recording call UPDATE message that is sent from local Cisco Unified Communications Manager for the party A (customer) voice. The agent IP phone forks the customer voice stream to the recorder.

Note the following particularities of call processing that apply in this use case:

• When the far-end party in a remote cluster transfers the call to a party in the local cluster, Cisco Unified Communications Manager sends a SIP UPDATE message with far-endaddr for the transferred-to local-cluster party.

• This transfer specifies a hairpin transfer: the far-end address changed to the local DN 1000 in the UPDATE message.

Note the header information of the INVITE messages from step 6 and step 14. The SIP header enhancement feature adds the information in bold text to the INVITE and UPDATE message headers.

**Step 6 INVITE Message Header Information**
```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB; x-farendrefci=ci1;x-farenddevice=PrivQSIGGW;x-farendaddr=3000>;tag=fromtag1
```

**Step 14 UPDATE Message Header Information**
```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB; x-farendrefci=ci1;x-farenddevice=PrivQSIGGW;x-farendaddr=1000>;tag=fromtag1
```

**Remote PBX Far-End Party Transfers Call to Local Phone**

In this use case for automatic call recording, a remote PBX far-end party transfers the call to a local phone by using path replacement. The following figure illustrates this use case.

**Figure 19: Remote PBX Far-End Party Transfers Call to Local Phone With Path Replacement**
In this use case, the following entities participate:

- The customer call originates from DN 3000 device D in PBX1.
- The agent receives the call at DN 2000 device B in cluster Cisco Unified CM1.
- Agent D transfers the call to DN 1000 device A in cluster Cisco Unified CM1.

During an automatic call recording session where the far-end (agent) party in a remote PBX transfers the call to a phone in the local cluster by using path replacement, the following steps take place:

1. Party D (far-end party = customer in remote PBX) calls party B (near-end party = agent) in local cluster by dialing 82000.
2. The remote PBX sends an setup message to the local cluster (Cisco Unified CM1) through a PRI QSIG gateway. The message contains information about party D.
3. Party B (near-end party = agent in local cluster) answers the call.
4. Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager first makes a recording call to the built-in bridge (BIB) of the party B (agent) IP phone for the agent voice.
5. Cisco Unified Communications Manager makes the second recording call to the BIB of the party B (agent) IP phone for the customer voice.
6. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone starts to fork the party B (agent) voice stream to the recorder.
7. The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone starts to fork the party D (customer) voice stream to the recorder.
8. Party D in the remote PBX initiates a consultation transfer (presses Transfer) and dials DN 81000 device A, which is in the local cluster.
9. The remote PBX sends an setup message to the local cluster (Cisco Unified CM1) through a PRI QSIG gateway. The message contains information about party D.
10. Local party A answers the call from party D.
11. Remote party D presses Transfer to complete the transfer.
12. Remote PBX sends UPDATE.
13. Remote PBX sends UPDATE.
14. Because party B is now connecting to a new far-end party, party A, local Cisco Unified Communications Manager sends two UPDATE messages to the recorder.
15. The recorder receives and answers the recording call UPDATE message that is sent from local Cisco Unified Communications Manager for the party B (agent) voice through a SIP trunk. The agent IP phone forks the agent voice stream to the recorder.
16. The recorder receives and answers the recording call UPDATE message that is sent from local Cisco Unified Communications Manager for the party A (customer) voice. The agent IP phone forks the customer voice stream to the recorder.
17. Local Cisco Unified Communications Manager initiates path replacement process directly connects device A with device B and eliminates routing through the remote PBX.
18. Because party B is now connecting to a new far-end party, party A, local Cisco Unified Communications Manager sends two UPDATE messages to the recorder.
TherecorderreceivesandanswerstherecordingcallUPDATEmessage thatissentfromlocalCiscoUnifiedCommunications ManagerforthepartyB(agent)voicethroughaSIPtrunk.TheagentIPphoneforks theagentvoicestreamtotherecorder.

The recorder receives and answers the recording call UPDATE message that is sent from local Cisco Unified Communications Manager for the party A (customer) voice. The agent IP phone forks the customer voice stream to the recorder.

Note the following particularities of call processing that apply in this use case:

• Path replacement replaces a hairpin call to remote PBX so that party A and party B are directly connected without routing through the remote PBX.
• The far-end call information gets updated when transfer completes.
• When path replacement completes, the far-end device also gets updated.

Note the header information of the INVITE messages from step 6, step 14, and step 17. The SIP header enhancement feature adds the information in bold text to the INVITE and UPDATE message headers.

**Step 6 INVITE Message Header Information**

From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;
x=farendrefci=ci1;x-farenddevice=PriQSIGGW;x-farendaddr=3000>;tag=fromtag1

**Step 14 UPDATE Message Header Information**

From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;
x=farendrefci=ci1;x-farenddevice=PriQSIGGW;x-farendaddr=1000>;tag=fromtag1

**Step 17 UPDATE Message Header Information**

From: <sip:2000@ucm1;x-nearend;x-refci=ci4;x-nearenddevice=deviceA;x-farendaddr=1000>;tag=fromtag1

**Far-End Party Transfers Call Across DMS Gateway**

In this use case for automatic call recording, the far-end party transfers a call across a DMS gateway. The following figure illustrates this use case.

**Figure 20: Far-End Party Transfers Call Across DMS Gateway**

In this use case, the following entities participate:
• The customer call originates from DN 9725550001 device D that connects to a DMS switch.
• The agent receives the call at DN 2000 device B in cluster Cisco Unified CM1.
• Agent D transfers the call to DN 9725550002 device E that connects to a DMS switch.

During an automatic call recording session where the far-end (agent) party that connects through a DMS switch transfers the call to a phone that also connects through a DMS switch, the following steps take place:

1. Party D (far-end party = customer across DMS switch) calls party B (near-end party = agent) in local cluster by dialing 82000.
2. The DMS switch sends a PriSetup message to the local cluster (Cisco Unified CM1) through a PRI DMS gateway. The message contains information about party D.
3. Party B (near-end party = agent in local cluster) answers the call.
4. Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager first makes a recording call to the built-in bridge (BIB) of the party B (agent) IP phone for the agent voice.
5. Cisco Unified Communications Manager makes the second recording call to the BIB of the party B (agent) IP phone for the customer voice.
6. The recorder receives and answers the recording call setup messages that are sent from local Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone starts to fork the party B (agent) voice stream to the recorder.
7. The recorder receives and answers the recording call setup messages that are sent from local Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone starts to fork the party D (customer) voice stream to the recorder.
8. Party D across the DMS gateway initiates a consultation transfer (presses Transfer) and dials DN 9725550002 device E, which is also across the DMS gateway.
9. Party E answers the call from party D.
10. The DMS switch sends a PriNotify message to the local cluster (Cisco Unified CM1) through a PRI DMS gateway. The message contains information about party E.
11. Because party B is now connecting to a new far-end party, party E, local Cisco Unified Communications Manager sends two UPDATE messages to the recorder.
12. The recorder receives and answers the recording call UPDATE message that is sent from local Cisco Unified Communications Manager for the party B (agent) voice through a SIP trunk. The agent IP phone forks the agent voice stream to the recorder.
13. The recorder receives and answers the recording call UPDATE message that is sent from local Cisco Unified Communications Manager for the party E (customer) voice. The agent IP phone forks the customer voice stream to the recorder.

Note the following particularities of call processing that apply in this use case:

• In general, when a far-end party on the PSTN side transfers a call, Cisco Unified Communications Manager does not know about the transferring nor transferred-to parties. If the DMS switch or PBX supports PriNotify, however, Cisco Unified Communications Manager receives the PriNotify message when the far-end party changes and can update the far-end information to the recorder.

Note the header information of the INVITE messages from step 6 and step 11. The SIP header enhancement feature adds the information in bold text to the INVITE and UPDATE message headers.
Step 6 INVITE Message Header Information
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;x-farendrefci=ci1;x-farenddevice=PriDMSGW;x-farendaddr=9725550001;tag=fromtag1>

Step 11 UPDATE Message Header Information
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;x-farendrefci=ci1;x-farenddevice=PriDMSGW;x-farendaddr=9725550002;tag=fromtag1>

Desktop Pickup of Mobile Phone Call
In this use case for automatic call recording, mobile phone user sends call to desk phone for pickup. The following figure illustrates this use case.

Figure 21: Desktop Pickup of Mobile Phone Call

In this use case, the following entities participate:

- The customer call originates from mobile device UserACell, enterprise extension 1000 and mobile number 9725551000.
- The agent receives the call at DN 2000 device B.
- The customer resumes the call from the customer enterprise phone DN 1000 device A.

During an automatic call recording session where desktop pickup of a mobile phone call occurs, the following steps take place:

1. UserACell calls party B DN 2000 device B.
2. Cisco Unified Mobile Communicator client sends SETUP message
3. SETUP message travels through H.323 gateway to local Cisco Unified CM1 cluster.
4. Party B answers the incoming call from UserACell.
5. Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager first makes a recording call to the built-in bridge (BIB) of the party B (agent) IP phone for the agent voice.
6. Cisco Unified Communications Manager makes the second recording call to the BIB of the party B (agent) IP phone for the customer voice.
The recorder receives and answers the recording INVITE messages that are sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone starts to fork the party B (agent) voice stream to the recorder.

The recorder receives and answers the recording INVITE messages that are sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone starts to fork the party UserACell (customer) voice stream to the recorder.

UserACell presses Enterprise Hold on the user A mobile phone.

User A presses Resume on the user A desk phone.

The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone forks the agent voice stream to the recorder.

The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone forks the customer voice stream to the recorder.

Note the following particularities of call processing that apply in this use case:

- Mobile phone uses the shared-line concept. When the mobile phone call gets put on hold and the desktop phone resumes the call, the far-end party changes. Cisco Unified Communications Manager sends an update to the recorder.
- The user picks up the call from the user desk phone to continue the conversation that started on the user mobile phone. To do so, use Cisco Unified Communications Manager to place the call on hold (enterprise hold) through the mobile phone data channel; then, resume the call from the desk phone.

Note the header information of the INVITE messages from step 7 and step 11. The SIP header enhancement feature adds the information in bold text to the INVITE and UPDATE message headers.

**Step 7 INVITE Message Header Information**

```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;x-farendrefci=ci1;x-farenddevice=UserACell;x-farendaddr=1000>;tag=fromtag1
```

**Step 11 UPDATE Message Header Information**

```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;x-farendrefci=ci1;x-farenddevice=deviceA;x-farendaddr=1000>;tag=fromtag1
```
Far-End Party Sends Call to Mobile Phone

In this use case for automatic call recording, the far-end party sends a call to the user mobile phone for mobile phone pickup. (This scenario specifies the opposite of the scenario that the Desktop Pickup of Mobile Phone Call, on page 32 specifies.) The following figure illustrates this use case.

Figure 22: Far-End Party Sends Call to Mobile Phone for Mobile Phone Pickup

In this use case, the following entities participate:

- The customer call originates from DN 1000 device A.
- The agent receives the call at DN 2000 device B.
- The customer resumes the call from the customer mobile phone device UserACell enterprise extension 1000 mobile number 9725551000.

During an automatic call recording session where an enterprise user sends a call to the user mobile phone, the following steps take place:

1. Enterprise user far-end party A calls party B DN 2000 device B from DN 1000 device A.
2. Party B DN 2000 answers the incoming call from far-end party A.
3. Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager first makes a recording call to the built-in bridge (BIB) of the party B (agent) IP phone for the agent voice.
4. Cisco Unified Communications Manager makes the second recording call to the BIB of the party B (agent) IP phone for the customer voice.
5. The recorder receives and answers the recording INVITE messages that are sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone starts to fork the party B (agent) voice stream to the recorder.
6. The recorder receives and answers the recording INVITE messages that are sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone starts to fork the party A (customer) voice stream to the recorder.
7. Party A presses Send to Mobile on the user A desk phone.
8. Cisco Unified Communications Manager sends a Setup message to the User A mobile phone.
9. User A presses answers the ringing call on device UserACell.
10 The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone forks the agent voice stream to the recorder.

11 The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone forks the customer voice stream to the recorder.

Note the following particularities of call processing that apply in this use case:

- When party UserACell answers the call, the party UserACell information is sent in a SIP UPDATE message to the recorder.

Note the header information of the INVITE messages from step 5 and step 10. The SIP header enhancement feature adds the information in bold text to the INVITE and UPDATE message headers.

**Step 5 INVITE Message Header Information**

```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;x-farendrefci=ci1;x-farenddevice=deviceA;x-farendaddr=1000>;tag=fromtag1
```

**Step 10 UPDATE Message Header Information**

```
From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;x-farendrefci=ci3;x-farenddevice=UserACell;x-farendaddr=1000>;tag=fromtag1
```

**Far-End Party in Remote Cluster Creates Conference**

In this use case for automatic call recording, the far-end party in a remote cluster creates a conference. The following figure illustrates this use case.

**Figure 23: Far-End Party in Remote Cluster Creates Conference**

In this use case, the following entities participate:

- The far-end customer call originates from DN 3000 device D.
- The near-end agent receives the call at DN 2000 device B.
- Party D creates a conference by conferencing in DN 3100 device E.

During an automatic call recording session where the far-end party in a remote cluster creates a conference, the following steps take place:

1 Party D (far-end party = customer in remote cluster) calls party B (near-end party = agent) by dialing 82000.
The INVITE message passes over the SIPTrunkToCluster2 SIP trunk.

Party B (near-end party = agent in local cluster) answers the call.

Because the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. Cisco Unified Communications Manager first makes a recording call to the built-in bridge (BIB) of the party B (agent) IP phone for the agent voice.

Cisco Unified Communications Manager makes the second recording call to the BIB of the party B (agent) IP phone for the customer voice.

The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the agent voice through a SIP trunk. The agent IP phone starts to fork the party B (agent) voice stream to the recorder.

The recorder receives and answers the recording call setup messages that are sent from Cisco Unified Communications Manager for the customer voice through a SIP trunk. The agent IP phone starts to fork the party D (customer) voice stream to the recorder.

Party D initiates a conference by pressing Confn and dialing DN 3100.

Party E DN 3100 device E answers the call.

Party D completes the conference by pressing Confn again.

The UPDATE message passes over the SIPTrunkToCluster2 SIP trunk.

Because party B is now connected to a new far-end party, CFB_2, Cisco Unified Communications Manager sends two UPDATE messages to the recorder.

The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for the party B (agent) voice through a SIP trunk. The agent IP phone forks the agent voice stream to the recorder.

The recorder receives and answers the recording call UPDATE message that is sent from Cisco Unified Communications Manager for CFB_2 (conference bridge) through a SIP trunk. The agent IP phone forks the conference voice stream to the recorder.

Note the following particularities of call processing and configuration that apply in this use case:

- Cisco Unified Communications Manager has very limited view if a far-end party creates a conference in a remote cluster. The far-end party device is always the trunk that links the two clusters.

- After the remote conference gets established, the remote Cisco Unified CM2 cluster delivers the b-number (conference bridge identifier) in the SIP UPDATE to the local cluster, Cisco Unified CM1. The Cisco Unified CM1 cluster sends the update to the recorder with the b-number and isfocus indicator.

- In the figure, the Cisco Unified CM1 cluster gets configured with a SIP trunk, SIPTrunkToCluster2, that links the Cisco Unified CM1 cluster to the Cisco Unified CM2 cluster. The corresponding SIP trunk that is configured in the Cisco Unified CM2 cluster specifies SIPTrunkToCluster1.

- When the conference gets created by the far-end party with DN 3000, the conference bridge identifier, b001234567, does not get passed to cluster Cisco Unified CM1 by default. If the identifier is not passed, the Cisco Unified CM1 cluster can still include the isfocus flag for the far-end party in the From header to the recorder, but the far-end party address will be empty.

- To allow the conference bridge identifier (the b-number of the conference bridge) to pass from cluster Cisco Unified CM2 to cluster Cisco Unified CM1, the administrator creates a SIP profile in cluster Cisco Unified CM2, checks the Deliver Conference Bridge Identifier check box, and assigns the SIP profile to SIPTrunkToCluster1 in cluster Cisco Unified CM2. The administrator also creates a SIP profile in cluster Cisco Unified CM1 and assigns this SIP profile to the SIPTrunkToCluster2 in cluster Cisco Unified CM1.

- For more details about configuring SIP profiles, see the Cisco Unified Communications Manager Administration Guide.
Note the header information of the INVITE messages from step 6 and step 12. The SIP header enhancement feature adds the information in bold text to the INVITE and UPDATE message headers.

**Step 6 INVITE Message Header Information**

From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;x-farendrefci=ci1;x-farenddevice=SIPTrunkToCluster2;x-farendaddr=3000>;tag=fromtag1

**Step 12 UPDATE Message Header Information**

From: <sip:2000@ucm1;x-nearend;x-refci=ci2;x-nearenddevice=deviceB;x-farendrefci=ci1;x-farenddevice=SIPTrunkToCluster2;x-farendaddr=b001234567;isfocus>;tag=fromtag1

The UPDATE message in step 12 includes isfocus. This isfocus indicates that the near-end party is participating in a conference call. The UPDATE message also includes a b-number as the new far-end address. The b-number specifies the DN of the conference bridge (CFB).

**Selective Call Recording**

In selective call recording, recording can be initiated using a softkey or programmable line key assigned to the device, a CTI-enabled application, or both interchangeably.

Selective call recording supports two modes:

**Silent Recording**

In the silent recording mode the call recording status is *not* reflected on the Cisco IP device display. Silent recording is typically used in a call center environment to enable a supervisor to record an agent call. A CTI-enabled application running on the supervisor desktop is generally used to start and stop the recording for the agent-customer call.

**User Recording**

In the user recording mode the call recording status is reflected on the Cisco IP device display. A recording may be started or stopped using a softkey, programmable line key, or CTI-enabled application running on the user desktop.

**Selective Call Recording Silent Recording mode**

In this use case for selective call recording silent recording mode, the supervisor manages the recording session from a CTI-enabled desktop.
The following figure illustrates selective call recording silent recording mode.

**Figure 24: Selective Call Recording Silent Recording Mode**

1. Dial 2000
2. Answer
3. Press start record
4. Invite near-end-relay record-invoker-silent
5. Invite far-end-relay record-invoker-silent
6. Press the StopRecord Button
7. Bye
8. Bye

In this use case:

- Device A (DN 1000) calls device B (DN 2000).
- The user on device B answers the call.
- The supervisor uses a CTI-enabled application to start and stop the call recording session.

The supervisor starts the recording session using a CTI-enabled desktop. Cisco Unified Communications Manager makes two recording calls to the built-in bridge (BIB) of the Cisco IP device (one call for the near-end media; another call for the far-end media). The recording calls are automatically and silently answered.

Cisco Unified Communications Manager extends both recording calls to the recording server connected by a SIP trunk.

The recorder receives and answers both recording calls and the user Cisco IP device starts to fork both media streams to the recorder. The call recording status is not reflected on the Cisco IP device display.
The CTI-enabled application is used to stop the recording session. Cisco Unified Communications Manager clears both recording calls.

**Selective Call Recording User Recording Mode—Recording Session Managed From a CTI-enabled Application**

In this use case for selective call recording user recording mode, the user manages the recording session from a CTI-enabled desktop. The following figure illustrates selective call recording user recording mode—recording session managed from a CTI-enabled application.

*Figure 25: User Recording Mode Managed from a CTI-Enabled Application*

In this use case:

- Device A (DN 1000) calls device B (DN 2000).
- The user on device B answers the call.
- The user on device B uses a CTI-enabled application to start and stop the call recording session.

The user starts the recording session using a CTI-enabled desktop. Cisco Unified Communications Manager makes two recording calls to the built-in bridge (BIB) of the Cisco IP device (one call for the near-end media; another call for the far-end media). The recording calls are automatically and silently answered.

Cisco Unified Communications Manager extends both recording calls to the recording server connected by a SIP trunk.
The recorder receives and answers both recording calls and the user Cisco IP device starts to fork both media streams to the recorder. The Cisco IP device display updates to reflect Recording. The key toggles from Record to StopRec.

The CTI-enabled application is used to stop the recording session. Cisco Unified Communications Manager clears both recording calls.

Selective Call Recording User Recording Mode—Recording Session Managed From a Cisco IP Device

In this use case for selective call recording user recording mode, the user presses a softkey or programmable line key on a Cisco IP device to start and stop the call recording session.

The following figure illustrates selective call recording user recording mode—recording session managed from a Cisco IP device.

Figure 26: User Recording Mode Managed from a Cisco IP Device

In this use case:

- Device A (DN 1000) calls device B (DN 2000).
- The user on device B answers the call.
• The user on device B uses the softkey or programmable line key on the Cisco IP device to start and stop the call recording session.

The user presses the *Record* key to start the recording session. Cisco Unified Communications Manager makes two recording calls to the built-in bridge (BIB) of the Cisco IP device (one call for the near-end media; another call for the far-end media). The recording calls are automatically and silently answered.

Cisco Unified Communications Manager extends both recording calls to the recording server connected by a SIP Trunk.

The recorder receives and answers both recording calls and the user Cisco IP device starts to fork both media streams to the recorder. The Cisco IP device display updates to reflect *Recording*. The key toggles from *Record* to *StopRec*.

The User presses the *StopRec* key to stop the recording session.

Cisco Unified Communications Manager clears both recording calls.
Cisco has more than 200 offices worldwide. Addresses, phone numbers, and fax numbers are listed on the Cisco Website at www.cisco.com/go/offices.