Network-Based Recording Using Cisco UBE

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The Network-Based Recording Using Cisco UBE feature supports software-based forking for Real-time Transport Protocol (RTP) streams. Media forking provides the ability to create midcall multiple streams (or branches) of audio associated with a single call and then send the streams of data to different destinations. You can enable Network-Based Recording using Cisco Unified Border Element (Cisco UBE) by configuring specific CLI commands on Cisco UBE or through a call agent. Cisco UBE acts as a recording client and MediaSense Session Initiation Protocol (SIP) recorder acts a recording server.

Functionalities of the recording client, Cisco UBE that is present in signaling and media path of the communication session, are as follows:

- Acts as a SIP user agent and sets up a recording session (SIP dialog) with the recording server.
- Acts as the source of the recorded media and forwards the recorded media to the recording server.
- Sends information periodically to a server that helps the recording server associate the call with media streams and identifies the participants of the call. This information sent to the recording server is called metadata.

MediaSense SIP recorder acts as the recording server. A recording server is a SIP user agent that archives media for extended durations. It provides search and retrieval of the archived media. The recording server is a storage place of the recorded session metadata.

The metadata carried in the SIP session between the recording client and the recording server is to:

- Carry the communication session data that describes the call.
- Send the metadata to the recording server. The recording server uses the metadata to associate communication sessions involving two or more participants with media streams.

The call leg that is created between the recording client and the recording server is known as the recording session.

- Prerequisites for Network-Based Recording Using Cisco UBE, page 2
- Restrictions for Network-Based Recording Using Cisco UBE, page 2
- Information About Network-Based Recording Using Cisco UBE, page 2
- How to Configure Network-Based Recording using Cisco UBE, page 5

- Configuration Examples for Network-Based Recording using Cisco UBE, page 14
- Feature Information for Network-Based Recording using Cisco UBE, page 15

Prerequisites for Network-Based Recording Using Cisco UBE

You must have an ISR G2 router equipped with the unified communication technology package configured as a Cisco UBE in flow-through mode for the Network-Based Recording Using Cisco UBE feature to function.

Cisco Unified Border Element

- Cisco IOS Release 15.2(1)T or a later release must be installed and running on your Cisco Unified Border Element.

Cisco Unified Border Element (Enterprise)

- Cisco IOS XE Release 3.8S or a later release must be installed and running on your Cisco ASR 1000 Series Router.

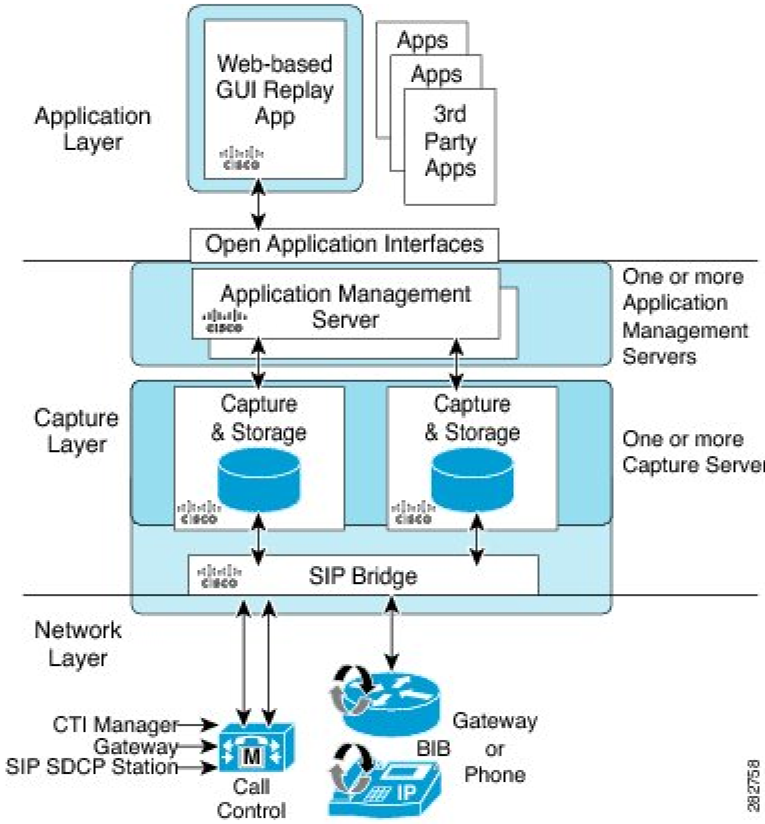
Restrictions for Network-Based Recording Using Cisco UBE

- The Network-Based Recording Using Cisco UBE feature is supported only for SIP-SIP call flows.
- The Network-Based Recording Using Cisco UBE feature is supported only on ISR G2 platforms (2901, 2911, 2921, 2951, 3945, 3945E).

Information About Network-Based Recording Using Cisco UBE

Open Recording Architecture

The Open Recording Architecture (ORA) comprises of elements, such as application management server and SIP bridge, to support IP-based recording. The ORA IP enables recording by solving topology issues, which accelerates the adoption of Cisco unified communication solutions.



Following are the three layers of the ORA architecture:

Network Layer

The ORA network layer is comprises call control systems, media sources, and IP foundation components, such as routers and switches.

Capture and Media Processing Layer

The ORA capture and media processing layer includes core functions of ORA--terminating media streams, storage of media and metadata, and speech analytics that can provide real-time events for applications.

Application Layer

The ORA application layer supports in-call and post-call applications through open programming interfaces. In-call applications include applications that make real-time business decisions like whether to record a particular call or not, controls pause and resume from Interactive Voice Response (IVR) or agent desktop systems, and performs metadata tagging and encryption key exchange at the call setup.

Post-call applications include the following:

- Traditional compliance search, replay, and quality monitoring.
- Advanced capabilities, such as speech analytics, transcription, and phonetic search.

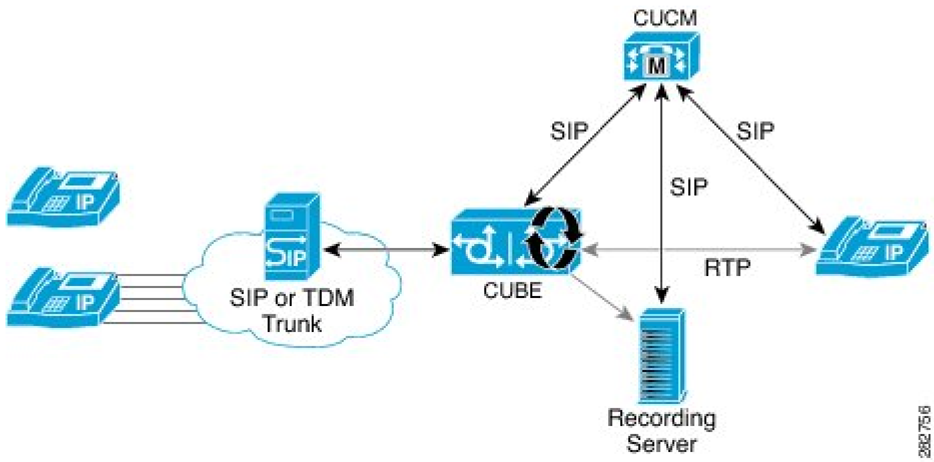
- Custom enterprise integration.
- Enterprise-wide policy management.

Media Forking Topologies

The following topologies support media forking:

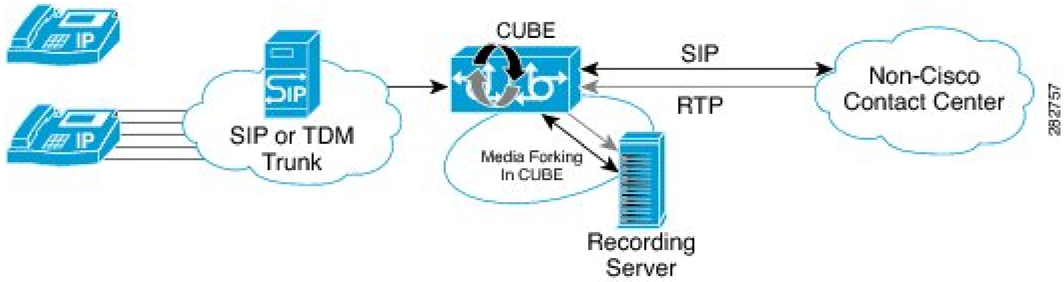
ORA SBC-Based Media Forking with Cisco UCM

ORA SBC-Based Media Forking with Cisco UCM shows the ORA SBC-based media forking with Cisco Unified CallManager (Cisco UCM) topology for media forking. This topology supports replication of media packets to allow recording the caller agent. It also enables Cisco UBE to establish full-duplex communication with the recording server. In this topology, SIP recording trunk is enhanced to have additional call metadata.



ORA SBC-Based Media Forking without Cisco UCM

ORA SBC-Based Media Forking without Cisco UCM shows the ORA SBC-based media forking without the Cisco UCM topology. This topology supports static configuration on Cisco UBE and the replication of media packets to allow recording caller-agent and full-duplex interactions at an IP call recording server.



SIP Recorder Interface

SIP is used as a protocol between Cisco UBE and the MediaSense SIP server. Extensions are made to SIP to carry the recording session information needed for the recording server. This information carried in SIP sessions between the recording client and the recording server is called metadata.

Metadata

Metadata is the information that is passed by the recording client to the recording server in a SIP session. Metadata describes the communication session and its media streams.

Metadata is used by the recording server to:

- Identify participants of the call.
- Associate media streams with the participant information. Each participant can have one or more media streams, such as audio, and video.
- Identify the participant change due to transfers during the call.

The recording server uses the metadata information along with other SIP message information, such as dialog ID and time and date header, to derive a unique key. The recording server uses this key to store media streams and associate the participant information with the media streams.

How to Configure Network-Based Recording using Cisco UBE

You can configure and verify the Network-Based Recording using Cisco UBE feature by using one of the following methods:

Configuring the Media Profile Recorder

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **media profile recorder** *profile-tag*
4. **media-recording** *dial-peer-tag* [*dial-peer-tag2...dial-peer-tag5*]
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable**	Enables privileged EXEC mode.

	Command or Action	Purpose
	Example: `Device> enable`	• Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.
Step 3	**media profile recorder** *profile-tag* **Example:** `Device(config)# media profile recorder 100`	Configures the media profile recorder and enters media profile configuration mode.
Step 4	**media-recording** *dial-peer-tag* [*dial-peer-tag2...dial-peer-tag5*] **Example:** `Device(cfg-mediaprofile)# media-recording 2000`	Sets voice-class recording parameters. **Note** You can specify a maximum of five dial-peer tags.
Step 5	**end** **Example:** `Device(cfg-mediaprofile)# end`	Exits media profile configuration mode.

Configuring the Media Class Globally

You can configure a media class globally by performing one of the following tasks:

Configuring a Media Class Using the Media Profile Recorder

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **media class** *tag*
4. **recorder profile** *tag*
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.
Step 3	**media class** *tag* **Example:** `Device(config)# media class 100`	Configures a media class and enters media class configuration mode.
Step 4	**recorder profile** *tag* **Example:** `Device(cfg-mediaclass)# recorder profile 100`	Configures the media profile recorder.
Step 5	**end** **Example:** `Device(cfg-mediaclass)# end`	Exits media class configuration mode.

Configuring Media Class Using the Recorder Parameter

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **media class** *tag*
4. **recorder parameter**
5. **media-recording** *dial-peer-tag*
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.
Step 3	**media class** *tag* **Example:** `Device(config)# media class 100`	Configures the media class and enters media class configuration mode.
Step 4	**recorder parameter** **Example:** `Device(cfg-mediaclass)# recorder parameter`	Enters media class recorder parameter configuration mode to enable you to configure recorder-specific parameters.
Step 5	**media-recording** *dial-peer-tag* **Example:** `Device(cfg-mediaclass-recorder)# media-recording 28`	Configures voice-class recording parameters. **Note** You can specify a maximum of five dial-peer tags.
Step 6	**end** **Example:** `Device(cfg-mediaclass-recorder)# end`	Exits media class recorder parameter configuration mode.

Configuring the Media Class for a Dial Peer

Before You Begin

You must configure a dial peer to connect to Cisco MediaSense. This dial peer is matched with Cisco Unified Border Element and a call is set up to Cisco MediaSense.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **session protocol sipv2**
5. **incoming called-number** *string*
6. **media-class** *tag*
7. **codec** *codec* [**bytes** *payload-size*] [**fixed-bytes**] [**mode** {**independent** | **adaptive**} [**bit-rate** *value*] [**framesize** {**30** | **60**} [**fixed**]]]
8. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* **voip** **Example:** `Device(config)# dial-peer voice 24 voip`	Defines a particular dial peer and enters dial peer voice configuration mode.
Step 4	**session protocol sipv2** **Example:** `Device(config-dial-peer)# session protocol sipv2`	Specifies SIP version 2 for calls between local and remote routers using the packet network.
Step 5	**incoming called-number** *string* **Example:** `Device(config-dial-peer)# incoming called-number 9845`	Specifies a digit string that can be matched with an incoming call to associate the call with a dial peer.
Step 6	**media-class** *tag* **Example:** `Device(config-dial-peer)# media-class 100`	Configures media class on a dial peer.

	Command or Action	Purpose
Step 7	**codec** *codec* [**bytes** *payload-size*] [**fixed-bytes**] [**mode** {**independent** | **adaptive**} [**bit-rate** *value*] [**framesize** {**30** | **60**} [**fixed**]]] **Example:** `Device(config-dial-peer)# codec g711ulaw`	Specifies the voice coder rate of speech for a dial peer.
Step 8	**end** **Example:** `Device(config-dial-peer)# end`	Exits dial peer configuration mode and returns to privileged EXEC mode.

Verifying the Network-Based Recording Using Cisco UBE Configuration

Perform this task to verify the configuration of the Network-Based Recording Using Cisco UBE Configuration feature. The **show** commands can be entered in any order.

SUMMARY STEPS

1. **enable**
2. **show voip rtp connections**
3. **show voip recmsp session**
4. **show voip recmsp session detail call-id** *call-id*
5. **show voip rtp forking**
6. **show call active voice compact**
7. **show sip-ua calls**

DETAILED STEPS

Step 1 **enable**

Enables privileged EXEC mode.

Example:

```
Device> enable
```

Step 2 **show voip rtp connections**

Real-Time Transport Protocol (RTP)-named.

Example:

```
Device# show voip rtp connections
```

```
VoIP RTP active connections :
No. CallId     dstCallId       LocalRTP RmtRTP     LocalIP                                RemoteIP
1     140        141            18792    18638     9.42.30.10                              9.42.30.32
2     141        140           19256    26184     9.42.30.10                              9.42.30.189
3     145         143            18648    38526     9.42.30.10                              9.42.29.7
4     146         143            17780    50482     9.42.30.10                              9.42.29.7
```

Step 3 **show voip recmsp session**

Displays active recording Media Service Provider (MSP) session information.

Example:

```
Device# show voip recmsp session
RECMSP active sessions:
MSP Call-ID             AnchorLeg Call-ID        ForkedLeg Call-ID
143                     141                              145
Found 1 active sessions
```

Step 4 **show voip recmsp session detail call-id** *call-id*

Displays detailed information about the recording MSP Call ID.

Example:

```
Device# show voip recmsp session detail call-id 145
RECMSP active sessions:
Detailed Information
=========================
Recording MSP Leg Details:
Call ID: 143
GUID : 7C5946D38ECD
AnchorLeg Details:
Call ID: 141
Forking Stream type: voice-nearend
Participant: 708090
Non-anchor Leg Details:
Call ID: 140
Forking Stream type: voice-farend
Participant: 10000
Forked Leg Details:
Call ID: 145
Near End Stream CallID 145
Stream State ACTIVE
Far End stream CallID 146
Stream State ACTIVE
Found 1 active sessions
```

Step 5 **show voip rtp forking**

Displays RTP media-forking connections.

Example:

```
Device# show voip rtp forking
VoIP RTP active forks :
 Fork 1
   stream type voice-only (0): count 0
   stream type voice+dtmf (1): count 0
   stream type dtmf-only (2): count 0
   stream type voice-nearend (3): count 1
     remote ip 9.42.29.7,  remote port 38526,  local port 18648
       codec g711ulaw,  logical ssrc 0x53
       packets sent 29687,  packets received 0
   stream type voice+dtmf-nearend (4): count 0
   stream type voice-farend (5): count 1
```

```
     remote ip 9.42.29.7,  remote port 50482,  local port 17780
       codec g711ulaw,  logical ssrc 0x55
       packets sent 29686,  packets received 0
   stream type voice+dtmf-farend (6): count 0
   stream type video (7): count
```

Step 6 **show call active voice compact**

Displays a compact version of voice calls in progress.

Example:

```
Device# show call active voice compact
<callID>  A/O FAX T<sec> Codec       type        Peer Address       IP R<ip>:<udp>
Total call-legs: 3
       140 ANS     T644   g711ulaw    VOIP         P10000       9.42.30.32:18638
       141 ORG     T644   g711ulaw    VOIP         P708090      9.42.30.189:26184
       145 ORG     T643   g711ulaw    VOIP         P595959        9.42.29.7:38526
```

Step 7 **show sip-ua calls**

Displays active user agent client (UAC) and user agent server (UAS) information on SIP calls.

Example:

```
Device# show sip-ua calls
Total SIP call legs:3, User Agent Client:2, User Agent Server:1
SIP UAC CALL INFO
Call 1
SIP Call ID                : 99EA5118-506211E0-80C6E01B-4C27AA62@9.42.30.10
   State of the call       : STATE_ACTIVE (7)
   Substate of the call    : SUBSTATE_NONE (0)
   Calling Number          : 10000
   Called Number           : 708090
   Bit Flags               : 0xC04018 0x10000100 0x80
   CC Call ID              : 141
   Source IP Address (Sig ): 9.42.30.10
   Destn SIP Req Addr:Port : [9.42.30.5]:5060
   Destn SIP Resp Addr:Port: [9.42.30.5]:5060
   Destination Name        : 9.42.30.5
   Number of Media Streams : 1
   Number of Active Streams: 1
   RTP Fork Object         : 0x0
   Media Mode              : flow-through
   Media Stream 1
     State of the stream      : STREAM_ACTIVE
     Stream Call ID           : 141
     Stream Type              : voice+dtmf (1)
     Stream Media Addr Type   : 1
     Negotiated Codec         : g711ulaw (160 bytes)
     Codec Payload Type       : 0
     Negotiated Dtmf-relay    : rtp-nte
     Dtmf-relay Payload Type  : 101
     QoS ID                   : -1
     Local QoS Strength       : BestEffort
     Negotiated QoS Strength  : BestEffort
     Negotiated QoS Direction : None
     Local QoS Status         : None
     Media Source IP Addr:Port: [9.42.30.10]:19256
     Media Dest IP Addr:Port  : [9.42.30.189]:26184
Options-Ping    ENABLED:NO    ACTIVE:NO
Call 2
SIP Call ID                : 9A6D8922-506211E0-80CEE01B-4C27AA62@9.42.30.10
   State of the call       : STATE_ACTIVE (7)
   Substate of the call    : SUBSTATE_NONE (0)
   Calling Number          :
   Called Number           : 595959                              Recoding server number
   Bit Flags               : 0xC04018 0x10800100 0x80
```

```
    CC Call ID               : 145
    Source IP Address (Sig ): 9.42.30.10
    Destn SIP Req Addr:Port : [9.42.29.7]:5060
    Destn SIP Resp Addr:Port: [9.42.29.7]:5060
    Destination Name        : 9.42.29.7
    Number of Media Streams : 2
    Number of Active Streams: 2
    RTP Fork Object         : 0x0
    Media Mode              : flow-through
    Media Stream 1
      State of the stream      : STREAM_ACTIVE
      Stream Call ID           : 145
      Stream Type              : voice-nearend (3)
      Stream Media Addr Type   : 1
      Negotiated Codec         : g711ulaw (160 bytes)
      Codec Payload Type       : 0
      Negotiated Dtmf-relay    : inband-voice
      Dtmf-relay Payload Type  : 0
      QoS ID                   : -1
      Local QoS Strength       : BestEffort
      Negotiated QoS Strength  : BestEffort
      Negotiated QoS Direction : None
      Local QoS Status         : None
      Media Source IP Addr:Port: [9.42.30.10]:18648
      Media Dest IP Addr:Port  : [9.42.29.7]:38526
    Media Stream 2
      State of the stream      : STREAM_ACTIVE
      Stream Call ID           : 146
      Stream Type              : voice-farend (5)
      Stream Media Addr Type   : 1
      Negotiated Codec         : g711ulaw (160 bytes)
      Codec Payload Type       : 0
      Negotiated Dtmf-relay    : inband-voice
      Dtmf-relay Payload Type  : 0
      QoS ID                   : -1
      Local QoS Strength       : BestEffort
      Negotiated QoS Strength  : BestEffort
      Negotiated QoS Direction : None
      Local QoS Status         : None
      Media Source IP Addr:Port: [9.42.30.10]:17780
      Media Dest IP Addr:Port  : [9.42.29.7]:50482
 Options-Ping    ENABLED:NO    ACTIVE:NO
    Number of SIP User Agent Client(UAC) calls: 2
 SIP UAS CALL INFO
 Call 1
 SIP Call ID                : 7CF44DF3-506611E0-8ED2B9D4-CA68C314@9.42.30.32
    State of the call       : STATE_ACTIVE (7)
    Substate of the call    : SUBSTATE_NONE (0)
    Calling Number          : 10000
    Called Number           : 708090
    Bit Flags               : 0x8C4401C 0x10000100 0x4
    CC Call ID              : 140
    Source IP Address (Sig ): 9.42.30.10
    Destn SIP Req Addr:Port : [9.42.30.32]:5060
    Destn SIP Resp Addr:Port: [9.42.30.32]:52757
    Destination Name        : 9.42.30.32
    Number of Media Streams : 1
    Number of Active Streams: 1
    RTP Fork Object         : 0x0
    Media Mode              : flow-through
    Media Stream 1
      State of the stream      : STREAM_ACTIVE
      Stream Call ID           : 140
      Stream Type              : voice+dtmf (0)
      Stream Media Addr Type   : 1
      Negotiated Codec         : g711ulaw (160 bytes)
      Codec Payload Type       : 0
      Negotiated Dtmf-relay    : rtp-nte
      Dtmf-relay Payload Type  : 101
      QoS ID                   : -1
      Local QoS Strength       : BestEffort
```

```
      Negotiated QoS Strength  : BestEffort
      Negotiated QoS Direction : None
      Local QoS Status         : None
      Media Source IP Addr:Port: [9.42.30.10]:18792
      Media Dest IP Addr:Port  : [9.42.30.32]:18638
Options-Ping    ENABLED:NO    ACTIVE:NO
   Number of SIP User Agent Server(UAS) calls: 1
```

Configuration Examples for Network-Based Recording using Cisco UBE

Example: Configuring the Media Profile Recorder

```
Device> enable
Device# configure terminal
Device(config)# media profile recorder 100
Device(cfg-mediaprofile)# media-recording 2000
Device(cfg-mediaprofile)# end
```

Example: Configuring the Media Class Recorder Globally

Example: Configuring Media Class Using the Media Profile Recorder

```
Device> enable
Device# configure terminal
Device(config)# media class 100
Device(cfg-mediaclass)# recorder profile 100
Device(cfg-mediaclass)# end
```

Example: Configuring Media Class Using the Recorder Parameter

```
Device> enable
Device# configure terminal
Device(config)# media class 100
Device(cfg-mediaclass)# recorder parameter
Device(cfg-mediaclass-recorder)# media-recording 28
Device(cfg-mediaclass-recorder)# end
```

Example: Configuring the Media Class for a Dial Peer

```
Device> enable
Device# configure terminal
Device(config)# dial-peer voice 24 voip
Device(config-dial-peer)# session protocol sipv2
```

```
Device(config-dial-peer)# incoming called-number 9845
Device(config-dial-peer)# media-class 100
Device(config-dial-peer)# codec g711ulaw
Device(config-dial-peer)# end
```

Example: Configuring the Dial Peer to Connect to MediaSense

```
Device> enable
Device# configure terminal
Device(config)# dial-peer voice 24 voip
Device(config-dial-peer)# destination-pattern 595959
Device(config-dial-peer)# session protocol sipv2
Device(config-dial-peer)# session target ipv4:10.42.29.7
Device(config-dial-peer)# session transport tcp
```

Feature Information for Network-Based Recording using Cisco UBE

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Table 1: Feature Information for Network-Based Recording using Cisco UBE

Feature Name	Releases	Feature Information
Network-Based Recording Using Cisco UBE	15.2(1)T	The Network-Based Recording Using Cisco UBE feature supports software-based forking for RTP streams. The following commands were introduced or modified: **media class**, **media profile recorder**, **media-recording**, **recorder parameter**, **recorder profile**, **show voip recmsp session**.

Feature Name	Releases	Feature Information
Network-Based Recording Using Cisco UBE	Cisco IOS XE Release 3.8S	The Network-Based Recording Using Cisco UBE feature supports software-based forking for RTP streams. The following commands were introduced or modified: **media class**, **media profile recorder**, **media-recording**, **recorder parameter**, **recorder profile**, **show voip recmsp session**.