WebEx Telepresence Media Support Over Single SIP Session

The WebEx Telepresence Media Support over Single SIP Session feature provides support for end-to-end negotiation of up to 6 m-lines or media lines over a single Session Initiation Protocol (SIP) session. The media types can be audio, video, or application.

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.

Restrictions for WebEx Telepresence Media Support Over Single SIP Session

- High availability is not supported with multiple m-lines.
- Only single dynamic payload type in the m-line for H.224 protocol is supported.
- Payload type interworking for Aggregation Service Routers (ASR) is not supported, so dynamic payload type is negotiated end-to-end.

Information About WebEx Telepresence Media Support Over Single SIP Session

The WebEx Telepresence Media Support over Single SIP Session feature provides the following support:

- End-to-end negotiation of multiple m-lines.
- Negotiation of Binary Floor Control Protocol (BFCP), IX, and H.224 protocol m-lines (m=application) and creation of Real-time Transport Protocol (RTP) or UDP streams for the same.
- Early-Offer (EO-EO) and Delayed-Offer (DO-DO) calls' support by the Cisco Unified Border Element (Cisco UBE) with multiple m-lines.
- End-to-end negotiation of multiple m-lines of same media type for video and application (but not audio).
- Mid-call escalation and de-escalation for multiple application and video m-lines.
- Secure RTP (SRTP) passthrough for all RTP streams (audio, video, and application).
- SRTP-RTP interworking for video (ASR only).
- Multiple dynamic payload types in the same m-line for the H.264 codec.

You can use the **show voip rtp connections** and **show call active video compact** commands to see the details about additional video and application streams.

Monitoring WebEx Telepresence Media Support Over Single SIP Session

Perform this task to see the details about additional video and application streams. The **show** commands can be entered in any order.

SUMMARY STEPS

1. **enable**
2. **show call active video compact**
3. **show voip rtp connections**
4. **show sip-ua calls**

DETAILED STEPS

Step 1 **enable**
Enables privileged EXEC mode.

Example:
```
Device> enable
```

Step 2 **show call active video compact**

Displays a compact version of call information for Skinny Call Control Protocol (SCCP), SIP, and H.323 video calls in progress. The codec type, negotiated codec, and remote media ports are displayed.

Example:

```
Device# show call active video compact

<callID>  A/O FAX T<sec> Codec       type        Peer Address       IP R<ip>:<udp>
Total call-legs: 2
         1 ANS     T5     H264        VOIP-VIDEO  P332211        9.45.38.39:2448
         6 ORG     T5     H264        VOIP-VIDEO  P1111          9.45.38.39:2438
```

Step 3 **show voip rtp connections**

Displays RTP named event packets. In the following sample output, two RTP connections are displayed for each m-line and a total of 10 RTP connections are displayed for 5 m-lines.

Example:

```
Device# show voip rtp connections

VoIP RTP active connections :
No. CallId     dstCallId  LocalRTP  RmtRTP   LocalIP                      RemoteIP
1     1          6          16384    54024  192.0.2.123                  192.0.2.39
2     2          7          16386    2448   192.0.2.123                  192.0.2.39
3     3          8          16400    5070   192.0.2.123                  192.0.2.39
4     4          9          16388    2450   192.0.2.123                  192.0.2.39
5     5          10         16402    2452   192.0.2.123                  192.0.2.39
6     6          1          16390    58121  192.0.2.123                  192.0.2.39
7     7          2          16392    2438   192.0.2.123                  192.0.2.39
8     8          3          16394    5070   192.0.2.123                  192.0.2.39
9     9          4          16396    2440   192.0.2.123                  192.0.2.39
10    10         5          16398    2442   192.0.2.123                  192.0.2.39
Found 10 active RTP connections
```

Step 4 **show sip-ua calls**

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

Example:

```
Device# show sip-ua calls

Total SIP call legs:2, User Agent Client:1, User Agent Server:1
SIP UAC CALL INFO
Call 1
SIP Call ID                : 72B6C784-753E11E2-FFFFFFFF8008B555-FFFFFFFFE340699E@9.45.47.123
   State of the call       : STATE_ACTIVE (7)
   Substate of the call    : SUBSTATE_NONE (0)
   Calling Number          : 332211
   Called Number           : 1111
   Bit Flags               : 0xC04018 0x10000100 0x80
   CC Call ID              : 6
   Source IP Address (Sig ): 9.45.47.123
   Destn SIP Req Addr:Port : [9.45.38.39]:5267
   Destn SIP Resp Addr:Port: [9.45.38.39]:5267
   Destination Name        : 9.45.38.39
   Number of Media Streams : 5
   Number of Active Streams: 5
   RTP Fork Object         : 0x0
   Media Mode              : flow-through
Media Stream 1
     State of the stream      : STREAM_ACTIVE
     Stream Call ID           : 6
     Stream Type              : voice-only (0)
```

```
      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 : NoneLocal QoS Status         : None
      Media Source IP Addr:Port: [9.45.47.123]:16390
      Media Dest IP Addr:Port  : [9.45.38.39]:58121
 Media Stream 2
      State of the stream      : STREAM_ACTIVE
      Stream Call ID           : 7
      Stream Type              : video (7)
      Stream Media Addr Type   : 1
      Negotiated Codec         : h263 (0 bytes)
      Codec Payload Type       : 97
      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: [9.45.47.123]:16392
      Media Dest IP Addr:Port  : [9.45.38.39]:2438
 Media Stream 3
      State of the stream      : STREAM_ACTIVE
      Stream Call ID           : 8
      Stream Type              : application (8)
      Stream Media Addr Type   : 1
      Negotiated Codec         : No Codec    (0 bytes)
            Codec Payload Type       : 255 (None)
      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: [9.45.47.123]:16394
      Media Dest IP Addr:Port  : [9.45.38.39]:5070
 Media Stream 4
      State of the stream      : STREAM_ACTIVE
      Stream Call ID           : 9
      Stream Type              : video (7)
      Stream Media Addr Type   : 1
      Negotiated Codec         : h263 (0 bytes)
      Codec Payload Type       : 97
      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: [9.45.47.123]:16396
      Media Dest IP Addr:Port  : [9.45.38.39]:2440
 Media Stream 5
      State of the stream      : STREAM_ACTIVE
      Stream Call ID           : 10
      Stream Type              : application (8)
      Stream Media Addr Type   : 1
      Negotiated Codec         : H.224 (0 bytes)
      Codec Payload Type       : 107
      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: [9.45.47.123]:16398
     Media Dest IP Addr:Port  : [9.45.38.39]:2442

Options-Ping    ENABLED:NO    ACTIVE:NO
   Number of SIP User Agent Client(UAC) calls: 1
```

Feature Information for WebEx Telepresence Media Support Over Single SIP Session

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.

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Table 1: Feature Information for WebEx Telepresence Media Support Over Single SIP Session

Feature Name	Releases	Feature Information
WebEx Telepresence Media Support Over Single SIP Session	15.3(2)T	The WebEx Telepresence Media Support over Single SIP Session feature provides support for end-to-end negotiation of up to 6 m-lines or media lines over a single Session Initiation Protocol (SIP) session. The media types can be audio, video, or application.
WebEx Telepresence Media Support Over Single SIP Session	Cisco IOS XE Release 3.9S	The WebEx Telepresence Media Support over Single SIP Session feature provides support for end-to-end negotiation of up to 6 m-lines or media lines over a single Session Initiation Protocol (SIP) session. The media types can be audio, video, or application.