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Profile: Cisco TIP Triple Screen Endpoint 1.6b Implementation Profile (TIPv6)

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Cisco TIP
Triple Screen Endpoint 1.6b
Implementation Profile
(for use with TIP v6)

Modification History

Revision	Date	Originator	Comments
1.0	04/05/2010	Cisco Systems, Inc.	Initial document
1.1	04/28/2010	Cisco Systems, Inc.	Changed MUST to SHOULD for the Conference ID in 5.1.13 and modifications made software releases in 11.1. Editorial only changes to introduction and clarification on previous use of the "compliant" term.
1.2	07/01/2010	Cisco Systems, Inc.	Several corrections and clarifications outlined in section 11.3 through 11.9

1 Contents

1. Introduction

The Telepresence Interoperability Protocol (TIP) is a signaling protocol used by high end video conferencing devices to signal and negotiate capabilities complimentary to those which are signaled and negotiated in SIP/SDP [1] [3]. TIP signaling utilizes the Real-time Transport Protocol (RTP) and RTP Control Protocol (RTCP)."

Though TIP as a protocol does not restrict device capabilities or media configurations, these set of accompanying documents define the different Cisco TIP implementation profiles, which indicate device capabilities or media configurations needed to achieve interoperability with Cisco TelePresence devices. These documents will be updated from time to time as Cisco TelePresence capabilities evolve, indicated by a new revision number that generally track Cisco software releases.

TIP systems built to be integrated in to a Cisco TelePresence system deployment will need to operate seamlessly with the Cisco Unified Communications Manager (CUCM) architecture and products, appearing as CUCM SIP devices with audio and video capabilities. CUCM allows SIP endpoints to connect as generic, standard endpoints per the publically available document(s) listed [10] in the References section. No licenses for proprietary SIP extensions are required.

A TIP device is capable of handling multiple audio and video streams. The devices negotiate the number of media streams they will exchange, and media sources are switched when necessary to always present the viewer with the active session participants.

This document specifies the 1.6b TIP profile for a triple screen TIP endpoint. The profile includes requirements related to SIP signaling, TIP signaling, media encoding, and general behavior. The document takes into consideration scenarios in which a TIP-compliant triple screen endpoint is communicating with another TIP-compliant triple screen endpoint, single screen endpoint or multi-point server.

2 Triple screen endpoint general capabilities

This section provides general requirements for a triple screen telepresence. More details will be covered in later sections. The general requirements include:

1. SIP signaling
2. H264 video encoding and decoding at 1080P and/or 720P as well as XGA and CIF resolutions
3. AAC-LD audio encoding and decoding
4. G711 audio encoding and decoding
5. G722 encoding and decoding support is optional
6. Three input video devices that can be encoded using H264 codec at either 720P or 1080P resolution as well as CIF resolution simultaneously
7. An auxiliary video device (aka. presentation) that can be encoded using H264 codec at an XGA resolution
8. Three input audio devices that can be encoded using AAC-LD codec
9. An auxiliary audio device (aka. presentation) that can be encoded using AAC-LD codec
10. Ability to mix all audio inputs and encode them using G711 codec at the same time as encoding the individual inputs using AAC-LD. G722 can be optionally used to encode the mix if G722 is supported by the endpoint
11. Ability to receive and decode three H264 encoded streams at 1080P, 720P, or CIF resolutions as well as an H264 encoded stream at XGA resolution
12. Ability to receive and decode four AAC-LD audio streams or three AAC-LD streams and one G711/G722 stream
13. Ability to mix up to four of the received and decoded AAC-LD and G711/G722 audio streams
14. RTP/RTCP support using AVP profile

3 SIP signaling

1. A TIP-compliant endpoint is a UC SIP line device that MUST implement the UC SIP line specifications [10].
2. A TIP-compliant endpoint MUST offer and negotiate a single audio media line and a single video media line in SDP [2] [3].
3. A TIP-compliant endpoint MUST be able to transmit DTMF tones using KPML [13] or 2833 [12]
4. A TIP-compliant endpoint SHOULD NOT transmit SIP INFO [14] for codec control when TIP has been successfully negotiated.
5. Standard SIP features such as hold/resume MUST be supported as specified in the UC SIP line specification [10].
6. A TIP-compliant endpoint SHOULD NOT during hold or depend on receiving RTCP packets.
7. A TIP-compliant endpoint MUST send and receive mid-call INVITEs that specify updated video bit rates following a successful TIP negotiation. More detailed requirements on this behavior are included later in this document.

3.1 SDP Audio Signaling

1. A TIP-compliant endpoint MUST specify AAC-LD as the first codec in the SDP audio media line. In other words only an AAC-LD codec can be negotiated in SIP for devices supporting TIP.
2. It is STRONGLY RECOMMENDED that a TIP-compliant endpoint uses 96 as its default AAC-LD dynamic payload type. If another payload type is used, a TIP-compliant endpoint MUST support asymmetric payload types and MUST support dynamic changes in payload types while media is active.
3. A TIP-compliant endpoint MUST use RFC 3640 [15] and the aac-hbr mode in its AAC-LD SDP signaling.
4. The AAC-LD requirements specified in 10 MUST be supported even when contradictory to what is received in the audio SDP ftmp line.

```
m=audio 16384 RTP/AVP 96 0 99 101
b=TIAS:64000
a=rtpmap:96 mpeg4-generic/48000
a=fmtp:96 profile-level-
id=16;streamtype=5;mode=AAC-
hbr;config=11B0;sizeLength=13;indexLength
=3;indexDeltaLength=3;constantDuration=4
80
a=rtpmap:0 PCMU/8000
a=rtpmap:99 L16/48000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Figure 1: Sample audio SDP for first TIP endpoint offer

3.2 SDP video signaling

1. A TIP-compliant endpoint MUST signal H264 codec in its video media line [7].
2. A TIP-endpoint MUST support the reception of a profile-level-id value of “420028” as well as a placeholder value of “ABCDEF”.
3. It is RECOMMENDED that a TIP-compliant endpoint indicates a profile-level-id value of 420028 and packetization-mode =1 in is SDP signaling.
4. It is STRONGLY RECOMMENDED that a TIP-compliant endpoint uses 112 as its default dynamic H264 payload type. If another payload type is used then the endpoint MUST support asymmetric payload types and MUST support dynamic changes in payload types while media is active.
5. A TIP-compliant endpoint MUST include a b=TIAS [16] field associated with the video media line in every SDP offer and answer. For the initial call setup and after resume (i.e., first reINVITE transaction resuming the media), the value of the b=TIAS field MUST be that of a single HD video stream.
6. The b=TIAS value for the initial call setup and after the resume MUST be equal to or larger than .936 Mbps. The negotiated bit rate value is calculated as the minimum of the b=TIAS values in the SDP offer and answer. The negotiated bit rate value determines the maximum video resolution for the call. A negotiated value above 2.25 Mbps will result in 1080P resolution. A negotiated value between .936 and 2.25 will result in a 720P resolution. Values below .936 Mbps will cause the call to be dropped.

Negotiated HD video bit rates	HD video resolution
2.25 – 4 >	1080P
.936- 2.25	720P
< .936	Call drop

Table 1: Bit rate to video resolution mapping

7. A TIP-compliant endpoint MUST NOT initiate any mid-call INVITE transactions after call setup or after resume transactions until TIP negotiation has completed.
8. All mid-call INVITEs, after TIP negotiation has completed and until the completion of a hold transaction, MUST have the b=TIAS value set to the sum of the bit rates of all HD video streams negotiated in TIP plus the presentation bit rate according to the first presentation frame rate negotiated in the current TIP session. Note that the first presentation bit rate is included irrespective of whether the presentation is active or not and irrespective of whether another presentation frame rate/bit rate has been negotiated. Below is the mapping between presentation frame rates and presentation bit rates.

Presentation frame rate	Presentation bit rate
30	4 Mbps
5	0.5 Mbps
1	100 Kbps

Table 2: Frame rates to bit rates mappings

9. A TIP-compliant endpoint MUST send a mid-call INVITE once TIP negotiation is completed, if it has initiated the call and the remote peer has not specified itself as a multi-point server in the TIP negotiation.
10. The negotiated bit rate MAY be changed mid-call by sending a mid-call INVITE with a new b=TIAS value.
11. A TIP-compliant endpoint MUST recalculate the negotiated bit rate for a single video

stream for each mid-call INVITE transaction and adjust its video encoder accordingly.

```
m=video 16386 RTP/AVP 112
      b=TIAS:2250000
a=rtpmap:112 H264/90000
      a=fmtp:112 profile-level-
      id=ABCDEF;packetization-
            mode=1
```

Figure 2: Sample TIP-compliant video SDP

4 Sample SIP call flows

4.1 Endpoint calling another endpoint

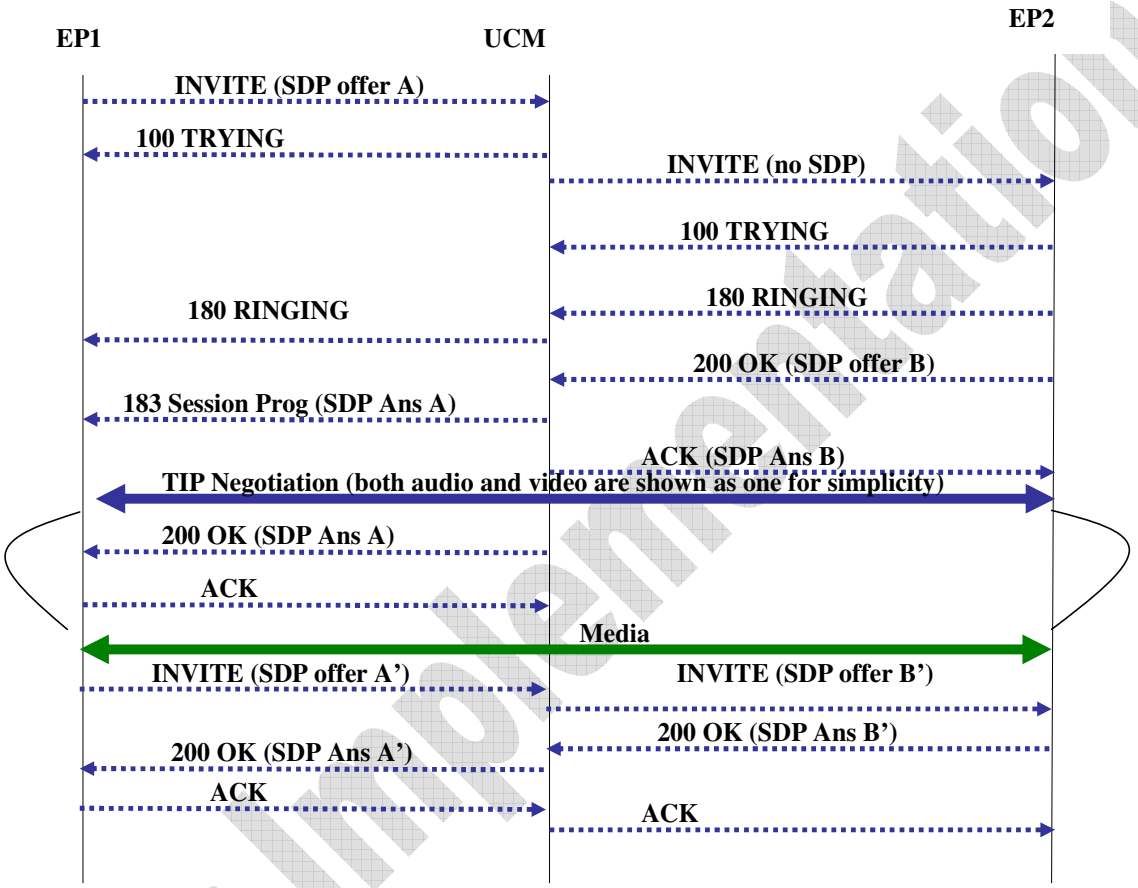

Figure 3: TIP endpoint calling another endpoint

Figure 3 shows a TIP endpoint dialing another TIP endpoint. The first INVITE transaction negotiates a single audio and video media line with H264 and AAC. The video resolution is determined by the bit rates specified by the b=TIAS parameter. As stated in 3 the video bit rate in this offer/answer exchange represents a single HD video stream.

Once each device has the media information of the other side, at the end of the first SDP offer/answer exchange, TIP negotiation MUST start on both the audio and the video channels.

Once TIP has completed on the audio channel (see 5 for definition of TIP negotiation completion), endpoints can start audio transmission of all audio segments negotiated in TIP. Once TIP has completed on the video channel, endpoints can start video transmission on all video segments negotiated in TIP. Moreover once TIP has completed on the video channel, the calling endpoint MUST send a mid-call INVITE which updates the video bit rate according to the number of video streams and the presentation frame rate negotiated in TIP (see 3 for more details on bit rate negotiation). Note that the mid-call bit rate negotiation could end up downgrading the video resolution or bit rate from that negotiated in the initial call setup, due to policies enforced by UCM.

4.2 TIP endpoint calling a TIP multi-point server

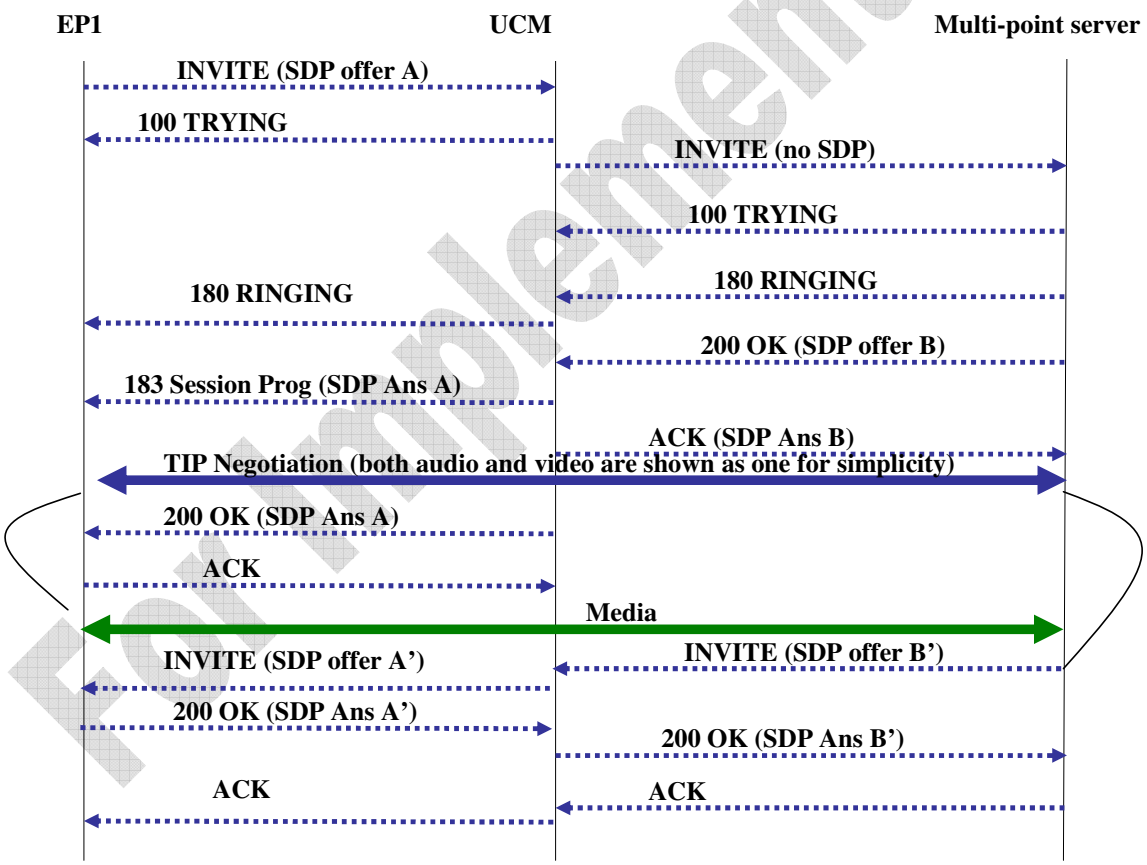

Figure 4: TIP endpoint calling a TIP multi-point server

Figure 4 shows a TIP endpoint calling a TIP multi-point server to join an ongoing multi-point session. This call flow is similar to that of two endpoints calling each other except that it is the multi-point server that ALWAYS initiates the mid-call INVITE to update the media bit rates, once the TIP negotiation has completed on the video channel.

4.3 TIP endpoint calling a TIP multi-point server as the first participant

Figure 5 shows a TIP endpoint dialing into a TIP multi-point server as the first participant. This flow is similar to the previous flow except that once TIP finishes, the multi-point server will send a mid-call INVITE putting the call on hold and stopping media transmission. The call will be resumed once a second participant joins the multi-point session. One thing to note is that the SDP hold message will carry the video bit rate in the b=TIAS parameter, representing the value negotiated in TIP. The endpoint SDP answer will do the same.

4.4 Resuming a TIP call after hold

This call flow is identical to Figure 4. At resume time the TIP negotiation MUST BE repeated and the same strategy for bit rate negotiation is repeated. No decisions are made at resume time based on what was negotiated before the hold. A call is identifed as on hold if the SDP at the audio media line level has connection information (c=) specifying a 0.0.0.0 IP address or has a send-only or inactive attribute. The same information specified at the session level while ommitted at the audio media line level will also indicate hold.

A call is resumed when a non-hold SDP is received following a hold SDP.

4.5 Reducing the bit rate in a call

In this example, a TIP endpoint has joined a multi-point conference and the negotiated bit rate for a single video stream was 4 Mbps. At some point the TIP multi-point server needs to reduce the bit rate of the conference to 2.5 Mbps, because a new endpoint, which can only handle 2.5 Mbps per video stream, has joined the conference. The multi-point server will send a mid-call INVITE to the TIP endpoint, similar to the mid-call INVITE in Figure 4, specifying a lower bit rate than the one negotiated in the previous INVITE transaction.

Note that the new bit rate will be the desired bit rate per stream multiplied by the number of HD video streams negotiated in TIP plus the presentation bit rate. As before, the presentation bit rate will be added irrespective of whether or not the presentation is active at this time. Also, the presentation bit rate will be calculated based on the frame rate first negotiated in the TIP session irrespective of whether this frame rate has been downgraded later or not.

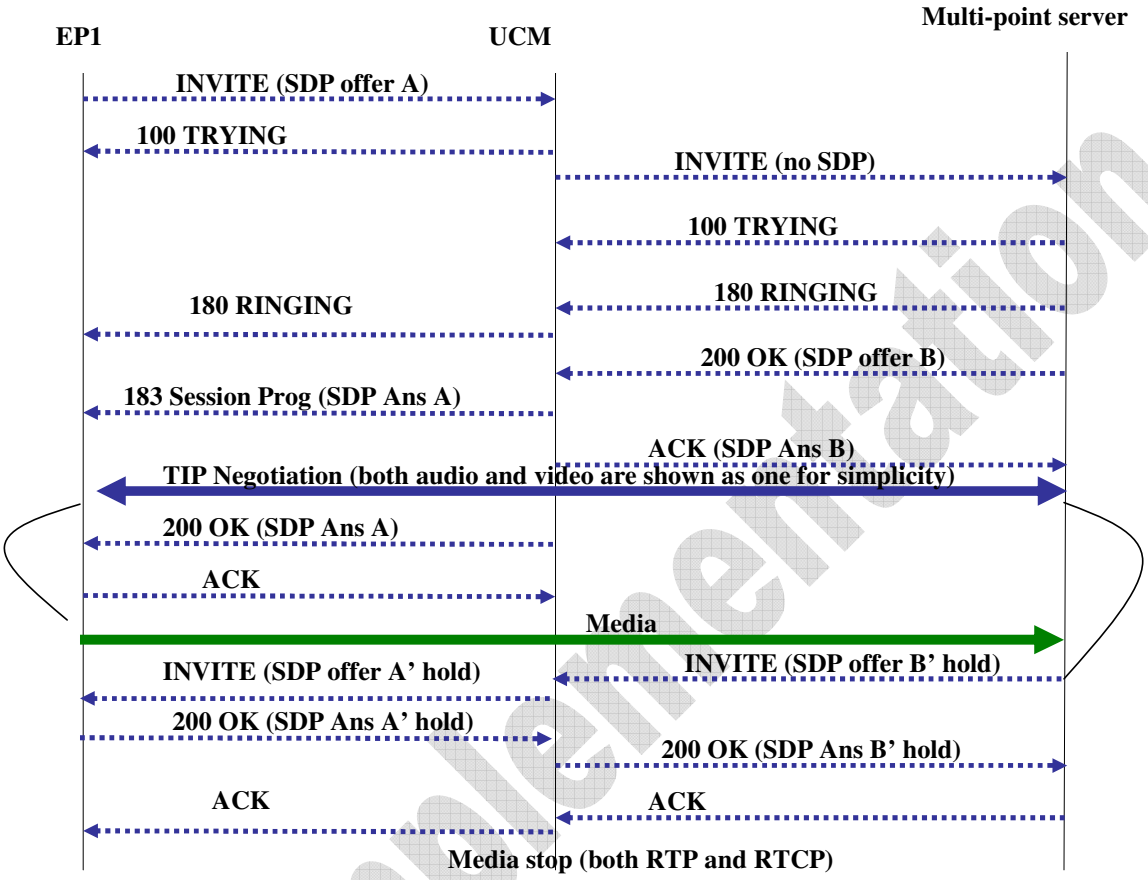

Figure 5: TIP endpoint joining a multi-point session as first participant

5 Initial TIP signaling

Triple screen endpoints that are TIP-compliant MUST adhere to all of the MUST requirements specified in the TIP document [11] as well as the MUST requirements in this document.

5.1 MUXCTRL Packet

1. In a video session, the AVPF profile MUST be supported and signaled in the MUXCTRL packet as specified in the TIP specs [11]. This in turn mandates support for the

transmission and reception of the RTCP video feedback packets as specified in the TIP specs [11]

2. In an audio session, the profile MUST be AVP.
3. The options field of the MUXCTRL packet MUST be set to 0x00.
4. The number of transmitted streams MUST be set to 6 for a video session when no presentation is to be transmitted, and set to 7 when presentation transmission is requested.
5. The number of streams MUST be set to 5 for an audio session.
6. The number of received video streams MUST be set to 4.
7. The number of received audio streams MUST be set to 5.
8. The transmit video positions MUST contain only: center, left, right, legacy center, legacy left and legacy right, when no presentation transmission is requested.
9. The auxiliary transmit video position MUST be set when presentation transmission is requested.
10. All transmit audio positions (left, right, center, legacy mix and auxiliary) MUST be set.
11. The receive video positions MUST contain center, left, right and auxiliary.
12. The receive audio positions MUST contain center, left, right, legacy and auxiliary.
13. TIP endpoints SHOULD set the conference identifier field, or if the identifier is not known, set the field to 0.

5.2 MediaOpts Packet

1. TIP endpoints MUST indicate support for transmission of Audio Activity Metric and MUST NOT indicate support for reception of Audio Activity Metric.
2. TIP endpoints MUST support the reception of Audio Dynamic Output Channel.
3. TIP endpoints MUST NOT indicate support for transmission of the Audio Dynamic Output Channel and MUST set output positions in all TIP-CSRCs to zero.
4. TIP endpoints MAY indicate support and use of G722 encoding for legacy audio streams.
5. TIP endpoints MUST indicate support for the transmission of the video Refresh flag but MUST NOT indicate support for the reception of the flag.
6. TIP endpoints MUST indicate support for Inband Parameter Sets (SPS/PPS) both for transmission and reception.
7. TIP endpoints MAY signal support for CABAC (Please see exception note in 11.1)
8. TIP endpoints MAY signal support for LTRP and GDR.

9. TIP endpoints MUST signal support for the maximum auxiliary frame rate that they can support. Note that frame rates are symmetric and the maximum can be 30, 5 or 1.
10. TIP endpoints are RECOMMENDED to support 30 fps as their maximum frame rate.
11. TIP endpoints MUST support 5 and 1 fps auxiliary frame rates.
12. TIP endpoints MAY signal a Satellite deployment Transmitter profile if it is known that this endpoint is connected using a satellite link.

5.3 TIP Negotiation and general mid-session signaling rules

1. A TIP endpoint MUST NOT start media transmission until the TIP negotiation has finished. TIP negotiation is finished once the following has completed:
    a. Local MUXCTRL packet has been sent and a remote ACK was received.
    b. Remote MUXCTRL has been received and an ACK was sent.
    c. Local MediaOpts packet has been sent and an ACK was received.
    d. Remote MediaOpts packet has been received and ACK was sent.

    Note that TIP can (and will) complete at one peer prior to the other. Implementation MUST be able to handle such scenarios.
2. The number of transmitted streams MUST NOT exceed the number of streams the remote peer is capable of receiving
3. Only positions indicated as supported by the remote peer can be used in the CSRC received position
4. Typically the number of transmitted streams is the minimum of what the local endpoint can transmit and what the remote peer can receive. The exception is for legacy streams, which MUST NOT be transmitted, except if the remote peer indicates its capability to receive the legacy streams AND the remote peer indicates that it is a multi-point server by signaling the isfocus parameter in its MUXCTRL packet.
5. Below is the maximum number of streams that will be sent and received when a triple TIP endpoint communicates with a remote peer supporting one of the TIP profiles:
    a. Remote peer is a triple screen TIP endpoint
        i. Transmitted audio streams: 4
        ii. Received audio streams: 4
        iii. Transmitted video streams: 4
        iv. Received video streams: 4
    b. Remote peer is a single screen TIP endpoint

i. Transmitted audio streams: 4
ii. Received audio streams: 2
iii. Transmitted video streams: 2
iv. Received video streams: 2

c. Remote peer is a multi-point server requiring legacy stream support:
i. Transmitted audio streams: 5
ii. Received audio streams: 5
iii. Transmitted video streams: 7
iv. Received video streams: 4

d. Remote peer is a multi-point server requiring no legacy stream support:
i. Transmitted audio streams: 4
ii. Received audio streams: 4
iii. Transmitted video streams: 4
iv. Received video streams: 4

6. When a triple screen endpoint communicates with a single screen endpoint, the sender has more positions and streams to transmit than the remote peer can receive. In this case, the triple screen endpoint SHOULD switch between the signals coming from its three main video inputs, based on criteria such as the associated audio input level.
7. TIP endpoints MUST NOT generate a second MediaOpts packet within the same TIP session[1] in which there have been changes to the supported presentation frame rate or any other media capabilities.
8. TIP endpoints MUST support the ability to receive MediaOpts packet that change the supported auxiliary frame rate, but only within the limit of the initially negotiated maximum auxiliary frame rate.
9. TIP endpoints MUST silently discard any received TIP packet with an unrecognized RTCP APP packet subtype.
10. TIP endpoints MUST silently discard any received TIP packet with an RTCP APP name other than “xcts” .
11. TIP endpoints MUST respond to ECHO requests as specified in the TIP specs.
12. TIP endpoints MUST support the reception of the media flow control request and process it according to the TIP specs [11].

[1] A TIP session is initiated at call setup and after a hold or resume transaction. It is identified by a unique TIP SSRC value.

13. TIP endpoints MUST NOT generate media flow control requests.
14. TIP endpoints MUST support the reception of the video refresh request and process it according to the TIP specs.
15. TIP endpoints MUST be able to rceive a video refresh request that does not include the Flags field. On receiving such a request, the endpoint can generate an IDR or a GDR. Note that GDRs can only be generated if GDR support has been negotiated for the current TIP session.
16. TIP endpoints MAY generate video refresh requests though it is NOT RECOMMENDED to use this mechanism when packet losses are encountered. Alternatively the video feedback mechanism should be used for reporting loss.

6 Presentation Signaling

Permission to present is requested and released using the video session MUXCTRL packet with the auxiliary video *transmission* position. The video session MUXCTRL packet with the auxiliary video *receiving* position is used ONLY by multi-point servers to grant permission to present.

A TIP endpoint can receive a MUXCTRL packet that does not change the presentation state. The endpoint MUST process such packets with no disruption to presentation media.

6.1 Transmitting and receiving permission-to-present requests in a multi-point session

To request permission to present, a TIP endpoint MUST send a MUXCTRL packet on the video session setting the transmit video auxiliary position to 1 and the number of transmitted video streams to 4.

If the request is granted, the multi-point server will send a MUXCTRL packet on the video session setting the receive auxiliary video position to 1. Once the endpoint ACKs the request it can start transmitting its presentation video and audio.

If the multi-point server denies the request due to conflict with another endpoint that already has permission to transmit, the multi-point server will send a MUXCTRL packet setting the auxiliary *receive* video position to 0. This request will typically have the auxiliary video *transmit* position set to 1 indicating that the multi-point server has another source ready to transmit.

A TIP endpoint receiving a request for permission to present MUST grant the request, via acking it and starting its presentation receivers. If the endpoint was already transmitting presentation prior to receiving the permission-to-present request, it MUST stop its presentation transmission on both audio and video channels and ACK the request.

TIP endpoints MUST NOT deny permission-to-transmit requests. In other words TIP endpoints MUST NOT send a MUXCTRL packet that has the auxiliary receive position set to zero.

6.2 Transmitting and receiving permission-to-present requests in a point-to-point session

To request permission to present, a TIP endpoint MUST send a MUXCTRL packet on the video session, setting the transmit video auxiliary position to 1 and incrementing the number of transmitted video streams by 1.

The request is granted once the remote endpoint ACKs the request. Once the ACK packet is received, the endpoint can start transmitting its presentation video and audio.

A presentation conflict can occur when a MUXCTRL is received from the remote peer requesting permission to present at the same time that a local permission-to-transmit has been sent. TIP resolves the conflict by specifying the endpoint that has initiated the call as the winner of the conflict. To achieve such a result the following behavior is a MUST in the case of a presentation conflict:

1. Before it considers itself the new presenter, the endpoint that has initiated the call MUST wait for a new MUXCTRL packet from the remote peer releasing its permission-to-present request (see release permission-to-present section below).
2. The endpoint that has not initiated the call MUST send a new MUXCTRL packet releasing its permission-to-present request
3. Both endpoints MUST ACK all MUXCTRL packets received

A TIP endpoint receiving a permission-to-present request MUST grant the request, via acking it and starting its presentation receivers, except in the case of a conflict, where the above stated behavior for conflict resolution MUST be followed. If the endpoint was already transmitting presentation prior to receiving the permission-to-present request, it MUST stop its presentation transmission on both audio and video channels and ACK the request.

A TIP endpoint MUST NOT deny any permission-to-transmit requests by sending a MUXCTRL that has the auxiliary receive position set to zero.

6.3 Transmitting and receiving permission-to-present release requests

To release permission to present, the endpoint MUST send a MUXCTRL packet on the video session, setting the transmit video auxiliary position to 0 and the number of transmitted video streams to 3.

Note that in a multi-point session an endpoint which has previously sent a permission-to-present that was not granted MUST still release the permission-to-present when it is no longer interested in transmitting presentation.

In point-to-point scenarios when the endpoint receives permission-to-present release request, it MUST ACK the request. If the endpoint is interested in presentation transmission then it MUST send a new MUXCTRL requesting permission to present.

6.4 Receiving a request-to-present grant in a multi-point session

A multi-point server can grant permission to present to an endpoint after it has denied the request initially. This is the case when a conflict occurs between multiple endpoints requesting to present at the same time and resulting in the multi-point server granting only one request while denying others. Once the winning endpoint releases its permission to present, the multi-point might grant the permission to present to the endpoint whose request was previously denied. In this case the endpoint will receive a MUXCTRL packet with the receive auxiliary video position set to 1 and the auxiliary transmit video position set to 0. If the endpoint is still interested in presentation transmission, it can start transmitting presentation. Otherwise the endpoint MUST send a new MUXCTRL packet requesting a permission-to-present release.

7 Presentation frame rate negotiation

7.1 Initial frame rate negotiation

As specified in the TIP specs [11], the MediaOpts packet is used for frame rate negotiation. It is RECOMMENDED that TIP endpoints support and signal a 30 fps capability in the MediaOpts packet. Once a TIP endpoint has signaled its initial presentation frame rate capability, it MUST NOT change it during the same TIP session. The negotiated initial frame rate is calculated as the minimum of what was signaled by both peers. Note that the initial frame rate is what will be used to determine the presentation bit rate included in the SDP video b=TIAS [16] line irrespective of whether or not the frame rate has changed later within the TIP session.
It is important to note that when an endpoint signals a frame rate capability, such as 30 fps, this endpoint MUST also support all lower frame rates such as 5 and 1 fps. However, an endpoint with a negotiated frame rate such as 30 fps MUST NOT transmit presentation using a different frame rate such as 5 fps without first receiving a MediaOpts packet negotiating the new frame rate.

7.2 Signaling a new frame rate

TIP endpoints MUST NOT dynamically change their signaled frame rate. However TIP endpoints MUST support receiving mid-session MediaOpts packets that can lower or increase the negotiated frame rate. Note that the negotiated frame rate MUST NOT be higher than the initially negotiated frame rate in the TIP session. Once the endpoint ACKs the MediaOpts packet, any new presentation transmitted MUST use the new frame rate.

8 Audio mixing and synchronization

In a multi-point session, when audio streams are sent to a TIP triple screen endpoint, the multi-point server maintains a mapping between a source segment/position and a receive segment/position. In other words if a triple endpoint has received the audio from a single endpoint on its left segment, the triple endpoint will always receive the same single endpoint audio on its left segment.

As a result, a triple screen endpoint could receive three audio streams from a multi-point server all to be played on the same segment/position e.g., left speaker. The receiver positions in the

CSRCs of these audio streams will be different while the dynamic audio output position will be the same indicating that these are simultaneous streams to be mixed rather than alternative streams to be switched. The output positions in the CSRCs of these audio streams will be the same, indicating that these streams need to be mixed and played out together. A TIP endpoint MUST mix, prior to playing out, audio streams that have the same output position and different receiver positions.

Note that some, or all, of the audio streams to be mixed could be coming from the same remote endpoint. Such audio streams can be identified as they share the same TIP-CSRC sampling clock ID. A TIP endpoint SHOULD use the audio SRs to synchronize audio between streams that share the same CSRC sampling clock ID.

9 Media

1. TIP endpoints MUST handle dynamic switches of SSRCs [4] and their associated CSRCs on received video and audio streams. The CSRC receive position identifies that this new SSRC is to replace the previous SSRC that was associated with the CSRC with the same receive position.
2. All audio and video SSRCs MUST NOT have only zeros in their last 8 bits.
3. TIP endpoints MUST change their audio and video streams' SSRCs and CSRCs after a hold/resume transaction.
4. Video SSRCs and CSRCs values MUST change when switching presentation streams from 30 fps to 1 or 5 fps and vice versa.
5. TIP endoints MUST support remote RTP senders and receivers corresponding to the same position yet using different SSRCs. For example a TIP endpoint might receive an RTCP RR or an RTCP feedback packet from a center position receiver with a source SSRC different from the source SSRC of the center position's RTP packets.
6. TIP endpoints MUST support the flow control TIP packet, both for transmitting and receiving video streams. The receive flow control will be used by a multi-point server to inform the endpoint that no media will be transmitted by the server using the receive position specified in the flow control packet target CSRC. The transmit flow control will be used by the multi-point server to inform the endpoint that no media should be transmitted using the transmit position specified in the flow control packet target CSRC.
7. TIP Endpoints MUST always transmit media using all source positions negotiated in TIP, unless a transmit flow control packet to turn off that specific position was received.
8. TIP endpoints MUST transmit RTCP Sender Reports [4].
9. TIP endpoints MUST silently discard any non-RTP packets, such as DTLS or STUN Keepalive packets received on the video and audio RTP sessions.

10. TIP endpoints MUST silently discard any RTP packets with an unknown payload type.

9.1 Audio

1. All non-legacy audio streams MUST be transmitted using AAC-LD (see 10 for codec requirements) and using RFC 3640 with AAC-hbr mode [15]
2. TIP endpoints MUST transmit 2 AAC audio frames (10 msec per frame) msec in AAC audio packets
3. TIP endpoints MUST send audio activity metric byte for all AAC audio streams as specified in the TIP specs [11] when the Audio Activity Metric support is negotiated in TIP
4. TIP endpoints MUST set the Audio Activity Metric associated with presentation audio to a value of 99
5. TIP endpoints MUST NOT send the audio activity metric byte for G711 or G722 legacy streams
6. TIP endpoints MUST be able to deal with periods where no audio packets are received on one or more of its active audio positions without disrupting the call. The recommendation is to use video streams to detect network problems and act on them. Video streams will receive a receive flow control off packet when no video packets are available for transmission on a specific position.
7. TIP endpoints MUST be able to deal with receiving the same audio stream with discontinuous sequence numbers. Note that for video sequence numbers will only be discontinuous during an SSRC switch whereas in the case of audio this might not always be true
8. TIP endpoints MUST support mixing of audio streams that are sent using the same output position and different receive positions
9. TIP endpoints SHOULD use received audio Sender Reports to synchronize audio between different SSRCs that share the same CSRC sampling clock ID.
10. TIP endpoints MUST generate accurate and frequent audio RTCP Sender Reports for audio sources to allow remote peers to perform cross-media synchronization
11. TIP endpoints MUST send all three main audio streams using the same sampling clock ID
12. When dynamic output channels are not negotiated, TIP endpoints MUST include zero in the 4 TIP-CSRC bits corresponding to output positions

9.2 Video

1. TIP endpoints MUST include zero in the 4 video TIP-CSRC bits, corresponding to the dynamic output positions.

2. All HD and auxiliary H264 video steams MUST be packetized using packetization mode "1" [7]. The following NAL types are supported:
    - 1 (non-IDR)
    - 5 (IDR)
    - 6 (SEI, only subtype 6 [recovery point] is supported for GDRs)
    - 7 (SPS)
    - 8 (PPS)
3. All CIF H264 video streams MUST be packetized using packetization mode 0 [7].
4. All video streams except for auxiliary MUST be transmitted using a 30 fps or 29.97 fps.
5. Auxiliary streams MUST be transmitted using the frame rate signaled in the TIP session.
6. TIP endpoints' video frames sampling instances ,and their corresponding RTP timestamps, MUST always correspond to regular intervals that is based on the video clock rate and the video frame rate. For example a 30 fps TIP video stream using a 90KHz clock can only generate video frames every 3000 clock ticks. A 5 fps TIP video stream using a 90 KHz clock can only generate video frames every 18000 clock ticks.
7. TIP endpoints MUST send the video refresh flag for all video streams when support for video refresh flag is negotiated in TIP.
8. TIP endpoints MUST support in-band transmission of the SPS and PPS as the first two NALs in each I-frame. TIP endpoints MUST support in-band transmission of the SPS, PPS, and SEI recovery point NAL as the first three NALS in each GDR.
9. TIP endpoints MUST support transmission and reception of RTCP video feedback as specified in the TIP SPECS. [11]
10. TIP endpoints MUST transmit a repair frame as soon as a negative video feedback report is processed.

10 Codec Specs

10.1 TIP HD Video Encoder Requirements

1. MUST generate H264 Main Profile compatible bit stream
2. MUST support CALVC
3. MAY support CABAC as a negotiated option
4. MUST support 1280x720 or 1920x1072 resolution
5. MUST support fixed frame rate: 30 fps or 29.97 fps
6. MUST generate one macro block-row per slice

   -1280x720: 45 slices per frame, each slice must be 80 macro blocks in length

   -1920x1072: 67 slices per frame, each slice must be 120 macro blocks in length

7. MUST support deblocking_filter_control_present = 1 & disable_deblocking_filter_idc = 1
8. MUST only use interprediction blocks of size 16x16
9. MUST only use one reference picture
10. Each NAL MUST be less than 3320 bytes in size
11. POC MUST increment by 2 every frame
12. **MUST NOT USE SPS IDs in the range of 10 to 20**
13. Long term reference pictures (LTRPs) MAY be used for error concealment. LTRPs require a four frame picture buffer. LTRPs use H.264 standard picture buffer management. LTRPs are optional and can be disabled.
14. Gradual decoder refresh pictures (GDRs) MAY be used at the start of a sequence instead of an instantaneous decoder refresh picture (IDR). A recovery point SEI precedes the GDR. The SEI always has the broken_link_flag set to 0. GDRs are optional and can be disabled.

10.2 TIP CIF Video Encoder Requirements

1. MUST support H264 Baseline Profile compatible bit stream
2. MUST support 352x288 resolution
3. May support variable frame rate up to 30 fps
4. Uses one reference picture from previous frame
5. Each NAL MUST be less than 3320 bytes in size

10.3 TIP Presentation Video Encoder Requirements (1 & 5 fps)

1. MUST support H264 Baseline Profile
2. MUST support 1024x768 resolution
3. MUST support Fixed frame rate: 1 or 5 fps
4. MUST support the use of one reference picture from previous frame
5. MUST support one macro block-row per slice,
-1024x768: 48 slices per frame, each slice must be 64 macro blocks in length
6. MUST support deblocking_filter_control_present = 1 & disable_deblocking_filter_idc = 1
7. MUST only support inter-prediction blocks of size 16x16
8. Each NAL MUST be less than 3320 bytes in size
9. POC MUST increment by 2 every frame

10.4 TIP Presentation Video Encoder (30fps) Requirements

1. MUST support H264 Main Profile compatible bit stream
2. MUST support CALVC
3. MAY support CABAC as a negotiated option
4. MUST support 1024x768 resolution
5. MUST support fixed frame rate: 30 fps or 29.97 fps
6. MUST support one macro block-row per slice
7. -1024x768: 48 slices per frame, each slice must be 64 macro blocks in length
8. MUST support deblocking_filter_control_present = 1 & disable_deblocking_filter_idc = 1
9. MUST only support inter-prediction blocks of size 16x16
10. MUST support one reference picture
11. Each NAL MUST be less than 3320 bytes in size
12. POC MUST increment by 2 every frame
13. **MUST NOT USE SPS IDs in the range of 10 to 20**
14. Long term reference pictures (LTRPs) MAY be used for error concealment. LTRPs require a four frame picture buffer. LTRPs use H264 standard picture buffer management. LTRPs are optional and can be disabled.
15. Gradual decoder refresh pictures (GDRs) MAY be used at the start of a sequence instead of an instantaneous decoder refresh picture (IDR). A recovery point SEI precedes the GDR. The SEI always has the broken_link_flag set to 0. GDRs are optional and can be disabled.

10.5 TIP Audio Encoder Requirements

Codec: G.711 (u-law)
Frame Size: 80 samples
CNG is OFF
Packetization interval: 20ms

Codec: G.722
Frame Size: 160 samples
Sampling Rate: 16KHz
Bit Rate: 64 kbps
CNG is OFF
Packetization interval: 20ms

Codec: AACLD
Frame Size: 480 samples
Sampling Rate: 48KHz
Bit Rate: 64 kbps
Mono channel
Transmux = 0
PNS (Perceptual Noise Substitution): enabled
LTP (Long Term Prediction): disabled
Packetization interval: 20ms

10.6 TIP Audio Activity Metric Calculation

The audio activity metric is an integer value from 0-99 computed by comparing the current audio power level to a measured noise floor. If the current power is less than or equal to the floor then the metric value should be 0. If the current power is between 1dB and 20dB above the floor then the metric value should be between 1 and 89. If the current power is between 20dB and 40dB above the floor then the metric value should be between 90 and 98. If the current power is greater than 40dB above the floor then the metric value should be 99.

The calculation of the noise floor should be dynamic during the course of the call, although historical information may be used as the initial value. The goal of the noise floor is to eliminate artificially high audio activity metric values due to constant background noise.

11 Changes from 1.6 profile

1. CABAC is now optional for HD and 30 fps auxiliary encoders. MediaOpts packet signaling has changed to reflect this update. However, CABAC is used by default and can only be turned off for multi-point sessions in deployments using the Cisco TelePresence 1.6.4 software release or later for the CTMS multipoint server, and CTS 1.6.5 or later for the endpoints.
2. Added restrictions on the SPS IDs which were missing from the 1.6 profile.
3. Added clarification for what constitutes a hold/resume scenario that triggers a TIP re-negotiation
4. Mandated SSRC and CSRC changes for a resumed call
5. Added support for RTP senders and receivers of the same endpoint using different SSRCs
6. Corrected the fact that not all audio streams transmitted by TIP endpoints are AAC-LD

7. Clarified that the video activity metric byte is not to be transmitted for G711/G722 streams
8. Added RTP video timestamps requirement
9. Indicated the possibility of receiving a video refresh request without a FLAG field and how an endpoint should deal with such request

12 References

[1] IETF RFC 3261 "SIP: Session Initiation Protocol"
[2] IETF RFC 3264 "An Offer/Answer Model with the Session Description Protocol (SDP)"
[3] IETF RFC 2327 "SDP: Session Description Protocol"
[4] IETF RFC 3550 "RTP: A Transport Protocol for Real-Time Applications"
[5] IETF RFC 3551 "RTP Profile for Audio and Video Conferences with Minimal Control"
[6] IETF RFC 4961 "Symmetric RTP/RTP Control Protocol (RTCP)"
[7] IETF RFC 3984 "RTP Payload Format for H.264 Video"
[8] IETF RFC 4585 "Extended RTP Profile for Real-time Transport Control Protocol (RTCP)-Based Feedback (RTP/AVPF)"
[9] IETF DRAFT draft-ietf-avt-rtp-and-rtcp-mux-07 "Multiplexing RTP Data and Control Packets on a Single Port"
[10] "Cisco Unified Communications Manager SIP Line Messaging Guide (Standard)" http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/sip_msg/line_std/8_0_1/siplst801.pdf
[11] Telepresence Interoperability Protocol 6.0
[12] IETF RFC 2833 "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals"
[13] IETF RFC 4730 "A Session Initiation Protocol (SIP) Event Package for Key Press Stimulus (KPML)"
[14] IETF RFC 5168 "XML Schema for Media Control"
[15] IETF RFC 3640 "RTP Payload Format for Transport of MPEG-4 Elementary Streams"
[16] IETF RFC 3890 "A Transport Independent Bandwidth Modifier for the Session Description Protocol (SDP)"

END OF DOCUMENT