CLUE
Internet-Draft
Intended status: Standards Track
Expires: January 18, 2013

R. Hansen
A. Krishna
A. Romanow
Cisco Systems

July 17, 2012

Call Flow for CLUE
draft-romanow-clue-call-flow-02

Abstract

This draft shows the CLUE call flow demonstrating the use of CLUE in SIP. It also provides information about the message and syntax for CLUE transport establishment and other extensions in SDP. In CLUE participants act as both providers and consumers. The draft includes a detailed example of a typical use case which includes both static and switched captures.

Status of this Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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1. Background

The purpose of this draft is to examine the call flow for implementing the CLUE framework using SIP and other IETF protocols. Although much of the CLUE Framework [I-D.ietf-clue-framework] is reasonably well agreed upon in the CLUE WG, in order to develop a feasible call flow, we had to make assumptions about many aspects of CLUE that have not yet been decided: what the transport will be, what the message format will be, etc. While these assumptions are not definitive final answers, we have tried to make sensible decisions based on existing drafts and common sense. Perhaps by examining in more detail how the call flow could be handled some light will be shed light on some of the proposals.

The following assumptions related to UDP are being made: these are also stated in [I-D.romanow-clue-sdp-usage]

- For each CLUE media type and content type (audio, video-main, or video-slides) , there will be at most one corresponding SDP media stream ("m=" line). Note: video-main and video-slides are both using SDP "video" media type.
- Multiple media captures of the same CLUE media and content type, or different encodings of a given media capture will all use the same SDP media stream, i. e., via RTP multiplexing.
- It is acceptable to use more than one offer/answer exchange to setup the whole Telepresence session.
- The Telepresence session is established by setting up sets of unidirectional streams (not bidirectional streams).

Today telepresence endpoints actively negotiate audio and video SDP media streams using the SDP offer/answer model during call establishment and during mid-call features, such as hold resume. The CLUE framework adds additional parameters to a telepresence session. Each party in the CLUE conference is usually both a provider and a consumer, whether the conference is point-to-point or multipoint. The CLUE parameter values for one provider may well not be the same values for the other provider. Thus each party needs to advertise its provider encoding parameter values to the other side. This is not the typical communication model for SDP offer/answer, which is more often a bidirectional agreement on parameter values.

As specified in the CLUE framework, parties in a telepresence conference do not negotiate a single value between them; therefore Offer Answer Model with SDP [RFC3264] is not used for the full negotiation of all encoding related values. Rather, we propose, sending CLUE parameter values via a new CLUE signaling protocol which will be negotiated via SIP. The CLUE messages consist of provider advertisements and consumer requests. [Edt. We use consumer request

and consumer configuration interchangeably. The framework uses the term "configuration", we tend to use "request" here because we are talking about the message usually, rather than the concept. Hopefully, it will not cause any confusion.]

2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119 [RFC2119] and indicate requirement levels for compliant implementations.

3. Solution Overview

A number of aspects of CLUE remain to be fully defined, including major issues such as the transport mechanism and message structure. In order to provide illustrative examples, this draft makes a number of assumptions and guesses about the final format of CLUE. Where possible an attempt has been made to use existing drafts (such as for the multiplexing methodology), or similar developments in other working groups (such as the use of SCTP over UDP as a transport from RTCWeb). As such, this draft should not be taken as an attempt to specify such aspects, but merely to illustrate how they would be used to convey the necessary data: were other decisions to be made on aspects such as transport, the mechanisms would be different, but the data needing to be conveyed would remain the same.

We have assumed about the call flow that:

1. SIP is used to establish the telepresence conference, similarly to using SIP for non-CLUE video conferencing.
2. The media parameters for the call are established using SDP.
3. A new and separate CLUE signaling protocol is used for carrying CLUE messages.(Investigation of SDP for CLUE [I-D.romanow-clue-sdp-usage] looked at carrying CLUE messages in SDP and concluded it was infeasible.)
4. Usage of the CLUE protocol is established through SDP
5. The transport for the CLUE protocol is SCTP over UDP, or SCTP over DTLS over UDP, following the RTCWeb specification.
6. The transport for CLUE protocol, SCPT over UDT or SCTP over DTLS over UDP is setup in SDP, as illustrated in the following diagram and discussed below.

The following illustration Figure 1 shows a high level diagram of a call flow between a 3 screen endpoint and an MCU, for example. [Edt. The diagram shows SCTP over UDP, it could also show SCTP over DTLS over UDP.]

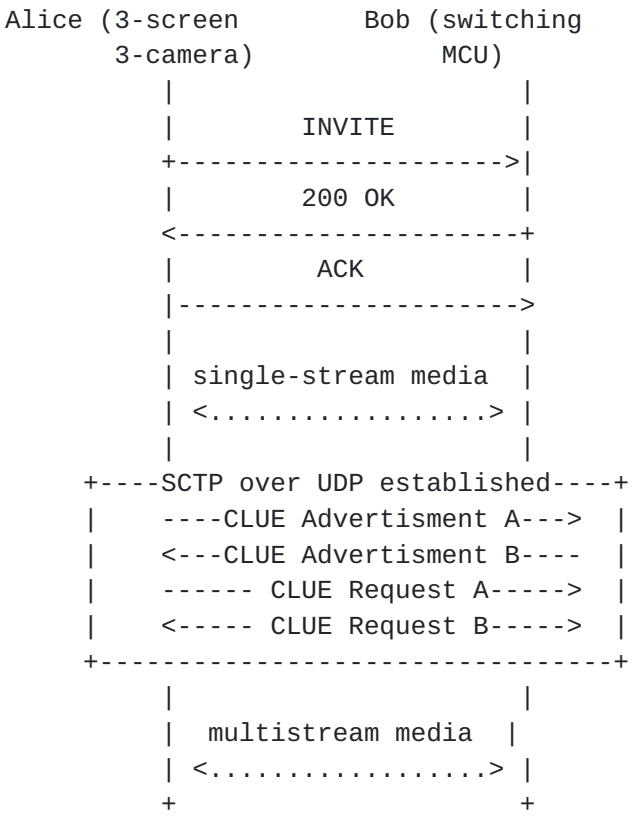


Figure 1: Call setup overview

The next Section 4 discusses the transport for carrying CLUE signaling protocol, following the direction that RTCWeb has taken. Then the telepresence call establishment using SDP is considered in Section 5, including RTP header extension, initial offer answer exchange, and bandwidth considerations. Then Section 5.5 describes CLUE message exchange following the call setup, including provider advertisements and consumer requests. A more detailed coverage of sending and receiving media follows. Interoperability with non-clue implementations is considered in Section 5.7 Then mid-call changes in provider advertisements and consumer requests are considered in Section 6. Section 7 describes a case study which includes both static and dynamic switched captures. Security considerations follows, and there is an appendix with example call flow messages.

4. Transport for carrying CLUE signaling protcol

This section discusses a feasible choice for the transport of CLUE signaling protocol. We are not suggesting the decision on transport be made in this document, but rather we made an assumption based on the requirements and discussion in [I-D.wenger-clue-transport] and what RTCWeb has specified. Most of this section gives a brief overview of the choices made in RTCWeb- SCTP over DTLS over UDP. In this document we have also included the possibility of SCTP over UDP.

4.1. SCTP over DTLS over UDP or SCTP over UDP

There have been several discussion in the CLUE WG suggesting that the transport requirements for CLUE and RTCWeb are sufficiently similar so that CLUE should consider following what RTCWeb has decided. This section briefly reviews RTCWeb's transport specification relevant for CLUE.

RTCWeb WG has reached a general consensus on using SCTP Stream Control Transmission Protocol [RFC4960] encapsulated on DTLS Datagram Transport Layer Security Version 1.2 [RFC6347] encapsulated on UDP for handling non-media data types in the context of RTCWeb. The approach is described in RTCWeb Datagram Connection [I-D.ietf-rtcweb-data-channel].

The transport protocol stack is illustrated in the diagram below Figure 2.

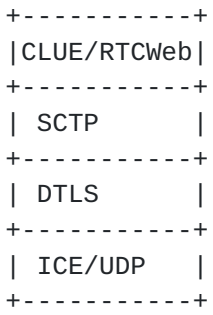


Figure 2: Protocol layers

The services offered by this stack are described in [I-D.ietf-rtcweb-data-channel].

The encapsulation of SCTP over DTLS over ICE over UDP provides a NAT traversal solution together with confidentiality, source authenticated, integrity protected transfers. This data transport

> service operates in parallel to the media transports, and all of them can eventually share a single transport-layer port number. SCTP provides multiple streams natively with reliable, unreliable and partially-reliable delivery modes.

DTLS Encapsulation of SCTP Packets for RTCWEB [I-D.tuexen-tsvwg-sctp-dtls-encaps] describes the encapsulation of SCTP by DTLS.

> The Stream Control Transmission Protocol (SCTP) is a transport protocol originally defined to run on top of the network protocols IPv4 or IPv6. This memo document specifies how SCTP can be used on top of the Datagram Transport Layer Security (DTLS) protocol. SCTP over DTLS is used by the RTCWeb protocol suite for transporting non-media data between browsers.

In addition a third RTCWeb related draft, Stream Control Transmission Protocol (SCTP)Based Media Transport in the Session Description Protocol (SDP) [I-D.ietf-mmusic-sctp-sdp] describes in detail how SDP can handle setting up SCTP and DTLS. (The draft does not consider SCTP over UDP.)

> SCTP (Stream Control Transmission Protocol) is a transport protocol used to establish associations between two endpoints. This document describes how to express media transport over SCTP in SDP (Session Description Protocol). This document defines the 'SCTP', 'SCTP/DTLS' and 'DTLS/SCTP' protocol identifiers for SDP. We have followed the syntax in this document.

We have followed this syntax here.

5. Initial Call Establishment

This section is not intended to prescribe the flows and messages precisely as they are shown, but rather illustrates the principles.

5.1. First offer-answer exchange

The flow illustrated below shows Alice calling Bob offering SDP parameters that are typical of an offer. The new extensions related to CLUE are highlighted below in Figure 3.

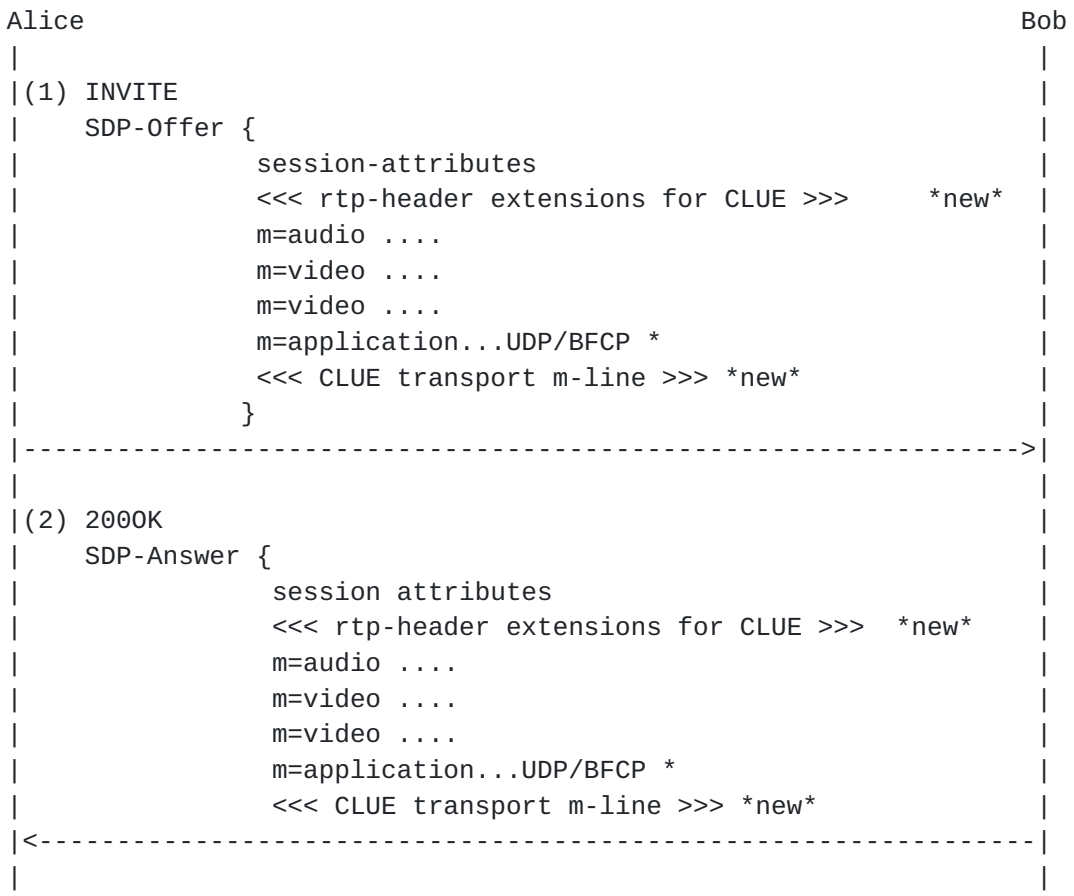


Figure 3: First SDP offer answer exchange

In order to support CLUE using SIP several issues need to be considered during the initial call setup. These include:

1. Extension of the RTP header for telepresence calls. As part of CLUE usage, two such potential requirements are under consideration. Refer to sec 5.2.
2. Establishment of CLUE signaling channel using a separate application m-line. Sec 5.3
3. Bandwidth considerations during initial setup. Sec.5.4

5.2. RTP-header extension and SDP support for CLUE calls

Currently 2 proposals are under consideration for mechanisms that require using RTP extension headers A General Mechanism for RTP Header Extensions [RFC5285]. Although neither proposal has been accepted, we are showing how each would be handled if they are added to the CLUE framework [I-D.ietf-clue-framework].

The first mechanism that requires an RTP header extension is for

associating RTP streams with CLUE captures as described in RTP Usage for Telepresence Sessions [I-D.lennox-clue-rtp-usage]. The second mechanism is using an audio rendering tag to associate audio captures with video captures as described in [I-D.romanow-clue-audio-rendering-tag]. Both use session-level SDP parameters as recommended by [RFC5285].

5.2.1. RTP extension header to associate RTP stream with CLUE capture, CLUE demux id

Telepresence calls multiplex encoded streams from multiple similar media sources on a single RTP session for the same media-type/and content-type. For example, Alice has a 3 camera system but in SDP she negotiates a single RTP session for main video. How does she associate these separate RTP streams with the CLUE capture ids to enable Bob, the receiver, to know where to send the audio and video streams? How do the RTP streams get routed to the appropriate decoders and output devices? This is considered in detail in [I-D.lennox-clue-rtp-usage].

The proposal is to extend the RTP header to embed a unique id in each RTP packet, referred to here as the CLUE demux id. This requires negotiating a unique "id" out of band in SDP so that multiple such extensions could co-exist. This is achieved by using the extmap extension in SDP as specified in [RFC5285].

The diagram in Figure 5 illustrates the call flow for the identifier being used to signal the CLUE demux id information in RTP extension header. In this diagram and subsequent ones, we have used the following drawing to represent a camera, by a circle, and a screen, by a square.

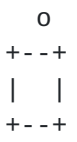


Figure 4: Representation of and endpoint camera and screen

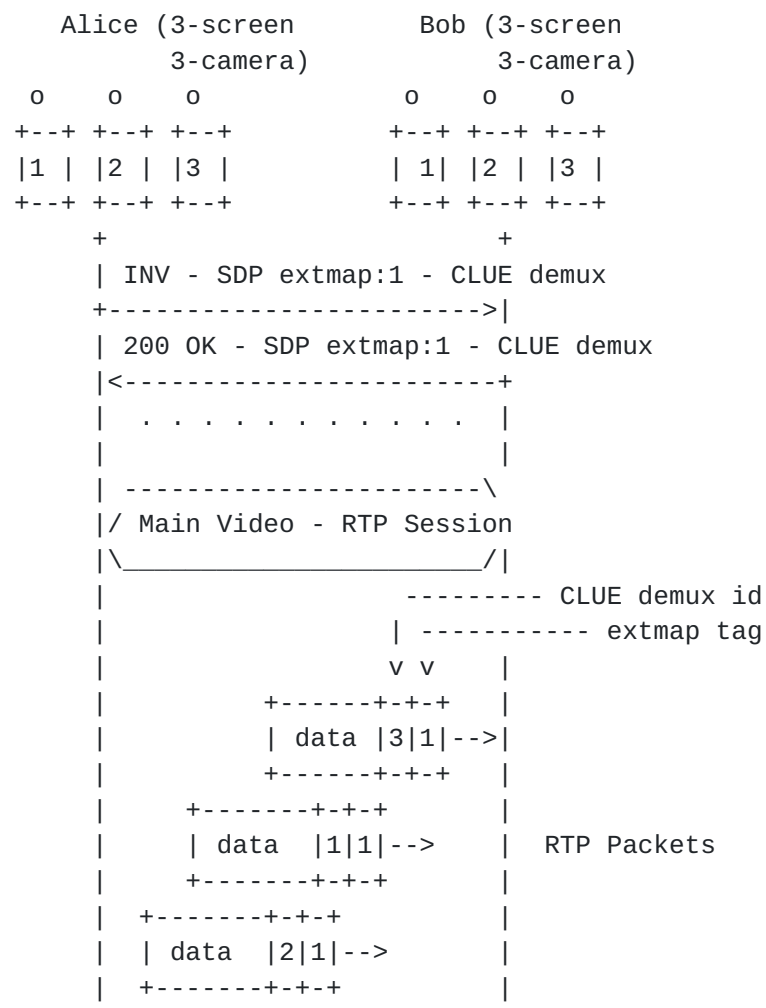


Figure 5: SDP example using extmap id-1

The extmap is a SDP session level attribute in the following example using extmap-1:

```
v=0
o=3ss 40471 40471 IN IP4 10.47.197.103
c=IN IP4 10.47.197.103
t=0 0
a=extmap:1 urn:ietf:params:clue:demux
```

5.2.2. RTP extension header for audio rendering tag

As explained in [I-D.romanow-clue-audio-rendering-tag], to support directional audio the receiver needs to know where to render audio in relationship to video captures. In some circumstances the spatial

information is not available in the provider advertisement, so the association cannot be made in that way. The proposal for these cases is that the consumer optionally tells the provider an audio tag value corresponding to each of its chosen video captures, which enables received audio to be associated with the correct video stream, even when the set of audible participants changes.

In the example provided below, the packet shows dual header extensions - one for associating RTP stream with capture as above, and one for associating audio capture with video capture.

The SDP session level attribute extmap id-3 below depicts the identifier being used to signal the audio tag field in the RTP extension header.

a=extmap:3 urn:ietf:params:clue:audiotag

5.3. CLUE channel establishment using SDP

Most importantly, the initial offer/answer negotiates the port and transport information for establishing a CLUE signaling channel. The new application m-line details the necessary information. As explained in Section 4, the CLUE signaling protocol will ride on top of SCTP transport.

Figure 6 shows an example call flow for establishing an unsecure CLUE channel.

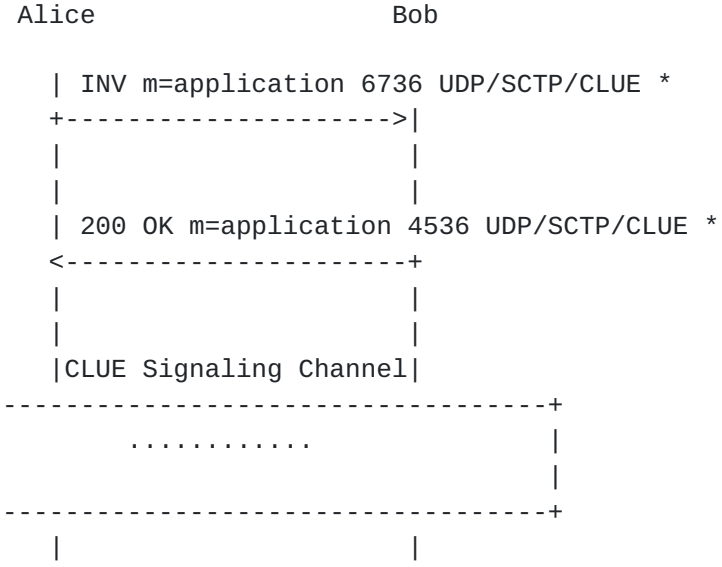


Figure 6: CLUE transport channel establishment

```
SDP Offer:
INVITE bob@biloxi.net SIP/2.0
...
Content-Length: 1000

v=0
....
m=audio...
...
m=video...
...
m=application 6736 UDP/SCTP/CLUE *
a=setup:actpass
a=connection:new

SDP Answer:
SIP/2.0 200 OK
...
Content-Length: 1000

v=0
....
m=audio...
m=video...
....
m=application 4534 UDP/SCTP/CLUE *
a=setup:active
a=connection:new
```

The same example encrypted with DTLS shall look as shown in Figure 7. A fingerprint hash attribute is included at the SDP session level in order for the other side to authenticate the identity while exchanging the certificate during the connection setup.

```
SDP Offer:
a=fingerprint:
SHA-1 64:02:2A:20:92:CD:DB:80:9F:68:0D:EF:AC:99:95:34:89:C6:7D:34
...
m=application 6736 UDP/DTLS/SCTP/CLUE *
a=setup:actpass
a=connection:new

SDP Answer:
a=fingerprint:
SHA-1 4B:AC:B7:B1:3F:82:18:3B:54:02:12:DF:3E:5D:49:6B:19:E5:7C:BA
...
m=application 4534 UDP/DTLS/SCTP/CLUE *
a=setup:passive
a=connection:new
```

Figure 7: Secure CLUE channel establishment

In an unencrypted call the 'setup' parameter in SDP is used to negotiate which participant will listen for an incoming SCTP connection, while the other participant initiates the connection. In an encrypted call using DTLS the 'setup' parameter negotiates which participant will establish the DTLS connection (the same participant then establishes SCTP over the DTLS channel once DTLS has been established).

5.4. Bandwidth considerations for CLUE

We discuss two models of bandwidth allocation for CLUE telepresence calls. In the first mechanism bandwidth is allocated as needed through successive re-INVITEs. This has the advantage of being bandwidth efficient but may require multiple re-INVITEs. In the second approach the maximum allowable bandwidth is allocated initially. This saves on re-INVITEs but may allocate bandwidth not needed. If desired the participant can-reinvite with a lower bandwidth.

5.4.1. Allocate as-you-go

The bandwidth initially offered by Alice is set to some default value (say for a single-screen system). The bandwidth requirement is bound to change as the call progresses on learning the remote end capabilities and the selections made thereafter. Any increase in the usage of bandwidth based on the selected captures will result in the telepresence endpoint sending out a new re-INVITE requesting the intermediaries to allocate extra bandwidth from their pool. The call flow is illustrated in Figure 8.

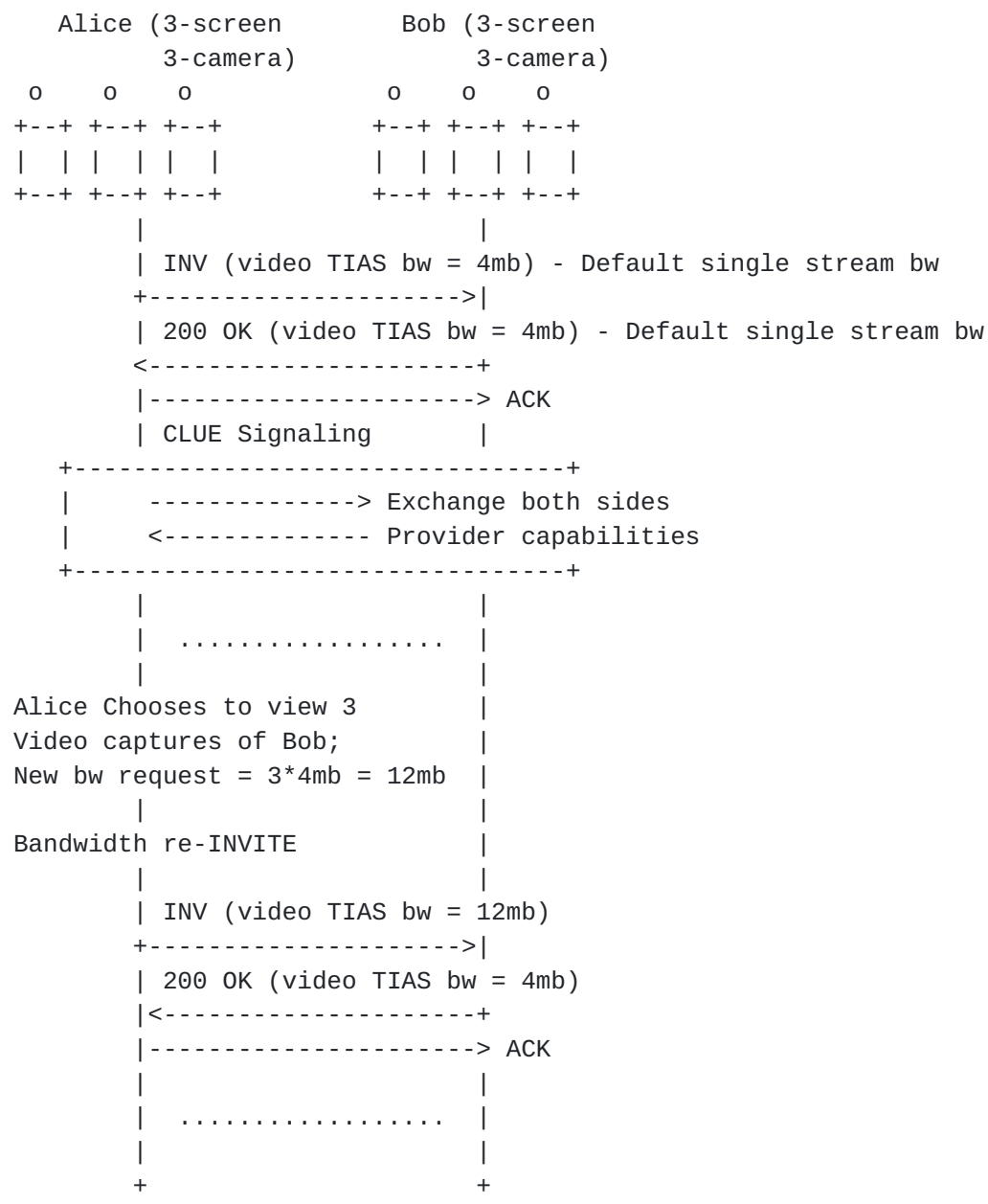


Figure 8: Allocate bandwidth as needed

The advantage of this approach is the usage efficiency, as during the initial call setup the remote endpoint's capabilities are still unknown. Also, even after learning the remote end's capabilities through CLUE, there is no guarantee that Alice shall decide to view all or a subset of the captures thereafter. So the reserved network bandwidth is the amount that is currently being used.

The drawback is the potential need for one or more re-INVITEs for re-negotiating new bandwidth numbers from either side if they choose to

view additional captures or choose captures in additional encoding formats. Also due to the asynchronous nature of the consumer requests, the flow is susceptible to glare re-INVITEs.

The glare would be more pronounced in the case after the first CLUE provider advertisement exchange. This may be due to these systems being pre-programmed to re-configure themselves immediately after learning each other's capabilities. Figure 9 illustrates the glare condition issue.

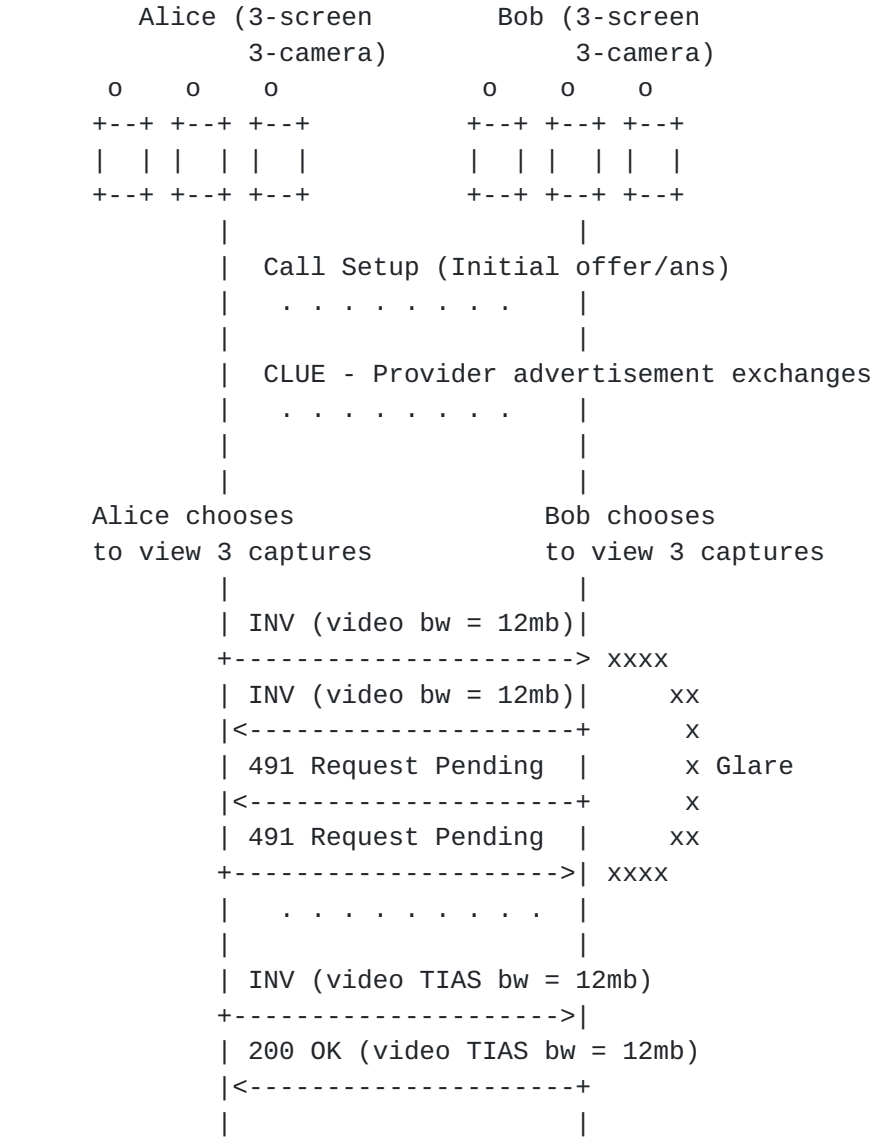


Figure 9: Glare condition in allocate bandwidth as needed

5.4.2. Pre-allocation of bandwidth

An alternate method to avoid the need for multiple re-INVITEs to increase bandwidth is to offer maximum bandwidth size that this device is capable of handling in the initial offer/answer. The call flow is illustrated in Figure 10. However this approach risks unnecessary call failures in the case the network bandwidth is provisioned and policed.

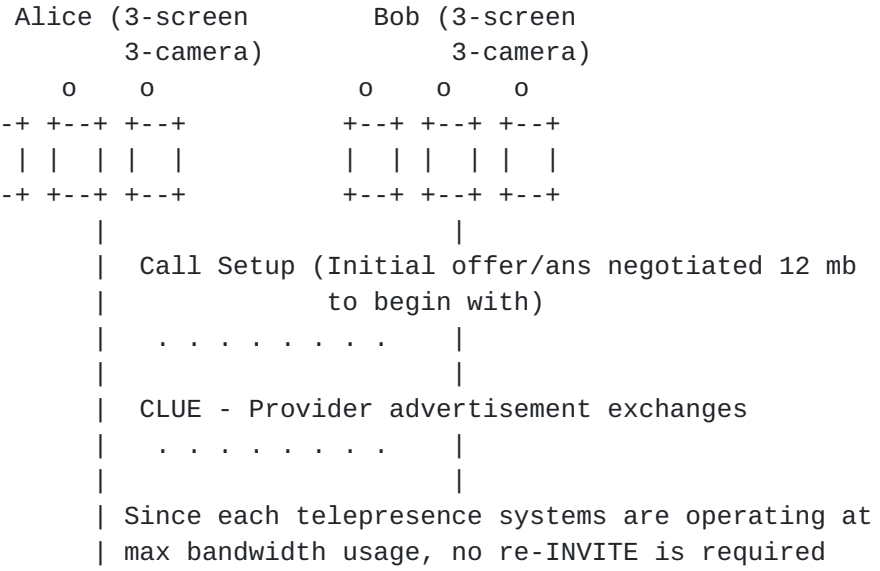


Figure 10: Maximum bandwidth pre-allocation

Implementations must be able to cope with the remote endpoints operating in either mode.

5.5. CLUE message exchange

This section describes the call flow for CLUE messages following call setup.

Following the SIP and SDP call setup, CLUE messages are exchanged. There are only 2 types of CLUE messages: provider advertisements and consumer requests.

Each participant in a CLUE telepresence conference, whether an endpoint or an MCU, is most likely to act as both provider and consumer, as discussed above. Thus each will send provider advertisements and receive consumer requests for those advertisements; and each will also receive provider advertisements

and send consumer requests.

The provider advertisements will consist of the following data elements:

- captures
- capture scenes
- encoding groups
- encodings
- simultaneous transmission set

The consumer requests will consist of the following data elements:

- captures
- encodings

5.5.1. Message sequencing

In considering the sequencing of messages, either participant could be first to send a provider advertisement. The next message, sent by the other participant, could be either a provider advertisement or a consumer request in response to the provider advertisement. This is illustrated in the 2 examples below. The first example shows Bob's provider advertisement sent before he responds to Alice's provider advertisement. The second example shows Bob sending a consumer request in response to Alice's provider advertisement before sending his own provider advertisement.

The important thing to keep in mind is that these messages are sent asynchronously and can be sent not only at the beginning of a call, but any time throughout the call when a parameter value has changed.

Figure 11 shows an example of CLUE messages with provider advertisements from each side following one another.

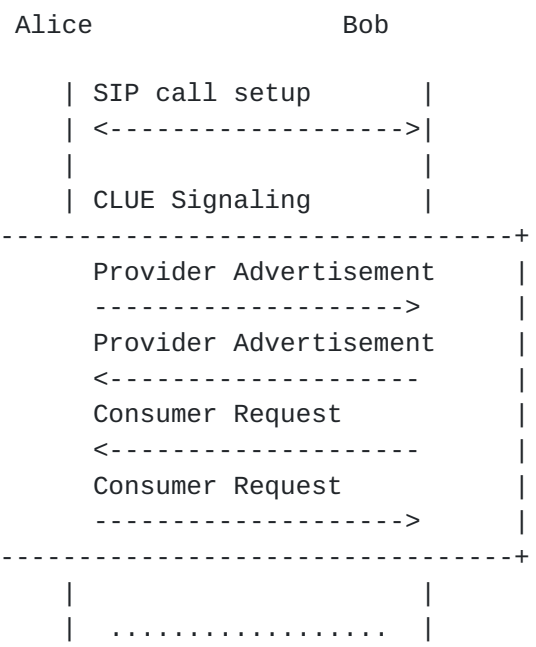


Figure 11: Back-to-back provider advertisements

Figure 12 shows an example of CLUE messages with provider advertisement directly followed by consumer request.

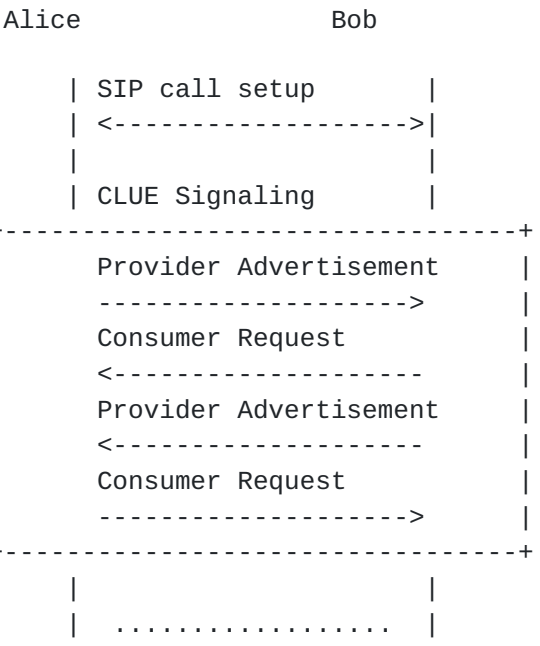


Figure 12: Provider advertisement followed by consumer request

Appendix A shows example provider advertisement and consumer request messages related to Section 7. Keep in mind these are not proposals, just examples using data elements taken from the current versions of the [I-D.romanow-clue-data-model] and [I-D.ietf-clue-framework].

5.5.2. Signalling bandwidth allocation

Call agents and other intermediaries responsible for bandwidth allocation may grant the bandwidth requested in the consumer request. Or, an intermediary may not allow the requested bandwidth. In this case, a feedback mechanism is desirable so that the receiver can adjust accordingly. This is a more generic SIP issue and is thus beyond the scope of this document.

5.6. Sending and receiving media

This section discusses CLUE's use of RTP extension header and RTCP.

5.6.1. RTP

CLUE allows multiple media streams to be multiplexed onto a single RTP session, to reduce the complexity of negotiating independent ports for an asymmetric and variable number of media streams, and aid NAT and SBC traversal [I-D.lennox-clue-rtp-usage]. Demultiplexing these media streams is achieved via the urn:ietf:params:clue:demux RTP header extension negotiated in the SIP messages. The provider MUST include this header extension in all media packets sent due to a successfully processed CLUE consumer request.

The value of the header extension for a given stream MUST match the value of the 'CLUE demux id' element for the associated stream in the consumer request. Figure 13 shows an example of an RTP packet containing an RTP header extension

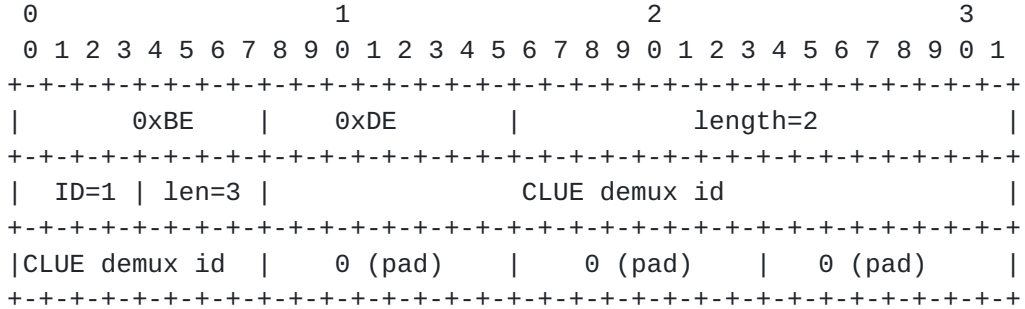


Figure 13: RTP packet showing extension header for CLUE demux id

See [I-D.lennox-clue-rtp-usage] for the specifics of the demultiplexing process.

A receiving device which has sent at least one CLUE consumer request MUST examine RTP packet headers for the presence of this header extension. If it is not present then the packet should be treated as part of a single-stream media stream. If the extension header is present then the value of the CLUE demux id allows the various multiplexed streams to be differentiated. Packets with a CLUE demux id that does not correspond to one of the ids sent in the most recent valid CLUE consumer request MUST be discarded.

5.6.2. RTCP

As described in the preceding session, multiple media streams are multiplexed onto a single RTP session. This RTP session SHOULD be matched by a single set of RTCP messages, sent in a conventional fashion. A device SHOULD send RTCP receiver reports, sender reports, SDES and other packets as it would in the single-stream case. However, when multiple media streams are present each RTCP packet SHOULD contain multiple chunks; each chunk can be used to identify a specific SSRC and include the information relevant to that media stream. This allows standard statistics, packet loss indications and other RTCP metrics to be used. CNAME, as part of the RTCP SDES message, can be used to indicate synchronization between multiple multiplexed media streams when they are generated by encoders sharing a common clock, in the same way it can be used to synchronize streams received on different ports.

5.7. Interoperability with non-CLUE systems

CLUE support in a remote device is identified by the presence of a valid SDP media stream advertising the ability to negotiate the SCTP-based CLUE protocol - absence of this requirement indicates that either the remote device does not support CLUE, or that a middle box in the signaling path has suppressed the ability to do so. A device that receives an SDP offer or answer without a valid media stream advertising the CLUE protocol MUST fall back to a single-stream call until such time as new SDP messages are able to negotiate a CLUE protocol media stream between the devices - see Section 5.7 for details on interoperability with non-CLUE devices. If both sides successfully support CLUE then the call flow proceeds as shown in this draft.

Note that as per Offer Answer Model with SDP [RFC3264], both devices MUST be prepared to receive media for any media streams they have advertised as recvonly or sendrecv. The devices SHOULD also begin sending single-stream media once the initial SDP negotiation is

complete, while the SCTP channel is being set up and before any CLUE messages are sent or received. This minimizes the time in which a basic single-stream call is established to match that of a conventional non-CLUE system, and ensures a baseline level of interoperability. Once CLUE messages have been received and processed the call will then be 'upgraded' to multistream.

5.7.1. Backward compatibility with non-CLUE endpoints

One of the requirements of CLUE is that it allows backwards compatibility with devices that are not CLUE-aware, ensuring that a call between a CLUEful device and a conventional non-CLUEful SIP will result in a conventional, single-stream call.

5.7.1.1. Handling of CLUE SDP by conventional SIP devices

The SIP messages sent by CLUEful endpoint are compatible with conventional SIP compliant devices. Audio and video remain on seperate ports, which not only allows non-CLUE devices to establish calls as normal, but allows seperate QoS settings to be applied to the different media types if desired. All the CLUE-specific changes are in the form of SDP extensions which must either be ignored (new attributes) or answered back appropriately indicating their lack of support (new media line) by the non-CLUE recipient. As such a compliant SIP device without CLUE support will be able to establish a conventional audio-video call on receiving a CLUEful INVITE.

5.7.1.2. CLUE Option Tag usage

It may prove useful to explicitly signal CLUE support; the use of a 'clue' option-tag makes the SIP devices explicit about CLUE support. Endpoints could use this tag to make the registrar aware of their ability to support CLUE, as shown in Figure 14.

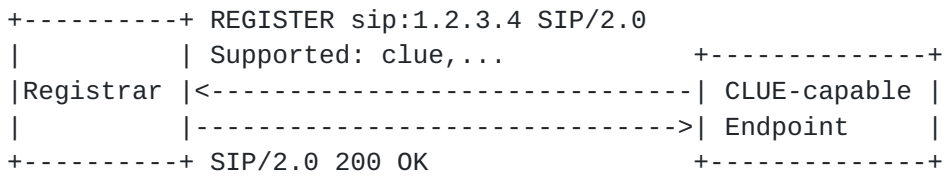


Figure 14: CLUE option tag for registration

Further, the option-tag also allows CLUE endpoints to specify that they do not wish to fall back to single stream behaviour (by including the 'clue' option-tag in a Require field) in cases where

they contact a non-CLUE endpoint:

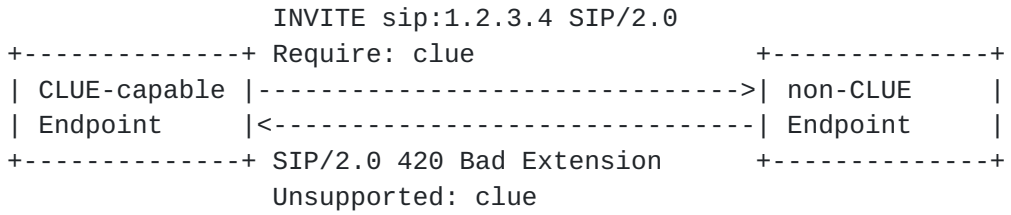


Figure 15: CLUE option tag for require

Finally, the option-tag may be used in an OPTIONS message to signal that a device supports CLUE even without the presence of an SDP, as show in Figure 16.

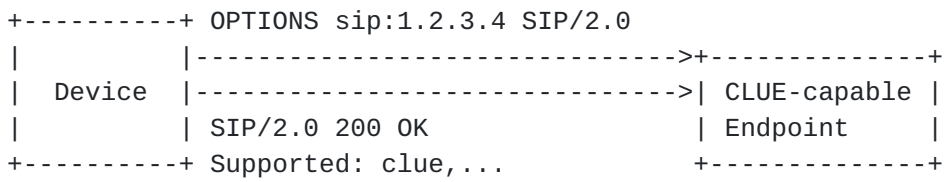


Figure 16: CLUE option tag for options

5.7.1.3. Non-compliant devices and usage of CLUE option-tag

There are non-compliant SIP devices which may react badly when receiving SDP extensions they do not understand, responding with a 400 Bad Request or in some other fashion. While the principal aim of this document is not to work around devices that do not implement specified behaviour, there are a number of potential strategies for dealing with such legacy devices:

1. The calling device may attempt to send a second INVITE without any CLUE parameters.
2. A back-to-back-user-agent in the call path to the legacy device may recognise its lack of CLUE support (via static configuration, lack of a 'Supported: clue' options tag in a REGISTER message or lack of CLUE parameters in the response to an OPTIONS message) and strip CLUE parameters from INVITE messages directed to that device.

6. Mid-call CLUE messages and feature interactions

A provider may send a new advertiesment at any time, and a consumer may send a new request at any time (once it has received an initial advertisement). There is not a one-to-one mapping between advertisements and requests - a consumer may send multiple requests to a single advertisement. Advertisements contain a sequence number, while consumer requests contain both a sequence number and a reference to the sequence number of the advertisement to which they correspond - the provider can use these to detect and discard requests relating to a previous, now invalid advertisement.

Consumer re-configuration can occur anytime during the course of a call. Possible triggers include but are not restricted to:

- New provider advertisement to indicate a changes in captures or encodings.
- Change in device configuration (such as availability of new screen) or user selection at the consumer end.
- Invocation of features such as Hold/Resume can potentially alter the call topology/view.

The bandwidth requirement and need for re-INVITE are based on the increase or decrease in usage. Implementations can choose an appropriate bandwidth strategy as discussed previously. However, they must ensure necessary bandwidth is in place before choosing any new captures.

6.1. CLUE messages triggered due to change in device capabilities

The following is a mid-call CLUE provider advertisement from Bob advertising capture-scene with 3 captures as opposed to 2 previously (could be triggered by turning on the 3rd camera. In order to keep the call-flow simple we assume both Alice and Bob allocate a large bandwidth (20Mb). The consumer request is within that bandwidth boundary. The call flow is shown in Figure 17.

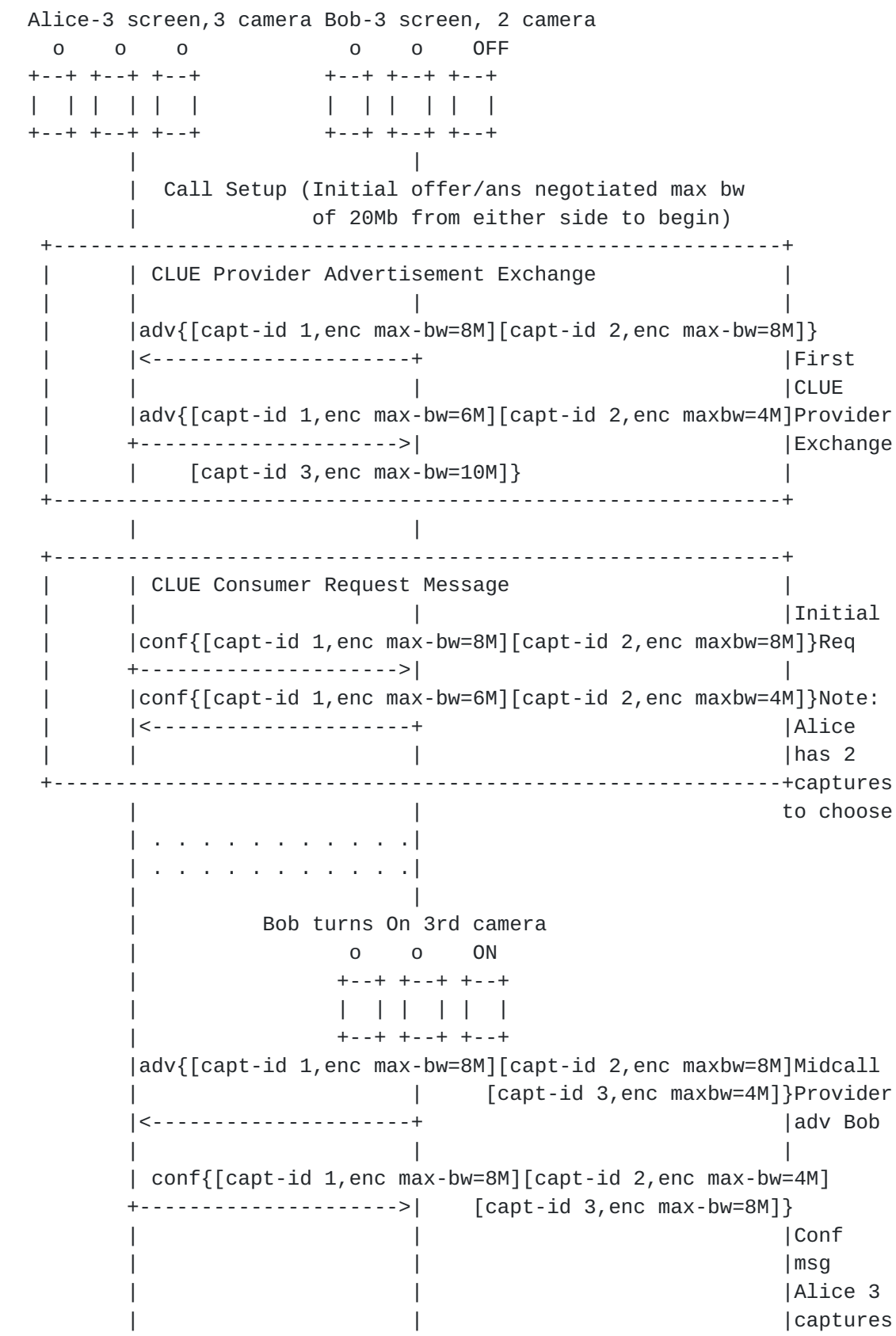

Alice-3 screen,3 camera Bob-3 screen, 2 camera
OFF
Call Setup (Initial offer/ans negotiated max bw
of 20Mb from either side to begin)
CLUE Provider Advertisement Exchange
adv{[capt-id 1,enc max-bw=8M][capt-id 2,enc max-bw=8M]}
adv{[capt-id 1,enc max-bw=6M][capt-id 2,enc maxbw=4M]
[capt-id 3,enc max-bw=10M]}
First
CLUE
Provider
Exchange
CLUE Consumer Request Message
conf{[capt-id 1,enc max-bw=8M][capt-id 2,enc maxbw=8M]}
Initial
Req
conf{[capt-id 1,enc max-bw=6M][capt-id 2,enc maxbw=4M]}
Note:
Alice
has 2
captures
to choose
Bob turns On 3rd camera
ON
adv{[capt-id 1,enc max-bw=8M][capt-id 2,enc maxbw=8M]
[capt-id 3,enc maxbw=4M]}
Midcall
Provider
adv Bob
conf{[capt-id 1,enc max-bw=8M][capt-id 2,enc max-bw=4M]
[capt-id 3,enc max-bw=8M]}
Conf
msg
Alice 3
captures

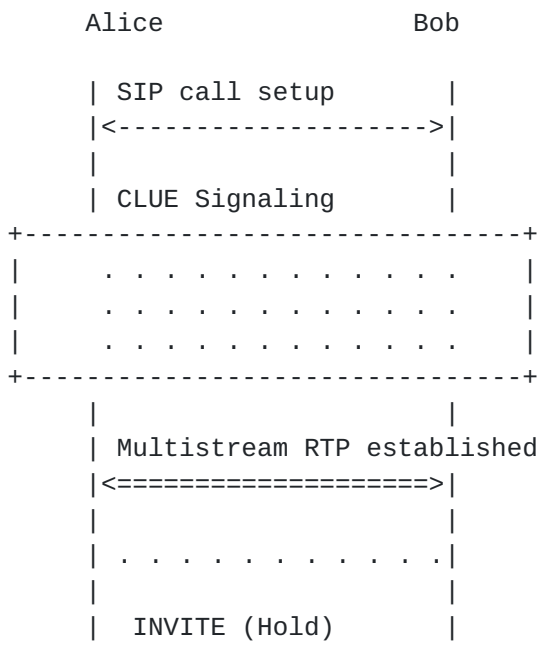
Figure 17: Mid-call change in provider advertisement

6.2. Closure and Re-establishment of the SCTP channel

A mid-call SDP offer or response may result in the closure of the SCTP channel, or a re-establishment of a channel that is either active or inactive. Reasons for these changes may include an endpoint going on or coming off hold, a device re-routing media to a different recipient, or an endpoint no longer wishing to continue to use multistream functionality. If the SCTP channel is reestablished it is not safe to assume that the other end of the SCTP channel has knowledge of the previous state of CLUE messaging - in some cases the new recipient of the channel may be an entirely new individual.

Depending on the intention of the initiator of hold, the CLUE connection could be preserved or re-established. For the shared-line and presence scenarios it is recommended to teardown and re-establish. If the connection was preserved across hold-resume the previous CLUE advertisements and configuration state continues to be valid. The offerer/and or answerer could choose to reuse the existing connection by setting a=connection:existing. If not the connection could be re-established by either piggybacking on the same SDP offer/answer for resume or seperately. The CLUE provider advertisment and configuration is exchanged all over again for establishing multistream.

Figure 18 depicts the Hold-Resume in which the CLUE signaling channel is re-established.



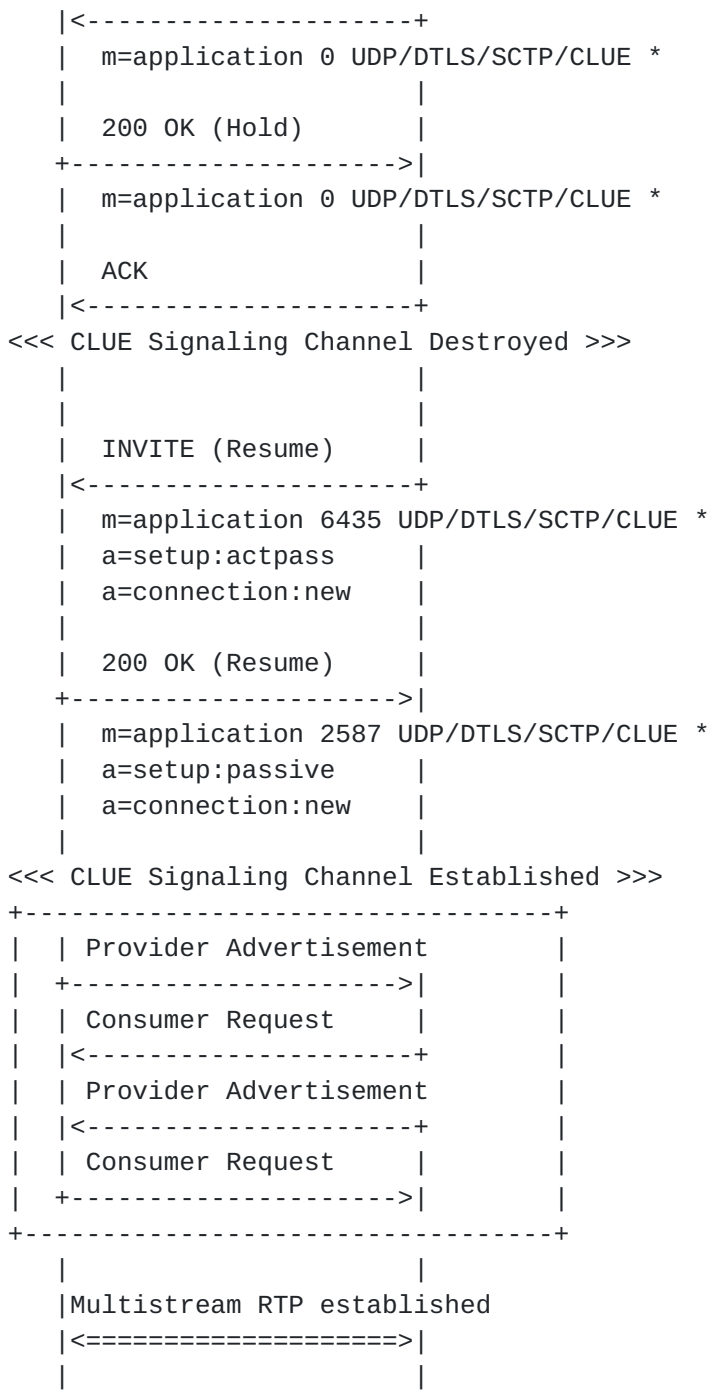


Figure 18: HOLD/RESUME with re-establish

7. Case study: example call flow

To help illustrate how all the aspects of a multistream call using CLUE tie together, a complete set of SIP and CLUE message is provided

for a sample call between two participants. In this example one participant is an endpoint and the other is an MCU. If both participants were endpoints, the call flow wold be the same although some of the content may be different.

Alice in this example is a three-screen endpoint, with three cameras. She is able to send all three camera streams individually, or can send a single switched video stream (based on the current active speaker, or most recent speaker if no one in her room is speaking). Her system also has three microphones, but only advertises a single, composed audio stream made up of the streams of three microphones mixed together. Bob, in contrast, is an MCU that switches media between different participants in a conference. Bob advertises up to three video streams and three audio streams, with alternate offerings for one- and two-screen systems. Figure 19 shows an overview of the call flow.

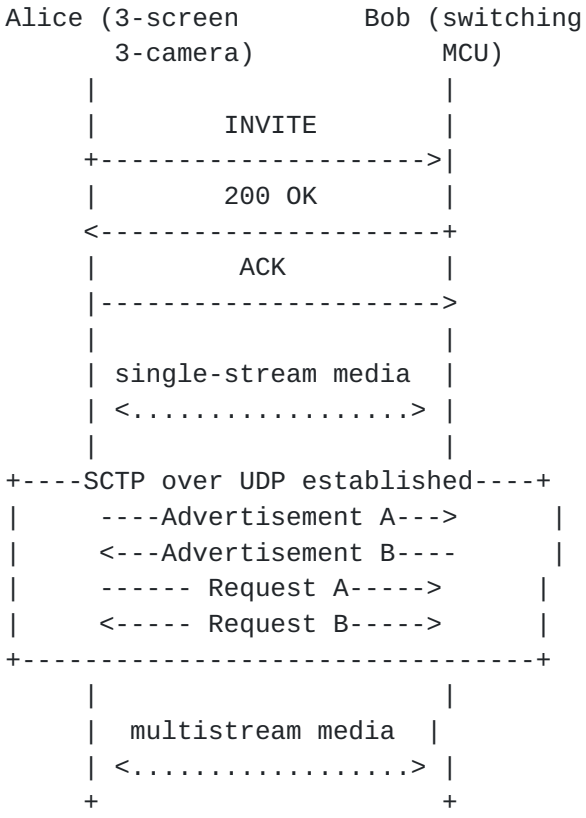


Figure 19: Example call flow

The flow begins with Alice sending an INVITE to Bob (Appendix A.1.) The SDP includes an audio stream, offering AAC and G.711(u), and a video stream offering H264. Alice is limited to receiving 6M of

bandwidth. Alice also includes a media line for CLUE over SCTP over UDP (for simplicity in this example the stream is unencrypted). The attributes for the stream show that this is a new connection, and that Alice is willing to be either active or passive with respect to establishment of the media session. Finally, Alice includes an SDP 'extmap' parameter, advertising that she understands the multiplex id RTP header extension, and that Bob should use an ID of 1 to identify such extensions; this parameter is included at the session level, as it applies to both audio and video media streams.

Bob responds with a 200 OK (Appendix A.2. - any prior 1xx messages have been elided for brevity). The SDP also includes an audio stream supporting AAC and G.711(u) and a video stream offering H264. Bob is able to receive up to 10M of media. Bob includes a media line for CLUE over SCTP over UDP that matches Alice's; the 'connection:new' attribute corresponds to hers, and Bob chooses to be passive in establishment of the stream. Bob also includes an SDP 'extmap' parameter, in this case specifying that Alice should send multiplexed media with an RTP extension header with an ID of 3.

Alice sends an ACK, not included in the appendix.

Having completed the initial SIP call setup, Alice and Bob begin sending H264 video and AAC audio in a conventional single-stream fashion. Further, Alice initiates a STCP connection to Bob on port 3782, encapsulated in UDP, as was negotiated in the SDP exchange.

With the CLUE transport established both sides are free to begin sending CLUE messages. The call flow shows Alice sending an advertisement before Bob, but in practice there is no correlation between the sending of these messages; Bob may send his message first, or both may be sent simultaneously. Both sides of the CLUE message exchange are independent of one another.

Alice's CLUE advertisement includes four video captures (Appendix A.3.) The first three are static captures that correspond to the three cameras of the system, while the final one is a switched capture to show the current active speaker. The static captures give the physical coordinates of the area of capture, in millimetres, while the switched capture does not include an area of capture. There is a single, mixed audio capture, again with no area of capture. The entries for the capture scene allow the recipient to understand which captures provide 'equivalent' views. In this case it illustrates that to get a 'complete' view of the scene the recipient should subscribe to either captures 1, 2 and 3, or to capture 4. Finally, the information in the encoding groups shows that while Alice can supply any given video stream at up to 4M, she can only supply a total of 6M of video.

In contrast, as a switching MCU Bob has no static captures (Appendix A.4.) Instead, Bob wants to be able to supply one-, two- and three-screen switched capture views. He does this by advertising six video captures with non-physical area of capture coordinates; these coordinates are used to illustrate the fraction of the overall scene a capture represents (and how to render the captures to ensure their adjacency is correct and no multi-screen systems are rendered back-to-front). A similar approach is taken for audio. This is reinforced in the entries, which illustrate the captures to which a recipient should subscribe to receive a 'correct' view of the conference. The encoding information supplied by Bob shows that he is able to supply up to three video streams, with the only encoding maximum that of the 10M advertised in his SDP for the stream total.

Having received Bob's CLUE advertisement, Alice then sends her consumer request (Appendix A.5.) As she is a three-screen system she requests video captures 4, 5 and 6, which correspond to the switched captures for a three-screen system. She includes a 'max-bandwidth' element, limiting each stream to 2M. Having two speakers she subscribes to audio captures 8 and 9, which correspond to the two-channel audio advertised by Bob.

Having received Alice's CLUE advertisement, Bob sends his initial consumer request (Appendix A.6.) Bob's request is straightforward, requesting the three static camera streams and the mixed audio stream, and adds no additional encoder limitations beyond those already imposed by his SDP.

Both Bob and Alice now send multistream audio and video, the RTP packets including the multiplex extension header with CLUE demux ids specified in the consumer requests. Note again that, though the example illustrates Alice and Bob sending their CLUE messages in order, this is not a requirement; Alice might send her advertisement, receive Bob's request and begin sending multistream media before ever receiving Bob's advertisement. At any time Alice or Bob may send further CLUE advertisement or request messages, which invalidate previous messages. They may also send SIP messages to update or alter media parameters.

8. Security Considerations

This document provides an overview of the call-flow of a CLUE multistream telepresence call, and hence is not an appropriate place to definitively identify and deal with security considerations. However, it should be clear from the above that CLUE does pose a number of security challenges: for example, the transport conveying CLUE messages must be encryptable in a fashion that limits the

potential for man-in-the-middle attacks, while the multiplexing of RTP streams must not interfere with the ability to secure these streams against interception or spoofing. These security considerations will be addressed as part of the specification of these aspects of CLUE, but it is believed that well-understood methods (such as DTLS for the CLUE protocol and SRTP for the media) can be used to secure these channels.

9. Acknowledgements

10. IANA Considerations

This draft includes examples of a number of features which, if agreed on by consensus, would have IANA considerations as part of their specification. The examples used here are purely for illustrative purposes, and do not represent the final format of these features; as such this draft has no IANA considerations.

11. References

11.1. Normative References

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Appendix A. Example call flow messages

These message examples are entirely illustrative - they are not meant as a proposal for messages. As much as possible they follow the nomenclature in the [I-D.ietf-clue-framework]. However, there are a few elements in the messages which are not in the framework, such as "mixed" for audio, and sequence numbers which will be necessary in any independent signaling protocol. [Edt. Hopefully, these few additions will not be distracting from the benefit of an example.]

A.1. INVITE from Alice

```
INVITE sip:bob@example.com SIP/2.0
Via: SIP/2.0/TCP 10.47.197.7:5060;branch=z9hG4bK7251784nf
Call-ID: 3ebf09e73b83408f@10.47.197.7
CSeq: 1 INVITE
Contact: <sip:alice@example.com>
From: "Alice" <sip:alice@example.com>;tag=6d1e0bf6d948f0b8
To: <sip:bob@example.com>
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: 525

v=0
o=alice 1 3 IN IP4 10.47.197.7
s=-
c=IN IP4 10.47.197.7
b=AS:6000
t=0 0
a=extmap:1 urn:ietf:params:clue:demux
m=audio 1000 RTP/AVP 101 0
b=TIAS:64000
a=rtpmap:101 MP4A-LATM/90000
a=fmtp:101 profile-level-id=25;object=23;bitrate=64000
a=rtpmap:0 PCMU/8000
a=sendrecv
m=video 1002 RTP/AVP 98
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42e016;max-mbps=35000;max-fs=3600
a=sendrecv
m=application 1004 UDP/SCTP/CLUE *
a=setup:actpass
a=connection:new
```

A.2. 200 OK from Bob

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.47.197.7:5060;branch=z9hG4bK7251784nf
Call-ID: 3ebf09e73b83408f@10.47.197.7
CSeq: 1 INVITE
From: "Alice" <sip:alice@example.com>;tag=6d1e0bf6d948f0b8
To: <sip:bob@example.com>;tag=b91d0ea4
Contact: <sip:bob@example.com>;isfocus
Content-Type: application/sdp
Content-Length: 474

v=0
o=bob 45727 45727 IN IP4 10.47.196.161
s=-
c=IN IP4 10.47.196.161
b=AS:10000
t=0 0
a=extmap:3 urn:ietf:params:clue:demux
m=audio 3778 RTP/AVP 101 0
a=rtpmap:101 MP4A-LATM/90000
a=fmtp:101 profile-level-id=25;object=23;bitrate=64000
a=sendrecv
m=video 3780 RTP/AVP 98 99 34 31
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42e016;max-mbps=244800;max-fs=8160
a=sendrecv
m=application 3782 UDP/SCTP/CLUE *
a=setup:passive
a=connection:new
```

A.3. CLUE advertisement A (from Alice)

```
<advertisement sequence='100'>
  <capture-scene id='1'>
    <spatial-description>
      <scale>millimeters</scale>
    </spatial-description>
    <capture id='1'>
      <media-type>video</media-type>
      <content-type>main</content-type>
      <encoding-group-id>1</encoding-group-id>
      <spatial-description>
        <point-of-capture>
          <point x='0' y='0' z='0'/>
```

```
          </point-of-capture>
          <capture-area>
            <point x='-1000' y='0' z='3000'/>
            <point x='1000' y='-1125' z='3000'/>
            <point x='-1000' y='-1125' z='3000'/>
            <point x='1000' y='0' z='3000'/>
          </capture-area>
        </spatial-description>
      </capture>
      <capture id='2'>
        <media-type>video</media-type>
        <content-type>main</content-type>
        <encoding-group-id>1</encoding-group-id>
        <spatial-description>
          <point-of-capture>
            <point x='0' y='0' z='0'/>
          </point-of-capture>
          <capture-area>
            <point x='1000' y='0' z='3000'/>
            <point x='2600' y='-1125' z='1800'/>
            <point x='1000' y='-1125' z='3000'/>
            <point x='2600' y='0' z='1800'/>
          </capture-area>
        </spatial-description>
      </capture>
      <capture id='3'>
        <media-type>video</media-type>
        <content-type>main</content-type>
        <encoding-group-id>1</encoding-group-id>
        <spatial-description>
          <point-of-capture>
            <point x='0' y='0' z='0'/>
          </point-of-capture>
          <capture-area>
            <point x='-1000' y='0' z='3000'/>
            <point x='-2600' y='-1125' z='1800'/>
            <point x='-1000' y='-1125' z='3000'/>
            <point x='-2600' y='0' z='1800'/>
          </capture-area>
        </spatial-description>
      </capture>
      <capture id='4'>
        <media-type>video</media-type>
        <content-type>main</content-type>
        <encoding-group-id>1</encoding-group-id>
        <switched>true</switched>
      </capture>
      <capture id='5'>
```

```
        <media-type>audio</media-type>
        <content-type>main</content-type>
        <encoding-group-id>2</encoding-group-id>
        <mixed>true</mixed>
      </capture>
      <entries>
        <entry>
          <capture id='1' />
          <capture id='2' />
          <capture id='3' />
        </entry>
        <entry>
          <capture id='4' />
        </entry>
        <entry>
          <capture id='5' />
        </entry>
      </entries>
    </capture-scene>
    <simultaneous-transmission-set />
    <encoding-group id='1'>
      <max-bandwidth>6000000</max-bandwidth>
      <encoding id='1'>
        <max-bandwidth>4000000</max-bandwidth>
      </encoding>
      <encoding id='2'>
        <max-bandwidth>4000000</max-bandwidth>
      </encoding>
      <encoding id='3'>
        <max-bandwidth>4000000</max-bandwidth>
      </encoding>
    </encoding-group>
    <encoding-group id='2'>
      <encoding id='4'>
        <max-bandwidth>64000</max-bandwidth>
      </encoding>
    </encoding-group>
  </advertisement>
```

A.4. CLUE advertisement B (from Bob)

```
  <advertisement sequence='1'>
    <capture-scene id='1'>
      <spatial-description>
        <scale>No Scale</scale>
        <scene-area>
```

```
          <point x='0' y='0' z='0'/>
          <point x='1' y='1' z='0'/>
          <point x='0' y='1' z='0'/>
          <point x='1' y='0' z='0'/>
        </scene-area>
      </spatial-description>
      <capture id='1'>
        <media-type>video</media-type>
        <content-type>main</content-type>
        <encoding-group-id>1</encoding-group-id>
        <switched>true</switched>
        <spatial-description>
          <capture-area>
            <point x='0' y='0' z='0'/>
            <point x='1' y='1' z='0'/>
            <point x='0' y='1' z='0'/>
            <point x='1' y='0' z='0'/>
          </capture-area>
        </spatial-description>
      </capture>
      <capture id='2'>
        <media-type>video</media-type>
        <content-type>main</content-type>
        <encoding-group-id>1</encoding-group-id>
        <switched>true</switched>
        <spatial-description>
          <capture-area>
            <point x='0' y='0' z='0'/>
            <point x='0.5' y='1' z='0'/>
            <point x='0' y='1' z='0'/>
            <point x='0.5' y='0' z='0'/>
          </capture-area>
        </spatial-description>
      </capture>
      <capture id='3'>
        <media-type>video</media-type>
        <content-type>main</content-type>
        <encoding-group-id>1</encoding-group-id>
        <switched>true</switched>
        <spatial-description>
          <capture-area>
            <point x='0.5' y='0' z='0'/>
            <point x='1' y='1' z='0'/>
            <point x='0.5' y='1' z='0'/>
            <point x='1' y='0' z='0'/>
          </capture-area>
        </spatial-description>
      </capture>
```

```
        <capture id='4'>
          <media-type>video</media-type>
          <content-type>main</content-type>
          <encoding-group-id>1</encoding-group-id>
          <switched>true</switched>
          <spatial-description>
            <capture-area>
              <point x='0' y='0' z='0'/>
              <point x='0.33' y='1' z='0'/>
              <point x='0' y='1' z='0'/>
              <point x='0.33' y='0' z='0'/>
            </capture-area>
          </spatial-description>
        </capture>
        <capture id='5'>
          <media-type>video</media-type>
          <content-type>main</content-type>
          <encoding-group-id>1</encoding-group-id>
          <switched>true</switched>
          <spatial-description>
            <capture-area>
              <point x='0.33' y='0' z='0'/>
              <point x='0.67' y='1' z='0'/>
              <point x='0.33' y='1' z='0'/>
              <point x='0.67' y='0' z='0'/>
            </capture-area>
          </spatial-description>
        </capture>
        <capture id='6'>
          <media-type>video</media-type>
          <content-type>main</content-type>
          <encoding-group-id>1</encoding-group-id>
          <switched>true</switched>
          <spatial-description>
            <capture-area>
              <point x='0.67' y='0' z='0'/>
              <point x='1' y='1' z='0'/>
              <point x='0.67' y='1' z='0'/>
              <point x='1' y='0' z='0'/>
            </capture-area>
          </spatial-description>
        </capture>
        <capture id='7'>
          <media-type>audio</media-type>
          <content-type>main</content-type>
          <encoding-group-id>2</encoding-group-id>
          <switched>true</switched>
          <spatial-description>
```

```
          <capture-area>
            <point x='0' y='0' z='0'/>
            <point x='1' y='1' z='0'/>
            <point x='0' y='1' z='0'/>
            <point x='1' y='0' z='0'/>
          </capture-area>
        </spatial-description>
      </capture>
      <capture id='8'>
        <media-type>audio</media-type>
        <content-type>main</content-type>
        <encoding-group-id>2</encoding-group-id>
        <switched>true</switched>
        <spatial-description>
          <capture-area>
            <point x='0' y='0' z='0'/>
            <point x='0.5' y='1' z='0'/>
            <point x='0' y='1' z='0'/>
            <point x='0.5' y='0' z='0'/>
          </capture-area>
        </spatial-description>
      </capture>
      <capture id='9'>
        <media-type>audio</media-type>
        <content-type>main</content-type>
        <encoding-group-id>2</encoding-group-id>
        <switched>true</switched>
        <spatial-description>
          <capture-area>
            <point x='0.5' y='0' z='0'/>
            <point x='1' y='1' z='0'/>
            <point x='0.5' y='1' z='0'/>
            <point x='1' y='0' z='0'/>
          </capture-area>
        </spatial-description>
      </capture>
      <capture id='10'>
        <media-type>audio</media-type>
        <content-type>main</content-type>
        <encoding-group-id>2</encoding-group-id>
        <switched>true</switched>
        <spatial-description>
          <capture-area>
            <point x='0' y='0' z='0'/>
            <point x='0.33' y='1' z='0'/>
            <point x='0' y='1' z='0'/>
            <point x='0.33' y='0' z='0'/>
          </capture-area>
```

```
        </spatial-description>
      </capture>
      <capture id='11'>
        <media-type>audio</media-type>
        <content-type>main</content-type>
        <encoding-group-id>2</encoding-group-id>
        <switched>true</switched>
        <spatial-description>
          <capture-area>
            <point x='0.33' y='0' z='0'/>
            <point x='0.67' y='1' z='0'/>
            <point x='0.33' y='1' z='0'/>
            <point x='0.67' y='0' z='0'/>
          </capture-area>
        </spatial-description>
      </capture>
      <capture id='12'>
        <media-type>audio</media-type>
        <content-type>main</content-type>
        <encoding-group-id>2</encoding-group-id>
        <switched>true</switched>
        <spatial-description>
          <capture-area>
            <point x='0.67' y='0' z='0'/>
            <point x='1' y='1' z='0'/>
            <point x='0.67' y='1' z='0'/>
            <point x='1' y='0' z='0'/>
          </capture-area>
        </spatial-description>
      </capture>
      <entries>
        <entry>
          <capture id='1' />
        </entry>
        <entry>
          <capture id='2' />
          <capture id='3' />
        </entry>
        <entry>
          <capture id='4' />
          <capture id='5' />
          <capture id='6' />
        </entry>
        <entry>
          <capture id='7' />
        </entry>
        <entry>
          <capture id='8' />
```

```
          <capture id='9' />
        </entry>
        <entry>
          <capture id='10' />
          <capture id='11' />
          <capture id='12' />
        </entry>
      </entries>
    </capture-scene>
    <simultaneous-transmission-set />
    <encoding-group id='1'>
      <encoding id='1' />
      <encoding id='2' />
      <encoding id='3' />
    </encoding-group>
    <encoding-group id='2'>
      <encoding id='1' />
      <encoding id='2' />
      <encoding id='3' />
    </encoding-group>
  </advertisement>
```

A.5. CLUE response A (from Alice)

```
<configure sequence='10' advertise-sequence='1'>
  <capture id='4'>
    <multiplex-id>11</multiplex-id>
    <encoding-id>1</encoding-id>
    <max-bandwidth>2000000</max-bandwidth>
  </capture>
  <capture id='5'>
    <multiplex-id>22</multiplex-id>
    <encoding-id>2</encoding-id>
    <max-bandwidth>2000000</max-bandwidth>
  </capture>
  <capture id='6'>
    <multiplex-id>44</multiplex-id>
    <encoding-id>3</encoding-id>
    <max-bandwidth>2000000</max-bandwidth>
  </capture>
  <capture id='8'>
    <multiplex-id>111</multiplex-id>
    <encoding-id>1</encoding-id>
  </capture>
  <capture id='9'>
    <multiplex-id>222</multiplex-id>
    <encoding-id>2</encoding-id>
  </capture>
</configure>
```

A.6. CLUE response B (from Bob)

```
<configure sequence='1' advertise-sequence='100'>
  <capture id='1'>
    <multiplex-id>1</multiplex-id>
    <encoding-id>1</encoding-id>
  </capture>
  <capture id='2'>
    <multiplex-id>2</multiplex-id>
    <encoding-id>2</encoding-id>
  </capture>
  <capture id='3'>
    <multiplex-id>3</multiplex-id>
    <encoding-id>3</encoding-id>
  </capture>
  <capture id='5'>
    <multiplex-id>1</multiplex-id>
    <encoding-id>4</encoding-id>
  </capture>
</configure>
```

Appendix B. Changes From Earlier Versions

Note to the RFC-Editor: please remove this section prior to publication as an RFC.

B.1. Changes From Draft -00

- Editorial changes including proper references.

B.2. Changes From Draft -01

- More editorial changes.
- Edited mid-call messages and feature interactions section. Introduced option tags.
- Cleaned up discussion of tags in RTP

Authors' Addresses

Robert Hansen
Cisco Systems
Langley,
UK

Email: rohanse2@cisco.com

Arun Krishna
Cisco Systems
San Jose, CA
USA

Email: arukrish@cisco.com

Allyn Romanow
Cisco Systems
San Jose, CA 95134
USA

Email: allyn@cisco.com