SIPREC
Internet-Draft
Intended status: Standards Track
Expires: January 31, 2016
Ram Mohan. Ravindranath
Cisco Systems, Inc.
Parthasarathi. Ravindran
Nokia Networks
Paul. Kyzivat
Huawei
July 30, 2015

Session Initiation Protocol (SIP) Recording Call Flows
draft-ietf-siprec-callflows-05

Abstract

Session recording is a critical requirement in many communications environments such as call centers and financial trading. In some of these environments, all calls must be recorded for regulatory, compliance, and consumer protection reasons. Recording of a session is typically performed by sending a copy of a media stream to a recording device. This document lists call flows that has snapshot of metadata sent from SRC to SRS. This is purely an informational document that is written to support the model defined in the metadata draft.

Table of Contents

1. Overview

[I-D.ietf-siprec-metadata] document focuses on the Recording metadata which describes the communication session. The document lists a few examples and shows the snapshots of metadata sent from SRC to SRS. For the sake of simplicity the entire SIP [RFC3261] messages are not shown at various points, instead only a snippets of the SIP/SDP messages and the XML snapshot of metadata is shown.

2. Terminology

The terms using in this defined are defined in [I-D.ietf-siprec-metadata]. No new terms/definitions are introduced in this document.

3. Metadata XML schema Instances

This section describes the metadata model XML instances for different use cases of SIPREC. For the sake of simplicity the complete SIP/SDP snippets are NOT shown here.

3.1. Sample Call flow

The following is a sample call flow that shows the SRC establishing a recording session towards the SRS. The SRC in this example could be part of any one of the architectures described in section 3 of [RFC7245].

```
               SRC                                                    SRS
                |                                                      |
                |(1) INVITE (metadata snapshot)   F1                   |
                |----------------------------------------------------->|
                |                            200 OK                    |
                |<-----------------------------------------------------|
                |(3) ACK                                               |
                |----------------------------------------------------->|
                |(4) RTP                                               |
                |=====================================================>|
                |=====================================================>|
                |=====================================================>|
                |=====================================================>|
                |(5) UPDATE/RE-INVITE (metadata update 1)     F2       |
                |----------------------------------------------------->|
                |                      200 OK                          |
                |<-----------------------------------------------------|
                | .................................................... |
                | .................................................... |
                |                                                      |
                |=====================================================>|
                |=====================================================>|
                |(7) UPDATE/RE-INVITE (metadata update n-1) Fn-1       |
                |----------------------------------------------------->|
                |                       200 OK                         |
                |<-----------------------------------------------------|
```

For the sake of simplicity, ACKs to RE-INVITES and BYEs are not shown. The subsequent sections describes the snapshot of metadata sent from SRC to SRS for each of the above transactions (F1 ... Fn-1). There may be multiple UPDATES/RE-INVITES mid call to indicates snapshots of different CS changes. Depending on the architecture described in section 3 of [RFC7245] an SRC may be a endpoint or B2BUA or as part of MEDIACTRL or Conference Focus. The subsequent sections in this document tries to list some example metadata snapshots for three major categories.

- o SRC recording streams unmixed to SRS. This includes cases where SRC is SIP UA or B2BUA.
- o SRC recording mixed streams to SRS. This includes cases where SRC is part of SIP conference model explained in [RFC4353].
- o SRC having a persistent RS with SRS.
- o Special flows like Turrent flows.

Note that only those examples for which metadata changes are listed in each category. For some of the call flows the snapshots may be

same (like in case of endpoint or B2BUA acting as SRC) and the same is mentioned in the text preceding the example.

3.2. Call Scenarios with SRC recording streams with out mixing

The section covers the models mentioned in the architecture document in section 3 of [RFC7245] where an SRC may be a SIP-UA or B2BUA. The SRS here could be a SIP-UA or an entity part of MEDIACTRL architecture described in [RFC6230].

3.2.1. Example 1: Basic Call

Basic call between two Participants Alice and Bob who are part of one CS. In this use case each participant sends two Media Streams. Media Streams sent by each participant are received all other participants in this use-case. In this example the SRC is a B2BUA in the path between Alice and Bob as described in section 3.1.1 of [RFC7245]. Below is the initial snapshot sent by SRC in the INVITE to SRS that has complete metadata. For the sake of simplicity only snippets of SIP/SDP are shown. The SRCs records the streams of each participant to SRS with out mixing in this example.

```
F1  INVITE SRC --------------> SRS

INVITE sip:recorder@example.com SIP/2.0
Via: SIP/2.0/TCP src.example.com;branch=z9hG4bKdf6b622b648d9
From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
To: <sip:recorder@example.com>
Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
Session-ID: ab30317f1a784dc48ff824d0d3715d86
 ;remote=00000000000000000000000000000000
CSeq: 101 INVITE
Max-Forwards: 70
Require: siprec
Accept: application/sdp, application/rs-metadata,
application/rs-metadata-request
Contact: <sip:2000@src.example.com>;+sip.src
Content-Type: multipart/mixed;boundary=foobar
Content-Length: [length]

--foobar
Content-Type: application/SDP
...
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

```
   a=label:96
   a=sendonly
   ...
   m=video 49174 RTP/AVPF 96
   a=rtpmap:96 H.264/90000
   a=label:97
   a=sendonly
   ...
   m=audio 51372 RTP/AVP 0
   a=rtpmap:0 PCMU/8000
   a=label:98
   a=sendonly
   ...
   m=video 49176 RTP/AVPF 96
   a=rtpmap:96 H.264/90000
   a=label:99
   a=sendonly
   ....
--foobar
Content-Type: application/rs-metadata
Content-Disposition: recording-session
```

```
<?xml version="1.0" encoding="UTF-8"?>
<recording xmlns='urn:ietf:params:xml:ns:recording:1'>
  <datamode>complete</datamode>
        <group group_id="7+OTCyoxTmqmqyA/1weDAg==">
                <associate-time>2010-12-16T23:41:07Z</associate-time>
                <!-- Standardized extension -->
                <call-center xmlns='urn:ietf:params:xml:ns:callcenter'>
                        <supervisor>sip:alice@atlanta.com</supervisor>
                </call-center>
                <mydata xmlns='http://example.com/my'>
                        <structure>FOO!</structure>
                        <whatever>bar</whatever>
                </mydata>
        </group>
        <session session_id="hVpd7YQgRW2nD22h7q60JQ==">
                <sipSessionID>ab30317f1a784dc48ff824d0d3715d86;
                  remote=47755a9de7794ba387653f2099600ef2</sipSessionID>
                <group-ref>7+OTCyoxTmqmqyA/1weDAg==
                </group-ref>
                <!-- Standardized extension -->
                <mydata xmlns='http://example.com/my'>
                        <structure>FOO!</structure>
                        <whatever>bar</whatever>
                </mydata>
        </session>
        <participant
              participant_id="srfBElmCRp2QB23b7Mpk0w==">
```

```
                <nameID aor="sip:alice@atlanta.com">
                        <name xml:lang="it">Alice</name>
                </nameID>
                <!-- Standardized extension -->
                <mydata xmlns='http://example.com/my'>
                        <structure>FOO!</structure>
                        <whatever>bar</whatever>
                </mydata>
        </participant>
        <participant
               participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
                <nameID aor="sip:bob@biloxy.com">
                        <name xml:lang="it">Bob</name>
                </nameID>
                <!-- Standardized extension -->
                <mydata xmlns='http://example.com/my'>
                        <structure>FOO!</structure>
                        <whatever>bar</whatever>
                </mydata>
        </participant>
        <stream stream_id="UAAMm5GRQKSCMVvLyl4rFw=="
               session_id="hVpd7YQgRW2nD22h7q60JQ==">
                <label>96</label>
        </stream>
        <stream stream_id="i1Pz3to5hGk8fuXl+PbwCw=="
               session_id="hVpd7YQgRW2nD22h7q60JQ==">
                <label>97</label>
        </stream>
        <stream stream_id="8zc6e0lYTlWIINA6GR+3ag=="
               session_id="hVpd7YQgRW2nD22h7q60JQ==">
                <label>98</label>
        </stream>
        <stream stream_id="EiXGlc+4TruqqoDaNE76ag=="
               session_id="hVpd7YQgRW2nD22h7q60JQ==">
                <label>99</label>
        </stream>
        <sessionrecordingassoc session_id="hVpd7YQgRW2nD22h7q60JQ==">
                <associate-time>2010-12-16T23:41:07Z</associate-time>
        </sessionrecordingassoc>
        <participantsessionassoc
              participant_id="srfBElmCRp2QB23b7Mpk0w=="
              session_id="hVpd7YQgRW2nD22h7q60JQ==">
                <associate-time>2010-12-16T23:41:07Z</associate-time>
        </participantsessionassoc>
        <participantsessionassoc
               participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
               session_id="hVpd7YQgRW2nD22h7q60JQ==">
                <associate-time>2010-12-16T23:41:07Z</associate-time>
```

```
        </participantsessionassoc>
        <participantstreamassoc
              participant_id="srfBElmCRp2QB23b7Mpk0w==">
                <send>i1Pz3to5hGk8fuXl+PbwCw==</send>
                <send>UAAMm5GRQKSCMVvLyl4rFw==</send>
                <recv>8zc6e0lYTlWIINA6GR+3ag==</recv>
                <recv>EiXGlc+4TruqqoDaNE76ag==</recv>
        </participantstreamassoc>
        <participantstreamassoc
               participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
                <send>8zc6e0lYTlWIINA6GR+3ag==</send>
                <send>EiXGlc+4TruqqoDaNE76ag==</send>
                <recv>UAAMm5GRQKSCMVvLyl4rFw==</recv>
                <recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>
        </participantstreamassoc>
</recording>
```


3.2.2. Example 2: Hold/resume

Assume a call between two Participants Alice and Bob is established and a RS is created for recording as in example1. This is the continuation of above use-case. One of the participants Bob puts Alice hold and then resumes as part of the same session. The send and recv XML elements of a participant is used to indicate whether a participant is sending not sending a particular media stream whose properties are represented by stream element. SRC sends a snapshot with only the changed XML elements.

During hold

```
F2 mid call  RE-INVITE SRC-------------------->SRS

INVITE sip:recorder@example.com SIP/2.0
Via: SIP/2.0/TCP src.example.com;branch=z9hG4bKdf6b622b648d9
From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
To: <sip:recorder@example.com>
Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
Session-ID: ab30317f1a784dc48ff824d0d3715d86
 ;remote=f81d4fae7dec11d0a76500a0c91e6bf6
CSeq: 101 INVITE
Max-Forwards: 70
Require: siprec
Accept: application/sdp, application/rs-metadata,
application/rs-metadata-request
```

```
      Contact: <sip:2000@src.example.com>;+sip.src
      Content-Type: multipart/mixed;boundary=foobar
      Content-Length: [length]

      --foobar
      Content-Type: application/SDP
      ...
      m=audio 49170 RTP/AVP 0
      a=rtpmap:0 PCMU/8000
      a=label:96
      a=sendonly
      ...
      m=video 49174 RTP/AVPF 96
      a=rtpmap:96 H.264/90000
      a=label:97
      a=sendonly
      ...
      m=audio 51372 RTP/AVP 0
      a=rtpmap:0 PCMU/8000
      a=label:98
      a=sendonly
      ...
      m=video 49176 RTP/AVPF 96
      a=rtpmap:96 H.264/90000
      a=label:99
      a=sendonly
      ....

      --foobar
   Content-Type: application/rs-metadata
   Content-Disposition: recording-session

      <?xml version="1.0" encoding="UTF-8"?>
        <recording xmlns='urn:ietf:params:xml:ns:recording:1'>
          <datamode>partial</datamode>
            <participantstreamassoc
             participant_id="srfBElmCRp2QB23b7Mpk0w==">
             <recv>8zc6e0lYTlWIINA6GR+3ag==</recv>
             <recv>EiXGlc+4TruqqoDaNE76ag==</recv>
            </participantstreamassoc>
            <participantstreamassoc
             participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
              <send>8zc6e0lYTlWIINA6GR+3ag==</send>
              <send>EiXGlc+4TruqqoDaNE76ag==</send>
             </participantstreamassoc>
           </recording>
```

In the above snippet, Alice with participant_id srfBElmCRp2QB23b7Mpk0w== only receives media streams and does not send any media. The same is indicated by the absence of send XML element. Bob(participant_id zSfPoSvdSDCmU3A3TRDxAw==) on the other hand would be sending media but does not receive any media from Alice and so recv XML element is absent in this instance.

During resume

The snapshot now has send and recv XML elements for both Alice and Bob indicating that both are receiving and sending media.

```
F3 mid call  RE-INVITE SRC-------------------->SRS

INVITE sip:recorder@example.com SIP/2.0
Via: SIP/2.0/TCP src.example.com;branch=z9hG4bKdf6b622b648d9
From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
To: <sip:recorder@example.com>
Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
Session-ID: ab30317f1a784dc48ff824d0d3715d86
 ;remote=f81d4fae7dec11d0a76500a0c91e6bf6
CSeq: 101 INVITE
Max-Forwards: 70
Require: siprec
Accept: application/sdp, application/rs-metadata,
application/rs-metadata-request
Contact: <sip:2000@src.example.com>;+sip.src
Content-Type: multipart/mixed;boundary=foobar
Content-Length: [length]

--foobar
Content-Type: application/SDP
...
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=label:96
a=sendonly
...
m=video 49174 RTP/AVPF 96
a=rtpmap:96 H.264/90000
a=label:97
a=sendonly
...
m=audio 51372 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

```
      a=label:98
      a=sendonly
      ...
      m=video 49176 RTP/AVPF 96
      a=rtpmap:96 H.264/90000
      a=label:99
      a=sendonly
      ....
      --foobar
   Content-Type: application/rs-metadata
   Content-Disposition: recording-session
```


```
      <?xml version="1.0" encoding="UTF-8"?>
        <recording xmlns='urn:ietf:params:xml:ns:recording:1'>
          <datamode>partial</datamode>
            <participantstreamassoc
             participant_id="srfBElmCRp2QB23b7Mpk0w==">
             <recv>8zc6e0lYTlWIINA6GR+3ag==</recv>
             <recv>EiXGlc+4TruqqoDaNE76ag==</recv>
             <send>i1Pz3to5hGk8fuXl+PbwCw==</send>
             <send>UAAMm5GRQKSCMVvLyl4rFw==</send>
            </participantstreamassoc>
            <participantstreamassoc
             participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
              <send>8zc6e0lYTlWIINA6GR+3ag==</send>
              <send>EiXGlc+4TruqqoDaNE76ag==</send>
              <recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>
             <recv>UAAMm5GRQKSCMVvLyl4rFw==</recv>
             </participantstreamassoc>
           </recording>
```


3.2.3. Example 3:Call Transfer (RE-INVITE and REFER based)

Basic call between two Participants Alice and Bob is connected and SRC(B2BUA as per section 3.1.1 of [RFC7245]) has sent a snapshot as in example 1. Transfer is initiated by one of the participants(Alice). After the transfer is completed, SRC sends a snapshot of the participant changes to SRS. In this transfer scenario, Alice drops out after transfer is completed and Bob and Carol gets connected and recording of media between Bob and Carol is done by SRC. There may be two cases here as described below.

Transfer with in the same session - (.e.g. RE-INVITE based transfer). Participant Alice drops out and Carol is added to the same session. No change to session/group element. A participantsessassoc element indicating that Alice has disassociated from the CS will be present in the snapshot. A new participant XML

element representing Carol with mapping to the same RS SDP stream used for mapping earlier Alice's stream is sent in the snapshot. A new sipSessionID XML element that has UUID tuples which corresponds to Bob and Carol is sent in the snapshot from SRC to SRS. Note that one half of the session ID that corresponds to Bob remains same.

```
mid call  RE-INVITE SRC--------------------->SRS


INVITE sip:recorder@example.com SIP/2.0
Via: SIP/2.0/TCP src.example.com;branch=z9hG4bKdf6b622b648d9
From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
To: <sip:recorder@example.com>
Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
Session-ID: ab30317f1a784dc48ff824d0d3715d86
 ;remote=f81d4fae7dec11d0a76500a0c91e6bf6
CSeq: 101 INVITE
Max-Forwards: 70
Require: siprec
Accept: application/sdp, application/rs-metadata,
application/rs-metadata-request
Contact: <sip:2000@src.example.com>;+sip.src
Content-Type: multipart/mixed;boundary=foobar
Content-Length: [length]

--foobar
Content-Type: application/SDP
...
m=audio 49180 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=label:96
a=sendonly
...
m=video 49182 RTP/AVPF 96
a=rtpmap:96 H.264/90000
a=label:97
a=sendonly
...
m=audio 51374 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=label:98
a=sendonly
...
m=video 49178 RTP/AVPF 96
a=rtpmap:96 H.264/90000
a=label:99
```

```
     a=sendonly
     ....
  --foobar
  Content-Type: application/rs-metadata
  Content-Disposition: recording-session
```


```
  <?xml version="1.0" encoding="UTF-8"?>
      <recording xmlns='urn:ietf:params:xml:ns:recording:1'>
             <datamode>partial</datamode>
              <session session_id="hVpd7YQgRW2nD22h7q60JQ==">
           <sipSessionID>3363127f0d084c10876dddd4f8e5eeb9
           ;remote=2272bb7e70fe41dba0025ae9a26d54cf</sipSessionID>
        </session>
         <participant
            participant_id="Atnm1ZRnOC6Pm5MApkrDzQ==">
            <nameIDaor="sip:carol@example.com">
             <name xml:lang="it">Carol</name>
           </nameID>
         </participant>
         <participantsessionassoc
            participant_id="Atnm1ZRnOC6Pm5MApkrDzQ=="
            session_id="hVpd7YQgRW2nD22h7q60JQ==">
            <associate-time>2013-12-16T23:41:07Z</associate-time>
         </participantsession>
         <participantsessionassoc
            participant_id="srfBElmCRp2QB23b7Mpk0w=="
            session_id="hVpd7YQgRW2nD22h7q60JQ==">
             <disassociate-time>2013-12-16T23:41:07Z</disassociate-time>
          </participantsessionassoc>
          <participantstreamassoc
             participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
            <send>8zc6e0lYTlWIINA6GR+3ag==</send>
            <send>EiXGlc+4TruqqoDaNE76ag==</send>
            <recv>60JAJm9UTvik0Ltlih/Gzw==</recv>
            <recv>AcR5FUd3Edi8cACQJy/3JQ==</recv>
         </participantstreamassoc>
         <participantstreamassoc
            participant_id="Atnm1ZRnOC6Pm5MApkrDzQ==">
           <send>60JAJm9UTvik0Ltlih/Gzw==</send>
           <send>AcR5FUd3Edi8cACQJy/3JQ==</send>
           <recv>8zc6e0lYTlWIINA6GR+3ag==</recv>
           <recv>EiXGlc+4TruqqoDaNE76ag==</recv>
           <associate-time>2013-12-16T23:42:07Z</associate-time>
         </participantstreamassoc>
    </recording>
```


Transfer with new session - (.e.g. REFER based transfer). In this

case a new session (CS) is created and shall be part of same CS-group (done by SRC).

SRC first sends an optional snapshot indicating disassociation of participant from the old CS. Please note this is a optional message. An SRC may choose to just send a INVITE with a new session element to implicitly indicate that the participants are now part of a different CS with out sending disassociation from the old CS. Also note that the SRC in this example uses the same RS. In case it decides to use a new RS, it will tear down the current RS using normal SIP procedures (BYE) with metadata as in example 4.

```
mid call  RE-INVITE SRC-------------------->SRS


INVITE sip:recorder@example.com SIP/2.0
Via: SIP/2.0/TCP src.example.com;branch=z9hG4bKdf6b622b648d9
From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
To: <sip:recorder@example.com>
Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
Session-ID: ab30317f1a784dc48ff824d0d3715d86
 ;remote=f81d4fae7dec11d0a76500a0c91e6bf6
CSeq: 101 INVITE
Max-Forwards: 70
Require: siprec
Accept: application/sdp, application/rs-metadata,
application/rs-metadata-request
Contact: <sip:2000@src.example.com>;+sip.src
Content-Type: multipart/mixed;boundary=foobar
Content-Length: [length]

--foobar
Content-Type: application/SDP
...
m=audio 49180 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=label:96
a=sendonly
...
m=video 49182 RTP/AVPF 96
a=rtpmap:96 H.264/90000
a=label:97
a=sendonly
...
m=audio 51374 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

```
     a=label:98
     a=sendonly
     ...
     m=video 49178 RTP/AVPF 96
     a=rtpmap:96 H.264/90000
     a=label:99
     a=sendonly
     ....

   --foobar
  Content-Type: application/rs-metadata
  Content-Disposition: recording-session
```


```
        <?xml version="1.0" encoding="UTF-8"?>
    <recording xmlns='urn:ietf:params:xml:ns:recording:1'>
           <datamode>Partial</datamode>
       <session session_id="hVpd7YQgRW2nD22h7q60JQ==">
         <stop-time>2010-12-16T23:41:07Z</stop-time>
       </session>
       <participantsessionassoc
          participant_id="srfBElmCRp2QB23b7Mpk0w=="
          session_id="hVpd7YQgRW2nD22h7q60JQ==">
           <disassociate-time>2010-12-16T23:41:07Z</disassociate-time>
        </participantsessionassoc>
       <participantsessionassoc
          participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
          session_id="hVpd7YQgRW2nD22h7q60JQ==">
          <disasociate-time>2010-12-16T23:41:07Z</disassociate-time>
        </participantsessionassoc>
  </recording>
```


Note in the above snapshot the participantsessionassoc element is optional as indicating session XML element with a stop-time implicitly means that all the participants associated with that session have been disassociated.

SRC sends another snapshot to indicate the participant change (due to REFER) and new session information after transfer. In this example it is assumed SRC uses the same Recording Session to continue recording the call. Also note that the sipSessionID XML element in metadata snapshot now indicates Alice and Carol in the (local, remote) uuid pair.

```
    mid call  RE-INVITE SRC-------------------->SRS
```

```
      INVITE sip:recorder@example.com SIP/2.0
      Via: SIP/2.0/TCP src.example.com;branch=z9hG4bKdf6b622b648d9
      From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
      To: <sip:recorder@example.com>
      Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
      Session-ID: ab30317f1a784dc48ff824d0d3715d86
       ;remote=f81d4fae7dec11d0a76500a0c91e6bf6
      CSeq: 101 INVITE
      Max-Forwards: 70
      Require: siprec
      Accept: application/sdp, application/rs-metadata,
      application/rs-metadata-request
      Contact: <sip:2000@src.example.com>;+sip.src
      Content-Type: multipart/mixed;boundary=foobar
      Content-Length: [length]

      --foobar
      Content-Type: application/SDP
      ...
      m=audio 49180 RTP/AVP 0
      a=rtpmap:0 PCMU/8000
      a=label:96
      a=sendonly
      ...
      m=video 49182 RTP/AVPF 96
      a=rtpmap:96 H.264/90000
      a=label:97
      a=sendonly
      ...
      m=audio 51374 RTP/AVP 0
      a=rtpmap:0 PCMU/8000
      a=label:98
      a=sendonly
      ...
      m=video 49178 RTP/AVPF 96
      a=rtpmap:96 H.264/90000
      a=label:99
      a=sendonly
      ....

    --foobar
   Content-Type: application/rs-metadata

   <?xml version="1.0" encoding="UTF-8"?>
       <recording xmlns='urn:ietf:params:xml:ns:recording:1'>
              <datamode>complete</datamode>
          <session session_id="bfLZ+NTFEeCNxQTuRyQBmw==">
            <start-time>2010-12-16T23:41:07Z</start-time>
```

```
    <sipSessionID>3363127f0d084c10876dddd4f8e5eeb9
    ;remote=2272bb7e70fe41dba0025ae9a26d54cf</sipSessionID>
  </session>
  <participant
      participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
      <nameID aor="sip:Bob@biloxy.com"/>
   </participant>
   <participant
     participant_id="Atnm1ZRnOC6Pm5MApkrDzQ==">
     <nameID aor="sip:carol@example.com"/>
  </participant>
   <stream stream_id="60JAJm9UTvik0Ltlih/Gzw=="
      session_id="bfLZ+NTFEeCNxQTuRyQBmw==">
      <label>96</label>
  </stream>
  <stream stream_id="AcR5FUd3Edi8cACQJy/3JQ=="
      session_id="bfLZ+NTFEeCNxQTuRyQBmw==">
      <label>97</label>
  </stream>
  <stream stream_id="8zc6e0lYTlWIINA6GR+3ag=="
      session_id="bfLZ+NTFEeCNxQTuRyQBmw==">
      <label>98</label>
  </stream>
  <stream stream_id="EiXGlc+4TruqqoDaNE76ag=="
      session_id="bfLZ+NTFEeCNxQTuRyQBmw==">
      <label>99</label>
  </stream>
  <participantsessionassoc
      participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
      session_id="bfLZ+NTFEeCNxQTuRyQBmw==">
      <associate-time>2010-12-16T23:32:03Z</associate-time>
   </participantsessionassoc>
  <participantsessionassoc
     participant_id="Atnm1ZRnOC6Pm5MApkrDzQ=="
     session_id="bfLZ+NTFEeCNxQTuRyQBmw==">
   <associate-time>2010-12-16T23:41:07Z</associate-time>
  </participantsessionassoc>
  <participantstreamassoc
      participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
    <send>8zc6e0lYTlWIINA6GR+3ag==</send>
    <send>EiXGlc+4TruqqoDaNE76ag==</send>
    <recv>60JAJm9UTvik0Ltlih/Gzw==</recv>
    <recv>AcR5FUd3Edi8cACQJy/3JQ==</recv>
  </participantstreamassoc>
  <participantstreamassoc
     participant_id="Atnm1ZRnOC6Pm5MApkrDzQ==">
   <send>60JAJm9UTvik0Ltlih/Gzw==</send>
   <send>AcR5FUd3Edi8cACQJy/3JQ==</send>
```

```
          &lt;recv&gt;8zc6e0lYTlWIINA6GR+3ag==&lt;/recv&gt;
          &lt;recv&gt;EiXGlc+4TruqqoDaNE76ag==&lt;/recv&gt;
          &lt;/participantstreamassoc&gt;
    &lt;/recording&gt;
```

3.2.4. Example 4: Call disconnect

This example shows a snapshot of metadata sent by an SRC at CS disconnect where the participants of CS are Alice and Bob

```
         SRC                                                 SRS
              |                                                      |
              |(1) BYE (metadata snapshot)   F1                      |
              |----------------------------------------------------->|
              |                            200 OK    F2              |
              |<-----------------------------------------------------|
```

```
F1  BYE SRC  -----------> SRS

BYE sip:2001@example.com SIP/2.0
Via: SIP/2.0/UDP src.example.com;branch=z9hG4bK47c8eb30
From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
To: <sip:recorder@example.com>
Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
Session-ID: ab30317f1a784dc48ff824d0d3715d86
    ;remote=f81d4fae7dec11d0a76500a0c91e6bf6
CSeq: 102 BYE
Max-Forwards: 70
Require: siprec
Accept: application/rs-metadata,
application/rs-metadata-request
Contact: <sip:2000@src.example.com>;+sip.src
Content-Type: multipart/mixed;boundary=foobar
Content-Length: [length]

--foobar
Content-Type: application/rs-metadata
```


```
<?xml version="1.0" encoding="UTF-8"?>
    <recording xmlns='urn:ietf:params:xml:ns:recording:1'>
           <datamode>Partial</datamode>
       <session session_id="hVpd7YQgRW2nD22h7q60JQ==">
          <stop-time>2010-12-16T23:41:07Z</stop-time>
       </session>
       <participantsessionassoc
          participant_id="srfBElmCRp2QB23b7Mpk0w=="
          session_id="hVpd7YQgRW2nD22h7q60JQ==">
         <disassociate-time>2010-12-16T23:41:07Z</disassociate-time>
        </participantsessionassoc>

       <participantsessionassoc
          participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
         session_id="hVpd7YQgRW2nD22h7q60JQ==">
          <disasociate-time>2010-12-16T23:41:07Z</disassociate-time>
        </participantsessionassoc>
   </recording>
```

3.3. Call Scenarios with SRC recording streams by mixing

The section covers the models mentioned in the architecture document in section 3 of [RFC7245] where an SRC may be part of Conference model either as Focus or a participant in Conference. The SRS here could be a SIP UA or an entity part of MEDIACTRL architecture. Note that the disconnect case is not shown since the metadata snapshot will be same as for a non-mixing case.

3.3.1. Example 1: Basic call with SRC mixing streams

Basic call between two Participants Alice and Bob who are part of one CS. In this use case each participant call into a conference server (say, an MCU) to attend one of many conferences hosted on or managed by that servers. Media Streams sent by each participant is received by the other participant. Below is the initial snapshot sent by SRC in the INVITE to SRS that has complete metadata. For the sake of simplicity only snippets of SIP/SDP are shown. The SRCs records the streams of each participant to SRS by mixing in this example. The SRC here is part of conference model described in section 3 of [RFC7245] as a Focus and does mixing. The SRC here is not a participant by itself and hence it does not contribute to streams.

```
                          F1  INVITE SRC --------------> SRS

   INVITE sip:recorder@example.com SIP/2.0
   Via: SIP/2.0/TCP src.example.com;branch=z9hG4bKdf6b622b648d9
   From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
   To: <sip:recorder@example.com>
   Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
   Session-ID: a358d2b81a444a8c8fb05950cef331e7
    ;remote=00000000000000000000000000000000
   CSeq: 101 INVITE
   Max-Forwards: 70
   Require: siprec
   Accept: application/sdp, application/rs-metadata,
   application/rs-metadata-request
   Contact: <sip:2000@src.example.com>;+sip.src
   Content-Type: multipart/mixed;boundary=foobar
   Content-Length: [length]

   --foobar
   Content-Type: application/SDP
   ...
   m=audio 49170 RTP/AVP 0
   a=rtpmap:0 PCMU/8000
```

```
      a=label:96
      a=sendonly

      ....
   --foobar
   Content-Type: application/rs-metadata
   Content-Disposition: recording-session
   <?xml version="1.0" encoding="UTF-8"?>
       <recording xmlns='urn:ietf:params:xml:ns:recording:1'>
              <datamode>complete</datamode>
         <session session_id="hVpd7YQgRW2nD22h7q60JQ==">
            <start-time>2010-12-16T23:41:07Z</start-time>
            <sipSessionID>fa3b60f27e91441e84c55a9a0095f075
             ;remote=a358d2b81a444a8c8fb05950cef331e7</sipSessionID>
             <sipSessionID>ca718e1430474b5485a53aa5d0bea45e
             ;remote=68caf509b9284b7ea45f84a049febf0a</sipSessionID>
         </session>
          <participant
             participant_id="srfBElmCRp2QB23b7Mpk0w==">
            <nameID aor="sip:alice@atlanta.com">
             <name xml:lang="it">Alice</name>
            </nameID>
          </participant>
          <participant
              participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
              <nameID aor="sip:bob@biloxy.com">
               <name xml:lang="it">Bob</name>
            </nameID>
          </participant>
          <stream stream_id="i1Pz3to5hGk8fuXl+PbwCw=="
              session_id="hVpd7YQgRW2nD22h7q60JQ==">
              <label>96</label>
          </stream>
          <participantsessionassoc
             participant_id="srfBElmCRp2QB23b7Mpk0w=="
             session_id="hVpd7YQgRW2nD22h7q60JQ==">
           <associate-time>2010-12-16T23:41:07Z</associate-time>
          </participantsessionassoc>
          <participantstreamassoc
             participant_id="srfBElmCRp2QB23b7Mpk0w==">
             <send>i1Pz3to5hGk8fuXl+PbwCw==</send>
             <recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>
          </participantstreamassoc>

          <participantsessionassoc
              participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
                   session_id="hVpd7YQgRW2nD22h7q60JQ==">
             <associate-time>2010-12-16T23:41:07Z</associate-time>
```

```
          </participantsessionassoc>
          <participantstreamassoc
              participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
            <send>i1Pz3to5hGk8fuXl+PbwCw==</send>
            <recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>
          </participantstreamassoc>

     </recording>
```


In the above example there are two participants Alice and Bob in the conference. Among other things, SRC sends Session-ID in the metadata snapshot. There are two Session-ID's here, one that corresponds to the SIP session between Alice and Conference Focus and the other for the SIP session between Bob and Conference Focus. In this use-case, since Alice and Bob calls into the conference these Session-ID's are different.

3.3.2. Example 2: Hold/resume with SRC recording by mixing streams

Assume a call between two Participants Alice and Bob is established and a RS is created for recording as in example 5. This is the continuation of above use-case. One of the participants Bob puts Alice hold and then resumes as part of the same session. The send and recv XML elements of a participant is used to indicate whether a participant is sending not sending a particular media stream whose properties are represented by stream element. The metadata snapshot looks as below:

During hold

```
      mid call hold   RE-INVITE SRC --------------> SRS

      INVITE sip:recorder@example.com SIP/2.0
      Via: SIP/2.0/TCP src.example.com;branch=z9hG4bKdf6b622b648d9
      From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
      To: <sip:recorder@example.com>
      Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
      Session-ID: a358d2b81a444a8c8fb05950cef331e7
       ;remote=f81d4fae7dec11d0a76500a0c91e6bf6
      CSeq: 101 INVITE
      Max-Forwards: 70
      Require: siprec
      Accept: application/sdp, application/rs-metadata,
      application/rs-metadata-request
      Contact: <sip:2000@src.example.com>;+sip.src
      Content-Type: multipart/mixed;boundary=foobar
      Content-Length: [length]

      --foobar
      Content-Type: application/SDP
      ...
      m=audio 49170 RTP/AVP 0
      a=rtpmap:0 PCMU/8000
      a=label:96
      a=sendonly

      ....
   --foobar
   Content-Type: application/rs-metadata
   Content-Disposition: recording-session
   <?xml version="1.0" encoding="UTF-8"?>
       <recording xmlns='urn:ietf:params:xml:ns:recording:1'>
              <datamode>partial</datamode>
              <stream stream_id="i1Pz3to5hGk8fuXl+PbwCw=="
              session_id="hVpd7YQgRW2nD22h7q60JQ==">
              <label>96</label>
          </stream>
          <participantstreamassoc
             participant_id="srfBElmCRp2QB23b7Mpk0w==">
              <recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>
          </participantstreamassoc>
          <participantstreamassoc
              participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
            <send>i1Pz3to5hGk8fuXl+PbwCw==</send>
          </participantstreamassoc>
     </recording>
```

During resume a snapshot shown below will be sent from SRC to SRS

```
          mid call resume   RE-INVITE SRC --------------> SRS

      INVITE sip:recorder@example.com SIP/2.0
      Via: SIP/2.0/TCP src.example.com;branch=z9hG4bKdf6b622b648d9
      From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
      To: <sip:recorder@example.com>
      Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
      Session-ID: a358d2b81a444a8c8fb05950cef331e7
       ;remote=f81d4fae7dec11d0a76500a0c91e6bf6
      CSeq: 101 INVITE
      Max-Forwards: 70
      Require: siprec
      Accept: application/sdp, application/rs-metadata,
      application/rs-metadata-request
      Contact: <sip:2000@src.example.com>;+sip.src
      Content-Type: multipart/mixed;boundary=foobar
      Content-Length: [length]

      --foobar
      Content-Type: application/SDP
      ...
      m=audio 49170 RTP/AVP 0
      a=rtpmap:0 PCMU/8000
      a=label:96
      a=sendonly

      ....
   --foobar
   Content-Type: application/rs-metadata
   Content-Disposition: recording-session
   <?xml version="1.0" encoding="UTF-8"?>
       <recording xmlns='urn:ietf:params:xml:ns:recording:1'>
              <datamode>partial</datamode>
              <stream stream_id="i1Pz3to5hGk8fuXl+PbwCw=="
              session_id="hVpd7YQgRW2nD22h7q60JQ==">
              <label>96</label>
          </stream>
         <participantstreamassoc
             participant_id="srfBElmCRp2QB23b7Mpk0w==">
             <send>i1Pz3to5hGk8fuXl+PbwCw==</send>
             <recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>
          </participantstreamassoc>
          <participantstreamassoc
              participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
            <send>i1Pz3to5hGk8fuXl+PbwCw==</send>
           <recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>
          </participantstreamassoc>
     </recording>
```

3.3.3. Example 3: Metadata snapshot of joining/dropping of a participant to a session

In a conference model, participants can join and drop a session any time during the session. The below shows a snapshot sent from SRC to SRC in these case. Note the SRC here can be a focus or a participant in the conference. In case where the SRC is a participant it may learn the information required for metadata by subscribing to conference event package [RFC4575]. Assume Alice and Bob were in the conference and a third participant Carol joins, then SRC sends the below snapshot with the indication of new participant.

```
        mid call resume   RE-INVITE SRC --------------> SRS

   INVITE sip:recorder@example.com SIP/2.0
   Via: SIP/2.0/TCP src.example.com;branch=z9hG4bKdf6b622b648d9
   From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
   To: <sip:recorder@example.com>
   Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
   Session-ID: a358d2b81a444a8c8fb05950cef331e7
    ;remote=f81d4fae7dec11d0a76500a0c91e6bf6
   CSeq: 101 INVITE
   Max-Forwards: 70
   Require: siprec
   Accept: application/sdp, application/rs-metadata,
   application/rs-metadata-request
   Contact: <sip:2000@src.example.com>;+sip.src
   Content-Type: multipart/mixed;boundary=foobar
   Content-Length: [length]

   --foobar
   Content-Type: application/SDP
   ...
   m=audio 49170 RTP/AVP 0
   a=rtpmap:0 PCMU/8000
   a=label:96
   a=sendonly

   ....
--foobar
Content-Type: application/rs-metadata
Content-Disposition: recording-session
<?xml version="1.0" encoding="UTF-8"?>
    <recording xmlns='urn:ietf:params:xml:ns:recording:1'>
           <datamode>partial</datamode>
           <session session_id="hVpd7YQgRW2nD22h7q60JQ==">
```

```
           <sipSessionID>fa3b60f27e91441e84c55a9a0095f075
            ;remote=a358d2b81a444a8c8fb05950cef331e7</sipSessionID>
            <sipSessionID>ca718e1430474b5485a53aa5d0bea45e
            ;remote=68caf509b9284b7ea45f84a049febf0a</sipSessionID>
            <sipSessionID>497c0f13929643b4a16858e2a3885edc
            ;remote=0e8a82bedda74f57be4a4a4da54167c4</sipSessionID>
        </session>
             <participant
            participant_id="Atnm1ZRnOC6Pm5MApkrDzQ==">
            <nameID aor="sip:carol@example.com">
             <name xml:lang="it">Carol</name>
           </nameID>
         </participant>
         <participantsessionassoc
            participant_id="Atnm1ZRnOC6Pm5MApkrDzQ=="
            session_id="hVpd7YQgRW2nD22h7q60JQ==">
            <associate-time>2013-12-16T23:41:07Z</associate-time>
         </participantsession>
         <participantstreamassoc
            participant_id="Atnm1ZRnOC6Pm5MApkrDzQ==">
            <send>i1Pz3to5hGk8fuXl+PbwCw==</send>
            <recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>
         </participantstreamassoc>
    </recording>
```


Assume Alice drops after some time from the conference. SRC generates a new snapshot showing Alice disassociating from the session

```
           mid call resume   RE-INVITE SRC --------------> SRS

      INVITE sip:recorder@example.com SIP/2.0
      Via: SIP/2.0/TCP src.example.com;branch=z9hG4bKdf6b622b648d9
      From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
      To: <sip:recorder@example.com>
      Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
       Session-ID: a358d2b81a444a8c8fb05950cef331e7
       ;remote=f81d4fae7dec11d0a76500a0c91e6bf6
      CSeq: 101 INVITE
      Max-Forwards: 70
      Require: siprec
      Accept: application/sdp, application/rs-metadata,
      application/rs-metadata-request
      Contact: <sip:2000@src.example.com>;+sip.src
      Content-Type: multipart/mixed;boundary=foobar
      Content-Length: [length]

      --foobar
      Content-Type: application/SDP
      ...
      m=audio 49170 RTP/AVP 0
      a=rtpmap:0 PCMU/8000
      a=label:96
      a=sendonly

      ....
   --foobar
   Content-Type: application/rs-metadata
   Content-Disposition: recording-session
   <?xml version="1.0" encoding="UTF-8"?>
       <recording xmlns='urn:ietf:params:xml:ns:recording:1'>
              <datamode>partial</datamode>
              <session session_id="hVpd7YQgRW2nD22h7q60JQ==">
             <sipSessionID>ca718e1430474b5485a53aa5d0bea45e
             ;remote=68caf509b9284b7ea45f84a049febf0a</sipSessionID>
             <sipSessionID>497c0f13929643b4a16858e2a3885edc
             ;remote=0e8a82bedda74f57be4a4a4da54167c4</sipSessionID>
         </session>
              <participantsessionassoc
             participant_id="srfBElmCRp2QB23b7Mpk0w=="
             session_id="hVpd7YQgRW2nD22h7q60JQ==">
           <associate-time>2010-12-16T23:41:07Z</associate-time>
          </participantsessionassoc>
      </recording>
```

3.3.4. Example 4: Call disconnect

When a CS is disconnected, SRC sends BYE with a snapshot of metadata having session stop-time and participant dis-associate times. The snapshot looks same as listed in section 3.2.4

3.4. Call scenarios with persistent RS between SRC and SRS

The section shows the snapshots of metadata for the cases there a persistent RS exists between SRC and SRS. An SRC here may be SIP UA or a B2BUA or may be part of Conference model either as Focus or a participant in Conference. The SRS here could be a SIP UA or an entity part of MEDIACTRL architecture. Except disconnect case, the snapshot remains same as one of the examples mentioned in previous sections.

3.4.1. Example 1: Metadata snapshot during CS disconnect with persistent RS between SRC and SRS

```
RE-INVITE sent from SRC  -----------> SRS

INVITE sip:2001@example.com SIP/2.0
Via: SIP/2.0/UDP src.example.com;branch=z9hG4bK47c8eb30
From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
To: <sip:recorder@example.com>
Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
 Session-ID: ab30317f1a784dc48ff824d0d3715d86
    ;remote=f81d4fae7dec11d0a76500a0c91e6bf6
CSeq: 101 INVITE
Max-Forwards: 70
Require: siprec
Accept: application/rs-metadata,
application/rs-metadata-request
Contact: <sip:2000@src.example.com>;+sip.src
Content-Type: multipart/mixed;boundary=foobar
Content-Length: [length]

--foobar
Content-Type: application/rs-metadata

<?xml version="1.0" encoding="UTF-8"?>
    <recording xmlns='urn:ietf:params:xml:ns:recording:1'>
           <datamode>Partial</datamode>
       <session session_id="hVpd7YQgRW2nD22h7q60JQ==">
          <stop-time>2010-12-16T23:41:07Z</stop-time>
       </session>
       <participantsessionassoc
          participant_id="srfBElmCRp2QB23b7Mpk0w=="
          session_id="hVpd7YQgRW2nD22h7q60JQ==">
         <disassociate-time>2010-12-16T23:41:07Z</disassociate-time>
        </participantsessionassoc>

       <participantsessionassoc
          participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
         session_id="hVpd7YQgRW2nD22h7q60JQ==">
          <disasociate-time>2010-12-16T23:41:07Z</disassociate-time>
        </participantsessionassoc>
   </recording>
```

3.5. Turrent-Case: Multiple CS into single RS with mixed stream

In trading floor environments, in order to minimize storage and recording system resources, it may be preferable to mix multiple concurrent calls (Communication Sessions) on different handsets/ speakers on the same turret into a single recording session. This

would means media in each CS is mixed and recorded as part of single media stream and multiple such CSs are recording in one Recording Session from a SRC to SRS.

Lets take a example where there are two CS[CS1 and CS2]. Assume mixing is done in each of these CS and both these CS are recorded as part of single RS from a single SRC which is part of both the CS. There are three possibilities here:

- o CS1 and CS2 uses the same Focus for fixing and that Focus is also acting as SRC in each of the CS.
- o In of the CS(say CS1), SRC is Focus and in the other CS(say CS2), SRC is just one of the participant of the conference.
- o In both CS1 and CS2, SRC is just a participant of Conference.

The following example shows the first possibility where CS1 and CS2 uses the same Focus for fixing and that Focus is also acting as SRC in each of the CS.

```
   snapshot of metadata  INVITE SRC --------------> SRS

   INVITE sip:recorder@example.com SIP/2.0
   Via: SIP/2.0/TCP src.example.com;branch=z9hG4bKdf6b622b648d9
   From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
   To: <sip:recorder@example.com>
   Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
   Session-ID: a358d2b81a444a8c8fb05950cef331e7
    ;remote=00000000000000000000000000000000
   Content-Type: application/SDP
   ...
   m=audio 49170 RTP/AVP 0
   a=rtpmap:0 PCMU/8000
   a=label:96
   a=sendonly
   ...
<?xml version="1.0" encoding="UTF-8"?>
    <recording xmlns='urn:ietf:params:xml:ns:recording:1'>
           <datamode>complete</datamode>
       <group group_id="7+OTCyoxTmqmqyA/1weDAg==">
               <associate-time>2010-12-16T23:41:07Z</associate-time>
      </group>
      <session session_id="hVpd7YQgRW2nD22h7q60JQ==">
         <group-ref>7+OTCyoxTmqmqyA/1weDAg==</group-ref>
        <start-time>2010-12-16T23:41:07Z</start-time>
         <sipSessionID>fa3b60f27e91441e84c55a9a0095f075
```

```
             ;remote=a358d2b81a444a8c8fb05950cef331e7</sipSessionID>
             <sipSessionID>ca718e1430474b5485a53aa5d0bea45e
             ;remote=a358d2b81a444a8c8fb05950cef331e7</sipSessionID>
             <sipSessionID>497c0f13929643b4a16858e2a3885edc
             ;remote=a358d2b81a444a8c8fb05950cef331e7</sipSessionID>
          </session>
         <session session_id="zzlafnvvjlCHllaHF6mn8kkSS==">
             <group-ref>7+OTCyoxTmqmqyA/1weDAg==</group-ref>
            <start-time>2010-12-16T23:43:07Z</start-time>
            <sipSessionID>ae10731ca50343a5aaae2dd0904a65de
            ;remote=a358d2b81a444a8c8fb05950cef331e7</sipSessionID>
            <sipSession>33c77aac7deb414cbc8c10f363fccb71
            ;remote=a358d2b81a444a8c8fb05950cef331e7</sipSessionID>
            <sipSessionID>fd6932e9e5fc489fae2d5b3779723b7e
           ;remote=a358d2b81a444a8c8fb05950cef331e7</sipSessionID>
          </session>
          <participant
             participant_id="srfBElmCRp2QB23b7Mpk0w==">
            <nameID aor="sip:alice@atlanta.com">
             <name xml:lang="it">Alice</name>
            </nameID>
          </participant>
          <participant
              participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
              <nameID aor="sip:Bob@biloxy.com">
               <name xml:lang="it">Bob</name>
            </nameID>
          </participant>
          <participant
             participant_id="EiXGlc+4TruqqoDaNE76ag==">
            <nameID aor="sip:Carol@example.com">
             <name xml:lang="it">Carol</name>
            </nameID>
          </participant>
          <stream stream_id="UAAMm5GRQKSCMVvLyl4rFw=="
              session_id="hVpd7YQgRW2nD22h7q60JQ==">
              <label>96</label>
          </stream>
          <participantsessionassoc
             participant_id="srfBElmCRp2QB23b7Mpk0w=="
             session_id="hVpd7YQgRW2nD22h7q60JQ==">
           <associate-time>2010-12-16T23:41:07Z</associate-time>
          </participantsessionassoc>
          <participantsessionassoc
              participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
              session_id="hVpd7YQgRW2nD22h7q60JQ==">
             <associate-time>2010-12-16T23:41:07Z</associate-time>
          </participantsessionassoc>
```

```
          <participantsessionassoc
              participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
              session_id="zzlafnvvjlCHllaHF6mn8kkSS==">
             <associate-time>2010-12-16T23:43:07Z</associate-time>
          </participantsessionassoc>
          <participantsessionassoc
              participant_id="EiXGlc+4TruqqoDaNE76ag=="
              session_id="zzlafnvvjlCHllaHF6mn8kkSS==">
             <associate-time>2010-12-16T23:43:07Z</associate-time>
          </participantsessionassoc>

          <participantstreamassoc
             participant_id="srfBElmCRp2QB23b7Mpk0w==">
             <send>UAAMm5GRQKSCMVvLyl4rFw==</send>
             <recv>UAAMm5GRQKSCMVvLyl4rFw==</recv>
          </participantstreamassoc>
          <participantstreamassoc
              participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
            <send>UAAMm5GRQKSCMVvLyl4rFw==</send>
            <recv>UAAMm5GRQKSCMVvLyl4rFw==</recv>
          </participantstreamassoc>
          <participantstreamassoc
              participant_id="EiXGlc+4TruqqoDaNE76ag==">
            <send>UAAMm5GRQKSCMVvLyl4rFw==</send>
            <recv>UAAMm5GRQKSCMVvLyl4rFw==</recv>
          </participantstreamassoc>
     </recording>
```


4. Security Considerations

There is no security consideration as it is informational callflow document.

5. IANA Considerations

This document has no IANA considerations

6. Acknowledgement

Thanks to Ofir Rath, Charles Eckel and Yaron Pdut for their review comments.

7. References

7.1. Normative References

[RFC3261] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M., and E. Schooler, "SIP: Session Initiation Protocol", RFC 3261, DOI 10.17487/RFC3261, June 2002, <http://www.rfc-editor.org/info/rfc3261>.

7.2. Informative References

[I-D.ietf-siprec-metadata] R, R., Ravindran, P., and P. Kyzivat, "Session Initiation Protocol (SIP) Recording Metadata", draft-ietf-siprec-metadata-17 (work in progress), February 2015.

[RFC7245] Hutton, A., Ed., Portman, L., Ed., Jain, R., and K. Rehor, "An Architecture for Media Recording Using the Session Initiation Protocol", RFC 7245, DOI 10.17487/RFC7245, May 2014, <http://www.rfc-editor.org/info/rfc7245>.

[RFC4575] Rosenberg, J., Schulzrinne, H., and O. Levin, Ed., "A Session Initiation Protocol (SIP) Event Package for Conference State", RFC 4575, DOI 10.17487/RFC4575, August 2006, <http://www.rfc-editor.org/info/rfc4575>.

[RFC4353] Rosenberg, J., "A Framework for Conferencing with the Session Initiation Protocol (SIP)", RFC 4353, DOI 10.17487/RFC4353, February 2006, <http://www.rfc-editor.org/info/rfc4353>.

[RFC6230] Boulton, C., Melanchuk, T., and S. McGlashan, "Media Control Channel Framework", RFC 6230, DOI 10.17487/RFC6230, May 2011, <http://www.rfc-editor.org/info/rfc6230>.

Authors' Addresses

Ram Mohan Ravindranath
Cisco Systems, Inc.
Cessna Business Park,
Kadabeesanahalli Village, Varthur Hobli,
Sarjapur-Marathahalli Outer Ring Road
Bangalore, Karnataka 560103
India

Email: rmohanr@cisco.com

Parthasarathi Ravindran
Nokia Networks
Bangalore, Karnataka
India

Email: partha@parthasarathi.co.in

Paul Kyzivat
Huawei
Hudson, MA
USA

Email: pkyzivat@alum.mit.edu