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Session Initiation Protocol (SIP) Recording Call Flows
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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, the metadata format for which is described in [[I-D.ietf-siprec-metadata](#)]. This is purely an informational document that is written to support the model defined in the metadata draft.

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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 [[I-D.ietf-siprec-architecture](#)].


```

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 [[I-D.ietf-siprec-architecture](#)] 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 [[I-D.ietf-siprec-architecture](#)] 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. 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
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'>
    <datamode>complete</datamode>
    <session session_id="hVpd7YQgRW2nD22h7q60JQ==">
      <start-time>2010-12-16T23:41:07Z</start-time>
    </session>
    <participant
      participant_id="srfBEImCRp2QB23b7Mpk0w=="
      <nameID aor=sip:alice@atlanta.com>
        <name xml:lang="it">Alice</name>
      </nameID>
    </participant>
    <participantsessionassoc
      participant_id="srfBEImCRp2QB23b7Mpk0w=="
      session_id="hVpd7YQgRW2nD22h7q60JQ=="
      <associate-time>2010-12-16T23:41:07Z</associate-time>
    </participantsessionassoc>
    <participantstreamassoc
      participant_id="srfBEImCRp2QB23b7Mpk0w=="
      <send>i1Pz3to5hGk8fuXl+PbwCw==</send>
      <send>UAAMm5GRQKSCMVvLy14rFw==</send>
      <recv>8zc6e0lYtLWIINA6GR+3ag==</recv>
      <recv>EiXGlc+4TruqqoDaNE76ag==</recv>
    </participantstreamassoc>
    <participant
      participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
      <nameID aor=sip:bob@biloxi.com>
        <name xml:lang="it">Bob</name>
      </nameID>
    </participant>
    <participantsessionassoc
      participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
```



```

        session_id="hVpd7YQgRW2nD22h7q60JQ==">
      <associate-time>2010-12-16T23:41:07Z</associate-time>
    </participantsessionassoc>
    <participantstreamassoc
      participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
      <send>8zc6e0lYtLWIINA6GR+3ag==</send>
      <send>EiXGlc+4TruqqoDaNE76ag==</send>
      <recv>UAAMm5GRQKSCMVvLy14rFw==</recv>
      <recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>
    </participantstreamassoc>
    <stream id="UAAMm5GRQKSCMVvLy14rFw=="
      session_id="hVpd7YQgRW2nD22h7q60JQ==">
      <label>96</label>
    </stream>
    <stream id="i1Pz3to5hGk8fuXl+PbwCw=="
      session_id="hVpd7YQgRW2nD22h7q60JQ==">
      <label>97</label>
    </stream>
    <stream id="8zc6e0lYtLWIINA6GR+3ag=="
      session_id="zSfPoSvdSDCmU3A3TRDxAw==">
      <label>98</label>
    </stream>
    <stream id="EiXGlc+4TruqqoDaNE76ag=="
      session_id="zSfPoSvdSDCmU3A3TRDxAw==">
      <label>99</label>
    </stream>
  </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. The metadata snapshot looks as below

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
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'>
    <datamode>partial</datamode>
    <participantstreamassoc
      participant_id="srfBEImCRp2QB23b7Mpk0w==">
      <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 during Alice with participant_id srfBEImCRp2QB23b7Mpk0w== only receives media streams since Alice is put on hold 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
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'>
    <datamode>partial</datamode>
    <participantstreamassoc
      participant_id="srfBEImCRp2QB23b7Mpk0w==">
      <recv>8zc6e0lYtLWIINA6GR+3ag==</recv>
      <recv>EiXGlc+4TruqqoDaNE76ag==</recv>
      <send>i1Pz3to5hGk8fuXl+PbwCw==</send>
      <send>UAAMm5GRQKSCMVvLy14rFw==</send>
    </participantstreamassoc>
    <participantstreamassoc
      participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
      <send>8zc6e0lYtLWIINA6GR+3ag==</send>
      <send>EiXGlc+4TruqqoDaNE76ag==</send>
      <recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>
      <recv>UAAMm5GRQKSCMVvLy14rFw==</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 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.

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
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'>
    <datamode>partial</datamode>
    <participantsessionassoc
      participant_id="srfBEImCRp2QB23b7Mpk0w=="
      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>
    <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>
    <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
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'>
      <datamode>Partial</datamode>
      <session session_id="hVpd7YQgRW2nD22h7q60JQ==">
        <stop-time>2010-12-16T23:41:07Z</stop-time>
      </session>
      <participantsessionassoc
        participant_id="srfBEImCRp2QB23b7Mpk0w=="
        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.

```
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
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'>
```

```
<datamode>complete</datamode>
```

```
<session session_id="bfLZ+NTFEeCNxQTuRyQBmw==">
```

```
<start-time>2010-12-16T23:41:07Z</start-time>
```

```
</session>
```

```
<participant
```

```
participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
```

```
<nameID aor=sip:Bob@biloxi.com/>
```

```
</participant>
```

```
<participantsessionassoc
```

```
participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
```

```
session_id="bfLZ+NTFEeCNxQTuRyQBmw==">
```

```
<associate-time>2010-12-16T23:32:03Z</associate-time>
```

```
</participantsessionassoc>
```



```
<participantstreamassoc
  participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
  <send>8zc6e0lYtLWIINA6GR+3ag==</send>
  <send>EiXGlc+4TruqqoDaNE76ag==</send>
  <recv>60JAJm9UTvik0Ltlih/Gzw==</recv>
  <recv>AcR5FUd3Edi8cACQJy/3JQ==</recv>
</participantstreamassoc>
<participant
  participant_id="Atnm1ZRnOC6Pm5MApkrDzQ=="
  <nameID aor=sip:carol@example.com/>
</participant>
<participantsessionassoc
  participant_id="Atnm1ZRnOC6Pm5MApkrDzQ=="
  session_id="bfLZ+NTFEeCNxQTuRyQBmw=="
  <associate-time>2010-12-16T23:41:07Z</associate-time>
</participantsessionassoc>
<participantstreamassoc
  participant_id="Atnm1ZRnOC6Pm5MApkrDzQ=="
  <send>60JAJm9UTvik0Ltlih/Gzw==</send>
  <send>AcR5FUd3Edi8cACQJy/3JQ==</send>
  <recv>8zc6e0lYtLWIINA6GR+3ag==</recv>
  <recv>EiXGlc+4TruqqoDaNE76ag==</recv>
</participantstreamassoc>
<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>
</recording>
```

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@8.3.2.12:5060 SIP/2.0
Via: SIP/2.0/UDP 8.3.2.34:5060;branch=z9hG4bK47c8eb30
From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
To: <sip:recorder@example.com>
Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
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'>
    <datamode>Partial</datamode>
    <session session_id="hVpd7YQgRW2nD22h7q60JQ==">
      <stop-time>2010-12-16T23:41:07Z</stop-time>
    </session>
    <participantsessionassoc
      participant_id="srfBEImCRp2QB23b7Mpk0w=="
      session_id="hVpd7YQgRW2nD22h7q60JQ=="
      <disassociate-time>2010-12-16T23:41:07Z</disassociate-time>
    </participantsessionassoc>

    <participantsessionassoc
      participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
      session_id="hVpd7YQgRW2nD22h7q60JQ=="
      <disassociate-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 [[I-D.ietf-siprec-architecture](#)] 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 sends one Media Streams. 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 could be part of conference Focus model described in section 3 of [[I-D.ietf-siprec-architecture](#)].

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
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'>

<datamode>complete</datamode>

<session session_id="hVpd7YQgRW2nD22h7q60JQ==">

<start-time>2010-12-16T23:41:07Z</start-time>

</session>

<participant

participant_id="srfBEImCRp2QB23b7Mpk0w==">

<nameID aor=sip:alice@atlanta.com>

<name xml:lang="it">Alice</name>

</nameID>

</participant>

<participantsessionassoc

participant_id="srfBEImCRp2QB23b7Mpk0w=="

session_id="hVpd7YQgRW2nD22h7q60JQ==">

<associate-time>2010-12-16T23:41:07Z</associate-time>

</participantsessionassoc>

<participantstreamassoc

participant_id="srfBEImCRp2QB23b7Mpk0w==">

<send>i1Pz3to5hGk8fuXl+PbwCw==</send>

<recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>

</participantstreamassoc>

<participant

participant_id="zSfPoSvdSDCmU3A3TRDxAw==">

<nameID aor=sip:bob@biloxi.com>

<name xml:lang="it">Bob</name>

</nameID>

</participant>

<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>

<stream id="i1Pz3to5hGk8fuXl+PbwCw=="

session_id="hVpd7YQgRW2nD22h7q60JQ==">

<label>96</label>

</stream>

</recording>

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
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'>

<datamode>partial</datamode>

<participantstreamassoc

participant_id="srfBEImCRp2QB23b7Mpk0w==">

<recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>

</participantstreamassoc>

<participantstreamassoc

participant_id="zSfPoSvdSDCmU3A3TRDxAw==">

<send>i1Pz3to5hGk8fuXl+PbwCw==</send>

</participantstreamassoc>

<stream id="i1Pz3to5hGk8fuXl+PbwCw=="

session_id="hVpd7YQgRW2nD22h7q60JQ==">

<label>96</label>

</stream>

</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
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'>

<datamode>partial</datamode>

<participantstreamassoc

participant_id="srfBEImCRp2QB23b7Mpk0w==">

<send>i1Pz3to5hGk8fuXl+PbwCw==</send>

<recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>

</participantstreamassoc>

<participantstreamassoc

participant_id="zSfPoSvdSDCmU3A3TRDxAw==">

<send>i1Pz3to5hGk8fuXl+PbwCw==</send>

<recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>

</participantstreamassoc>

<stream id="i1Pz3to5hGk8fuXl+PbwCw=="

session_id="hVpd7YQgRW2nD22h7q60JQ==">

<label>96</label>

</stream>

</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
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'>

<datamode>partial</datamode>

<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>

<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
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'>
    <datamode>partial</datamode>
    <participantsessionassoc
      participant_id="srfBEImCRp2QB23b7Mpk0w=="
      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@8.3.2.12:5060 SIP/2.0
Via: SIP/2.0/UDP 8.3.2.34:5060;branch=z9hG4bK47c8eb30
From: <sip:2000@example.com>;tag=35e195d2-947d-4585-946f-09839247
To: <sip:recorder@example.com>
Call-ID: d253c800-b0d1ea39-4a7dd-3f0e20a
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'>
    <datamode>Partial</datamode>
    <session session_id="hVpd7YQgRW2nD22h7q60JQ==">
      <stop-time>2010-12-16T23:41:07Z</stop-time>
    </session>
    <participantsessionassoc
      participant_id="srfBEImCRp2QB23b7Mpk0w=="
      session_id="hVpd7YQgRW2nD22h7q60JQ==">
      <disassociate-time>2010-12-16T23:41:07Z</disassociate-time>
    </participantsessionassoc>

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

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

In trading turrent, audio mixing is done locally before forwarding to the recording server. Sender and receiver audio is mixed for each communication session. The audio from multiple communication sessions is also mixed, or multiplexed, in a single RTP session to the recording server.

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

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'>
    <datamode>complete</datamode>
    <group group_id="7+0TCyoxTmqmqyA/1weDAg==">
      <associate-time>2010-12-16T23:41:07Z</associate-time>
    </group>
    <session session_id="hVpd7YQgRW2nD22h7q60JQ==">
      <group-ref>7+0TCyoxTmqmqyA/1weDAg==</group-ref>
      <start-time>2010-12-16T23:41:07Z</start-time>
    </session>
    <session session_id="zzlafnvvj1CH1laHF6mn8kkSS==">
      <group-ref>7+0TCyoxTmqmqyA/1weDAg==</group-ref>
      <start-time>2010-12-16T23:43:07Z</start-time>
    </session>
    <participant
      participant_id="srfBE1mCRp2QB23b7Mpk0w=="
      <nameID aor=sip:ram@example.com>
        <name xml:lang="it">RamMohan R</name>
      </nameID>
    </participant>
    <participantsessionassoc
      participant_id="srfBE1mCRp2QB23b7Mpk0w=="
      session_id="hVpd7YQgRW2nD22h7q60JQ=="
      <associate-time>2010-12-16T23:41:07Z</associate-time>
    </participantsessionassoc>
    <participantstreamassoc
      participant_id="srfBE1mCRp2QB23b7Mpk0w=="
      <send>UAAMm5GRQKSCMVvLy14rFw==</send>
      <recv>UAAMm5GRQKSCMVvLy14rFw==</recv>
    </participantstreamassoc>
    <participant
      participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
      <nameID aor=sip:partha@example.com>
        <name xml:lang="it">Parthasarathi R</name>
      </nameID>
    </participant>
    <participantsessionassoc
      participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
      session_id="hVpd7YQgRW2nD22h7q60JQ==">
```



```
<associate-time>2010-12-16T23:41:07Z</associate-time>
</participantsessionassoc>
<participantsessionassoc
  participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
  session_id="zzlafnvvj1CH1laHF6mn8kkSS==">
  <associate-time>2010-12-16T23:43:07Z</associate-time>
</participantsessionassoc>
<participantstreamassoc
  participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
  <send>UAAMm5GRQKSCMVvLy14rFw==</send>
  <recv>UAAMm5GRQKSCMVvLy14rFw==</recv>
</participantstreamassoc>
<participant
  participant_id="EiXGlc+4TruqqoDaNE76ag==">
  <nameID aor=sip:paul@example.com>
    <name xml:lang="it">Paul</name>
  </nameID>
</participant>
<participantsessionassoc
  participant_id="EiXGlc+4TruqqoDaNE76ag=="
  session_id="zzlafnvvj1CH1laHF6mn8kkSS==">
  <associate-time>2010-12-16T23:43:07Z</associate-time>
</participantsessionassoc>
<participantstreamassoc
  participant_id="EiXGlc+4TruqqoDaNE76ag==">
  <send>UAAMm5GRQKSCMVvLy14rFw==</send>
  <recv>UAAMm5GRQKSCMVvLy14rFw==</recv>
</participantstreamassoc>

<stream id="UAAMm5GRQKSCMVvLy14rFw=="
  session_id="hVpd7YQgRW2nD22h7q60JQ==">
  <label>96</label>
</stream>
</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

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7. References

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