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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 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 snip of the SIP/SDP message and the XML snapshot of metadata is shown. This document is informational and is not normative on any aspect of SIP.

2. Terminology

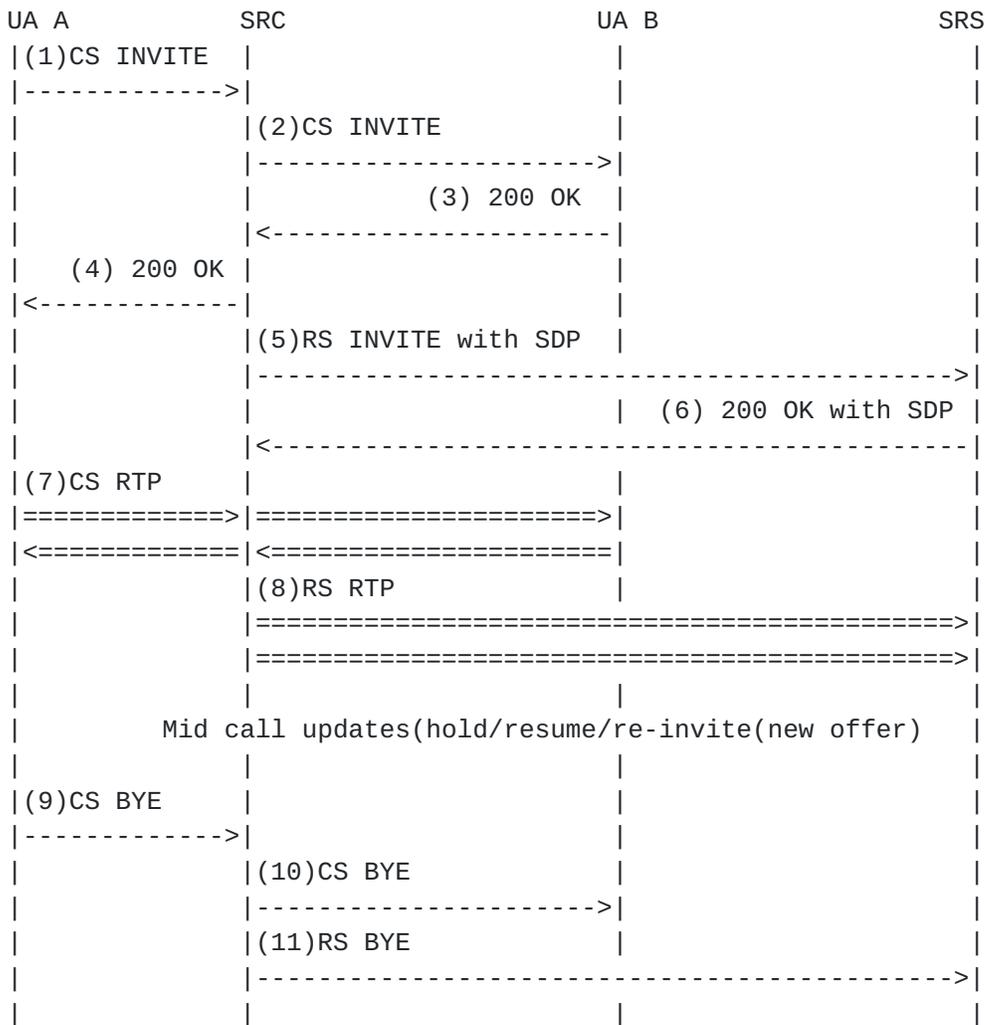
The terms using in this defined are defined in [[I-D.ietf-siprec-metadata](#)]. No new terms/definitions are introduced in this document.

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 messages are NOT shown here.

3.1. Sample Call flow

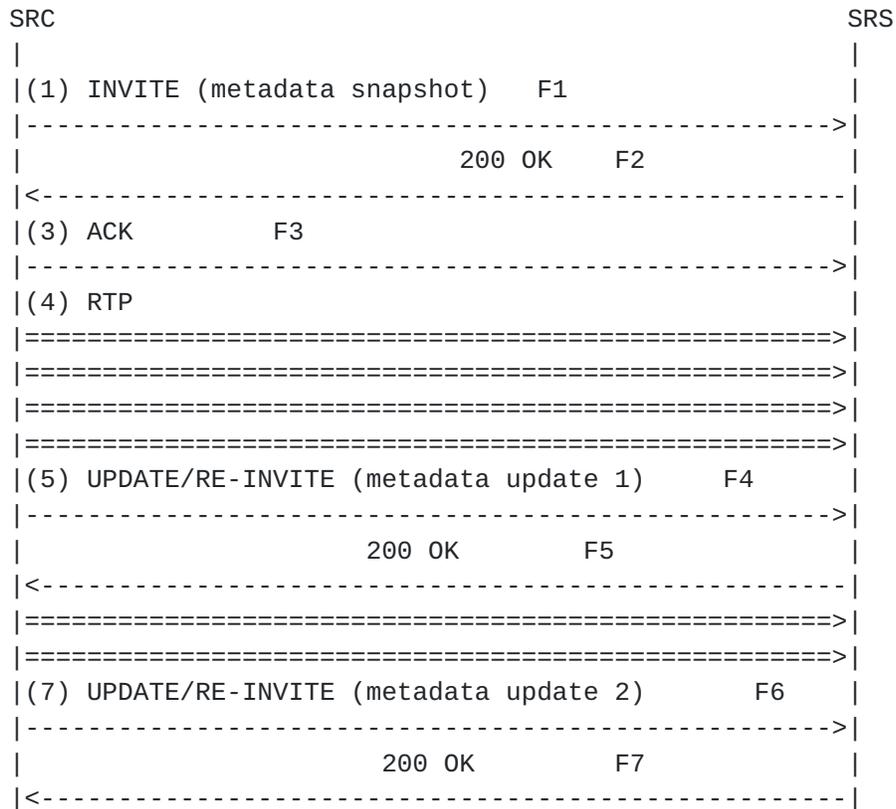
The following is a sample call flow that shows the SRC establishing a recording session towards the SRS. The call flow is essentially identical when the SRC is a B2BUA or as the endpoint itself. Note that the SRC can choose when to establish the Recording Session independent of the Communication Session, even though the following call flow suggests that the SRC is establishing the Recording Session (message #5) after the Communication Session is established.



The above call flow can also apply to the case of a centralized conference with a mixer. For clarity, ACKs to INVITES and 200 OKs to BYEs are not shown. The conference focus can provide the SRC functionality since the conference focus has access to all the media from each conference participant. When a recording is requested, the SRC delivers the metadata and the media streams to the SRS. Since the conference focus has access to a mixer, the SRC may choose to mix the media streams from all participants as a single mixed media stream towards the SRS.

An SRC can use a single recording session to record multiple communication sessions. Every time the SRC wants to record a new call, the SRC updates the recording session with a new SDP offer to add new recorded streams to the recording session, and correspondingly also update the metadata for the new call.

3.2. Example 1: Basic Call



Basic call between two Participants A(Ram) and B(Partha) who are part of one session. In this use case each participant sends two Media Streams. Media Streams sent by each participant is received all other participants of that CS 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 snippets of SDP is shown. Here the RS stream is unmixed.

F1 INVITE SRC -----> SRS

```

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
```

```
....
```

```
<?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:ram@example.com>
        <name xml:lang="it">RamMohan R</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:partha@example.com>
        <name xml:lang="it">Parthasarathi R</name>
      </nameID>
    </participant>
    <participantsessionassoc
      participant="zSfPoSvdSDCmU3A3TRDxAw=="
      <associate-time>2010-12-16T23:41:07Z</associate-time>
```



```
</participantsessionassoc>
<participantstreamassoc
  participant="zSfPoSvdSDCmU3A3TRDxAw==">
  <send>8zc6e0lYt1WIINA6GR+3ag==</send>
  <send>EiXGlc+4TruqqoDaNE76ag==</send>
  <recv>UAAMm5GRQKSCMVvLy14rFw==</recv>
  <recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>
</participantstreamassoc>
<stream id="UAAMm5GRQKSCMVvLy14rFw=="
  session="hVpd7YQgRW2nD22h7q60JQ==">
  <label>96</label>
</stream>
<stream id="i1Pz3to5hGk8fuXl+PbwCw=="
  session="hVpd7YQgRW2nD22h7q60JQ==">
  <label>97</label>
</stream>
<stream id="8zc6e0lYt1WIINA6GR+3ag=="
  session="zSfPoSvdSDCmU3A3TRDxAw==">
  <label>98</label>
</stream>
<stream id="EiXGlc+4TruqqoDaNE76ag=="
  session="zSfPoSvdSDCmU3A3TRDxAw==">
  <label>99</label>
</stream>
</recording>
```

3.3. Example 2: Hold/resume

Basic call between two Participants A and B. This is the continuation of above use-case. One of the participants(say A) goes on hold and then resumes as part of the same session. The metadata snapshot looks as below

During hold

F4 RE-INVITE SRC----->SRS

Content-Type: application/SDP

```

...
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=label:96
a=inactive
...
m=video 49174 RTP/AVPF 96
a=rtpmap:96 H.264/90000
a=label:97
a=inactive
...
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
....

```

```

<?xml version="1.0" encoding="UTF-8"?>
  <recording xmlns='urn:ietf:params:xml:ns:recording'>
    <dataMode>partial</dataMode>
    <participantstreamassoc
      participant="srfBEImCRp2QB23b7Mpk0w=="
      session="hVpd7YQgRW2nD22h7q60JQ==">
      <recv>8zc6e0lYtLWIINA6GR+3ag==</recv>
      <recv>EiXGlc+4TruqqoDaNE76ag==</recv>
    </participantstreamassoc>
    <participantstreamassoc
      participant="zSfPoSvdSDCmU3A3TRDxAw=="
      session="hVpd7YQgRW2nD22h7q60JQ==">
      <send>8zc6e0lYtLWIINA6GR+3ag==</send>
      <send>EiXGlc+4TruqqoDaNE76ag==</send>
    </participantstreamassoc>
  </recording>

```

During resume

The snapshot will look pretty much same as above with just the SDP

dir change.

F5 RE-INVITE SRC----->SRS

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

....

<?xml version="1.0" encoding="UTF-8"?>

<recording xmlns='urn:ietf:params:xml:ns:recording'>

<dataMode>partial</dataMode>

<participantstreamassoc

 participant="srfBEImCRp2QB23b7Mpk0w=="

 session="hVpd7YQgRW2nD22h7q60JQ==">

 <recv>8zc6e0lYTLWIINA6GR+3ag==</recv>

 <recv>EiXGlc+4TruqqoDaNE76ag==</recv>

</participantstreamassoc>

<participantstreamassoc

 participant="zSfPoSvdSDCmU3A3TRDxAw=="

 session="hVpd7YQgRW2nD22h7q60JQ==">

 <send>8zc6e0lYTLWIINA6GR+3ag==</send>

 <send>EiXGlc+4TruqqoDaNE76ag==</send>

</participantstreamassoc>

</recording>

3.4. Example 3: Basic Call with transfer

Basic call between two Participants A and B is connected as in Use-case 1. Transfer is initiated by one of the participants or by other entity(3PCC case). SRC sends a snapshot of the participant changes to SRS. In this instance participant A(Ram) drops out during the transfer and Participant C(Paul) joins the session. There can be two cases here, same session continues after transfer or a new session (e.g. REFER based transfer) is created

Transfer with same session retained - (.e.g. RE-INVITE based transfer). Participant A drops out and C is added to the same session. No change to session/group element. C will be new stream element which maps to RS SDP using the same labels in this instance.

```

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
....
<?xml version="1.0" encoding="UTF-8"?>
  <recording xmlns='urn:ietf:params:xml:ns:recording'>
    <dataMode>partial</dataMode>
    <participantstreamassoc
      participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
      <send>8zc6e0lYtLWIINA6GR+3ag==</send>
      <send>EiXGlc+4TruqqoDaNE76ag==</send>
      <recv>60JAJm9UTvik0Ltlh/Gzw==</recv>
      <recv>AcR5FUd3Edi8cACQJy/3JQ==</recv>
    </participantstreamassoc>

```



```

<participant
  participant_id="Atnm1ZRn0C6Pm5MApkrDzQ==">
  <nameIDaor=sip:paul@example.com>
  <name xml:lang="it">Paul Kyzivat</name>
  </nameID>
</participant>
<participantsessionassoc
  participant_id="Atnm1ZRn0C6Pm5MApkrDzQ=="
  session_id="hVpd7YQgRW2nD22h7q60JQ==">
  <associate-time>2010-12-16T23:41:07Z</associate-time>
</participantsession>
<participantstreamassoc
  participant_id="Atnm1ZRn0C6Pm5MApkrDzQ=="
  <send>60JAJm9UTvik0Ltlih/Gzw==</send>
  <send>AcR5FUd3Edi8cACQJy/3JQ==</send>
  <recv>8zc6e0lYtLWIINA6GR+3ag==</recv>
  <recv>EiXGlc+4TruqqoDaNE76ag==</recv>
  <associate-time>2010-12-16T23:41:07Z</associate-time>
</participantstreamassoc>
<stream stream_id="60JAJm9UTvik0Ltlih/Gzw=="
  session_id="hVpd7YQgRW2nD22h7q60JQ==">
  <label>96</label>
</stream>
<stream stream_id="AcR5FUd3Edi8cACQJy/3JQ=="
  session_id="hVpd7YQgRW2nD22h7q60JQ==">
  <label>97</label>
</stream>
<stream stream_id="8zc6e0lYtLWIINA6GR+3ag=="
  session_id="zSfPoSvdSDCmU3A3TRDxAw==">
  <label>98</label>
</stream>
<stream stream_id="EiXGlc+4TruqqoDaNE76ag=="
  session_id="zSfPoSvdSDCmU3A3TRDxAw==">
  <label>99</label>
</stream>
</recording>

```

Transfer with new session - (.e.g. REFER based transfer). In this case new session is part of same grouping (done by SRC).

SRC may send an optional snapshot indicating stop for the old 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="srfBE1mCRp2QB23b7Mpk0w=="
      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>

```

SRC sends a snapshot to indicate the participant change and new session information after transfer. In this example the same RS is used.

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
....
<?xml version="1.0" encoding="UTF-8"?>
  <recording xmlns='urn:ietf:params:xml:ns:recording'>

```



```
<dataMode>partial</dataMode>
<session session_id="bfLZ+NTFEeCNxQTuRyQBmw==">
  <start-time>2010-12-16T23:41:07Z</start-time>
</session>
<participant
  participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
  <nameID aor=sip:partha@example.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:paul@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>
```



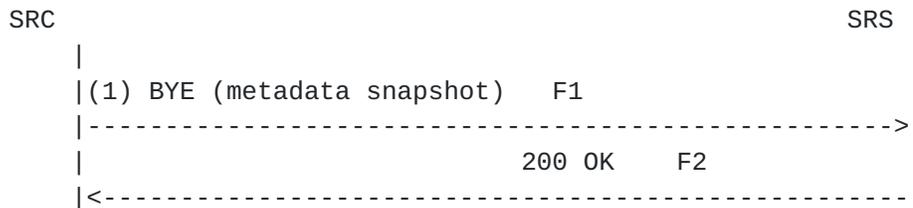
```

    <stream stream_id="EiXGlc+4TruqqoDaNE76ag=="
      session_id="bflZ+NTFEeCNxQTuRyQBmw==">
      <label>99</label>
    </stream>
  </recording>

```

3.5. Example 4: Call disconnect

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



F1 BYE SRC -----> SRS

```

<?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>

```


3.6. Example 5: Multiple CS into single RS with mixed stream

In trading turret, 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.

```
F1 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="srfBEImCRp2QB23b7Mpk0w=="
      <nameID aor=sip:ram@example.com>
        <name xml:lang="it">RamMohan R</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>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="zSfPoSvdSDCmU3A3TRDxAw=="
  session="hVpd7YQgRW2nD22h7q60JQ==">
  <associate-time>2010-12-16T23:41:07Z</associate-time>
</participantsessionassoc>
<participantsessionassoc
  participant="zSfPoSvdSDCmU3A3TRDxAw=="
  session="zzlafnvvj1CH11aHF6mn8kkSS==">
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<participantstreamassoc
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  <recv>UAAMm5GRQKSCMVvLyl4rFw==</recv>
</participantstreamassoc>
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  <nameID aor=sip:paul@example.com>
    <name xml:lang="it">Paul</name>
  </nameID>
</participant>
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</stream>
</recording>
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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 for his review comments.

7. References

7.1. Normative References

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