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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 organizations. In some of these environments, all calls must be recorded for regulatory, compliance, and consumer protection reasons. The recording of a session is typically performed by sending a copy of a media stream to a recording device. This document lists call flows with metadata snapshots sent from a Session Recording Client(SRC) to a Session Recording Server(SRS).

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[1.](#) Overview

Session recording is a critical requirement in many communications environments, such as call centers and financial trading organizations. In some of these environments, all calls must be recorded for regulatory, compliance, and consumer protection reasons. The recording of a session is typically performed by sending a copy of a media stream to a recording device. [[RFC7865](#)] focuses on the

recording metadata which describes the Communication Session(CS). This document lists few examples and shows the snapshots of metadata sent from a Session Recording Client(SRC) to Session Recording Server (SRS). For the sake of simplicity the entire Session Initiation Protocol (SIP) [[RFC3261](#)] messages are not shown, instead only

snippets of the SIP and Session Description Protocol (SDP) [[RFC4566](#)] messages and the XML snapshot of metadata is shown.

[2.](#) Terminology

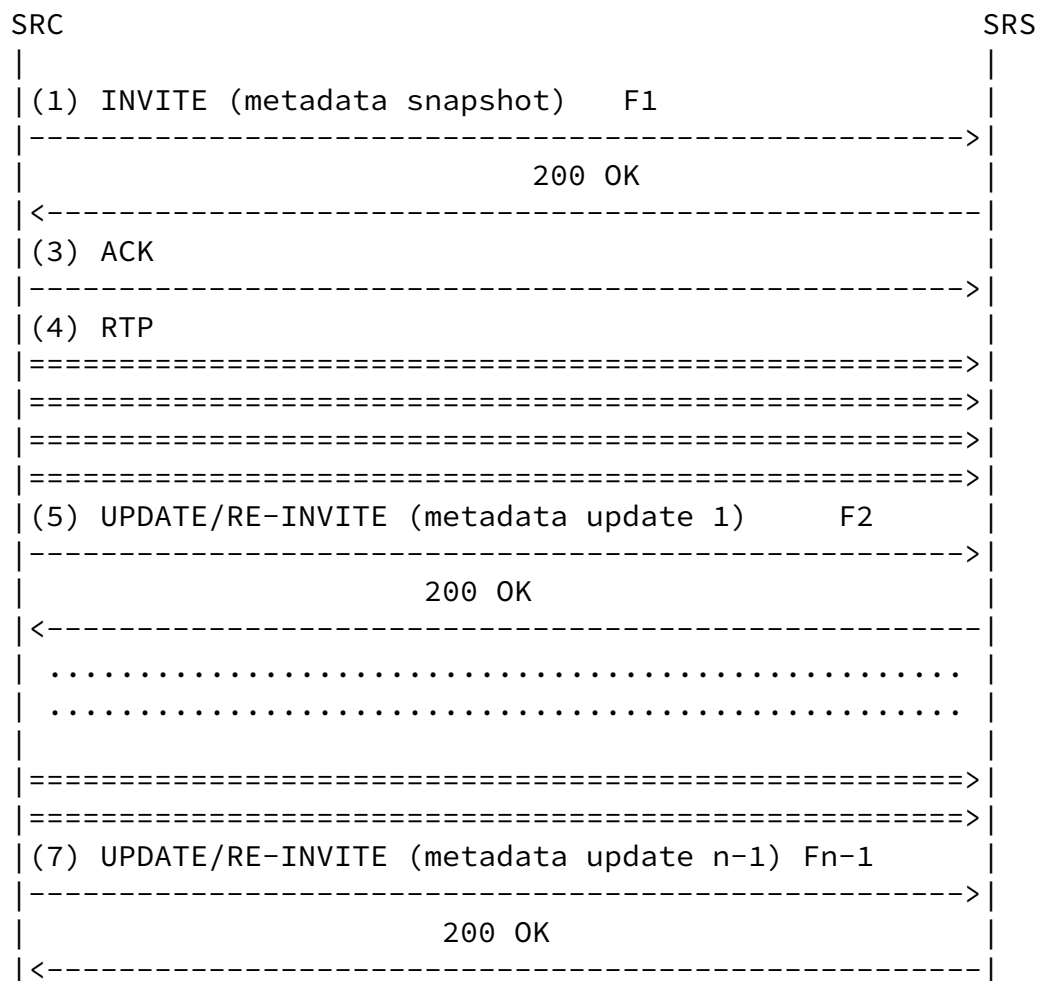
The terms using in this document are defined in [[RFC7865](#)] and [[RFC6341](#)]. No new definitions are introduced in this document.

[3.](#) Metadata XML Instances

The following sub-sections has examples showing the metadata snapshot sent from SRC to SRS. In all these use-cases, the SRC is a B2BUA.

[3.1.](#) Sample Call flow

The following is a sample call flow that shows the SRC establishing a Recording Session(RS) towards the SRS. The SRC in this example could be part of any one of the architectures described in [section 3 of](#) [\[RFC7245\]](#).



For the sake of simplicity, ACKs to RE-INVITES and BYEs are not shown. The subsequent sections describe 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 indicate 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 try 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 Turret 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 without mixing

This section describes example flows where SRC can be a SIP-UA or B2BUA as described in [section 3 of \[RFC7245\]](#). The SRS here can be a SIP-UA or an entity part of MEDIACTRL architecture described in [section 3 of \[RFC7245\]](#).

[3.2.1.](#) Example 1: Basic Call

Basic call between two participants Alice and Bob who are part of the same CS. In this use case each participant sends two media streams(audio and video). Media streams sent by each participant are received by the other participant 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. This snapshot has the complete metadata. For the sake of simplicity only snippets of SIP/SDP are shown. In

this example the SRCs records the streams of each participant to SRS without mixing.

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+0TCyoxTmqmqyA/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>F00!</structure>
        <whatever>bar</whatever>
      </mydata>
    </group>
    <session session_id="hVpd7YQgRW2nD22h7q60JQ==">
      <sipSessionID>ab30317f1a784dc48ff824d0d3715d86;
        remote=47755a9de7794ba387653f2099600ef2</sipSessionID>
      <group-ref>7+0TCyoxTmqmqyA/1weDAg==
      </group-ref>
      <!-- Standardized extension -->
      <mydata xmlns='http://example.com/my'>
        <structure>F00!</structure>
        <whatever>bar</whatever>
      </mydata>
    </session>
  <participant
    participant_id="srfBEImCRp2QB23b7Mpk0w==">
    <nameID aor="sip:alice@atlanta.com">
      <name xml:lang="it">Alice</name>
    </nameID>
    <!-- Standardized extension -->
    <mydata xmlns='http://example.com/my'>

```

```

</session>
<participant
  participant_id="srfBEImCRp2QB23b7Mpk0w==">
  <nameID aor="sip:alice@atlanta.com">
    <name xml:lang="it">Alice</name>
  </nameID>
  <!-- Standardized extension -->
  <mydata xmlns='http://example.com/my'>

```

```

                <structure>F00!</structure>
                <whatever>bar</whatever>
            </mydata>
        </participant>
        <participant
            participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
            <nameID aor="sip:bob@bilox.com">
                <name xml:lang="it">Bob</name>
            </nameID>
            <!-- Standardized extension -->
            <mydata xmlns='http://example.com/my'>
                <structure>F00!</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="srfBEImCRp2QB23b7Mpk0w=="
            session_id="hVpd7YQgRW2nD22h7q60JQ==">
            <associate-time>2010-12-16T23:41:07Z</associate-time>
        </participantsessionassoc>
        <participantsessionassoc

```



```

        session_id="hVpd7YQgRW2nD22h7q60JQ==">
        <associate-time>2010-12-16T23:41:07Z</associate-time>
</participantsessionassoc>
<participantstreamassoc
    participant_id="srfBEImCRp2QB23b7Mpk0w==">
    <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

A call between two participants Alice and Bob is established and a RS is created for recording as in example 1. One of the participants Bob puts Alice hold and then resumes as part of the same CS. The 'send' and 'recv' XML elements of a 'participantstreamassoc' XML element is used to indicate whether a participant is contributing to a media stream or not. 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>

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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="srfBEImCRp2QB23b7Mpk0w==">
      <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>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 acting as SRC as per [section 3.1.1 of \[RFC7245\]](#)) has sent a snapshot as described 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 are two flows that can happen 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' XML 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==">

<nameID aor="sip:carol@example.com">

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

</nameID>

```

</participant>
<participantsessionassoc
  participant_id="Atnm1ZRn0C6Pm5MApkrDzQ=="
  session_id="hVpd7YQgRW2nD22h7q60JQ==">
  <associate-time>2013-12-16T23:41:07Z</associate-time>
</participantsessionassoc>
<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>60JAJm9UTvik0Ltlh/Gzw==</recv>
  <recv>AcR5FUd3Edi8cACQJy/3JQ==</recv>
</participantstreamassoc>
<participantstreamassoc
  participant_id="Atnm1ZRn0C6Pm5MApkrDzQ==">
  <send>60JAJm9UTvik0Ltlh/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 an optional message. An SRC may choose to just send an INVITE with a new 'session' XML element to implicitly indicate that the participants are now part of a different CS without sending disassociation from the old CS. The SRC in this example uses the same RS. In case the SRC wishes 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="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>

```

In the above snapshot, the 'participantsessionassoc' XML 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 RS to continue recording the call. The 'sipSessionID' XML element in metadata snapshot now indicates Bob 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==">
```

<sipSessionID>3363127f0d084c10876dddd4f8e5eeb9

```
    ;remote=2272bb7e70fe41dba0025ae9a26d54cf</sipSessionID>
    <start-time>2010-12-16T23:41:07Z</start-time>
</session>
<participant
  participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
  <nameID aor="sip:Bob@bilox.com"/>
</participant>
<participant
  participant_id="Atnm1ZRnOC6Pm5MApkrDzQ=="
  <nameID aor="sip:carol@example.com"/>
</participant>
  <stream stream_id="60JAJm9UTvik0Ltlh/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>60JAJm9UTvik0Ltlh/Gzw==</recv>
    <recv>AcR5FUd3Edi8cACQJy/3JQ==</recv>
  </participantstreamassoc>
  <participantstreamassoc
    participant_id="Atnm1ZRnOC6Pm5MApkrDzQ==">
    <send>60JAJm9UTvik0Ltlh/Gzw==</send>
    <send>AcR5FUd3Edi8cACQJy/3JQ==</send>
  </participantstreamassoc>

```

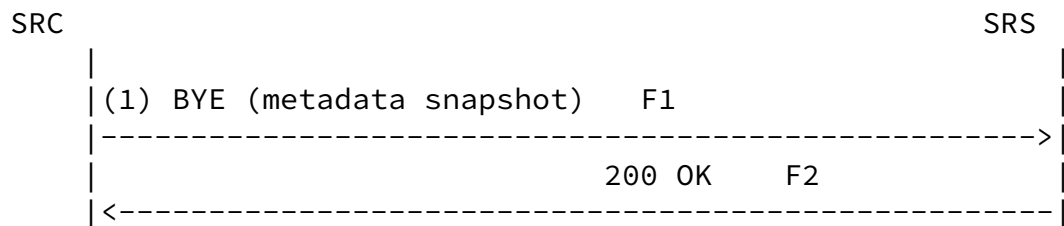
```

    <recv>8zc6e0lYTLWIINA6GR+3ag==</recv>
    <recv>EiXGlc+4TruqqoDaNE76ag==</recv>
  </participantstreamassoc>
</recording>

```

[3.2.4.](#) Example 4: Call disconnect

This example shows a snapshot of metadata sent by the SRC to SRS when a CS with Alice and Bob as participants is disconnected.



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>
```

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```
</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>
```

[3.3.](#) Call Scenarios with SRC recording streams by mixing

The section describes a few example call flows where SRC may be part of conference model either as focus or a participant in conference as explained in [section 3.1.5 of \[RFC7245\]](#). The SRS here can 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 calls into a conference server (say, an MCU) to attend one of many conferences hosted on or managed by that server. Media streams sent by each participant are received by all the other participants in the conference. Below is the initial snapshot sent by SRC in the INVITE to SRS that has the complete metadata. For the sake of simplicity only snippets of SIP/SDP are shown. The SRC 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 media.

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==">
      <sipSessionID>fa3b60f27e91441e84c55a9a0095f075
        ;remote=a358d2b81a444a8c8fb05950cef331e7</sipSessionID>
      <sipSessionID>ca718e1430474b5485a53aa5d0bea45e
        ;remote=68caf509b9284b7ea45f84a049febf0a</sipSessionID>
      <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>
    <participant
      participant_id="zSfPoSvdSDCmU3A3TRDxAw=="
      <nameID aor="sip:bob@biloxi.com">
        <name xml:lang="it">Bob</name>
      </nameID>
    </participant>
    <stream stream_id="i1Pz3to5hGk8fuXl+PbwCw=="
      session_id="hVpd7YQgRW2nD22h7q60JQ==">
```

```
      <label>96</label>
    </stream>
    <participantsessionassoc
      participant_id="srfBEImCRp2QB23b7Mpk0w=="
      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="srfBEImCRp2QB23b7Mpk0w==">
  <send>i1Pz3to5hGk8fuXl+PbwCw==</send>
  <recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>
</participantstreamassoc>

<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

This is the continuation of Example 1: Basic call with SRC mixing streams. Given a call between two participants Alice and Bob is established and a RS is created for recording as in example 5. One of the participants, Bob puts Alice hold and then resumes as part of the same CS. The 'send' and 'recv' XML elements of a 'participant' XML element are used to indicate whether a participant is contributing or not to a media stream. 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="srfBEImCRp2QB23b7Mpk0w==">
      <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==">
```

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```

    <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. Below is a snapshot sent from SRC to SRC in this case. Note the SRC here can be a focus or a participant in the conference. In the 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>
    </participantsessionassoc>
    <participantstreamassoc
      participant_id="Atnm1ZRnOC6Pm5MApkrDzQ=="
      <send>i1Pz3to5hGk8fuXl+PbwCw==</send>
      <recv>i1Pz3to5hGk8fuXl+PbwCw==</recv>
    </participantstreamassoc>
  </recording>
```

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

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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="srfBEImCRp2QB23b7Mpk0w=="
      session_id="hVpd7YQgRW2nD22h7q60JQ=="
      <disassociate-time>2010-12-16T23:41:07Z</disassociate-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 where 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 a conference. The SRS here could be a SIP UA or an entity part of MEDIACTRL architecture. Except in the disconnect case, the snapshot remains same as 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="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.](#) Turret-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 (each call is one CS) on different handsets/speakers on the same turret into a single RS. This would mean media

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

Taking an 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:


```
    ;remote=a358d2b81a444a8c8fb05950cef331e7</sipSessionID>
    <sipSessionID>497c0f13929643b4a16858e2a3885edc
    ;remote=a358d2b81a444a8c8fb05950cef331e7</sipSessionID>
    <group-ref>7+OTCyoxTmqmqyA/1weDAg==</group-ref>
    <start-time>2010-12-16T23:41:07Z</start-time>
  </session>
<session session_id="e6370VVGEEWAG6886p18uA==">
  <sipSessionID>ae10731ca50343a5aaae2dd0904a65de
  ;remote=a358d2b81a444a8c8fb05950cef331e7</sipSessionID>
  <sipSessionID>33c77aac7deb414cbc8c10f363fccb71
  ;remote=a358d2b81a444a8c8fb05950cef331e7</sipSessionID>
  <sipSessionID>fd6932e9e5fc489fae2d5b3779723b7e
  ;remote=a358d2b81a444a8c8fb05950cef331e7</sipSessionID>
  <group-ref>7+OTCyoxTmqmqyA/1weDAg==</group-ref>
  <start-time>2010-12-16T23:43:07Z</start-time>
</session>
<participant
  participant_id="srfBEImCRp2QB23b7Mpk0w==">
  <nameID aor="sip:alice@atlanta.com">
    <name xml:lang="it">Alice</name>
  </nameID>
</participant>
<participant
  participant_id="zSfPoSvdSDCmU3A3TRDxAw==">
  <nameID aor="sip:Bob@biloxi.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="srfBEImCRp2QB23b7Mpk0w=="
  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="e6370VVGEEWAG6886p18uA=="
  <associate-time>2010-12-16T23:43:07Z</associate-time>
</participantsessionassoc>
<participantsessionassoc
  participant_id="EiXGlc+4TruqqoDaNE76ag=="
  session_id="e6370VVGEEWAG6886p18uA=="
  <associate-time>2010-12-16T23:43:07Z</associate-time>
</participantsessionassoc>

<participantstreamassoc
  participant_id="srfBEImCRp2QB23b7Mpk0w=="
  <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

Security considerations mentioned in [\[RFC7865\]](#) and [\[RFC7866\]](#) has to be followed by SRC and SRS for setting up RS SIP dialog and sending metadata.

[5.](#) IANA Considerations

This document has no IANA considerations

[6.](#) Acknowledgement

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