

ControlONE Technical Guide

Recording Interface - SIPREC

v6.1

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Introduction

This document describes ControlONE Interface for external Call Recording using SIPREC standard, as described in RFC 6341.

Definitions

- CS = Call Session
- RS = Recording Session
- SRS = Session Recording Server. External Recorder.
- SRC = Session Recording Client. ControlONE Conference Server/Console

Interface Description

Using SIPREC interface, an external recorder system can be easily integrated to ControlONE. As a single channel, SIPREC can handle Call Information and Media in the same connection.

ControlONE supports up to 6 active recorders on each call. This way, a geographic distributed high-availability solution can be achieved.

The list of active SRS servers needs to be configured in ControlONE server WEB or CLI Interface.

Session Flow

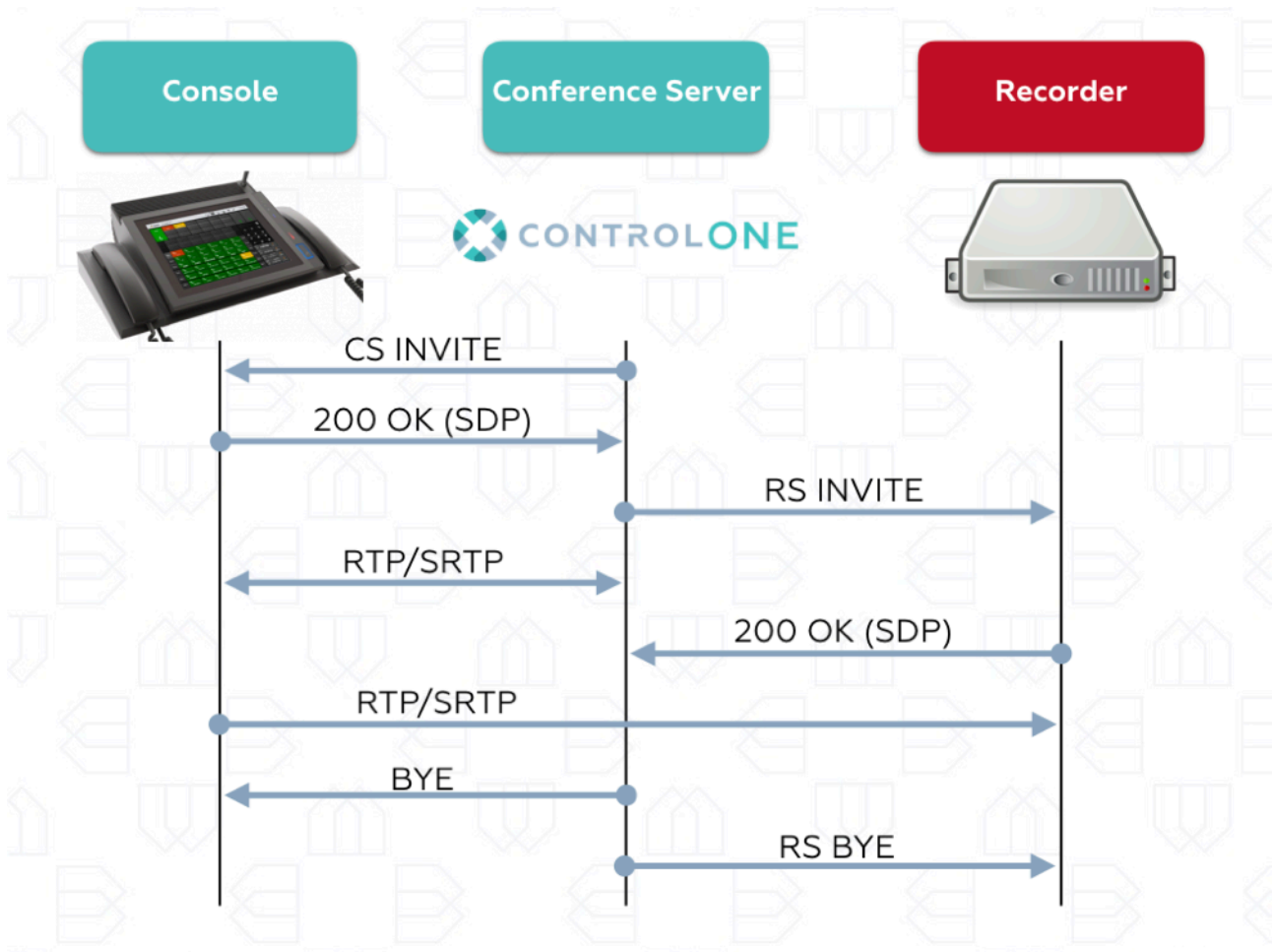
All calls (CS) are handled by ControlONE Conference Server, and for each one, a request to start RS is sent to SRS. RS starts right after call is connected to console, and before other party is connected.

RS signaling uses the standard SIP protocol (RFC 3261) and SDP (RFC 4566). As media (audio) is only sent from SRC to SRS, option "a=sendonly" is used in RS INVITE.

As signaling and audio are sent from different components, the RTP source IP address will be different, and is sent in "c= line" of SDP in RS INVITE.

Media port "m= line" in SIP INVITE are used as source port for RTP stream. Destination port of SRS is expected in SDP reply contained in "200 OK" message.

In sum, RS flow occurs as described in "RFC 8068", "3.3 Call Scenarios with SRC Recording Streams by Mixing". See diagram below.



Call Flow example

Call Information

Call Information can be sent in RS INVITE from SRC to SRS using 2 different methods:

1. SIP Headers

In this case, SIP Headers are sent in RS INVITE to SRS, with a "X-ControlONE-Info-" prefix.

In case of multiples participants, all participants is sent in a list, separated by comma.

2. RFC 7865 Metadata

Another option is to send metadata using standard defined by RFC 7865.

Every SRS can be configured in system to use SIP Headers, RFC 7865 Metadata or both methods to receive call information.

Media Session

Audio is always mixed by SRC (Console), and only one RTP Stream is sent per RS to SRS.

Transcoding is supported during SIP recording session, and SRS can select codec in "200 OK". CS codec is preferred, and sent as first in "m= line" list.

Security

SIP communication between SRC and SRS can use SIP over TLS. Key Exchange is done using standard SDES (RFC 4568). After a successful SDES negotiation, SRTP is sent from Console to SRS. AES 128 and 256 algorithms are available.

Certificate validation can be done by ControlONE, using a list of trusted Certification Authorities in system.

Nota that SRTP is only enable if SIP/TLS is used, for security reasons.

Even secure calls between Console, Conference Server and external SIP Servers can be recorded, as RS is a new call, with mixed audio, and no secret key is reused from CS.

Licensing

SIPREC interface requires one CO-LICE-LOG license per system, regardless of how many active recorders are configured.

Examples

RS INVITE using SIP headers for Call information

```
INVITE sip:recorder@srs.example.com:5060 SIP/2.0
Via: SIP/2.0/UDP src.example.com:5060;branch=z9hG4bK2bbb31b4
Max-Forwards: 70
From: "Console ID" <sip:2000@src.example.com>;tag=as753c344d
To: <sip: sip:recorder@srs.example.com:5060>
Contact: <sip:2000@src.example.com:5060>;+sip.src
Call-ID: 558591eb6e73a81f01fc992f311e8242
CSeq: 101 INVITE
User-Agent: BYNE Conference
Date: Fri, 01 May 2017 10:17:11 GMT
X-ControlONE-Info-ConsoleID: Console ID
X-ControlONE-Info-User: Logged User
X-ControlONE-Info-Calling: Calling Name <1000>
X-ControlONE-Info-Called: Called Name 1<2000>, Called Name 2<2001>
X-ControlONE-Info-CallID: 1973499061
Allow: INVITE, ACK, CANCEL, BYE
Require: siprec
Accept: application/sdp, application/rs-metadata
Content-Type: application/sdp
Content-Length: [length]

v=0
o=SIMB-Conference 1055606034 1055606034 IN IP4 127.127.127.127
s=SIMB Conference
c=IN IP4 127.127.127.127
t=0 0
m=audio 7038 RTP/AVP 18 0 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=label:182
a=sendonly
```

RS INVITE using RFC 7865 Metadata

```
INVITE sip:recorder@srs.example.com:5060 SIP/2.0
Via: SIP/2.0/UDP src.example.com:5060;branch=z9hG4bK2bbb31b4
Max-Forwards: 70
From: "Console ID" <sip:2000@src.example.com>;tag=as753c344d
To: <sip: sip:recorder@srs.example.com:5060>
Contact: <sip:2000@src.example.com:5060>
Call-ID: 558591eb6e73a81f01fc992f311e8242
CSeq: 101 INVITE
User-Agent: BYNE Conference
Date: Fri, 01 May 2017 10:17:11 GMT
Allow: INVITE, ACK, CANCEL, BYE
Require: siprec
Accept: application/sdp, application/rs-metadata
Content-Type: multipart/mixed;boundary=foobar
Content-Length: [length]
```

```
--foobar
Content-Type: application/SDP
v=0
o=SIMB-Conference 1055606034 1055606034 IN IP4 127.127.127.127
s=SIMB Conference
c=IN IP4 127.127.127.127
t=0 0
m=audio 7038 RTP/AVP 18 0 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
aptime:20
a=label:182
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="a41aaf39-690d-4a20-8970-ae53ffa4dd81">
      <sipSessionID>ab797d38-af92-4d8b-bd16-ddb8ff63a27d</
sipSessionID>
      <sipSessionID>f6c76919-8016-4b28-b216-cc8470eee4de</
sipSessionID>
      <sipSessionID>31e403cc-c390-4d4d-95f4-3364be4cbffa</
sipSessionID>
      <start-time>2017-05-01T10:17:11Z</start-time>
    </session>
  <participant
```

```
    participant_id="d436830c-c10b-497c-a999-a481f90693a6">
    <nameID aor="tel:1000">
    <name xml:lang="it">Calling Name</name>
    </nameID>
  </participant>
</participant>
  participant_id="561fef91-b9a4-4402-b1d1-e8bc35d47cca">
  <nameID aor="tel:2000">
  <name xml:lang="it">Called Name 1</name>
  </nameID>
</participant>
</participant>
  participant_id="47615330-a348-43c2-9f45-cac897216177">
  <nameID aor="tel:2001">
  <name xml:lang="it">Called Name 2</name>
  </nameID>
</participant>
</participant>
<stream stream_id="0edd55c5-0436-4afe-ac94-1e48cc41fd93"
  session_id="a41aaf39-690d-4a20-8970-ae53ffa4dd81">
  <label>182</label>
</stream>
<participantsessionassoc
  participant_id="d436830c-c10b-497c-a999-a481f90693a6"
  session_id="a41aaf39-690d-4a20-8970-ae53ffa4dd81">
  <associate-time>2017-05-01T10:17:11Z</associate-time>
</participantsessionassoc>
<participantsessionassoc
  participant_id="561fef91-b9a4-4402-b1d1-e8bc35d47cca"
  session_id="a41aaf39-690d-4a20-8970-ae53ffa4dd81">
  <associate-time>2017-05-01T10:17:15Z</associate-time>
</participantsessionassoc>
<participantsessionassoc
  participant_id="47615330-a348-43c2-9f45-cac897216177"
  session_id="a41aaf39-690d-4a20-8970-ae53ffa4dd81">
  <associate-time>2017-05-01T10:17:17Z</associate-time>
</participantsessionassoc>
<participantstreamassoc
  participant_id="d436830c-c10b-497c-a999-a481f90693a6">
  <send>0edd55c5-0436-4afe-ac94-1e48cc41fd93</send>
  <recv>0edd55c5-0436-4afe-ac94-1e48cc41fd93</recv>
</participantstreamassoc>
<participantstreamassoc
  participant_id="561fef91-b9a4-4402-b1d1-e8bc35d47cca">
  <send>0edd55c5-0436-4afe-ac94-1e48cc41fd93</send>
  <recv>0edd55c5-0436-4afe-ac94-1e48cc41fd93</recv>
</participantstreamassoc>
<participantstreamassoc
  participant_id="47615330-a348-43c2-9f45-cac897216177">
  <send>0edd55c5-0436-4afe-ac94-1e48cc41fd93</send>
  <recv>0edd55c5-0436-4afe-ac94-1e48cc41fd93</recv>
</participantstreamassoc>
</recording>
```


Related Documents

- Use Cases and Requirements for SIP-Based Media Recording (SIPREC) - RFC 6341, see <https://tools.ietf.org/html/rfc6341>
- Session Recording Protocol - RFC 7866, see <https://tools.ietf.org/html/rfc7866>
- Session Initiation Protocol (SIP) Recording Metadata - RFC 7865, see <https://tools.ietf.org/html/rfc7865>
- Session Initiation Protocol (SIP) Recording Call Flows - RFC 8068, see <https://tools.ietf.org/html/rfc8068>
- Session Description Protocol (SDP) Security Descriptions for Media Streams - RFC 4568, see <https://tools.ietf.org/html/rfc4568>