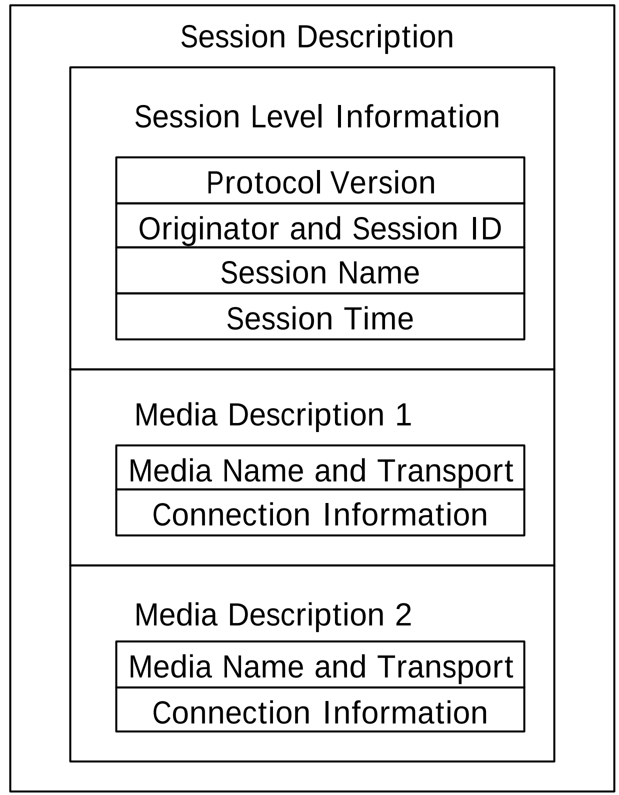
Session Initiation Protocol (SIP)

The Session Description Protocol

- The Most Common Message Body
  - Session information describing the media to be exchanged between the parties
  - SDP, RFC 2327 (initial publication)
    - A number of modifications to the protocol have been suggested.
- SIP uses SDP in an answer/offer mode.
  - An agreement between the two parties as to the types of media they are willing to share
  - RFC 3264 (An Offer/Answer Model with SDP)
    - To describe how SDP and SIP should be used together

The Structure of SDP

- SDP simply provides a format for describing session information to potential session participants.
- Text-based Protocol
- The Structure of SDP
  - Session Level Info
    - Name of the session
    - Originator of the session
    - Time that the session is to be active
  - Media Level Info
    - Media type
    - Port number
    - Transport protocol
    - Media format



SDP Syntax

- A number of lines of text
- In each line
  - field=value
  - field is exactly one character (case-significant)
- Session-level fields
- Media-level fields
  - Begin with media description field (m=)

Mandatory Fields

- v=(protocol version)
- o=(session origin or creator)
- s=(session name), a text string
  - For multicast conference
- t=(time of the session), the start time and stop time
  - For pre-arranged multicast conference
- m=(media)
  - Media type
  - The transport port
  - The transport protocol
  - The media format (typically an RTP payload format)

Optional Fields [1/3]

- Some optional fields can be applied at both session and media levels.
  - The value applied at the media level overrides that at the session level
- i=(session information)
  - A text description
  - At both session and media levels
  - It would be somewhat superfluous since SIP already supports the Subject header.
- u=(URI of description)
  - Where further session information can be obtained
  - Only at session level

Optional Fields [2/3]

- e=(e-mail address)
  - Who is responsible for the session
  - Only at the session level
- p=(phone number)
  - Only at the session level
- c=(connection information)
  - Network type, address type and connection address
  - At session or media level
- b=(bandwidth information)
  - In kilobits per second
  - At session or media level

Optional Fields [3/3]

- r=(repeat times)
  - For regularly scheduled session a session is to be repeated
  - How often and how many times
- z=(timezone adjustments)
  - For regularly scheduled session
  - Standard time and daylight savings time
- k=(encryption key)
  - An encryption key or a mechanism to obtain it for the purposes of encrypting and decrypting the media
  - At session or media level
- a=(attributes)
  - Describe additional attributes

Ordering of Fields

- Session Level
  - Protocol version (v)
  - Origin (o)
  - Session name (s)
  - Session information (i)
  - URI (u)
  - E-mail address (e)
  - Phone number (p)
  - Connection info (c)
  - Bandwidth info (b)
  - Time description (t)
  - Repeat info (r)
  - Time zone adjustments (z)
  - Encryption key (k)
  - Attributes (a)
- Media level
  - Media description (m)
  - Media info (i)
  - Connection info (c)
    - Optional if specified at the session level
  - Bandwidth info (b)
  - Encryption key (k)
  - Attributes (a)

Subfields [1/3]

- Field = <value of subfield1> <value of subfield2> <value of subfield3>
- Origin
  - Username, the originator's login id or "-"
  - Session ID
    - A unique ID
    - Make use of NTP timestamp
  - Version, a version number for this particular session
  - Network type
    - A text string
    - IN refers to Internet
  - Address type
    - IP4, IP6
  - Address, a fully-qualified domain name or the IP address

Subfields [2/3]

- Connection Data
  - The network and address at which media data will be received
  - Network type
  - Address type
  - Connection address
- Media Information
  - Media type
    - Audio, video, data, or control
  - Port
  - Format
    - List the various types of media format that can be supported
    - According to the RTP audio/video profile
  - m= audio 45678 RTP/AVP 15 3 0
    - G.728, GSM, G.711

Subfields [3/3]

- Attributes
  - To enable additional information to be included
  - Property attribute
    - a=sendonly
    - a=recvonly
  - Value attribute
    - a=orient:landscape used in a shared whiteboard session
  - Rtpmap attribute
    - The use of dynamic payload type
    - a=rtpmap:<payload type> <encoding name>/<clock rate> [/<encoding parameters>].
    - m=video 54678 RTP/AVP 98
    - a=rtpmap 98 L16/16000/2
      - 16-bit linear encoded stereo (2 channels) audio sampled at 16kHz

Usage of SDP with SIP

- SIP and SDP make a wonderful partnership for the transmission of session information.
- SIP provides the messaging mechanism for the establishment of multimedia sessions.
- SDP provides a structured language for describing the sessions.
  - The entity headers identifies the message body.

SIP Inclusion in SIP Messages

- Fig 5-15
  - G.728 is selected
- INVITE with multiple media streams
  - Unsupported should also be returned with a port number of zero
- An alternative
  - INVITE

    m=audio 4444 RTP/AVP 2 4 15

    a=rtpmap 2 G726-32/8000

    a=rtpmap 4 G723/8000

    a=rtpmap 15 G728/8000
  - 200 OK

    m=audio 6666 RTP/AVP 15

    a=rtpmap 15 G728/8000

Daniel<sip:Collins@station1.work.com>

Boss<sip:Manager@station2.work.com>

a

```
INVITE sip:Manager@station2.work.com SIP/2.0
From: Daniel<sip:Collins@station1.work.com>; tag = abcd1234
To: Boss<sip:Manager@station2.work.com>
CSeq: 1 INVITE
Content-Length: 213
Content-Type: application/sdp
Content-Disposition: session

v=0
o=collins 123456 001 IN IP4 station1.work.com
s=
c=IN IP4 station1.work.com
t=0 0
m=audio 4444 RTP/AVP 2
a=rtpmap 2 G726-32/8000
m=audio 4666 RTP/AVP 4
a=rtpmap 4 G723/8000
m=audio 4888 RTP/AVP 15
a=rtpmap 15 G728/8000
```

b

```
SIP/2.0 200 OK
...
```

Daniel<sip:Collins@station1.work.com>

Boss<sip:Manager@station2.work.com>

b

SIP/2.0 200 OK
From: Daniel<sip:Collins@station1.work.com>; tag = abcd1234
To: Boss<sip:Manager@station2.work.com>; tag = xyz789
CSeq: 1 INVITE
Content-Length: 163
Content-Type: application/sdp
Content-Disposition: session

v=0
o=collins 45678 001 IN IP4 station2.work.com
s=
c=IN IP4 station2.work.com
t=0 0
m=audio 0 RTP/AVP 2
m=audio 0 RTP/AVP 4
m=audio 6666 RTP/AVP 15
a=rtpmap 15 G728/8000

c

ACK sip:manager@station2.work.com SIP/2.0
From: Daniel<sip:Collins@station1.work.com>; tag = abcd1234
To: Boss<sip:Manager@station2.work.com>; tag = xyz789
CSeq: 1 ACK
Content-Length: 0

d

Conversation

SIP and SDP Offer/Answer Model

- Re-INVITE is issued when the server replies with more than one codec.
  - With the same dialog identifier (To and From headers, including tag values), Call-ID and Request-URI
  - The session version is increased by 1 in o= line of message body.
- A mismatch
  - 488 or 606
  - Not Acceptable
  - A Warning header with warning code 304 (media type not available) or 305 (incompatible media type)
  - Then the caller issues a new INVITE request.

Daniel<sip:Collins@station1.work.com> Boss<sip:Manager@station2.work.com>

a

INVITE sip:manager@station2.work.com SIP/2.0
CSeq: 1 INVITE
Content-Length: 183
Content-Type: application/sdp
Content-Disposition: session

v=0
o=collins 123456 001 IN IP4 station1.work.com
s=
c=IN IP4 station1.work.com
t=0 0
m=audio 4444 RTP/AVP 2 4 15
a=rtpmap 2 G726-32/8000
a=rtpmap 4 G723/8000
a=rtpmap 15 G728/8000
a=inactive

b

SIP/2.0 200 OK
CSeq: 1 INVITE
Content-Length: 157
Content-Type: application/sdp
Content-Disposition: session

v=0
o=collins 45678 001 IN IP4 station2.work.com
s=
c=IN IP4 station2.work.com
t=0 0
m=audio 6666 RTP/AVP 4 15
a=rtpmap 4 G723/8000
a=rtpmap 15 G728/8000
a=inactive

Daniel<sip:Collins@station1.work.com>

Boss<sip:Manager@station2.work.com>

c

ACK sip:manager@station2.work.com SIP/2.0
From: Daniel<sip:Collins@station1.work.com>; tag = abcd1234
To: Boss<sip:Manager@station2.work.com>; tag = xyz789
CSeq: 1 ACK
Content-Length: 0

d

INVITE sip:manager@station2.work.com SIP/2.0
CSeq: 2 INVITE
Content-Length: 126
Content-Type: application/sdp
Content-Disposition: session

v=0
o=collins 123456 002 IN IP4 station1.work.com
s=
c=IN IP4 station1.work.com
t=0 0
m=audio 4444 RTP/AVP 15
a=rtpmap 15 G728/8000

OPTIONS Method

- Determine the capabilities of a potential called party
- Accept Header
  - Indicate the type of information that the sender hopes to receive
- Allow Header
  - Indicate the SIP methods that Boss can handle
- Supported Header
  - Indicate the SIP extensions that can be supported

Daniel<sip:Collins@station1.work.com>

Boss<sip:Manager@station2.work.com>

a

OPTIONS sip:manager@station2.work.com SIP/2.0
Via: SIP/2.0/UDP Station1.work.com; branch=z9hG4bK7890123
From: Daniel<sip:Collins@work.com>; tag=lmnop123
To: Boss<sip:Manager@station2.work.com>
Call-ID: 123456@station1.work.com
Contact: Daniel<sip:Collins@station1.work.com>
CSeq: 1 OPTIONS
Accept: application/sdp
Content-Length: 0

b

SIP/2.0 200 OK
Via: SIP/2.0/UDP Station1.work.com; branch=z9hG4bK7890123
From: Daniel<sip:Collins@work.com>; tag=lmnop123
To: Boss<sip:Manager@station2.work.com>; tag=xyz5678
Call-ID: 123456@station1.work.com
CSeq: 1 OPTIONS
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE
Supported: newfield
Content-Length: 146
Content-Type: application/sdp

v=0
o=manager 45678 001 IN IP4 station2.work.com
s=
c=IN IP4 station2.work.com
t=0 0
m=audio 0 RTP/AVP 4 15
a=rtpmap 4 G723/8000
a=rtpmap 15 G728/8000