

11. - SIP SDP - Changing Session Parameters

In SIP SDP, the session parameters can be changed during the lifetime of a session by sending a re-INVITE message. A re-INVITE is an INVITE message sent within an established session that is used to modify session attributes.

When a re-INVITE message is sent, it includes a new SDP offer that specifies the updated session parameters. The remote party can then respond with a new SDP answer that acknowledges the changes or rejects them. The re-INVITE can be used to update media session attributes such as codecs, media formats, bandwidth, and transport protocols, as well as session-level attributes such as session name, session description, and session timing.

Re-INVITEs are commonly used in scenarios where a user wants to add or remove media streams from an established session, change the codec or format of an existing media stream, or modify other session parameters. The use of re-INVITEs provides a flexible mechanism for session negotiation and management in SIP-based communication systems.

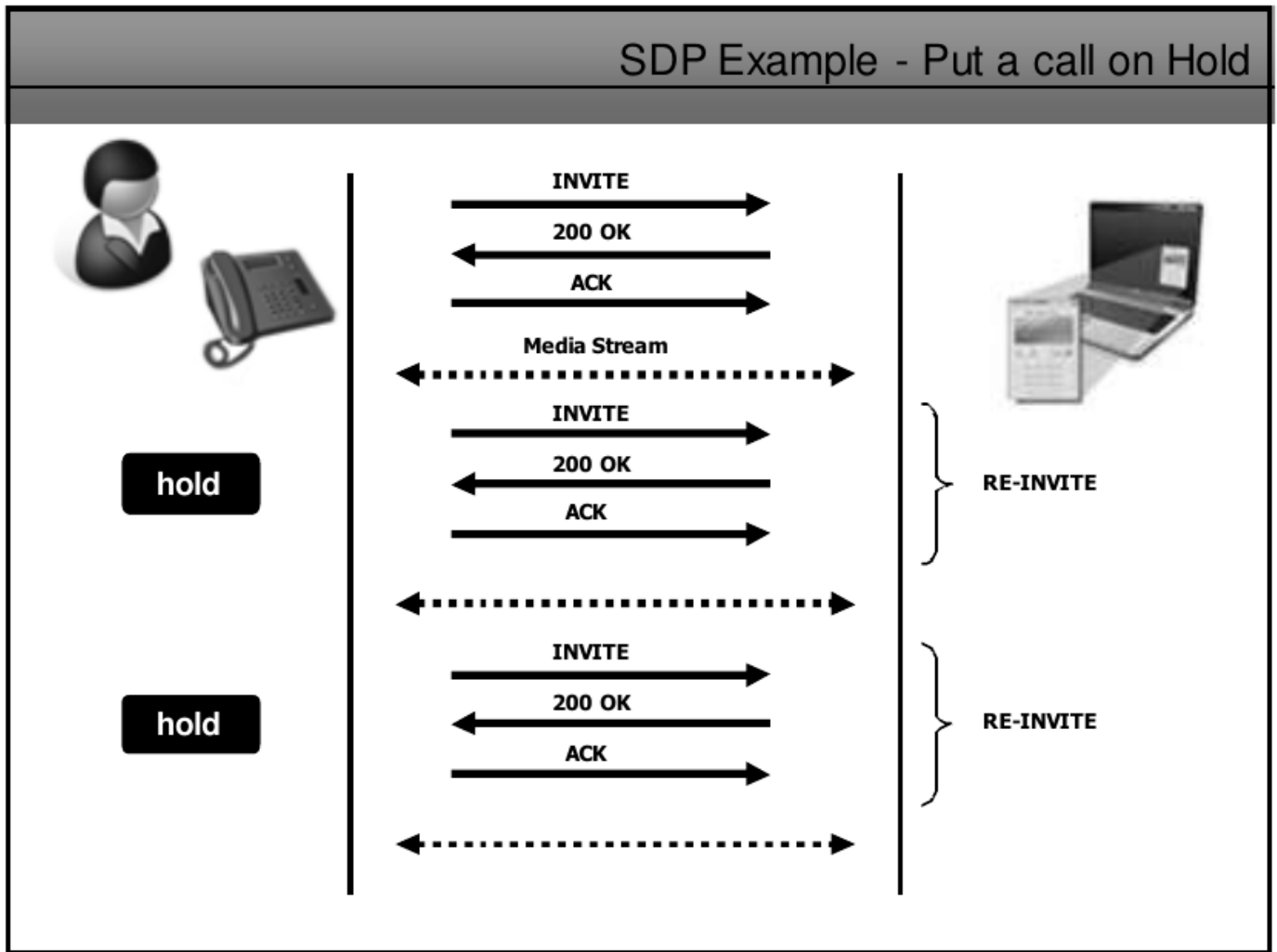
Other methods that will change or update SDP parameters

1. INVITE with new SDP: When a new INVITE is sent with a new SDP, it indicates that the session parameters are being updated.
2. UPDATE: The UPDATE method can be used to modify the session parameters without terminating the session. It carries a new SDP message that describes the updated parameters.
3. Re-INVITE: A Re-INVITE can be sent by either the caller or the callee to modify the session parameters. It is similar to the INVITE method, but it is sent while the call is in progress.
4. INFO: The INFO method can be used to send non-INVITE related information during a call. It can also be used to modify the session parameters by carrying a new SDP message.

These methods can be used to update session parameters such as codec types, media ports, IP addresses, bandwidth, and other attributes defined in the SDP.

Putting a call on Hold example

Here a call is placed on hold and re-invite messages are sent



the SDP message is updated with `a=sendonly` and the IP is changed to `0.0.0.0`, remote user updates his sdp with `200 ok` and puts himself on `a=recvonly`, here user in left will send (send only) music on hold and remote (in receive only mode) will hear music)

when unhold user on left will send new invite with his IP `192.168.100.102` and `rtpmap a=sendrecv`, remote user will respond with `200 ok` and change mode to `a=sendrecv`, this will reestablish audio both ways.

SDP Example - Call Hold Trace



hold



```
v=0
s=SIP Call
c=IN IP4 0.0.0.0
t=0 0
a=sendonly
m=audio 20034 RTP/AVP 18 96
```

```
v=0
s=call
c=IN IP4 82.36.89.78
t=0 0
a=recvonly
m=audio 57528 RTP/AVP 18 96
```

```
v=0
s=SIP Call
c=IN IP4 192.168.100.102
t=0 0
a=sendrecv
m=audio 20034 RTP/AVP 18 96
```

```
v=0
s=call
c=IN IP4 82.36.89.78
t=0 0
a=sendrecv
m=audio 57528 RTP/AVP 18 96
```

there are multiple ways of messages when placing a call on hold example below

RFC 2543

Specified that placing a user on hold was accomplished by setting the connection address to c=0.0.0.0 in the SDP body

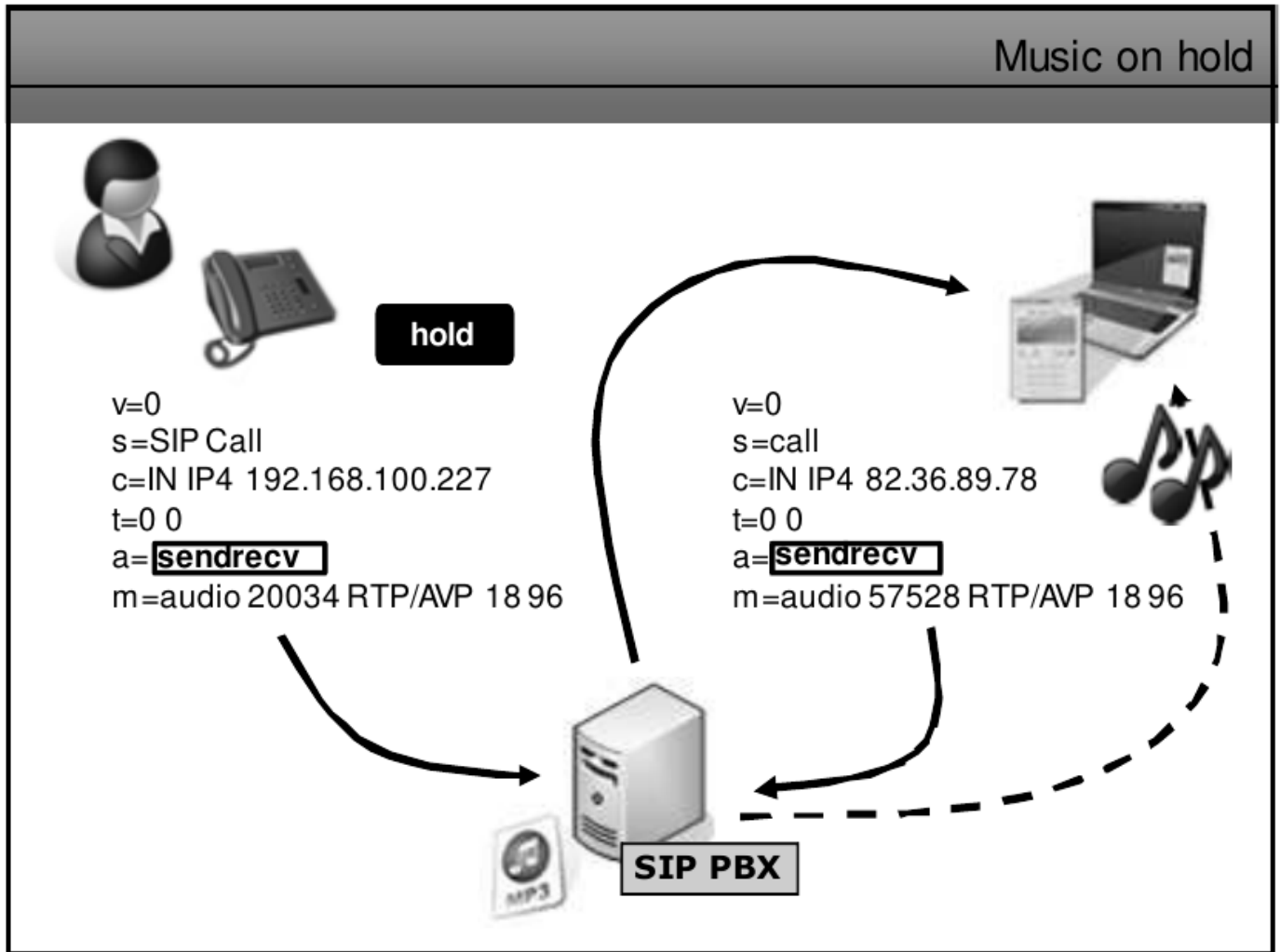
Though its usage for putting a call on hold is no longer recommended, since it doesn't allow for RTCP to be used with held streams, doesn't work with IPv6 and so on

RFC 3264

Extends options available with the following attributes and really discourages the use of c=0.0.0.0, though you may find that some devices support both RFCs to allow connection to devices that only support one of them

sendrecv	Used to establish a 2-way media stream.
recvonly	The SIP endpoint would only receive (listen mode) and not send media.
sendonly	The SIP endpoint would only send and not receive media.
inactive	The SIP endpoint would neither send nor receive media.

Music on hold



In SIP and SDP, putting a call on hold is achieved by the initiator of the hold sending a re-INVITE message to the other party with an SDP offer that indicates the call is being put on hold. The SDP offer typically sets the media streams to inactive state by setting the port to 0 and the IP address to 0.0.0.0.

For example, suppose Alice wants to put Bob on hold during a call. Alice would send a re-INVITE message to Bob with an SDP offer indicating that the audio and/or video streams are inactive, like so:

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bK776asdhd
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314160 INVITE
Contact: <sip:alice@pc33.atlanta.com>
```

```
User-Agent: BriaX 3.5.5 build 81887
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Allow-Events: presence, message-summary, refer, dialog
Session-Expires: 3600;refresher=uas
Content-Type: application/sdp
Content-Length: 218
```

```
v=0
o=- 123456 789012 IN IP4 pc33.atlanta.com
s=SIP Call
c=IN IP4 0.0.0.0
a=sendonly
t=0 0
m=audio 0 RTP/AVP 0
m=video 0 RTP/AVP 31
```

When Bob receives this message, he can either accept the hold request by sending a 200 OK response, or reject it by sending a 486 Busy Here response. If he accepts it, he will send a 200 OK response with his own SDP answer indicating that the media streams have been set to inactive state, like so:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bK776asdhds
To: Bob <sip:bob@biloxi.com>;tag=2482893830n
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314160 INVITE
Contact: <sip:bob@192.0.2.4:5060>
Content-Type: application/sdp
Content-Length: 102

v=0
o=- 123456 789012 IN IP4 192.0.2.4
s=SIP Call
c=IN IP4 0.0.0.0
t=0 0
a=recvonly
m=audio 0 RTP/AVP 0
m=video 0 RTP/AVP 31
```

Once the hold has been established, Alice and Bob can either resume the call by sending another re-INVITE message with an SDP offer indicating that the media streams are active, or terminate the call by sending a BYE message.

INVITE and reINVITE

In a SIP message, the type of message (e.g. INVITE, BYE, REGISTER, etc.) is specified in the first line, which is called the request line or status line. For example, in an INVITE message, the request line would start with "INVITE".

In the case of a re-INVITE message, it will also have the same request line format starting with "INVITE", but it will typically include additional headers such as "Call-ID", "From", "To", "CSeq" and "Max-Forwards". These headers provide information about the ongoing call session, allowing the recipient to identify it as a re-INVITE message and determine how to handle it appropriately.

It's also important to note that a re-INVITE message is always sent within an established SIP session and it typically contains a new SDP offer, which updates the media parameters of the session.

on the example below we can see the first 2 options are an Invite, to distinguish them we can look at the CSeq: number, they both are 1

the third attempt its a reINVITE as is using a different CSeq number.

INVITE and reINVITE

Original INVITE

INVITE sip:4791996@sipgate.co.uk;user=phone SIP/2.0
Via: SIP/2.0/UDP 82.36.89.78:2051;branch=z9hG4bK-ign0xpnjt3ku;rport
From: "Lynda Francis" <sip:8692812@sipgate.co.uk>;tag=7nk985tf rf
To: <sip:4791996@sipgate.co.uk;user=phone>
Call-ID: 3c28c57bcd14-3f bh2gw7ljev 9@model300-00041325A582
CSeq: 1 INVITE

Resent INVITE

INVITE sip:4791996@sipgate.co.uk;user=phone SIP/2.0
Via: SIP/2.0/UDP 82.36.89.78:2051;branch=z9hG4bK-ign0xpnjt3ku;rport
From: "Lynda Francis" <sip:8692812@sipgate.co.uk>;tag=7nk985tf rf
To: <sip:4791996@sipgate.co.uk;user=phone>
Call-ID: 3c28c57bcd14-3f bh2gw7ljev 9@model300-00041325A582
CSeq: 1 INVITE

reINVITE

INVITE sip:4791996@sipgate.co.uk;user=phone SIP/2.0
Via: SIP/2.0/UDP 82.36.89.78:2051;branch=z9hG4bK-ign0xpnjt3ku;rport
From: "Lynda Francis" <sip:8692812@sipgate.co.uk>;tag=7nk985tf rf
To: <sip:4791996@sipgate.co.uk;user=phone>
Call-ID: 3c28c57bcd14-3f bh2gw7ljev 9@model300-00041325A582
CSeq: 2 INVITE

In the next example, Alice sends two INVITEs with different branch IDs but the same CSeq number of 1 and 2 respectively. Bob can respond to each of these INVITEs with a different branch ID but the same CSeq number of 1 and 2 respectively.

After the call has been established, Alice sends a re-INVITE with a new branch ID and a CSeq number of 3 to

```
INVITE sip:bob@192.0.2.1 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.2:5060;branch=z9hG4bK-1234-5678
Max-Forwards: 70
To: Bob <sip:bob@192.0.2.1>
From: Alice <sip:alice@192.0.2.3>;tag=8765
Call-ID: 1234abcd@192.0.2.2
CSeq: 1 INVITE
Contact: <sip:alice@192.0.2.2>
Content-Type: application/sdp
Content-Length: 123
```

v=0
o=- 123456 789012 IN IP4 192.0.2.2
s=SIP Call
c=IN IP4 192.0.2.2
t=0 0
m=audio 3456 RTP/AVP 0 8

INVITE sip:bob@192.0.2.1 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.2:5060;branch=z9hG4bK-1234-5678
Max-Forwards: 70
To: Bob <sip:bob@192.0.2.1>
From: Alice <sip:alice@192.0.2.3>;tag=8765
Call-ID: 1234abcd@192.0.2.2
CSeq: 1 INVITE
Contact: <sip:alice@192.0.2.2>
Content-Type: application/sdp
Content-Length: 123

v=0
o=- 123456 789012 IN IP4 192.0.2.2
s=SIP Call
c=IN IP4 192.0.2.2
t=0 0
m=audio 3456 RTP/AVP 0 8

RE-INVITE sip:bob@192.0.2.1 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.2:5060;branch=z9hG4bK-3456-7890
Max-Forwards: 70
To: Bob <sip:bob@192.0.2.1>
From: Alice <sip:alice@192.0.2.3>;tag=8765
Call-ID: 1234abcd@192.0.2.2
CSeq: 3 RE-INVITE
Contact: <sip:alice@192.0.2.2>
Content-Type: application/sdp
Content-Length: 123

v=0
o=- 123456 789012 IN IP4 192.0.2.2
s=SIP Call
c=IN IP4 192.0.2.2

t=0 0

m=audio 3456 RTP/AVP 0 8

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