Offer & Answer interworking between JSEP & SIP
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Abstract

Real time communication Web (RTCWeb) workgroup defines the real time communication using JavaScript Session establishment protocol (JSEP) as an offer/answer mechanism. Session Initiation protocol (SIP) is IETF defined and well deployed protocol for real time communication. This document provides offer & answer interworking between JSEP and SIP.

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1. Introduction

Real time communication Web (RTCWeb) workgroup defines JavaScript Session establishment protocol (JSEP) [I-D.ietf-rtcweb-jsep] as an offer/answer mechanism. The transport to carry JSEP information shall be HTTP long polling [RFC6202] or WebSockets over TLS/TCP [RFC6455]. JSEP Offer/answer details shall be carried based on the application choice and some of the possible mechanism are REST or SIP or XMPP. HTTP based transport mechanism has the advantages of traversing through all NAT and firewall. JSEP offer/answer focuses on the offer/answer between two browsers wherein the offer/answer related feature like parallel forking in SIP is not provided.

Session Initiation protocol (SIP) [RFC3261] is IETF defined and well deployed protocol for real time communication. SIP offer/answer mechanism is based on [RFC3264] and few enhancements are done in the SIP layer.

RTCWeb clients (browser) and SIP UA supports Session Description protocol (SDP) [RFC4566] as the media description protocol. In case of mapping between these two protocols, SDP shall be reused.

SIP offer/answer supports the different RTP models [RFC3550] like RTP endpoint, RTP Mixer, RTP translator whereas JSEP supports RTP endpoint only.

As there are difference between SIP offer/answer and JSEP, there is a need of explicit mapping. This document provides the mapping between JSEP and SIP offer/answer mechanism. This mapping happens between SIP UA and RTCWeb entity which handles JSEP. RTCWeb entity in this document refers to the RTCWeb compliant browser or RTCWeb Server.

2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119]. This document only uses these key words when referencing normative statements in existing RFCs.

3. Definitions

- RTCWeb entity: RTCWeb compliant browser or RTCWeb Server.
- 18x response: 180 (ringing) or 183 (session progress) or any other response in the range of 180-189
4. SIP and JSEP offer/answer mapping architecture

The mapping between SIP and JSEP offer/answer is the building block described in this document. This mapping happens between SIP UA and RTCWeb entity which handles JSEP. RTCWeb entity in this document refers to the RTCWeb compliant browser or RTCWeb Server. The following diagrams illustrate the possible JSEP-SIP mapping architecture.

In the above figure 1, JSEP-SIP mapping is done at RTCWeb server. JSEP shall be transported from RTCWeb browser to RTCWeb Server by any signaling mechanism.

5. JSEP offer/answer to SIP offer/answer mapping

5.1. Basic JSEP session mapping

In RTCWeb browser, ICE is mandatorily supported. ICE calls for two offer/answer exchange wherein the initial offer indicates the set of ICE candidate pairs and the second offer/answer confirms the candidate pair which is used for the session. JSEP OFFER SHALL be mapped to INVITE with SDP and 200 OK with SDP SHALL be mapped to JSEP ANSWER.
### 5.2. Early media mapping

In SIP, early dialog is established using PRACK [RFC3262] method. ICE negotiation leads to OFFER2 from RTCWeb browser and it is triggered using UPDATE method [RFC3311] towards SIP UA. 18x (18x/183) response with SDP triggers the early media. SDP 200 for INVITE with Answer2 SHALL be ignored towards JSEP side as the answer2 is forwarded as part of 200 for UPDATE with Answer2.
PRANSWER state in JSEP is not used during ANSWER in the above callflow as OFFER2 is not allowed as per PRANSWER state in JSEP. OFFER in PRANSWER is an open item in JSEP.

5.3. Re-INVITE mapping

```
RTCWeb Browser          JSEP-SIP GW          SIP UA
<--Dialog is established with INV/200/ACK--
<-----OFFER----------<--RE-INVITE(OFFER)--
|                   |--------100--------->
|-------ANSWER-------|-----200 ANSWER----->
|                   |<----- ACK----------
```

Fig 5: RE-INVITE mapping

5.4. Offer JSEP- SIP Offer Race condition handling

It is possible for offer to be send by RTCWeb browser and SIP UA during the middle of the session. In [RFC3262], these race conditions are resolved using 491 response as shown below.
5.5. JSEP to SIP serial forking

In case of SIP serial forking with ICE exchange, second offer MAY be required in case the candidate pair is changing from the default during the early dialog and such earlier dialog offer/answer exchange has to happen in dialog1 using UPDATE mechanism. When the second INVITE is forked from proxy towards the UA3, the final response 200 OK from UA3 is received with different SDP as Answer3. Answer3 has to be mapped as Offer 3 towards JSEP in JSEP-SIP gateway as JSEP completed offer-answer exchange already. Also, PRANSWER state in JSEP is not used as OFFER2 with ICE update is not possible within PRANSWER state and it is an open item in JSEP. ANSWER4 SDP is same as ANSWER2, RE-INVITE towards UA shall be optimized or else RE-INVITE MUST be exchanged between JSEP-SIP GW to SIP-UA.
The assumption in Fig 5 is that ANSWER2 and ANSWER4 are same. In case they are different, there is a need of extra offer-answer between JSEP-SIP GW and SIP UA3

5.6. JSEP to SIP Parallel forking

JSEP does not support SIP parallel forking currently and it is the responsibility of the application to achieve the same. As JSEP-SIPGW is the application in this document, SIP parallel forking can be achieved as shown below. The point to be noted is that only one active RTP stream is possible in JSEP side for a given single m-line from JSEP offer and so, the other active RTP stream is updated as inactive using UPDATE offer/answer as shown in offer4. The rest of the callflow for parallel forking is same as SIP serial forking handling.
The assumption in Fig 8 is that ANSWER2 and ANSWER5 are same. In case they are different, there is a need of extra offer-answer between JSEP-SIP GW and SIP UA3

<table>
<thead>
<tr>
<th>RTCWeb Browser</th>
<th>JSEP-SIPGW</th>
<th>SIP Proxy</th>
<th>SIP UA2</th>
<th>SIP UA3</th>
</tr>
</thead>
<tbody>
<tr>
<td>--OFFER---&gt;</td>
<td>--INVITE(OFFER)-&gt;</td>
<td>--INVITE1---&gt;</td>
<td>OFFER</td>
<td></td>
</tr>
<tr>
<td></td>
<td>&lt;-----100--------</td>
<td>-----INVITE2---------</td>
<td>OFFER</td>
<td></td>
</tr>
<tr>
<td>&lt;-ANSWER---</td>
<td>&lt;-----18x ANSWER--</td>
<td>&lt;-18x ANSWER--</td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;OFFER2---&gt;</td>
<td>--PRACK---------&gt;</td>
<td>-----PRACK---------&gt;</td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;-ANSWER2--</td>
<td>&lt;-200 UPDATE</td>
<td>---- 200------</td>
<td>ANSWER2</td>
<td></td>
</tr>
<tr>
<td>&lt;OFFER3---&gt;</td>
<td>&lt;-18x ANSWER3---</td>
<td>&lt;-18x ANSWER3--------</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>--UPDATE OFFER3---&gt;</td>
<td>-----UPDATE---</td>
<td>OFFER2</td>
<td></td>
</tr>
<tr>
<td></td>
<td>&lt;-200 UPDATE</td>
<td>---- 200------</td>
<td>ANSWER2</td>
<td></td>
</tr>
<tr>
<td>&lt;ANSWER5---&gt;</td>
<td>&lt;200 UPDATE</td>
<td>---- 200------</td>
<td>ANSWER4</td>
<td></td>
</tr>
<tr>
<td></td>
<td>&lt;-200 ANSWER3----</td>
<td>&lt;200------------</td>
<td>ANSWER3</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>---- ACK---------&gt;</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>&lt;--- BYE1----&gt;</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Fig 8: Parallel Forking JSEP to SIP session mapping

6. SIP offer/answer to JSEP offer/answer mapping

SIP specific offer/answer mechanism is defined in [RFC3261].
6.1. Basic SIP-JSEP session mapping

INVITE with SDP offer is the first message in SIP which maps to JSEP Offer. The user alert information is not part of JSEP, HTTP application specific extension shall indicate the user alert status to the interworking entity which shall be later mapped to 18x mapping without SDP. When RTCWeb user accepts the session, JSEP send ANSWER towards interworking entity and ANSWER will be mapped to 200 OK message in SIP.

6.2. INVITE without SDP Offer to JSEP session mapping

It is possible in SIP that INVITE without SDP offer is the first message in SIP which has to initiate JSEP Offer from the client side. The mechanism by which JSEP offer initiate request between interworking entity and JSEP entity is outside the scope of the document. Once the JSEP offer is received in the interworking entity, it will be to 183 with offer towards UAC. PRACK with SDP answer is received.

6.3. SIP Non-ICE to JSEP ICE handling

SIP does not mandate for ICE [RFC5245] whereas JSEP in RTCWeb has to have mandatorily ICE handling. So, SIP-JSEP interworking entity has to make sure ICE to non-ICE handling as well as SDP specific changes for these interworking.

6.4. JSEP to SIP Trickle-ICE handling

Trickle ICE [I-D.rescorla-mmusic-ice-trickle] is a ICE extension mechanism wherein the candidates can be exchanged incrementally. This would allow ICE agents to exchange host candidates as soon as a session has been initiated. Connectivity checks for a media stream would also start as soon as the first candidates for that stream have become available.
### JSEP ICE to SIP Non-ICE Handling

Most of the deployed SIP devices do not support ICE whereas JSEP in RTCWeb always requires ICE. So, the interworking JSEP ICE to SIP Non-ICE has to be handled in case JSEP-SIP gateway. This document does not mandate the mechanism for this interworking but provides one of the possible solutions wherein ICE exchange is handled by JSEP-SIP GW and here how JSEP-SIP Gateway identify SIP UA does not support ICE is outside the scope of this document.
Here, RE-INVITE is not triggered for OFFER2 as the update is related to the selected candidate pair only and there is no other SDP change.

7. Security Considerations

The security consideration of SIP UA and JSEP MUST be considered for the interworking functionality. There is no need of any other extra security considerations for this document.

8. IANA Considerations

There is no IANA considerations for this draft.

9. Acknowledgement

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10. References

10.1. Normative References


10.2. Informative References


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