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Interworking between the Session Initiation Protocol (SIP) and the Extensible Messaging and Presence Protocol (XMPP): Instant Messaging draft-ietf-stox-im-13

Abstract

This document defines a bidirectional protocol mapping for the exchange of single instant messages between the Session Initiation Protocol (SIP) and the Extensible Messaging and Presence Protocol (XMPP).

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1. Introduction

In order to help ensure interworking between instant messaging (IM) systems that conform to the instant messaging / presence requirements [RFC2779], it is important to clearly define protocol mappings between such systems. Within the IETF, work has proceeded on two instant messaging technologies:

- Various extensions to the Session Initiation Protocol ([RFC3261]) for instant messaging, in particular the MESSAGE method extension [RFC3428].

- The Extensible Messaging and Presence Protocol (XMPP), which consists of a formalization of the core XML streaming protocols developed originally by the Jabber open-source community; the relevant specifications are [RFC6120] for the XML streaming layer and [RFC6121] for basic presence and instant messaging extensions.

One approach to helping ensure interworking between these protocols is to map each protocol to the abstract semantics described in [RFC3860]; that is the approach taken by [I-D.ietf-simple-cpim-mapping] and [RFC3922]. By contrast, the approach taken in this document is to directly map semantics from one protocol to another (i.e., from SIP/SIMPLE to XMPP and vice-versa), since that is how existing systems solve the interworking problem.

Both XMPP and IM-capable SIP systems enable entities to exchange "instant messages". The term "instant message" usually refers to a

message sent between two entities for delivery in close to real time (rather than a message that is stored and forwarded to the intended recipient upon request). This document covers single messages only (sometimes called "page-mode" messaging), since they form the lowest common denominator for IM. Separate documents cover "session-mode" instant messaging in the form of one-to-one chat sessions [I-D.ietf-stox-chat], as well as multi-party chat sessions [I-D.ietf-stox-groupchat]. In particular, session-mode instant messaging supports several features that are not part of page-mode instant messaging, such as a higher level of assurance regarding end-to-end message delivery.

The architectural assumptions underlying such direct mappings are provided in [RFC7247], including mapping of addresses and error conditions. The mappings specified in this document cover basic instant messaging functionality, i.e., the exchange of a single instant message between a SIP user and an XMPP user in either direction. Mapping of more advanced functionality is out of scope for this document, but other documents in this "series" cover such topics.

2. Intended Audience

The documents in this series are intended for use by software developers who have an existing system based on one of these technologies (e.g., SIP), and would like to enable communication from that existing system to systems based on the other technology (e.g., XMPP). We assume that readers are familiar with the core specifications for both SIP [RFC3261] and XMPP [RFC6120], with the base document for this series [RFC7247], and with the following IM-related specifications:

- Session Initiation Protocol (SIP) Extension for Instant Messaging [RFC3428]

- Extensible Messaging and Presence Protocol: Instant Messaging and Presence [RFC6121]

Note well that not all protocol-compliant messages are shown (such as SIP 100 TRYING messages), in order to focus the reader on the essential aspects of the protocol flows.

3. Terminology

A number of terms used here are explained in [RFC3261], [RFC3428], [RFC6120], and [RFC6121].

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119].

4. XMPP to SIP

As described in [RFC6121], a single instant message is an XML <message/> stanza of type "normal" sent over an XML stream (since "normal" is the default for the 'type' attribute of the <message/> stanza, the attribute is often omitted).

When the XMPP user Juliet <juliet@example.com> wants to send an instant message to Romeo, she interacts with her XMPP client, which generates an XMPP <message/> stanza. The syntax of the <message/> stanza, including required and optional elements and attributes, is defined in [RFC6121] (for single instant messages, Section 5.1 of [RFC6121] recommends that the value of the 'to' address be a "bare JID" of the form "localpart@domainpart"). The following is an example of such a stanza:

Example 1: XMPP user sends message

```
|  <message from='juliet@example.com/yn0cl4bnw0yr3vym'
|           to='romeo@example.net'>
|    <body>Art thou not Romeo, and a Montague?</body>
|  </message>
```

Upon receiving such a message stanza, the XMPP server needs to determine the identity of the domainpart in the 'to' address, which it does by following the procedures explained in Section 5 of [RFC7247]. If the domain is a SIP domain, the XMPP server will hand off the message stanza to an XMPP-to-SIP gateway that natively communicates with IM-aware SIP servers.

The XMPP-to-SIP gateway is then responsible for translating the XMPP message stanza into a SIP MESSAGE request from the XMPP user to the SIP user:

Example 2: XMPP user sends message (SIP transformation)

```
|  MESSAGE sip:romeo@example.net SIP/2.0
|  Via: SIP/2.0/TCP x2s.example.com;branch=z9hG4bK776sgdkse
|  Max-Forwards: 70
|  To: sip:romeo@example.net
|  From: <sip:juliet@example.com;gr=yn0cl4bnw0yr3vym>;tag=12345
|  Call-ID: D9AA95FD-2BD5-46E2-AF0F-6CFAA96BDDFA
|  CSeq: 1 MESSAGE
|  Content-Type: text/plain
|  Content-Length: 35
|
|  Art thou not Romeo, and a Montague?
```

The destination SIP server is responsible for delivering the message to the intended recipient, and the recipient is responsible for generating a response (e.g., 200 OK).

Example 3: SIP user agent indicates receipt of message

```
|  SIP/2.0 200 OK
|  Via: SIP/2.0/TCP x2s.example.com;branch=z9hG4bK776sgdkse
|  From: sip:juliet@example.com;tag=12345
|  To: sip:romeo@example.net;tag=vwxyz
|  Call-ID: D9AA95FD-2BD5-46E2-AF0F-6CFAA96BDDFA
|  CSeq: 1 MESSAGE
|  Content-Length: 0
```

As described in [RFC3428], a downstream proxy could fork a MESSAGE request, but it would return only one 200 OK to the gateway.

> Note: This document does not specify handling of the 200 OK by the XMPP-to-SIP gateway (e.g., to enable message acknowledgements). See [I-D.ietf-stox-chat] for a mapping of message acknowledgements in the context of one-to-one chat sessions.

The mapping of XMPP syntax to SIP syntax MUST be as shown in the following table.

Table 1: Message syntax mapping from XMPP to SIP

XMPP Element or Attribute	SIP Header or Contents
<body/>	body of MESSAGE
<subject/>	Subject
<thread/>	Call-ID
from	From (1)
id	transaction identifer
to	To or Request-URI
type	(no mapping) (2)
xml:lang	Content-Language

1. As shown in the foregoing example and described in [RFC7247], the XMPP-to-SIP gateway MUST map the bare JID ("localpart@domainpart") of the XMPP sender to the SIP From header and include the resourcepart of the full JID as the GRUU portion [RFC5627] of the SIP URI.

2. Because there is no SIP header field that matches the meaning of the XMPP message 'type' values ("normal", "chat", "groupchat", "headline", "error"), no general mapping is possible here.

5. SIP to XMPP

As described in [RFC3428], a single instant message is a SIP MESSAGE request sent from a SIP user agent to an intended recipient who is most generally referenced by an Instant Message URI of the form <im:user@domain> but who might be referenced by a SIP or SIPS URI of the form <sip:user@domain> or <sips:user@domain>.

When the SIP user Romeo <sip:romeo@example.net> wants to send an instant message to Juliet, he interacts with his SIP user agent, which generates a SIP MESSAGE request. The syntax of the MESSAGE request is defined in [RFC3428]. The following is an example of such a request:

Example 4: SIP user sends message

```
|  MESSAGE sip:juliet@example.com SIP/2.0
|  Via: SIP/2.0/TCP s2x.example.net;branch=z9hG4bKeskdgs677
|  Max-Forwards: 70
|  To: sip:juliet@example.com
|  From: sip:romeo@example.net;tag=vwxyz
|  Call-ID: 9E97FB43-85F4-4A00-8751-1124FD4C7B2E
|  CSeq: 1 MESSAGE
|  Content-Type: text/plain
|  Content-Length: 44
|
|  Neither, fair saint, if either thee dislike.
```

Section 5 of [RFC3428] stipulates that a SIP User Agent presented with an im: URI should resolve it to a sip: or sips: URI. Therefore we assume that the Request-URI of a request received by an IM-capable SIP-to-XMPP gateway will contain a sip: or sips: URI. Upon receiving the MESSAGE, the SIP server needs to determine the identity of the domain portion of the Request-URI or To header, which it does by following the procedures explained in Section 5 of [RFC7247]. If the domain is an XMPP domain, the SIP server will hand off the MESSAGE to an associated SIP-to-XMPP gateway that natively communicates with XMPP servers.

The SIP-to-XMPP gateway is then responsible for translating the request into an XMPP message stanza from the SIP user to the XMPP user and returning a SIP "200 OK" message to the sender:

Example 5: SIP user sends message (XMPP transformation)

```
|  <message from='romeo@example.net/dr4hcr0st3lup4c'
|           to='juliet@example.com'>
|    <body>Neither, fair saint, if either thee dislike.</body>
|  </message>
```

Note that the stanza handling rules specified in [RFC6121] allow the receiving XMPP server to deliver a message stanza whose 'to' address is a bare JID ("localpart@domainpart") to multiple connected devices. This is similar to the "forking" of messages in SIP.

The mapping of SIP syntax to XMPP syntax MUST be as shown in the following table.

Table 2: Message syntax mapping from SIP to XMPP

SIP Header or Contents	XMPP Element or Attribute
Call-ID	<thread/>
Content-Language	xml:lang
CSeq	(no mapping)
From	from (1)
Subject	<subject/>
Request-URI or To	to
body of MESSAGE	<body/>
transaction identifier	id

1. As shown in the foregoing example and described in [RFC7247], if the IM-capable SIP-to-XMPP gateway has information about the GRUU [RFC5627] of the particular endpoint that sent the SIP message then it MUST map the sender's address to a full JID ("localpart@domainpart/resourcepart") in the 'from' attribute of the XMPP stanza and include the GRUU as the resourcepart.

When transforming SIP pager-mode messages, an IM-capable SIP-to-XMPP gateway MUST specify no XMPP 'type' attribute or, equivalently, a 'type' attribute whose value is "normal" [RFC6121].

See Section 7 of this document about the handling of SIP message bodies that contain content types other than plain text.

6. Message Size

[RFC3428] specifies that (outside of a media session) the size of a MESSAGE request is not allowed to exceed 1300 bytes. Although in practice XMPP instant messages do not often exceed that size, neither [RFC6120] nor [RFC6121] sets an upper limit on the size of XMPP stanzas. However, XMPP server deployments usually do limit the size of stanzas in order to help prevent denial of service attacks, and [RFC6120] states that if a server sets a maximum stanza size then the limit is not allowed to be less than 10,000 bytes. Because of this mismatch, an XMPP-to-SIP gateway SHOULD return a <policy-violation/> stanza error if an XMPP user attempts to send an XMPP message stanza that would result in a SIP MESSAGE greater than 1300 bytes. Although such a gateway might decide to "upgrade" from page mode to session mode using the Message Session Relay Protocol (MSRP) and thus treating the instant message as part of a chat session as described in [I-D.ietf-stox-chat], such behavior is application-specific and this document provides no guidelines for how to complete such an upgrade.

7. Content Types

SIP requests of type MESSAGE are allowed to contain essentially any content type. The recommended procedures for SIP-to-XMPP gateways to use in handling these content types are as follows.

An IM-aware SIP-to-XMPP gateway MUST process SIP messages that contain message bodies of type "text/plain" and MUST encapsulate such message bodies as the XML character data of the XMPP <body/> element.

An IM-aware SIP-to-XMPP gateway SHOULD process SIP messages that contain message bodies of type "text/html"; if so, a gateway MUST transform the "text/html" content into XHTML content that conforms to the XHTML-IM Integration Set specified in [XEP-0071].

Although an IM-aware SIP-to-XMPP gateway MAY process SIP messages that contain message bodies of types other than "text/plain" and "text/html", the handling of such content types is a matter of implementation.

8. Internationalization Considerations

Both XMPP and SIP support the UTF-8 encoding [RFC3629] of Unicode characters [UNICODE] within messages, along with tagging of the language for a particular message (in XMPP via the 'xml:lang' attribute and in SIP via the Content-Language header). Gateways MUST map these language tagging mechanisms if they are present in the original message. Several examples follow, using the "XML Notation" [RFC3987] for Unicode characters outside the ASCII range.

Example 6: SIP user sends message

```
|  MESSAGE sip:juliet@example.com SIP/2.0
|  Via: SIP/2.0/TCP s2x.example.net;branch=z9hG4bKeskdgs677
|  Max-Forwards: 70
|  To: sip:juliet@example.com
|  From: sip:romeo@example.net;tag=vwxyz
|  Call-ID: 5A37A65D-304B-470A-B718-3F3E6770ACAF
|  CSeq: 1 MESSAGE
|  Content-Type: text/plain
|  Content-Length: 45
|  Content-Language: cs
|
|  Nic z ob&#xC3A9;ho, m&#xC3A1; d&#xC49B;vo spanil&#xC3A1;,
|  nenavid&#xC3AD;&#xC5A1;-li jedno nebo druh&#xC3A9;.
```

Example 7: SIP user sends message (XMPP transformation)

```
|  <message from='romeo@example.net'
|           to='juliet@example.com'
|           xml:lang='cs'>
|    <body>
|  Nic z ob&#xC3A9;ho, m&#xC3A1; d&#xC49B;vo spanil&#xC3A1;,
|  nenavid&#xC3AD;&#xC5A1;-li jedno nebo druh&#xC3A9;.
|    </body>
|  </message>
```

9. IANA Considerations

This document requests no actions of IANA.

10. Security Considerations

Detailed security considerations for instant messaging protocols are given in [RFC2779], for SIP-based instant messaging in [RFC3428] (see also [RFC3261]), and for XMPP-based instant messaging in [RFC6121] (see also [RFC6120]). The security considerations provided in [RFC7247] also apply.

This document specifies methods for exchanging "page-mode" instant messages through a gateway that translates between SIP and XMPP, and [I-D.ietf-stox-chat] specifies such methods for "session-mode" instant messaging between MSRP and XMPP. Such a gateway MUST be compliant with the minimum security requirements of the textual chat protocols for which it translates (i.e., SIP or MSRP and XMPP).

The addition of gateways to the security model of instant messaging specified in [RFC2779] introduces some new risks. In particular, end-to-end security properties (especially confidentiality and integrity) between instant messaging clients that interface through a gateway can be provided only if common formats are supported. Specification of those common formats is out of scope for this document. For instant messages, it is possible to use [RFC3862] and [RFC3923], but those methods are not widely implemented. A more widely implemented albeit unstandardized method for interoperable end-to-end encryption would be Off-the-Record Messaging [OTR].

11. References

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