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P. Saint-Andre
&yet
A. Houri
IBM
J. Hildebrand
Cisco Systems, Inc.
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Interworking between the Session Initiation Protocol (SIP) and the Extensible Messaging and Presence Protocol (XMPP): Instant Messaging
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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:

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

o 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).

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 "pager-mode" messaging), since they form the lowest common denominator for IM. Separate documents cover one-to-one chat sessions [I-D.ietf-stox-chat] and multi-party groupchat [I-D.ietf-stox-groupchat].

The architectural assumptions underlying such direct mappings are provided in [I-D.ietf-stox-core], 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 [I-D.ietf-stox-core], and with the following IM-related specifications:

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

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

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). In this document we will

assume that such a message is sent from an XMPP client to an XMPP server over an XML stream negotiated between the client and the server, and that the client is controlled by a human user (this is a simplifying assumption introduced for explanatory purposes only; the XMPP sender could be an automated client, a component such as a workflow application, a server, etc.).

When Juliet 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, the value of the 'to' address SHOULD 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/balcony'
|           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 [I-D.ietf-stox-core]. If the domain is a SIP domain, the XMPP server will hand off the message stanza to an XMPP-to-SIP gateway or connection manager that natively communicates with IM-aware SIP servers.

The XMPP-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=balcony>;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:romeo@example.net;tag=vwxyz
|  To: sip:juliet@example.com;tag=12345
|  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.

Informational Note: This document does not specify handling of the 200 OK by the XMPP-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 SHOULD be as shown in the following table. (Mappings for several aspects not mentioned here are specified in [I-D.ietf-stox-chat].)

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	(no mapping)
to	To or Request-URI
type	(no mapping) (2)
xml:lang	Content-Language

1. As shown in the foregoing example and described in [I-D.ietf-stox-core], the XMPP-SIP gateway SHOULD map the full JID (localpart@domainpart/resourcepart) of the XMPP sender to the SIP From header and include the resourcepart 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>. Here again we introduce the simplifying assumption that the user agent is controlled by a human user, whom we shall dub <romeo@example.net>.

When Romeo 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-XMPP gateway will contain a sip: or sips: URI. Upon receiving the MESSAGE, the SIP (MSRP) 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 [I-D.ietf-stox-core]. If the domain is an XMPP domain, the SIP server will hand off the MESSAGE to an associated SIP-XMPP gateway or connection manager 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/orchard'
|           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 SHOULD be as shown in the following table. (Mappings for several aspects not mentioned here are specified in [I-D.ietf-stox-chat].)

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/>

1. As shown in the foregoing example and described in [I-D.ietf-stox-core], if the IM-capable SIP-XMPP gateway has information about the GRUU [RFC5627] of the particular endpoint that sent the SIP message then it SHOULD 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-XMPP gateway SHOULD specify no XMPP 'type' attribute or, equivalently, a 'type' attribute whose value is "normal" [RFC6121].

See Section 6 of this document about the handling of SIP message bodies that contain content types other than plain text.

6. 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.

7. Internationalization Considerations

Both XMPP and SIP support the UTF-8 encoding [RFC3629] of Unicode characters [UNICODE] within messages, and signalling of the language for a particular message (in XMPP via the 'xml:lang' attribute and in SIP via the Content-Language header). Several examples follow, using the "XML Notation" for Unicode characters outside the ASCII range described in [RFC3987].

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: 9E97FB43-85F4-4A00-8751-1124FD4C7B2E
|  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>
```

8. IANA Considerations

This document requests no actions of IANA.

9. 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 [I-D.ietf-stox-core] also apply.

This document specifies methods for exchanging instant messages through a gateway that translates between SIP and XMPP. Such a gateway MUST be compliant with the minimum security requirements of the instant messaging protocols for which it translates (i.e., SIP 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 user agents that interface through an IM-capable SIP-XMPP gateway can be provided only if common formats are supported. Specification of those common formats is out of scope for this document, although it is preferred to use [RFC3862] for instant messages.

10. References

10.1. Normative References

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10.2. Informative References

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Appendix A. Acknowledgements

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Authors' Addresses

Peter Saint-Andre
&yet
P.O. Box 787
Parker, CO 80134
USA

Email: ietf@stpeter.im

Avshalom Houri
IBM
Rorberg Building, Pekris 3
Rehovot 76123
Israel

Email: avshalom@il.ibm.com

Joe Hildebrand
Cisco Systems, Inc.
1899 Wynkoop Street, Suite 600
Denver, CO 80202
USA

Email: jhildebr@cisco.com