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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-01**

**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 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, as developed within the SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE) Working Group; the relevant specification for instant messaging is [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-aware SIP systems enable entities to exchange "instant messages". The term "instant message" usually refers to messages sent between two entities for delivery in close to real time (rather than messages that are 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 instant messaging. One-to-one chat sessions and multi-party groupchat are covered in separate documents.

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.

The discussion venue for this document is the mailing list of the STOX WG; visit <https://www.ietf.org/mailman/listinfo/stox> for subscription information and discussion archives.



## 2. 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 [\[RFC2119\]](#).

## 3. 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 a bot-controlled client, a component such as a workflow application, a server, etc.). Continuing the tradition of Shakespeare examples in XMPP documentation, we will say that the XMPP user has an XMPP address of <juliet@example.com>.

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/resourcepart"). The following is an example of such a stanza:

Example: 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 stanza, the XMPP server to which Juliet has connected either delivers it to a local recipient (if the hostname in the 'to' attribute matches one of the hostnames serviced by the XMPP server) or attempts to route it to the foreign domain that services the hostname in the 'to' attribute. Naturally, in this document we assume that the hostname in the 'to' attribute is an IM-aware SIP service hosted by a separate server. As specified in [\[RFC6121\]](#), the



XMPP server needs to determine the identity of the foreign domain, which it does by performing one or more DNS SRV lookups [RFC2782]. For message stanzas, the order of lookups recommended by [RFC6121] is to first try the "\_xmpp-server" service as specified in [RFC6120] and to then try the "\_im" service as specified in [RFC3861]. Here we assume that the first lookup will fail but that the second lookup will succeed and return a resolution "\_im.\_simple.example.net.", since we have already assumed that the example.net hostname is running a SIP instant messaging service. (Note: The XMPP server may have previously determined that the foreign domain is a SIMPLE server, in which case it would not need to perform the SRV lookups; the caching of such information is a matter of implementation and local service policy, and is therefore out of scope for this document.)

Once the XMPP server has determined that the foreign domain is serviced by a SIMPLE server, it must determine how to proceed. We here assume that the XMPP server contains or has available to it an XMPP-SIMPLE gateway (such an architecture is described in [I-D.ietf-stox-core]). The XMPP server would then deliver the message stanza to the XMPP-SIMPLE gateway.

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

Example: 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
| Call-ID: Hr0zny9l3@example.com
| CSeq: 1 MESSAGE
| Content-Type: text/plain
| Content-Length: 35
|
| Art thou not Romeo, and a Montague?
```

The mapping of XMPP syntax elements to SIP syntax elements SHOULD be as shown in the following table. (Mappings for elements not mentioned are undefined.)



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
type	(no mapping)
xml:lang	Content-Language

1. As shown in the foregoing example and described in [\[I-D.ietf-stox-core\]](#), the XMPP-SIMPLE gateway SHOULD map the full JID (localpart@domainpart/resourcepart) of the XMPP sender to the SIP From header and include the resourcepart to the GRUU portion of the SIP URI [\[RFC5627\]](#).

#### 4. 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: SIP user sends message:

```
| MESSAGE sip:juliet@example.com SIP/2.0
| Via: SIP/2.0/TCP s2x.example.net;branch=z9hG4bKeskdg677
| Max-Forwards: 70
| To: sip:juliet@example.com
| From: sip:romeo@example.net
| Call-ID: M4spr4vdu@example.net
| 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 a SIMPLE-XMPP gateway will contain a sip: or sips: URI. The gateway SHOULD resolve that address to an im: URI for SIP MESSAGE requests, then follow the rules in [\[RFC3861\]](#) regarding the "\_im" SRV service for the target domain contained in the Request-URI. If SRV address resolution fails for the "\_im" service, the gateway MUST either attempt a lookup for the "\_xmpp-server" service as specified in [\[RFC6120\]](#) or return an error to the sender (the SIP "502 Bad Gateway" error seems most appropriate; see [\[I-D.ietf-stox-core\]](#) for details). If SRV address resolution succeeds, the gateway is 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: 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>
```

The mapping of SIP syntax elements to XMPP syntax elements SHOULD be as shown in the following table. (Mappings for elements not mentioned in the foregoing table are undefined.)



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	to
body of MESSAGE	<body/>

- As shown in the foregoing example and described in [\[I-D.ietf-stox-core\]](#), if the SIMPLE-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, a SIMPLE-XMPP gateway SHOULD specify no XMPP 'type' attribute or, equivalently, a 'type' attribute whose value is "normal" [\[RFC6121\]](#).

See [Section 5](#) of this document about the handling of SIP message bodies that contain content types other than plain text.

## 5. Content Types

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

A SIMPLE-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.

A SIMPLE-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 1.0 Integration Set specified in [\[XEP-0071\]](#).

Although a SIMPLE-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.



## 6. 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]).

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 a SIMPLE-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.

## 7. IANA Considerations

This document requests no actions of IANA.

## 8. References

### 8.1. Normative References

- [I-D.ietf-stox-core]  
Saint-Andre, P., Houri, A., and J. Hildebrand,  
"Interworking between the Session Initiation Protocol  
(SIP) and the Extensible Messaging and Presence Protocol  
(XMPP): Core", [draft-ietf-stox-core-02](#) (work in progress),  
August 2013.
- [RFC2119] Bradner, S., "Key words for use in RFCs to Indicate  
Requirement Levels", [BCP 14](#), [RFC 2119](#), March 1997.
- [RFC2782] Gulbrandsen, A., Vixie, P., and L. Esibov, "A DNS RR for  
specifying the location of services (DNS SRV)", [RFC 2782](#),  
February 2000.
- [RFC3261] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston,  
A., Peterson, J., Sparks, R., Handley, M., and E.  
Schooler, "SIP: Session Initiation Protocol", [RFC 3261](#),  
June 2002.



- [RFC3428] Campbell, B., Rosenberg, J., Schulzrinne, H., Huitema, C., and D. Gurle, "Session Initiation Protocol (SIP) Extension for Instant Messaging", [RFC 3428](#), December 2002.
- [RFC3861] Peterson, J., "Address Resolution for Instant Messaging and Presence", [RFC 3861](#), August 2004.
- [RFC5627] Rosenberg, J., "Obtaining and Using Globally Routable User Agent URIs (GRUUs) in the Session Initiation Protocol (SIP)", [RFC 5627](#), October 2009.
- [RFC6120] Saint-Andre, P., "Extensible Messaging and Presence Protocol (XMPP): Core", [RFC 6120](#), March 2011.
- [RFC6121] Saint-Andre, P., "Extensible Messaging and Presence Protocol (XMPP): Instant Messaging and Presence", [RFC 6121](#), March 2011.
- [XEP-0071] Saint-Andre, P., "XHTML-IM", XSF XEP 0071, November 2012.

## **[8.2. Informative References](#)**

- [I-D.ietf-simple-cpim-mapping] Rosenberg, J. and B. Campbell, "CPIM Mapping of SIMPLE Presence and Instant Messaging", [draft-ietf-simple-cpim-mapping-01](#) (work in progress), June 2002.
- [RFC2779] Day, M., Aggarwal, S., and J. Vincent, "Instant Messaging / Presence Protocol Requirements", [RFC 2779](#), February 2000.
- [RFC3860] Peterson, J., "Common Profile for Instant Messaging (CPIM)", [RFC 3860](#), August 2004.
- [RFC3862] Klyne, G. and D. Atkins, "Common Presence and Instant Messaging (CPIM): Message Format", [RFC 3862](#), August 2004.
- [RFC3922] Saint-Andre, P., "Mapping the Extensible Messaging and Presence Protocol (XMPP) to Common Presence and Instant Messaging (CPIM)", [RFC 3922](#), October 2004.

## **[Appendix A. Acknowledgements](#)**

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