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Interworking between the Session Initiation Protocol (SIP) and the
Extensible Messaging and Presence Protocol (XMPP): One-to-One Text Chat
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Abstract

This document defines a bidirectional protocol mapping for the exchange of instant messages in the context of a one-to-one chat session between a user of the Session Initiation Protocol (SIP) and a user of the Extensible Messaging and Presence Protocol (XMPP). Specifically for SIP text chat, this document specifies a mapping to the Message Session Relay Protocol (MSRP).

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behalf of Romeo.

Example: (F4) Gateway sends ACK to Romeo's UA

```
ACK sip:juliet@example.com SIP/2.0
To: <sip:romeo@example.net>;gr=orchard
From: <sip:juliet@example.com>
Contact: <sip:juliet@example.com>;gr=balcony
Call-ID: 711609sa
```

The XMPP-to-SIP gateway then transforms the original XMPP chat message into MSRP.

Example: (F5) Gateway transforms XMPP message to MSRP

```
MSRP a786hjs2 SEND
From-Path: msrp://x2s.example.com:7654/jshA7weztas/tcp
To-Path: msrp://s2x.example.net:12763/kjhd37s2s20w2a/tcp
Message-ID: 87652491
Byte-Range: 1-25/25
Content-Type: text/plain
```

```
Art thou not Romeo, and a Montague?
-----a786hjs2$
```

Romeo can then send a reply using his MSRP user agent.

Example: (F6) Romeo sends a reply

```
MSRP a786hjs2 SEND
To-Path: msrp://x2s.example.com:7654/jshA7weztas/tcp
From-Path: msrp://s2x.example.net:12763/kjhd37s2s20w2a/tcp
Message-ID: 87652491
Byte-Range: 1-25/25
Failure-Report: no
Content-Type: text/plain
```

```
Neither, fair saint, if either thee dislike.
-----a786hjs2$
```

The SIP-to-XMPP gateway would then transform that message into appropriate XMPP syntax for routing to the intended recipient.

Example: (F7) Gateway transforms MSRP message to XMPP

```
<message from='romeo@example.net/orchard'
  to='juliet@example.com/balcony'
  type='chat'>
  <thread>711609sa</thread>
  <body>Neither, fair saint, if either thee dislike.</body>
</message>
```

When the MSRP user wishes to end the chat session, the user's MSRP client sends a SIP BYE.

Example: (F8) Romeo terminates the chat session

```
BYE juliet@example.com sip: SIP/2.0
Max-Forwards: 70
From: <sip:romeo@example.net>;tag=087js
To: <sip:juliet@example.com>;tag=786
Call-ID: 711609sa
Cseq: 1 BYE
Content-Length: 0
```

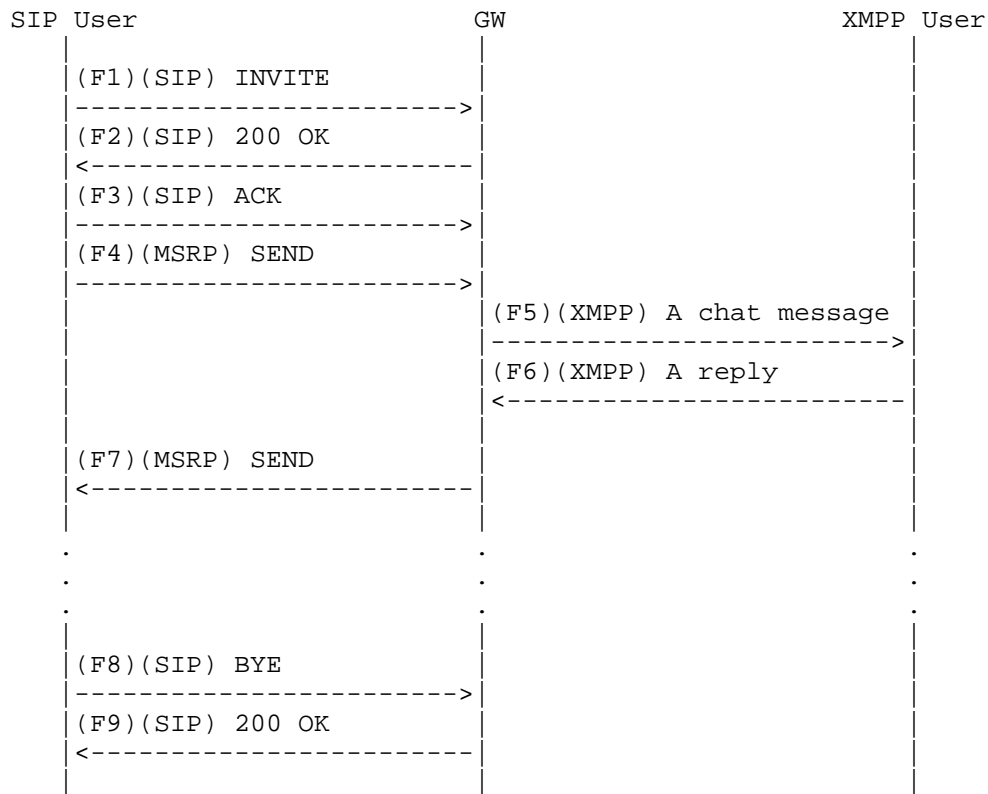
The BYE is then acknowledged by the XMPP-to-SIP gateway.

Example: (F9) Gateway acknowledges termination

```
SIP/2.0 200 OK
From: <sip:juliet@example.com>;tag=786
To: <sip:romeo@example.net>;tag=087js
Call-ID: 711609sa
CSeq: 1 BYE
Content-Length: 0
```

4. MSRP to XMPP

When an MSRP client sends messages through a gateway to an XMPP client that does not support formal sessions, the order of events is as follows.



Example: (F1) SIP user starts the session

```

INVITE sip:juliet@example.com SIP/2.0
To: <sip:juliet@example.com>
From: <sip:romeo@example.net>
Contact: <sip:romeo@example.net>;gr=orchard
Subject: Open chat with Romeo?
Call-ID: 742507no
Content-Type: application/sdp

c=IN IP4 s2x.example.net
m=message 7313 TCP/MSRP *
a=accept-types:text/plain
a=lang:en
a=lang:it
a=path:msrp://s2x.example.net:7313/ansp71weztas;tcp
  
```

Example: (F2) Gateway accepts session on Juliet's behalf

```
SIP/2.0 200 OK
To: <sip:romeo@example.net>;gr=orchard
From: <sip:juliet@example.com>
Contact: <sip:juliet@example.com>;gr=balcony
Call-ID: 742507no
Content-Type: application/sdp
```

```
c=IN IP4 x2s.example.com
m=message 8763 TCP/MSRP *
a=accept-types:text/plain
a=lang:it
a=path:msrp://x2s.example.com:8763/lkjh37s2s20w2a;tcp
```

Example: (F3) Romeo sends ACK

```
ACK sip:juliet@example.com SIP/2.0
To: <sip:juliet@example.com>;gr=balcony
From: <sip:romeo@example.net>
Contact: <sip:romeo@example.net>;gr=orchard
Call-ID: 742507no
```

Example: (F4) Romeo sends a message

```
MSRP ad49kswow SEND
To-Path: msrp://x2s.example.com:8763/lkjh37s2s20w2a;tcp
From-Path: msrp://s2x.example.net:7313/ansp7lweztas;tcp
Message-ID: 44921zaqwsx
Byte-Range: 1-32/32
Failure-Report: no
Content-Type: text/plain
```

```
I take thee at thy word ...
-----ad49kswow$
```

Example: (F5) Romeo sends a message (XMPP translation)

```
<message from='romeo@example.net'
  to='juliet@example.com'
  type='chat'>
  <thread>742507no</thread>
  <body>I take thee at thy word ...</body>
</message>
```


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.

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Appendix A. Acknowledgements

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a gateway. (This assumption not meant to discourage protocol translation within multi-protocol clients or servers; instead, this assumption is followed mainly to clarify the discussion and examples so that the protocol translation principles can be more easily understood and can be applied by client and server implementors with appropriate modifications to the examples and terminology.) Specifically, we assume that the protocol translation will occur within an "XMPP-to-SIP gateway" that translates XMPP syntax and semantics on behalf of an XMPP service when communicating with SIP services and/or within a "SIP-to-XMPP gateway" that translates SIP syntax and semantics on behalf of a SIP service when communicating with XMPP services (naturally, these logical functions could occur in one and the same actual translator).

This document assumes that a gateway will translate directly from one protocol to the other. For the sake of the examples, we further assume that protocol translation will occur within a gateway in the source domain, so that information generated by the user of an XMPP service will be translated by a gateway within the trust domain of that XMPP service, and information generated by the user of a SIP service will be translated by a gateway within the trust domain of that SIP service. However, nothing in this document ought to be taken as recommending against protocol translation at the destination domain.

An architectural diagram for a possible gateway deployment is shown below, where the entities have the following significance and the "#" character is used to show the boundary of a trust domain:

- o romeo@example.net -- a SIP user.
- o example.net -- a SIP service with a gateway ("GW") to XMPP.
- o juliet@example.com -- an XMPP user.
- o example.com -- an XMPP service with a gateway ("GW") to SIP.

```

#####
#                                     #
#           +-----+-----+       +-----+-----+       #
#           | example.net | GW |---#---| GW | example.com |   #
#           +-----+-----+       +-----+-----+       #
#           |                                     |             #
#           romeo@example.net                   juliet@example.com #
#                                     #
#####

```


3. Translate "\26" to "&", "\27" to "'", and "\2f" to "/" respectively (this is consistent with [XEP-0106]).
4. Determine if the foreign domain supports im: and pres: URIs (discovered via [RFC2782] lookup as specified in [RFC6121]), else assume that the foreign domain supports sip:/sips: URIs.
5. If converting into im: or pres: URI, for each byte, if the byte is in the set (), ., ; [\] or is a UTF-8 character outside the ASCII range then percent-encode that byte to "%hexhex" format. If converting into sip: or sips: URI, for each byte, if the byte is in the set # [\] ^ ` { | } or is a UTF-8 character outside the ASCII range then percent-encode that byte to "%hexhex" format.
6. Combine resulting local part with mapped hostname to form local@domain address.
7. Prepend with 'im:' scheme (for XMPP <message/> stanzas) or 'pres:' scheme (for XMPP <presence/> stanzas) if foreign domain supports these, else prepend with 'sip:' or 'sips:' scheme according to local service policy.
8. If the XMPP address included a resourcepart and the destination URI scheme is 'sip:' or 'sips:', optionally append the slash character '/' and then append the resourcepart (making sure to percent-encode any UTF-8 characters outside the ASCII range) as the "gr" URI parameter.

5. Error Condition Mapping

SIP response codes are specified in [RFC3261] and XMPP error conditions are specified in [RFC6120]. Because there is no equivalent in XMPP for the provisional (1xx) and successful (2xx) response codes in SIP, mappings are provided only for the redirection (3xx), request failure (4xx), server failure (5xx), and global failure (6xx) codes.

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Example: (F7) Chatroom presence information translated into XMPP

```
<presence from='verona@chat.example.org/Romeo'
  to='romeo@example.com'>
  <x xmlns='http://jabber.org/protocol/muc#user'>
    <item affiliation='none' role='participant' />
  </x>
</presence>

<presence from='verona@chat.example.org/Ben'
  to='romeo@example.com'>
  <x xmlns='http://jabber.org/protocol/muc#user'>
    <item affiliation='none' role='participant' />
  </x>
</presence>
```

Upon receiving these presence stanzas, if the MSRP conference server has already completed the subscription to the Conference Event package [RFC4575], the XMPP-to-SIP gateway MUST translate them in a SIP NOTIFY request containing the participant list (represented in the [RFC4575] format).

- [RFC4975] Campbell, B., Mahy, R., and C. Jennings, "The Message Session Relay Protocol (MSRP)", RFC 4975, September 2007.
- [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.
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Appendix A. Acknowledgements

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Some text in this document was borrowed from [I-D.ietf-stox-core] and from [XEP-0045].

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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.

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presence 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 presence 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 "_pres" 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 "_pres.simple.example.net.", since we have already assumed that the example.net hostname is running a SIP presence service. (Note: The XMPP server might have previously determined that the foreign domain is a SIMPLE server, e.g., when it sent a SIP SUBSCRIBE to the SIP user when Juliet sent initial presence to the XMPP 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 needs to determine how to proceed. We here assume that the XMPP server contains or has available to it an XMPP-SIMPLE gateway. The XMPP server would then deliver the presence stanza to the XMPP-SIMPLE gateway.

The XMPP-SIMPLE gateway is then responsible for translating the XMPP presence stanza into a SIP NOTIFY request and included PIDF document from the XMPP user to the SIP user.

server associated with the XMPP-SIMPLE gateway as described above).

6. IANA Considerations

This document requests no actions of IANA.

7. References

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