Transporting User to User Call Control Information in SIP for ISDN Interworking
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Abstract

Several approaches to transporting the ITU-T Q.931 User to User Information Element (UU IE) data in SIP have been proposed. As networks move to SIP it is important that applications requiring this data can continue to function in SIP networks as well as the ability to interwork the information to/from ISDN for end-to-end transparency. This extension will also be used for native SIP endpoints implementing similar services and interworking with ISDN services. This document discusses requirements and approaches and recommends a new header field User-to-User be standardized. Example use cases include an exchange between two User Agents, retargeting by a proxy, and redirection. An example application is in an Automatic
Call Distributor (ACD) in a contact center.

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1. Overview

This document describes the transport of User to User Information (UUI) in ISDN interworking scenarios using SIP [RFC3261]. Specifically, we discuss the transport of call control related ITU-T Q.931 User to User Information Element (UU IE) [Q931] data in SIP. Q.931 UUI is widely used in the PSTN today in contact center and call centers which are transitioning away from ISDN to SIP. This extension will also be used for native SIP endpoints implementing similar services and interworking with ISDN services. Five different use case call flows are discussed.

It is important to note that this extension will not be used for transport of information where the content is known and a specific SIP header field exists for carrying this information. For cases where the content of user to user information is relevant to SIP, it is expected a new SIP header field will be created to transport the information. For example, location information is an example of user to user information, and obviously it needs to be transported in a specifically named SIP header rather than transported using this generic mechanism. It is also important to note that standardization of the UUI content is explicitly out of scope for this specification.

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 BCP 14, RFC 2119 [RFC2119].

3. Requirements

This section discusses the requirements for the transport of call control related user to user information (UUI). We define call control UUI as information that is generated, transported, and consumed at the time of call setup (i.e. during a pending INVITE transaction). The information can be used for call routing, alerting, call distribution, or simply rendering. The exact usage and semantics of call control UUI is out of scope - SIP is simply providing the transport function for this that ISDN Provides in the PSTN. Non-call control UUI can be sent using the INFO method and not using the extensions described in this specification.

REQ-1: The mechanism will allow UAs to insert and receive ITU-T Q.931 User to User Information Element (referred to as UUI) data in SIP call setup requests and responses.
SIP messages covered by this include INVITE requests and end-to-end responses to the INVITE, which includes 18x and 200 responses.

REQ-2: The mechanism will allow UAs to insert and receive ITU-T Q.931 User to User Information Element (referred to as UUI) data in SIP call teardown requests and responses.

Q.931 UUI supports inclusion in release and release completion messages. SIP messages covered by this include BYE and 200 OK responses to a BYE.

REQ-3: The mechanism will allow UUI to be inserted and retrieved in SIP redirects to INVITEs.

SIP messages covered by this include 3xx responses to INVITE and REFER requests.

REQ-4: The mechanism will allow UUI to be able to survive proxy retargeting.

Retargeting is a common method of call routing in SIP, and must not result in the loss of user to user information.

REQ-5: The mechanism will allow the UUI to be transported directly in the SIP message.

Passing a pointer or link to the UUI information will not meet the real-time processing considerations.

REQ-6: The mechanism will minimize reliance on SIP extensions or uncommon SIP behavior.

REQ-7: The mechanism will allow the inserter of UUI to be sure that the recipient understands the call control UUI mechanism.

Understanding the mechanism means that the UAS will extract and utilize the UUI information transported. Understanding the format and nature of the actual UUI data is not covered by this requirement.

4. Use Cases

This section discusses four uses cases for the transport of call control related user to user information. What is not discussed here is the transport of non-call control UUI which can be done using the SIP INFO method. These use cases help explain the requirements from the previous section.
4.1. User Agent to User Agent

In this scenario, the originator UA includes UUI in the INVITE sent through a proxy to the terminating UA. The terminator can use the UUI in any way. If it is an ISDN gateway, it could map the UUI into the appropriate Q.931 element. Alternatively, it might render the information to the user, or use it for alerting or as a lookup for a screen pop. In this case, the proxy does not need to understand the UUI mechanism, but normal proxy rules should result in the UUI being forwarded without modification. This call flow is shown in Figure 1.

```
Originator | Proxy | Terminator
-------------|-------|-------------
INVITE (UUI) F1 --------------------------------->  INVITE (UUI) F2
          100 Trying F3 --------------------------------->
          200 OK F5 -------------------------------------->
          ACK F6 -------------------------------------->
          200 OK F4 -------------------------------------->
          ACK F7 -------------------------------------->

Figure 1. Call flow with UUI exchanged between Originator and Terminator.
```

This call flow utilizes REQ-1.

4.2. Proxy Retargeting

In this scenario, the originator UA includes UUI in the INVITE sent through a proxy to the terminating UA. The proxy retargets the INVITE, sending it to a different termination UA. The UUI information is then processed by the terminating UA. This call flow is shown in Figure 2.

```
Originator | Proxy | Terminator 2
-------------|------|-------------
INVITE (UUI) F1 --------------------------------->  INVITE (UUI) F2
          100 Trying F3 --------------------------------->
          200 OK F5 -------------------------------------->
          ACK F6 -------------------------------------->
          200 OK F4 -------------------------------------->
          ACK F7 -------------------------------------->

Figure 2. Call flow with UUI exchanged between Originator and Terminator 2.
```
Figure 2. Call flow with Proxy Retargeting.

This call flow utilizes REQ-1 and REQ-4.

4.3. Redirection

In this scenario, UUI is inserted by a redirect server which is then included in the INVITE sent by the Originator to the Terminator. In this case, the Originator does not necessarily need to support the UUI mechanism but does need to support the SIP redirection mechanism used to include the UUI information.

```
<table>
<thead>
<tr>
<th>Originator</th>
<th>Redirect Server</th>
<th>Terminator</th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE F1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>-------------------</td>
<td>------------------------</td>
<td>------------</td>
</tr>
<tr>
<td>302 Moved (UUI) F2</td>
<td></td>
<td></td>
</tr>
<tr>
<td>ACK F3</td>
<td></td>
<td></td>
</tr>
<tr>
<td>INVITE (UUI) F4</td>
<td></td>
<td></td>
</tr>
<tr>
<td>200 OK F5</td>
<td></td>
<td></td>
</tr>
<tr>
<td>ACK F6</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
```

Figure 3. Call flow with UUI exchanged between Redirect Server and Terminator

This call flow utilizes REQ-1 and REQ-3.

A common application of this call flow is an Automatic Call Distributer (ACD) in a PSTN contact center. The originator would be a PSTN gateway. The ACD would act as a Redirect Server, inserting UUI based on called number, calling number, time of day, and other information. The resulting UUI would be passed to the agent’s handset which acts as the Terminator. The UUI could be used to lookup information rendered to the agent at the time of call answering.

4.4. Referral

In this scenario, a Referrer UA causes an INVITE to be generated between the Originator and Terminator with UUI information inserted by the Referrer UA. Note that this REFER [RFC3515] could be part of a transfer operation or it might be unrelated to an existing call, such as out-of-dialog REFER call control. In some cases, this call
flow is used in place of the redirection call flow, but where immediately upon answer, the REFER is sent. This scenario is shown in Figure 4.

<table>
<thead>
<tr>
<th>Originator</th>
<th>Referrer</th>
<th>Terminator</th>
</tr>
</thead>
<tbody>
<tr>
<td>REFER (UUI) F1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>202 Accepted F2</td>
<td></td>
<td></td>
</tr>
<tr>
<td>NOTIFY (100 Trying) F3</td>
<td></td>
<td></td>
</tr>
<tr>
<td>200 OK F4</td>
<td></td>
<td></td>
</tr>
<tr>
<td>INVITE (UUI) F5</td>
<td></td>
<td></td>
</tr>
<tr>
<td>200 OK F6</td>
<td></td>
<td></td>
</tr>
<tr>
<td>ACK F7</td>
<td></td>
<td></td>
</tr>
<tr>
<td>NOTIFY (200 OK) F8</td>
<td></td>
<td></td>
</tr>
<tr>
<td>200 OK F9</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Figure 4. Call flow with transfer after answer.

Some scenarios involving referral have been proposed to use a REFER sent during an early dialog. This NOT RECOMMENDED call flow is shown in Figure 5. This flow is not recommended due to the number of messages exchanged (due to the REFER, CANCEL, and 487 responses) and the sending of the REFER in the early dialog.
5. Possible Mechanisms

Three mechanisms for transporting UUI will be described: MIME body, URI parameter, and header field transport.

5.1. Why INFO is Not Used

Since the INFO method [RFC2976], was developed for ISUP interworking of user-to-user information, it might seem to be the logical choice here. For non-call control user-to-user information, INFO can be utilized for end to end transport. However, for transport of call control user-to-user information, INFO can not be used. As the call
flows in the previous section show, the information is related to an attempt to establish a session and must be passed with the session setup request (INVITE), responses to that INVITE, or session termination requests. As a result, it is not possible to use INFO in these cases.

5.2. MIME body Approach

One method of transport is to transport the UUI information as a MIME body. This is in keeping with the SIP-T architecture [RFC3372] in which MIME bodies are used to transport ISUP information. Since the INVITE will normally have an SDP message body, the resulting INVITE with SDP and UUI will be multipart MIME. The insertion of a UUI message body by a redirect server or in a REFER is difficult. The body would need to be encoded in the Contact URI of the 3xx response or the Refer-To URI of a REFER. For example:

<allOneLine>
Contact: <sip:+12125551212@gateway.example.com?Content-Type=application/isup&body=ZeGl9i2icVqaNVailT6F5iJ90m6mvuTS40K05M0vDk0Q4Xs>
</allOneLine>

Note that the <allOneLine> tag convention from SIP Torture Test Messages [RFC4475] is used to show that there are no line breaks in the actual message syntax.

The MIME body approach meets REQs 1-5. However, it does not meet REQ-6 as support for Multipart MIME and escaped bodies in URIs is uncommon in SIP UAs.

5.3. URI Parameter

Another proposed approach is to encode the UUI as a URI parameter into the Contact or Refer-To URI.

<allOneLine>
Contact: <sip:+12125551212@gateway.example.com;uui=ZeGl9i2icVqaNVailT6F5iJ90m6mvuTS40K05M0vDk0Q4Xs>
</allOneLine>

The resulting INVITE would contain UUI in the Request-URI of the INVITE. The URI parameter has a drawback in that a URI parameter will not survive retargeting by a proxy as shown in Figure 2. That is, if the URI is an Address of Record instead of a Contact URI, the URI parameter will not be copied over the Contact URI, resulting in the loss of the information. As a result, this approach does not meet REQ-4.
5.4. Header Field Approach

Another approach that has been proposed is to use a header field to transport the UUI information. The header field would be included in INVITE requests and responses and BYE requests and responses, and would pass transparently through proxies. For redirection, the header field would be escaped into the Contact or Refer-To URI. This is commonly supported in UAs due to call transfer use cases. As a result, the header field approach supports REQs 1-6. In order to meet REQ-7, a SIP feature tag is needed which can be included in Supported and Require header fields.

The Call-Info header field is related to the UUI information. However, there are a number of important differences:

- Call-Info is typically used for rendering to the user. While some of the UUI information may ultimately be rendered to the user, most of the UUI information will be consumed by the end device.
- Call-Info usually contains a URI pointer the information instead of the actual information itself which does not REQ-5. It could be possible to use a data URI to carry the UUI directly in this header field.

Overall, the overloading of the Call-Info header field for carrying ISUP UUI does not seem like a good idea. For these reasons, a separate header field needs to be defined, described here as User-to-User. For example, here is an example User-to-User header field from message F1 in Figure 1:

```
User-to-User: 56a390f3d2b7310023a;encoding=hex
```

For example, here is an escaped User-to-User header field from the redirection response F2 of Figure 3:

```
Contact: <sip:+12125551212@gateway.example.com?User-to-User=56a390f3d2b7310023a%3Bencoding%3Dhex>
```

The resulting INVITE F5 would contain the User-to-User header field of the previous example.

An escaped User-to-User header field from the REFER message response F1 of Figure 4:
This would result in the INVITE F4 containing the User-to-User header field of the first example.

6. Recommendation

The recommendation is to define a new SIP header field "User-to-User" to transport UUI information in ISDN interworking applications since this mechanism best supports the requirements. A SIP feature tag "uui" also needs to be defined so that it can be used in Supported and Require header fields to meet REQ-6.

The format of the UUI information is a topic of future standardization. Currently, UUI is proprietary, requiring coordinated configuration between servers. Standardizing the format or providing content tags would provide additional benefits.

7. Appendix - Syntax for UUI Header Field

Editor’s Note: Eventually this text will move to a SIP Working Group document to define the new header field.

The User-to-User header field can be present in INVITE requests and responses only and in BYE requests and responses.

Current usage is to interoperate with ISDN User to User Signaling (UUS), a supplementary service in which manufacturer specific information is transported via the codeset 0 User-user Information IE. Three services are defined: service 1, service 2, and service 3. This draft only addresses the SIP equivalent of service 1 although it could easily be expanded later to address services 2 and 3. UUS Service 1 involves user to user signaling exchanged during call setup and clearing within the following Q.931 call control messages: SETUP, ALERT, CONNECT, DISCONNECT, RELEASE, and RELEASE COMPLETE. UUS Service 2 involves user to user signaling exchanged during call establishment (between ALERT and CONNECT) via the USER INFORMATION message. This service usually has a maximum of 2 USER INFORMATION messages in each direction. UUS Service 3 involves user to user signaling exchanged on an active call via the USER INFORMATION message.

The following syntax specification uses the augmented Backus-Naur
Form (BNF) as described in RFC 2234 and extends RFC 3261.

UUI         = "User-to-User" HCOLON uui-data *(SEMI uui-param)
uui-data   = token
uui-param   = enc-param | generic-param
enc-param   = "encoding=" ("hex" | token)

Only one User-to-User header field may be present in a request or response.

OPEN ISSUE: Should we allow multiple User-to-User header fields in SIP even though ISDN does not support this? If so, we would need to provide guidance for a SIP to ISDN gateway for mapping, such as map the first header field and drop the rest.

The only defined parameter for the User-to-User header field is the encoding parameter. "encoding=hex" is used to indicate that the UUI information is encoded as hex digits per the ISDN specification. The first octet is the protocol discriminator. Other encoding methods of encoding may also be standardized.

OPEN ISSUE: Is there an advantage in extracting the protocol discriminator and encoding in a header parameter? What would a SIP UA do with this information?

Note that ISDN limits UUI to 128 octets in length. While this header field has no such limitations, transporting UUI longer than 128 octets will result in interoperability failures when interworking with ISDN.

8. IANA Considerations

8.1. Registration of Header Field

This document defines a new SIP header field named "User-to-User".

The following row shall be added to the "Header Fields" section of the SIP parameter registry:

<table>
<thead>
<tr>
<th>Header Name</th>
<th>Compact Form</th>
<th>Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>User-to-User</td>
<td></td>
<td>[RFCXXXX]</td>
</tr>
</tbody>
</table>

Editor’s Note: [RFCXXXX] should be replaced with the designation of
8.2. Registration of Header Field Parameter

This document defines a parameter for the header field defined in the preceding section. The header field "User-to-User" can contain the parameter "encoding".

The following rows shall be added to the "Header Field Parameters and Parameter Values" section of the SIP parameter registry:

<table>
<thead>
<tr>
<th>Header Field</th>
<th>Parameter Name</th>
<th>Predefined Values</th>
<th>Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>User-to-User</td>
<td>encoding</td>
<td>No</td>
<td>[RFCXXXX]</td>
</tr>
</tbody>
</table>

Editor’s Note: [RFCXXXX] should be replaced with the designation of this document.

8.3. Registration of SIP Option Tag

This specification registers a new SIP option tag, as per the guidelines in Section 27.1 of [RFC3261].

This document defines the SIP option tag "uui".

The following row has been added to the "Option Tags" section of the SIP Parameter Registry:

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>uui</td>
<td>This option tag is used to indicate that a UA supports and understands the User-to-User header field.</td>
<td>[RFCXXXX]</td>
</tr>
</tbody>
</table>

Editor’s Note: [RFCXXXX] should be replaced with the designation of this document.

9. Security Considerations

User to user information can be exchanged over SIP on a hop-by-hop or end-to-end basis. In some cases, it may carry private information that may need confidential transport and integrity protection.
Standard SIP security mechanisms can be used to secure this header field. For example, TLS transport can provide single hop confidentiality and integrity protection. For multiple hop or end-to-end confidentiality and integrity protection, S/MIME can be utilized.

Information contained in a User-to-User header field should only be trusted if it is received within a trust domain. For example, Spec-T, defined in [RFC3324] could be used to define a trust domain. When utilized by a gateway to map information to or from ISDN Q.931, appropriate policy should be applied based on the PSTN trust domain.

10. Acknowledgements

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11. Informative References


Authors’ Addresses

Alan Johnston (editor)
Avaya
St. Louis, MO 63124

Email: alan@sipstation.com

Joanne McMillen
Avaya

Email: joanne@avaya.com