Network Working Group	A. Johnston, Ed.
Internet-Draft	J. McMillen
Intended status: Standards Track	Avaya
Expires: September 6, 2010	March 05, 2010

TOC

Transporting User to User Call Control Information in SIP for ISDN Interworking
draft-johnston-sipping-cc-uui-09

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 with this ISDN service for end-to- end transparency. This document discusses three mechanisms to meet the requirements defined in the Requirements for SIP Call Control UUI document. A new SIP header field which bests meets these requirements is proposed.

Status of this Memo

Copyright Notice

Table of Contents

---

1. Overview

TOC

This document describes the transport of User to User Information (UUI) in ISDN interworking scenarios using SIP [RFC3261] (Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M., and E. Schooler, “SIP: Session Initiation Protocol,” June 2002.). Specifically, we discuss the transport of call control related ITU-T Q.931 User to User Information Element (UU IE) [Q931] (, “ITU-T Q.931 User to User Information Element (UU IE),” .) and ITU-T Q.763 User to User Information Parameter [Q763] (, “ITU-T Q.763 Signaling System No. 7 - ISDN user part formats and codes,” .) data in SIP. UUI is widely used in the PSTN today in contact centers 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.

Part of the definition of this ISDN service is that the UUI information is not known and understood by the ISDN network that transports it. This is for two reasons. Firstly, this supports a strict layering of protocols and data. Providing information and understanding of the data to the transport layer would not provide any benefits and instead could create cross layer coupling and increase the complexity of the system. Secondly, either the originator or terminator of the service might be a simple PSTN gateway designed for scalability and lowest cost. As a result, it is neither feasible nor desirable for this device to understand the information but instead the goal is to pass the information as efficiently as possible to another application which does understand the data. Both of these arguments still apply to SIP, especially when one or both endpoints are gateways.

In the future, where both endpoints are intelligent SIP user agents, it may be possible for them to understand and interpret the UUI data.

There may be some cases where the UUI information is relevant to SIP. In this case, it might be worthwhile attempting to map UUI data to an appropriate SIP header field or to standardize a new header field. However, the requirements and use cases for this are different enough from those described in this document that these two situations should be examined separately. This document looks only at the requirements and mechanisms for replicating the existing, widely used and deployed ISDN UUI service.

The requirements, scenarios, and call flows for SIP call control UUI is discussed in [johnston-dispatch-sip-cc-uui] (Johnston, A. and J. McMillen, “Requirements for Transporting User to User Call Control Information in SIP for ISDN Interworking,” .). All references to requirement numbers (REQ-N) and figure numbers refer to this draft.

---

2. Terminology

TOC

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] (Bradner, S., “Key words for use in RFCs to Indicate Requirement Levels,” March 1997.).

---

3. Possible Mechanisms

TOC

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

3.1. Why INFO is Not Used

TOC

Since the INFO method [RFC2976] (Donovan, S., “The SIP INFO Method,” October 2000.), 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.

3.2. MIME body Approach

TOC

One method of transport is to transport the UUI information as a MIME body. This is in keeping with the SIP-T architecture [RFC3372] (Vemuri, A. and J. Peterson, “Session Initiation Protocol for Telephones (SIP-T): Context and Architectures,” September 2002.) 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. This is not ideal as many SIP UAs do not support multipart MIME INVITEs.

A bigger problem is the insertion of a UUI message body by a redirect server or in a REFER. The body would need to be encoded in the Contact URI of the 3xx response or the Refer-To URI of a REFER. Currently, no UAs support this capability today, and even defining this is problematic. For example, do all the Content-* header fields have to be escaped as well? What if the escaped Content-Length does not agree with the escaped body?

An example:

```
<allOneLine>
Contact: <sip:+12125551212@gateway.example.com?Content-Type=
application/uui&body=ZeGl9i2icVqaNVailT6F5iJ90m6mvuTS4OK05M0vDk0Q4Xs>
</allOneLine>
```

Note that the <allOneLine> tag convention from SIP Torture Test Messages (Sparks, R., Hawrylyshen, A., Johnston, A., Rosenberg, J., and H. Schulzrinne, “Session Initiation Protocol (SIP) Torture Test Messages,” May 2006.) [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.

3.3. URI Parameter

TOC

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=ZeGl9i2icVqaNVailT6
F5iJ90m6mvuTS4OK05M0vDk0Q4Xs>
</allOneLine>
```

An INVITE sent to this Contact URI would contain UUI in the Request-URI of the INVITE. The URI parameter has a drawback in that a URI parameter carried in a Request-URI will not survive retargeting by a proxy as shown in Figure 2 of [johnston-dispatch-sip-cc-uui] (Johnston, A. and J. McMillen, “Requirements for Transporting User to User Call Control Information in SIP for ISDN Interworking,” .). That is, if the URI is included with an Address of Record instead of a Contact URI, the URI parameter in the Reqeuest-URI will not be copied over to the Contact URI, resulting in the loss of the information. As a result, this approach does not meet REQ-4. Note that if this same URI was present in a Refer-To header field, the same loss of information would occur.

3.4. Header Field Approach

TOC

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 or by an application server.

* Call-Info usually contains a URI pointer to the information instead of the actual information itself which does not meet REQ-5. It could be possible to use a data URI to carry the UUI directly in this header field.

*The use of Call-Info for interworking to and from ISDN networks seems problematic.

Overall, the overloading of the Call-Info header field for carrying interworked UUI does not seem like a good idea. A separate header field allows for clear policy and authorization rules to be used. 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 of [johnston-dispatch-sip-cc-uui] (Johnston, A. and J. McMillen, “Requirements for Transporting User to User Call Control Information in SIP for ISDN Interworking,” .):

```
User-to-User: 56a390f3d2b7310023a;encoding=hex;purpose=isdn-interwork
 ;content=isdn-uui
```

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

```
<allOneLine>
Contact: <sip:+12125551212@gateway.example.com?User-to-User=
56a390f3d2b7310023a%3Bencoding%3Dhex%3Bpurpose%3Disdn-interwork%3B
content%3Disdn-uui>
</allOneLine>
```

The resulting INVITE F5 would contain:

```
User-to-User: 56a390f3d2b7310023a;encoding=hex;purpose=isdn-interwork
 ;content=isdn-uui
```

An escaped User-to-User header field from the REFER message response F1 of Figure 4:

```
<allOneLine>
Refer-To: <sip:+12125551212@gateway.example.com?User-to-User=
56a390f3d2b7310023a%3Bencoding%3Dhex%3Bpurpose%3Disdn-interwork%3B
content%3Disdn-uui>
</allOneLine>
```

This would result in the INVITE F4 containing:

```
User-to-User: 56a390f3d2b7310023a;encoding=hex;purpose=isdn-interwork
 ;content=isdn-uui
```

4. 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 in [johnston-dispatch-sip-cc-uui] (Johnston, A. and J. McMillen, “Requirements for Transporting User to User Call Control Information in SIP for ISDN Interworking,” .) as demonstrated by existing implementations and running code. 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-7.
To help tag and identify the UUI used with this header field, "purpose", "content", and "encoding" parameters are defined. This specification only defines "purpose=isdn-intework", "content=isdn-uui", and "encoding=hex". Other specifications can define other purposes and contents for this header field per the requirements of this document.

---

5. Syntax for UUI Header Field

TOC

The User-to-User header field can be present in INVITE requests and responses only and in BYE requests and responses.
This document defines the purpose usage of "isdn-interwork" which 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- to-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. For SS7, user-to-user information may be exchanged within the following Q.763 messages: INITIAL ADRESS MESSAGE, ADDRESS COMPLETE MESSAGE, CALL PROGRESS, CONNECT, ANSWER, and RELEASE. 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 | cont-param | purp-param | generic-param
enc-param   = "encoding=" ("hex" | token)
cont-param  = "content=" ("isdn-uui" | token)
purp-param  = "purpose=" ("isdn-interwork" | token)
```

If the encoding, content, or purpose parameters are not present, their default values of "hex", "isdn-uui", and "isdn-interwork" MUST be assumed. Only one User-to-User header field with purpose=isdn-interwork may be present in a request or response. The "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.
UUI data with purpose=isdn-interwork MUST be less than 129 octets in length. This is because ISDN limits UUI to 128 octets in length plus the single octet protocol discriminator. Transporting UUI longer than 128 octets will result in interoperability failures when interworking with ISDN. UUI used for other purposes may have other length constraints, defined by the specification for that purpose.
A UA that supports this feature and the "uui" option tag MUST support the call flows in [johnston-dispatch-sip-cc-uui] (Johnston, A. and J. McMillen, “Requirements for Transporting User to User Call Control Information in SIP for ISDN Interworking,” .). In redirection scenarios, if the Redirect Server is not in the same administrative domain as the Terminator, the Redirect Server MUST NOT remove or replace any UUI in the initial INVITE. In Figure 3 of [johnston-dispatch-sip-cc-uui] (Johnston, A. and J. McMillen, “Requirements for Transporting User to User Call Control Information in SIP for ISDN Interworking,” .), this means that if F1 included UUI, the Redirect Server could not modify or replace the UUI in F2. However, if the Redirect Server and the Terminator are part of the same administrative domain, they may have a policy allowing the Redirect Server to modify or rewrite UUI information. In fact, many UUI uses within an Enterprise rely on this feature to work today in ISDN.

5.1. Definition of New Parameter Values

TOC

This specification defines only the values of "hex", "isdn-uui", and "isdn- interwork" for the "encoding", "content", and "purpose" parameters respectively. New values can be defined and added to the IANA registry with a standards track RFC, which needs to discuss the issues in this section.
New "encoding" values must reference a common encoding scheme or define the exact new encoding scheme.

New "content" values must describe the content of the UUI and give some example use cases. The default "encoding" and other allowed encoding methods must be defined for this new content.
New "purpose" values must describe the new purpose and give some example use cases. The default "content" value and other allowed contents must be defined for this new purpose. Any restrictions on the size of the UUI data must be described for the new purpose.

6. IANA Considerations

TOC

6.1. Registration of Header Field

TOC

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:

Header Name	Compact Form	Reference
User-to-User		[RFCXXXX]

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

6.2. Registration of Header Field Parameters

TOC

This document defines the parameters for the header field defined in the preceding section. The header field "User-to-User" can contain the parameters "encoding", "content", and "purpose".
The following rows shall be added to the "Header Field Parameters and Parameter Values" section of the SIP parameter registry:

Header Field	Parameter Name	Predefined Values	Reference
User-to-User	encoding	hex	[RFCXXXX]
User-to-User	content	isdn-interwork	[RFCXXXX]
User-to-User	purpose	isdn-uui	[RFCXXXX]

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

6.3. Registration of SIP Option Tag

TOC

This specification registers a new SIP option tag, as per the guidelines in Section 27.1 of [RFC3261] (Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M., and E. Schooler, "SIP: Session Initiation Protocol," June 2002.). This document defines the SIP option tag "uui".
The following row has been added to the "Option Tags" section of the SIP Parameter Registry:

Name	Description	Reference
uui	This option tag is used to indicate that a UA supports and understands the User-to-User header field.	[RFCXXXX]

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

7. Security Considerations

TOC

User to user information can be exchanged over SIP on a hop-by-hop or end-to-end basis. In some cases, UUI may carry privacy information that would require confidentiality and message integrity. Standard SIP security mechanisms, viz., based on TLS, offer these properties per-hop. To preserve multi-hop or end-end confidentiality and integrity, S/MIME profile MUST be utilized. Since the security requirements and key management of the UUI information are likely to be quite different from the SIP signaling transport, another approach would be for the UUI information to be encrypted before being passed to SIP for transport. Received User-to-User information should only be trusted if it is authenticated or if it is received within a trust domain. For example, Spec-T, defined in [RFC3324] (Watson, M., “Short Term Requirements for Network Asserted Identity,” November 2002.) could be used to define a trust domain. When utilized by a gateway to map information to or from ISDN Q.931 and ISUP Q.763, appropriate policy should be applied based on the PSTN trust domain.

8. Acknowledgements

TOC

Thanks to Spencer Dawkins, Keith Drage, Vijay Gurbani, and Laura Liess for their review of the document. The authors wish to thank Francois Audet, Denis Alexeitsev, Paul Kyzivat, Cullen Jennings, and Mahalingam Mani for their comments.

9. Informative References

TOC

[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 (TXT).
[Q931]	“ITU-T Q.931 User to User Information Element (UU IE),” http://www.itu.int/rec/T-REC-Q.931-199805-I/en .
[Q763]	“ITU-T Q.763 Signaling System No. 7 - ISDN user part formats and codes,” http://www.itu.int/rec/T-REC-Q.931-199805-I/en .
[ANSII]	“ANSI T1.643-1995, Telecommunications-Integrated Services Digital Network (ISDN)-Explicit Call Transfer Supplementary Service.”
[ETSI]	“ETSI ETS 300 207-1 Ed.1 (1994), Integrated Services Digital Network (ISDN); Diversion supplementary services.”
[RFC2119]	Bradner, S., “Key words for use in RFCs to Indicate Requirement Levels,” BCP 14, RFC 2119, March 1997 (TXT, HTML, XML).
[RFC3372]	Vemuri, A. and J. Peterson, “Session Initiation Protocol for Telephones (SIP-T): Context and Architectures,” BCP 63, RFC 3372, September 2002 (TXT).
[RFC2976]	

	Donovan, S., "The SIP INFO Method," RFC 2976, October 2000 (TXT).
[RFC3515]	Sparks, R., "The Session Initiation Protocol (SIP) Refer Method," RFC 3515, April 2003 (TXT).
[RFC3324]	Watson, M., "Short Term Requirements for Network Asserted Identity," RFC 3324, November 2002 (TXT).
[RFC4475]	Sparks, R., Hawrylyshen, A., Johnston, A., Rosenberg, J., and H. Schulzrinne, "Session Initiation Protocol (SIP) Torture Test Messages," RFC 4475, May 2006 (TXT).
[johnston-dispatch-sip-cc-uui]	Johnston, A. and J. McMillen, "Requirements for Transporting User to User Call Control Information in SIP for ISDN Interworking," draft-johnston-dispatch-sip-cc-uui-01 .

Authors' Addresses

TOC

	Alan Johnston (editor)
	Avaya
	St. Louis, MO 63124
Email:	alan.b.johnston@gmail.com
	Joanne McMillen
	Avaya
Email:	joanne@avaya.com