Internet Engineering Task Force (IETF)
Request for Comments: 6567
Category: Informational
ISSN: 2070-1721
A. Johnston
Avaya
L. Liess
Deutsche Telekom AG
April 2012

Problem Statement and Requirements for Transporting User-to-User Call Control Information in SIP

Abstract

This document introduces the transport of call control User-to-User Information (UUI) using the Session Initiation Protocol (SIP) and develops several requirements for a new SIP mechanism. Some SIP sessions are established by or related to a non-SIP application. This application may have information that needs to be transported between the SIP User Agents during session establishment. In addition to interworking with the Integrated Services Digital Network (ISDN) UUI Service, this extension will also be used for native SIP endpoints requiring application UUI.

Status of This Memo

Copyright Notice

Table of Contents

1. Overview

This document describes the transport of User-to-User Information (UUI) during SIP [RFC3261] session setup. This section introduces UUI and explains how it relates to SIP.

We define SIP UUI data as application-specific information that is related to a session being established using SIP. It is assumed that the application is running in both endpoints in a two-party session. That is, the application interacts with both the User Agents in a SIP session. In order to function properly, the application needs a small piece of information, the UUI, to be transported at the time of session establishment. This information is essentially opaque data to SIP -- it is unrelated to SIP routing, authentication, or any other SIP function. This application can be considered to be operating at a higher layer on the protocol stack. As a result, SIP should not interpret, understand, or perform any operations on the UUI. Should this not be the case, then the information being transported is not considered UUI, and another SIP-specific mechanism will be needed to transport the information (such as a new header field). In particular, this mechanism creates no requirements on intermediaries such as proxies, Back-to-Back User Agents, and Session Border Controllers.

UUI is defined this way for two reasons. First, this definition supports a strict layering of protocols and data. Providing information and understanding of the UUI to the transport layer (SIP in this case) would not provide any benefits and instead could create cross-layer coupling. Second, it is neither feasible nor desirable for a SIP User Agent (UA) to understand the information; instead, the goal is for the UA to simply pass the information as efficiently as possible to the application that does understand the information.

An important application is the interworking with User-to-User Information (UUI) in ISDN, specifically the transport of the call-control-related ITU-T Q.931 User-to-User Information Element (UUIE) [Q931] and ITU-T Q.763 User-to-User Information Parameter [Q763] data in SIP. ISDN UUI is widely used in the Public Switched Telephone Network (PSTN) today in contact centers and call centers. These applications are currently transitioning away from using ISDN for session establishment to using SIP. Native SIP endpoints will need to implement a similar service and be able to interwork with this ISDN service.

Note that the distinction between call control UUI and non-call-control UUI is very important. SIP already has a mechanism for sending arbitrary UUI data between UAs during a session or dialog -- the SIP INFO [RFC6086] method. Call control UUI, in contrast, must be exchanged at the time of setup and needs to be carried in the INVITE and a few other methods and responses. Applications that exchange UUI but do not have a requirement that it be transported and processed during call setup can simply use SIP INFO and do not need a new SIP extension.

In this document, four different use case call flows are discussed. Next, the requirements for call control UUI transport are discussed.

2. Use Cases

This section discusses four use cases for the transport of call control User-to-User Information. These use cases will help motivate the requirements for SIP call control UUI.

2.1. User Agent to User Agent

In this scenario, the originating UA includes UUI in the INVITE sent through a proxy to the terminating UA. The terminating UA can use the UUI in any way. If it is an ISDN gateway, it could map the UUI into the appropriate DSS1 [Q933] information element, QSIG [QSIG] information element, or ISDN User Part (ISUP) parameter. Alternatively, the using application might render the information to the user or use it during 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.

```
      Originating UA           Proxy          Terminating UA
             |                   |                   |
             |   INVITE (UUI) F1 |                   |
             |------------------>|   INVITE (UUI) F2 |
             |      100 Trying F3|------------------>|
             |<------------------|         200 OK F4 |
             |          200 OK F5|<------------------|
             |<------------------|                   |
             |  ACK F6           |                   |
             |------------------>|            ACK F7 |
             |                   |------------------>|
```

Figure 1: Call Flow with UUI Exchanged between Originating and Terminating UAs

2.2. Proxy Retargeting

In this scenario, the originating UA includes UUI in the INVITE request sent through a proxy to the terminating UA. The proxy retargets the INVITE request, changing its Request-URI to a URI that addresses the terminating UA. The UUI data is then received and processed by the terminating UA. This call flow is identical to Figure 1 except that the proxy retargets the request, i.e., changes the Request-URI as directed by some unspecified process. The UUI in the INVITE request needs to be passed unchanged through this proxy retargeting operation. Note that the contents of the UUI is not used by the proxy for routing, as the UUI has only end-to-end significance between UAs.

2.3. Redirection

In this scenario, UUI is inserted by an application that utilizes a SIP Redirect Server. The UUI is then included in the INVITE request sent by the originating UA to the terminating UA. In this case, the originating UA does not necessarily need to support the UUI mechanism but does need to support the SIP redirection mechanism used to include the UUI data. Two examples of UUI with redirection (transfer and diversion) are defined in [ANSI] and [ETSI].

Note that this case may not precisely map to an equivalent ISDN service use case. This is because there is no one-to-one mapping between elements in a SIP network and elements in an ISDN network. Also, there is not an exact one-to-one mapping between SIP call control and ISDN call control. However, this should not prevent the usage of SIP call control UUI in these cases. Instead, these slight differences between the SIP UUI mechanism and the ISDN service need to be carefully noted and discussed in an interworking specification.

Figure 2 shows this scenario, with the Redirect Server inserting UUI that is then included in the INVITE request F4 sent to the terminating UA.

```
         Originating UA       Redirect Server      Terminating UA
                 |                    |                    |
                 |          INVITE F1 |                    |
                 |------------------->|                    |
                 | 302 Moved (UUI) F2 |                    |
                 |<-------------------|                    |
                 |            ACK F3  |                    |
                 |------------------->|                    |
                 |  INVITE (UUI) F4   |                    |
                 |---------------------------------------->|
                 |  200 OK F5                              |
                 |<----------------------------------------|
                 |  ACK F6                                 |
                 |---------------------------------------->|
```

Figure 2: Call Flow with UUI Exchanged between Redirect Server and Terminating UA

A common example 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 terminating UA. The UUI could be used to lookup information for rendering to the agent at the time of call answering.

This redirection scenario and the referral scenario in the next section are the most important scenarios for contact center applications. Incoming calls to a contact center almost always are redirected or referred to a final destination, sometimes multiple times, based on collected information and business logic. The ability to pass along UUI in these call redirection scenarios is critical.

2.4. Referral

In this scenario, the application uses a UA to initiate a referral, which causes an INVITE request to be generated between the originating UA and terminating UA with UUI data inserted by the referrer UA. Note that this REFER method [RFC3515] could be part of a transfer operation, or it might be unrelated to an existing call, such as out-of-dialog REFER request. In some cases, this call flow is used in place of the redirection call flow: the referrer immediately answers the call and then sends the REFER request. This scenario is shown in Figure 3.

```
     Originating UA         Referrer           Terminating UA
           |                      |                     |
           |  REFER (UUI) F1      |                     |
           |<---------------------|                     |
           |  202 Accepted F2     |                     |
           |--------------------->|                     |
           |  INVITE (UUI) F3     |                     |
           |------------------------------------------->|
           | NOTIFY (100 Trying) F4                     |
           |--------------------->|                     |
           |         200 OK F5    |                     |
           |<---------------------|                     |
           |  200 OK F6                                 |
           |<-------------------------------------------|
           |  ACK F7                                    |
           |------------------------------------------->|
           | NOTIFY (200 OK) F8   |                     |
           |--------------------->|                     |
           |        200 OK F9     |                     |
           |<---------------------|                     |
```

Figure 3: Call Flow with Referral and UUI

3. Requirements

This section states the requirements for the transport of call control User-to-User Information (UUI).

REQ-1: The mechanism will allow UAs to insert and receive 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, i.e., 18x and 200 responses. UUI data may also be inserted in 3xx responses to an INVITE. However, if a 3xx response is recursed on by an intermediary proxy, the resulting INVITE will not contain the UUI data from the 3xx response. In a scenario where a proxy forks an INVITE to multiple UAS who include UUI data in 3xx responses, if a 3xx response is the best response sent upstream by the proxy, it will contain the UUI data from only one 3xx response.

REQ-2: The mechanism will allow UAs to insert and receive UUI data in SIP dialog terminating 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 and referrals.

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

REQ-4: The mechanism will allow UUI to be able to survive proxy retargeting or redirection of the request.

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 should not require processing entities to dereference a URL in order to retrieve the UUI data.

Passing a pointer or link to the UUI data will not meet the real-time processing considerations and would complicate interworking with the PSTN.

REQ-6: The mechanism will support interworking with call-control-related DSS1 information elements or QSIG information elements and ISUP parameters.

REQ-7: The mechanism will allow a UAC to learn that a UAS understands the UUI mechanism.

REQ-8: The mechanism will allow a UAC to require that a UAS understands the call control UUI mechanism and have a request routed based on this information. If the request cannot be routed to a UAS that understands the UUI mechanism, the request will fail.

This could be useful in ensuring that a request destined for the PSTN is routed to a gateway that supports the UUI mechanism rather than an otherwise equivalent PSTN gateway that does not support the ISDN mechanism. Note that support of the UUI mechanism does not, by itself, imply that a particular application is supported (see REQ-10).

REQ-9: The mechanism will allow proxies to remove a particular application usage of UUI data from a request or response.

This is a common security function provided by border elements to header fields such as Alert-Info or Call-Info URIs. There is no requirement for UAs to be able to determine if a particular usage of UUI data has been removed from a request or response.

REQ-10: The mechanism will provide the ability for a UA to discover which application usages of UUI another UA understands or supports.

The creation of a registry of application usages for the UUI mechanism is implied by this requirement. The ISDN service utilizes a field known as the protocol discriminator, which is the first octet of the ISDN UUI data, for this purpose.

REQ-11: The UUI is a sequence of octets. The solution will provide a mechanism of transporting at least 128 octets of user data and a one-octet protocol discriminator, i.e., 129 octets in total.

There is the potential for non-ISDN services to allow UUI to be larger than 128 octets. However, users of the mechanism will need be cognizant of the size of SIP messages and the ability of parsers to handle extremely large values.

REQ-12: The recipient of UUI will be able to determine the entity that inserted the UUI. It is acceptable that this is performed implicitly where it is known that there is only one other end UA involved in the dialog. Where that does not exist, some other mechanism will need to be provided. The UUI mechanism does not introduce stronger authorization requirements for SIP; instead, the mechanism needs to be able to utilize existing SIP approaches for request and response identity.

This requirement comes into play during redirection, retargeting, and referral scenarios.

4. Security Considerations

The security requirements for the UUI mechanism are described in this section. It is important to note that UUI security is jointly provided at the application layer and at the SIP layer. As such, is important for application users of the UUI mechanism to know the level of security used and deployed in their particular SIP environments and not to assume that a standardized (but perhaps rarely deployed) security mechanism is in place.

There are three main security models that need to be addressed by the UUI mechanism. One model treats the SIP layer as untrusted and requires end-to-end integrity protection and/or encryption. This model can be achieved by providing these security services at a layer above SIP. In this case, the application integrity protects and/or encrypts the UUI data before passing it to the SIP layer. This method has two advantages: it does not assume or rely on end-to-end security mechanisms in SIP, which have virtually no deployment, and it allows an application that understands the contents of the UUI to apply a proper level of security.

The second approach is for the application to pass the UUI without any protection to the SIP layer and require the SIP layer to provide this security. This approach is possible in theory, although its practical use would be extremely limited.

The third model utilizes a trust domain and relies on perimeter security at the SIP layer. This is the security model of the PSTN and ISDN where UUI is commonly used today. This approach uses hop-by-hop security mechanisms and relies on border elements for filtering and application of policy. This approach is used today in UUI deployments. Within this approach, there is a requirement that intermediary elements can detect and remove a UUI element based on policy, but there is no requirement that an intermediary element be able to read or interpret the UUI (as the UUI contents only have end-to-end significance).

The next three requirements capture the UUI security requirements.

REQ-13: The mechanism will allow integrity protection of the UUI.

This allows the UAS to be able to know that the UUI has not been modified or tampered with by intermediaries. Note that there are tradeoffs between this requirement and requirement REQ-9 for proxies and border elements to remove UUI. One possible way to satisfy both of these requirements is to utilize hop-by-hop protection. This property is not guaranteed by the protocol in the ISDN application.

REQ-14: The mechanism will allow end-to-end privacy of the UUI.

Some UUI may contain private or sensitive information and may require different security handling from the rest of the SIP message. Note that this property is not available in the ISDN application.

REQ-15: The mechanism will allow both end-to-end and hop-by-hop security models.

The hop-by-hop model is required by the ISDN UUI service.

5. Acknowledgements

Thanks to Joanne McMillen, who was a co-author of earlier draft versions of this specification. Thanks to Spencer Dawkins, Keith Drage, Dale Worley, and Vijay Gurbani for their review of earlier draft versions of this document. The authors wish to thank Christer Holmberg, Frederique Forestie, Francois Audet, Denis Alexeitsev, Paul Kyzivat, Cullen Jennings, and Mahalingam Mani for their comments on this topic.

6. Informative References

[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.

[Q931] ITU-T, "ISDN user-network interface layer 3 specification for basic call control", ITU-T Recommendation Q.931, <http://www.itu.int/rec/T-REC-Q.931-199805-I/en>.

[Q763] ITU-T, "Signalling System No. 7 - ISDN User Part formats and codes", ITU-T Recommendation Q.763, <http://www.itu.int/rec/T-REC-Q.763-199912-I/en>.

[RFC6086] Holmberg, C., Burger, E., and H. Kaplan, "Session Initiation Protocol (SIP) INFO Method and Package Framework", RFC 6086, January 2011.

[Q933] ITU-T, "ISDN Digital Subscriber Signalling System No. 1 (DSS1) - Signalling specifications for frame mode switched and permanent virtual connection control and status monitoring", ITU-T Recommendation Q.933, <http://www.itu.int/rec/T-REC-Q.933/en>.

[QSIG] ECMA, "Private Integrated Services Network (PISN) - Circuit Mode Bearer Services - Inter-Exchange Signalling Procedures and Protocol (QSIG-BC)", Standard ECMA-143, December 2001.

[ANSI] ANSI, "Telecommunications-Integrated Services Digital Network (ISDN)-Explicit Call Transfer Supplementary Service", ANSI T1.643-1995.

[ETSI] ETSI, "Integrated Services Digital Network (ISDN); Diversion supplementary services", ETSI ETS 300 207-1, Ed. 1, 1994.

[RFC3515] Sparks, R., "The Session Initiation Protocol (SIP) Refer Method", RFC 3515, April 2003.

Authors' Addresses

Alan Johnston
Avaya
St. Louis, MO 63124

EMail: alan.b.johnston@gmail.com

Laura Liess
Deutsche Telekom AG

EMail: laura.liess.dt@gmail.com