Transporting User to User Information for Call Centers using SIP
draft-johnston-sipping-cc-uui-01

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Abstract

Several approaches to transporting User to User Information (UUI) in Contact Center scenarios have been implemented. This document discusses the approaches and recommends a single approach for standardization.
1. Overview

This document describes the transport of User to User Information (UUI) in Contact Center scenarios using SIP [1].

2. Roles

There are three roles in a typical Call Center scenario:

Originator - The party generating the call destined for the contact center.

ACD - The automatic call distributor which performs database queries and determines proper routing for the call.

Terminator - The destination of the call as determined by the ACD.

3. Approaches

Three approaches will be described: MIME body, header field, and URI parameter transport.

3.1. Call Flows

All three approaches use the same call flows, shown in Figures 1, 2, and 4 below. Figure 1 shows the ACD redirecting the call prior to it being answered using a 302 Moved Temporarily response. User to user information (UUI) is added by the ACD which is then included in the INVITE sent to the Terminator. The URI of the terminator is contained in the Contact header field of message F2.

<table>
<thead>
<tr>
<th>Originator</th>
<th>ACD</th>
<th>Terminator</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>INVITE F1</td>
<td></td>
</tr>
<tr>
<td></td>
<td>------------&gt;</td>
<td></td>
</tr>
<tr>
<td></td>
<td>302 Moved (UUI) F2</td>
<td></td>
</tr>
<tr>
<td></td>
<td>&lt;------------</td>
<td></td>
</tr>
<tr>
<td></td>
<td>ACK F3</td>
<td></td>
</tr>
<tr>
<td></td>
<td>------------&gt;</td>
<td></td>
</tr>
<tr>
<td></td>
<td>INVITE (UUI) F4</td>
<td></td>
</tr>
<tr>
<td></td>
<td>&lt;--------------------</td>
<td></td>
</tr>
<tr>
<td></td>
<td>200 OK F5</td>
<td></td>
</tr>
<tr>
<td></td>
<td>&lt;--------------------</td>
<td></td>
</tr>
<tr>
<td></td>
<td>ACK F6</td>
<td></td>
</tr>
</tbody>
</table>
Figure 1. Call flow with redirection prior to answer.

Figure 2 shows a call flow in which the call is answered by the ACD and then the call is transferred using a REFER. If the ACD does not care to receive notification of the outcome of the transfer, the BYE could be sent immediately, or the Refer-Sub: false header field added to the REFER. If the REFER is sent out of dialog, the Target-Dialog header field will need to be present.

![Call Flow Diagram](image)

Figure 2. Call flow with transfer after answer.

Note that there have been some proposals to use REFER for scenario of Figure 1. In this case, a REFER is sent by the ACD during the early dialog. This NOT RECOMMENDED call flow is shown in Figure 3. 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.

Figure 3. NOT RECOMMENDED call flow showing REFER prior to answer.

In Figure 4, the ACD is shown proxying the INVITE to the terminator instead of redirecting as in Figure 1.
Figure 4. Call flow with proxying.

In Figure 5, the ACD is shown proxying the INVITE and modifying the UII information provided by the Originator.

3.2. MIME body Approach

One method of transport is to transport the UII information as a MIME body. This is in keeping with the SIP-T architecture in which MIME bodies are used to transport ISUP information. However, in the SIP-T example, the body is added by the originator and used by the terminator. The insertion of the UII by an intermediary, the ACD, is difficult. The body would need to be encoded in the Contact URI of Figure 1 or the Refer-To URI of Figure 2. For example, if the application/acd MIME type were defined:

Contact: <sip:+12125551212@gateway.example.com?Content-Type=application/acd&body=ZeGl9i2icVqaNVailT6F5iJ90m6muT5496506vdckQ4Kx>

The resulting INVITE would then have two message bodies, one SDP, the other being the MIME object. In addition to the body, the escaped Content-Type header field should also be included in the resulting INVITE.
While the body approach would work with the redirection and REFER call flows, but does not work with the proxy call flow of Figure 4.

3.3. 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 escaped into the Contact or Refer-To URI. The header field approach will work with the redirection, REFER, and proxy call flows. The Call-Info header field is related to the UUI information. However, there are a number of important differences:

- Call-Info is 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 contains a URI pointer the information instead of the actual information itself.

The use of the data URI scheme [6] could be used to overcome the second issue, i.e.:

```
Call-Info: <data:application/acid,2eG19i2icVqaNVa11t6F5i1J90m6m6vUT40K5MV0vDk0Q4Xs>
```

For these reasons, a separate header field needs to be defined, described here as User-to-User:

```
Contact: <sip:+12125551212@gateway.example.com?User-to-User=ZeGl9i2icVqaNVa11t6F5i1J90m6m6vUT40K5MV0vDk0Q4Xs>
```

The resulting INVITE would contain the User-to-User header field.

3.4. URI Parameter

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

```
Contact: <sip:+12125551212@gateway.example.com;uui=ZeGl9i2icVqaNVa11t6F5i1J90m6m6vUT40K5MV0vDk0Q4Xs>
```

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. 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.

4. Recommendation

The recommendation is to define a new SIP header field User-to-User to transport UUI information in Contact Center applications.
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.

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.

5. 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.

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 uuidata *(SEMI generic-param)
uuidata         = token
```

6. IANA Considerations

TBD.

7. Security Considerations

TBD.
8. Informative References


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