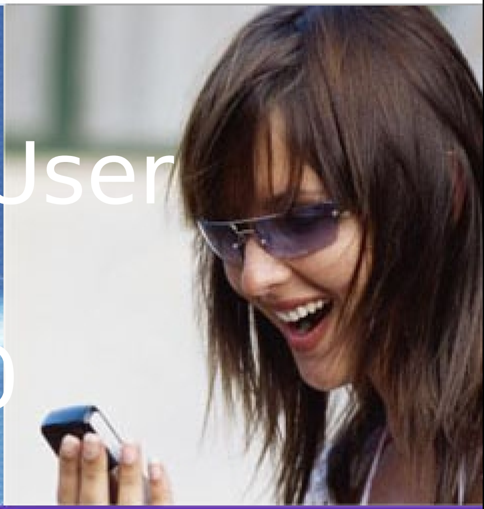
Interworking ISDN Call Control User Information with SIP
draft-drage-cuss-sip-uui-isdn-00



Keith Drage

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Purpose of draft-drage-cuss-sip-uui-isdn-00

Charter:

- “The working group will define guidelines for other applications to utilize the mechanism and the information that each application must specify to utilize the mechanism. New applications of the mechanism will require a standards-track document.”
- “The group will produce: … - An application usage specification for the ISDN UUI Service.”

Milestone in CUSS charter:

- “Jun 2011 ISDN UUI Service application usage specification to IESG (PS)”

Abstract

- “The motivation and use cases for interworking and transporting ITU-T DSS1 User-user information element data in SIP are described in the "Problem Statement and Requirements for Transporting User to User Call Control Information in SIP" document. 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 defines a usage of the User-to-User header field to enable interworking with this ISDN service.”

OPEN ISSUE 1

Key to defining such an application is to understand how the capabilities of the generic SIP User-to-User header field extension relate to those provide by the ISDN user-to-user signalling supplementary service. If they are the same as the ISDN User-to-user signalling supplementary service then interworking is solely a matter of mapping the construct in one protocol into the equivalent construct in the other protocol.

As the ISDN user-to-user signalling supplementary service is a somewhat restricted service, it is unlikely that the capabilities will be more than the generic SIP User-to-User header field extension. So we need to deal with the question of what occurs when (and if) the generic SIP User-to-User header field extension has wider capabilities than those of the ISDN user-to-user signalling supplementary service.

The capabilities of the ISDN user-to-user signalling supplementary service have been outlined in section 3.

The following open issues target what could occur if each of these capabilities are exceeded.

OPEN ISSUE 2

The maximum number of octets defined for the generic SIP User-to-User header field extension exceeds the 128 octets supported by the ISDN user-to-user signalling supplementary service. Obviously if the SIP sender sends more than that allowed, then mapping of the entire contents is impossible. Truncation should not occur (as the truncation cannot be signalled in the ISDN and the contents will probably end up meaningless) and therefore the only option is to discard the information and proceed without it.

Mapping in the opposite direction will never be a problem.

Is extra signalling needed to allow for this discard; it is believed that the answer is no.

If the SIP user knows that it is interworking with the ISDN, then the UUI application at the SIP endpoint should limit its communication to 128 octet packets, in the knowledge that discard will occur if it does not. The UUI application at the SIP endpoint has complete control over what occurs. It should be noted that this was exactly the envisaged operation when early ISDN implementations that only supported 32 octets interworked with those supporting 128 octets. It also corresponds to the interworking with ISDNs that do not support the supplementary service at all, as discard will occur in these circumstances as well.

Note that failure to include the user-user data into the ISDN SETUP message (when discard occurs) will result in the service being unavailable for the remainder of the call when UUS1 implicit operation is used.

OPEN ISSUE 3

The generic SIP User-to-User header field extension supports the description of more "protocol discriminators" that that supported by the ISDN user-to-user signalling supplementary service. Part of this depends on how these additional application identifiers (if any) are carried, and as a result whether the existing ISDN protocol discriminator is carried "as is". At the moment this is assumed, and therefore if a valid protocol discriminator value exists, then it is mapped. If one does not exist then it is not mapped, and discard of the entire user data occurs as in open issue 2.

It is believed that many of the considerations of Issue 2 apply, and therefore the sole reason for any additional signalling support is to identify a protocol discriminator that can be mapped (i.e. one that forms part of the set that exists in ISDN), from those that cannot.

Note that failure to include the user-user data into the ISDN SETUP message (when discard occurs) will result in the service being unavailable for the remainder of the call when UUS1 implicit operation is used.

OPEN ISSUE 4

It could be that more than one payload is allowed to be included in the generic SIP User-to-User header field extension whereas only one payload is supported by the ISDN user-to-user signalling supplementary service.

It is believed the considerations are identical to open issue 2.

OPEN ISSUE 5

If integrity protection or encryption is supported in the generic SIP User-to-User header field extension then it is unlikely this can be supported in the ISDN. It is assumed that the gateway will support nothing but the transparent mapping of payload, and indeed fulfils no useful function by performing any capability in regard to integrity protection or encryption.

Similar considerations apply as for open issue 2, i.e. that the UUI application at the SIP endpoint has complete control over what occurs.

OPEN ISSUE 6

Interworking depends on there being equivalent functionality existing in both protocols. The mapping for ISDN basic call to SIP is well established and deployed. It is believed that there is no issue in mapping the generic SIP User-to-User header field extension as supported in RFC 3261 to the ISDN user-to-user signalling supplementary service in this respect. Issues may occur when more complex SIP transactions are used such as the 3xx response and the REFER method. This is of course dependent on there being a mapping at the ISDN gateway of the the 3xx response or the REFER method in the first place. Many SIP deployments rely on some server converting REFER transactions to INVITE transactions within the SIP environment, therefore the interworking requirements are merely those of the INVITE dialog itself.

Are there other more complex scenarios that need to be studied for interworking?

Conclusion

Is the document an appropriate document to meet the identified WG milestone?

Other comments and discussion?