dispatch R. Jesske
Internet-Draft L. Liess
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Reason header filed in Session Initiation Protocol (SIP) responses
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Abstract

Although the use of the Reason header field in responses is considered in RFC3326, doing so is not specified for any existing response code. Nonetheless, the Reason header field has been widely used in responses to carry Q.850 reason codes for failure responses to INVITEs that have been gatewayed to PSTN systems. This document specifies and formally permits the use of the Reason header field in SIP responses to carry Q.850 reason codes for this and other purposes.

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1. 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 [RFC2119].

This document uses terms from [RFC3261].

2. Overview

With introducing the Session Initiation Protocol [RFC3261]into the IP Multimedia Subsystem (IMS) which is defined by the 3rd Generation Partnership Project the it was required to interoperate with the PSTN/ISDN. The European Telecommunication Standards Institute (ETSI) has defined a Next Generation Network (NGN) where a substantial part of it is based on the IMS.

ETSI has developed a number of requirements to support the usage of SIP in Next Generation Networks that interoperate, at the service level, with the Public Switched Telephone Network (PSTN), the Integrated Services Digital Network (ISDN), the 3GPP IP Multimedia Subsystem (IMS), and SIP networks and terminals that implement the service logic.

Also ITU-T has specified an interworking between SIP and PSTN/ISDN networks [ITU.Q1912.5.2004] and [TS29.163] where reason within responses are already supported.

In order to provide full support in SIP of existing services, extensions to SIP are needed.

This document proposes the use of the Reason header field in responses. This is needed for creating services that must be interoperable with the PSTN/ISDN network and the interoperability of traversing communications through SIP not using SIP-I.

The main used case for reason header within responses are interworking situations with PSTN/ISDN networks where the ISUP cause In many cases the mapping of specific cause values will result in a generic SIP Response.

[RFC3398] and other Interworking specifications like [RFC3326] are describing the mapping of ISUP Cause Values to SIP and vice versa. Looking on the specific mapping shows that information will be lost when the call traverses ISUP without using SIP-T.

3. Overall Applicability

The SIP procedures specified in this document are foreseen for networks providing simulation services and/or interworking to the PSTN/ISDN.

The document is describing the use of the Reason header in SIP responses. These procedures are only valuable if the reason contained in the element "protocol" is "Q.850". A inclusion of a SIP reason (protocol="SIP") is not helpful due to the fact that the response already provides the SIP reason. The Release Causes are described within [Q.850]. (Note: The ETSI specifications can be downloaded under http://pda.etsi.org/pda/queryform.asp free of charge.)

The appearance of the Reason header is applicable to final responses 3xx, 4xx, 5xx and 6xx and 18x and 199 provisional responses [I-D.ietf-sipcore-199].

4. Requirements

A UA may have the ability to display ISUP specific release causes or show a equivalent text.

In SIP-to-PSTN gateway scenarios, it is desirable to provide the UAC with the specific call release reason provided by the PSTN. To support this:

REQ-1: Provide in SIP responses a way to carry PSTN call release

codes, along with indication of any context or variant identification needed to interpret the code unambiguously.

REQ-2: Provide an extensibility mechanism so that further information about the call release can be specified.

5. Procedures

5.1. Procedures at the UA

A UA that supports the Reason header field can process the [Q.850] Cause Value and display it or an equivalent text. The inclusion of a Reason header field by UA is only for B2B UA interworking with the PSTN/ISDN or providing services foreseen.

5.2. Procedures at a SIP proxy

SIP proxies that receive a response containing a Reason header field is forwarding the response without changing the reason.

A SIP proxy receiving a request that includes a Reason header field can route the request to an application server for further analysis and base services on it.

Based on network policy a Proxy can remove a Reason header field send from a UAC.

5.3. Procedures at an application server

An application server that receives a SIP request that contains a response including a Reason header MAY analyze the SIP Reason and base further procedures on this analyses.

For Example the application server could use the reason for sending a announcement towards the originating entity of the session.

As an example the Anonymous Communication Rejection (ACR) service defined by ETSI Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN)

6. Procedures at an interworking point with ISUP

For interoperability reasons the Q.850 Cause Value of a Release shall be mapped to the Reason Header.

7. Security Considerations

The presence of the Reason header in a response does not affect the treatment of the response.

Including such a header by an untrusted entity could adulterate the reactions of the originating entities. E.G. sending back a cause value "87" can cause an announcement within the PSTN/ISDN saying that the call was rejected due to the Closed User Group service.

Therefore it is RECOMMENDED to include the Reason header information in Responses only by trusted entities as it is described within [RFC3325].

8. IANA Considerations

This document describes the use of the Reason header field described within [RFC3326] . No additional SIP elements are defined within this document. Therefore, this document does not provide any action to IANA.

9. Normative References

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[ITU.Q1912.5.2004] International Telecommunications Union, "Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part [ITU-T Recommendation Q.1912.5 (2004)]", ITU Recommendation Q.1912.5, April 2004.

[Q.850] "Usage of cause and location in the Digital

Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part [ITU-T Recommendation Q.850]", ITU Recommendation Q.850, April 1998.

[RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.

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[RFC3326] Schulzrinne, H., Oran, D., and G. Camarillo, "The Reason Header Field for the Session Initiation Protocol (SIP)", RFC 3326, December 2002.

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[TS29.163] "3rd Generation Partnership Project; Technical Specification Group Core Network and Terminals; Interworking between the IP

Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (Release 8)".

Authors' Addresses

Roland Jesske
Deutsche Telekom

Heinrich-Hertz-Strasse 3-7
Darmstadt,   64307
Germany

Phone: +4961516282766
EMail: r.jesske@telekom.de

Laura Liess
Deutsche Telekom
Heinrich-Hertz-Strasse 3-7
Darmstadt,   64307
Germany

Phone: +4961516282761
EMail: Laura.Liess@telekom.de