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Requirements for the use of the Reason header filed in Session Initiation Protocol (SIP) responses
draft-jesske-dispatch-req-reason-in-responses-02

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 like it is shown below.

[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.

Example:

[RFC3398] describes the mapping of following ISUP Causes to 503 and 408 like follows.

ISUP Cause value	SIP response
34 no circuit available	503 Service unavailable
38 network out of order	503 Service unavailable
41 temporary failure	503 Service unavailable
42 switching equipment congestion	503 Service unavailable
47 resource unavailable	503 Service unavailable
58 bearer capability not presently Available	503 Service unavailable
88 incompatible destination	503 Service unavailable
18 no user responding	408 Request Timeout

The mapping back is shown as follows:

Response received	Cause value in the REL
503 Service unavailable	41 Temporary failure

408 Request timeout	102 Recovery on timer expiry

The Example with 503 shows that a couple of different ISUP Cause values are interworked to only one SIP response. With 408 the meaning of the release cause is changed when interworked back to ISUP. Also Services built on Cause 18 (e.G. a 2nd call attempt on an other number, this service is like a sequential forking) will not work.

3. 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.

4. 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], the procedures for that are described within [draft-jesske-dispatch-reason-in-responses-02] .

5. Example

Figure 1 shows the example of SIP interworking with the PSTN/ISDN. Cause #87 is sent when the connecting user is not member of a Closed User Group.


```
            A                Gateway             Proxy               AS
            |        IAM        |                  |                  |
            |------------------>|     INVITE       |                  |
            |                   |----------------->|      INVITE      |
            |                   |     100 Trying   |----------------->|
            |                   |<-----------------|    100 Trying    |
            |                   |                  |<-----------------|
            |   ACK  SDP held   |                  |                  |
```

```
           |<------------------|                  |  603 Decline     |
           |                   |  603 Decline     | Reason Q850 #87  |
           |                   |  Reason Q850 #87 |                  |
           |   REL Cause #87   |                  |<-----------------|
           |                   |<-----------------|                  |
           |<----------------- |                  |                  |
           |                   |                  |                  |
           |                   |                  |                  |
           |                   |                  |                  |
```

Figure 1: ISUP-SIP Call

Figure 2 shows the example where the SIP network is used as transit between PSTN/ISDN networks. This avoids that the Mapping back to the Q.850 cause within ISUP change the meaning of the reason for release of the call.

```
           A                Gateway             Gateway              B
           |        IAM        |                  |                  |
           |------------------>|     INVITE       |                  |
           |                   |----------------->|      IAM         |
           |                   |     100 Trying   |----------------->|
           |                   |<-----------------|    100 Trying    |
           |                   |   603 Decline    |                  |
           |                   |  Reason Q850 #87 |  REL Cause #87   |
           |   REL Cause #87   |                  |<-----------------|
           |                   |<-----------------|                  |
           |<----------------- |                  |                  |
           |                   |                  |                  |
           |                   |                  |                  |
           |                   |                  |                  |
```

Figure 2: Transit case

Figure 3 shows the example where the SIP network puts an announcement

towards the UAB. The AS sends an announcement with a specific text back. After some Time the Response will be sent back to the UA A and closes all open transactions. With this possibility the SIP user can informed with more specific information than only the Response code.

```
A                   AS               Gateway              B
|        INVITE     |                 |                   |
|------------------>|     INVITE      |                   |
|                   |---------------->|       IAM         |
|                   |    100 Trying   |------------------>|
|                   |<----------------|                   |
|                   |   503 Decline   |                   |
|                   | Reason Q850 #41 |  REL Cause #41    |
|                   |                 |<------------------|
|  Announcement     |<----------------|                   |
|< ================ |                 |                   |
|                   |                 |                   |
|  503 after Timeout|                 |                   |
|<----------------- |                 |                   |
```

Figure 3: Transit case

Figure 3: Call Release within the PSTN with an announce played within the SIP network.

6. 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].

7. IANA Considerations

This document does not have any implications for IANA.

8. Normative References

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[draft-jesske-dispatch-reason-in-responses-02]
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