DISPATCH R. Jesske
Internet-Draft L. Liess
Intended status: Standards Track Deutsche Telekom
Expires: February 12, 2012 August 11, 2011

# Carrying Q.850 Codes in Reason Header Fields in SIP (Session Initiation Protocol) Responses

draft-jesske-dispatch-update3326-reason-responses-05

## Abstract

Although the use of the Reason header field in responses is considered in general in RFC3326, its use is not specified for any particular response code. Nonetheless, existing deployments have been using Reason header fields in responses to carry Q.850 cause codes for failure responses to INVITE requests that have been gatewayed to PSTN systems. This document normatively describes the use of the Reason header field in SIP responses to carry Q.850 cause codes.

## Status of This Memo



## Copyright Notice

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## 1. Overview

Although the use of the Reason header field in responses is considered in general in RFC3326[RFC3326], its use is not specified for any particular response code. Nonetheless, existing deployments have been using Reason header fields in responses to carry Q.850 [Q.850] cause codes for failure responses to INVITE requests that have been gatewayed to PSTN systems. This document normatively describes the use of the Reason header field in SIP responses to carry Q.850 [Q.850] cause codes.

## 2. 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].

## 3. Applicability

This document allows SIP responses to carry Reason header fields as follows:

Any SIP Response message, with the exception of a 100 (Trying) MAY contain a Reason header field with a Q.850 [Q.850] cause code.

The Reason header field is not needed in the the 100 (Trying) responses since they are transmitted hop-by-hop, not end-to-end. SIP responses with Reason header fields carrying values other than Q.850 [Q.850] cause code are outside of the scope of this document.

## 4. Security Considerations

This specification allows the presence of the Reason containing Q.850 [Q.850] cause codes in responses. The presence of the Reason header field in a response does not affect the treatment of the response. Nevertheless, there could be situations where a wrong Q.850 [Q.850] cause code could, for example, cause an announcement system to play the wrong information. To avoid such situations, it is RECOMMENDED that this header field is protected by a suitable integrity mechanism. The use of transport or network layer hop-by-hop security mechanisms, such as TLS or IPSec with appropriate cipher suites, can satisfy this requirement.

## 5. IANA Considerations

No IANA actions are required

## 6. Acknowledgments

Thanks to Gonzalo Camarillo and Mary Barnes for the detailed review of this document.

Thanks to Paul Kyzivat, Mary Barnes, John Elwell, Keith Drage, Thomas Belling who provided helpful comments, feedback and suggestions.

## 7. Normative References

[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.

[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.

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

## Authors' Addresses

Roland Jesske
Deutsche Telekom
Heinrich-Hertz-Strasse 3-7
Darmstadt, 64307
Germany

Phone: +4961515812766
EMail: r.jesske@telekom.de

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

Phone: +4961515812761
EMail: Laura.Liess@telekom.de

Network Working Group
Internet Draft
Intended status: Informational
Expires: November 20, 2011

Paul E. Jones
Gonzalo Salgueiro
James Polk
Parthasarathi Ravindran
Cisco Systems
Laura Liess
Roland Jesske
Deutsche Telekom
Salvatore Loreto
Ericsson
Hadriel Kaplan
Acme Packet
May 20, 2011

# Requirements for an End-to-End Session Identification in IP-Based Multimedia Communication Networks
draft-jones-ipmc-session-id-reqts-00.txt

## Abstract

This document specifies the requirements for an end-to-end session identifier in IP-based multimedia communication networks. This identifier would enable endpoints, intermediate devices, and management and monitoring systems to identify a session end-to-end, associate multiple endpoints with a given multipoint conference, track communication sessions when they are redirected, and associate one or more media flows with a given communication session.

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# 1. Introduction

IP-based multimedia communication systems like SIP [1] and H.323 [2] have the concept of a "call identifier" that is globally unique. The identifier is intended to help form a globally unique value that represents an end-to-end communication session. Such an identifier is useful for troubleshooting, billing, session tracking, and so forth.

Unfortunately, there are a number of factors that contribute to the fact that the current call identifiers defined in SIP and H.323 are not suitable for end-to-end session identification. Perhaps most significant is the fact that the syntax for the call identifier in SIP and H.323 is different between the two protocols. This important fact makes it impossible for call identifiers to be exchanged end-to-end when a network utilizes one or more session protocols.

Another reason why the current call identifiers are not suitable to identify the session end-to-end is that in real-world deployments devices like session border controllers often change the values as the session signaling passes through. This is true even when a single session protocol is employed and not a byproduct of protocol interworking.

Lastly, identifiers that might have been used to identify a session end-to-end fail to meet that need when sessions are manipulated through supplementary service interactions. For example, when a session is transferred or if a PBX joins two communication sessions together locally, the end-to-end properties of currently-defined identifiers are lost.

This draft specifies the requirements for an end-to-end session identifier. With this draft, the authors would like to encourage discussion and progress work in the dispatch working group or working group as designated by the IETF, the outcome of which will be a new RFC that defines a session ID in conformance with these requirements.

## 2. Terminology

SIP defines additional terms used in this document that are specific to the SIP domain such as "proxy"; "registrar"; "redirect server"; "user agent server" or "UAS"; "user agent client" or "UAC"; "back-to-back user agent" or "B2BUA"; "dialog"; "transaction"; "server transaction".

In this document, the word "session" refers to a "communication session" that may exist between two SIP user agents or that might pass through one or more intermediary devices, including B2BUAs or SIP Proxies.

The term "end-to-end" in this document means the communication session from the point of origin, passing through any number of intermediaries, to the ultimate point of termination. It is recognized that legacy devices may not support the "end-to-end" session identifier, though an identifier might be created by an intermediary when it is absent from the session signaling.

## 3. Requirements for the End-to-End Session Identifier

REQ1: It must be possible for an administrator or an external device which monitors the SIP-traffic to use the identifier to identify a set of dialogs which have a relationship with each other, such that they represent the same SIP session, with as high a probability as possible.

REQ2: It must be possible to identify the end-to-end session when a session is transferred or if two different sessions are joined together via an intermediary (e.g., a PBX).

REQ3: It must be possible to identify all sessions participating in a multipoint or multi-party conference by observing the end-to-end session identifiers of each session.

REQ4: It must be possible to pass the identifier unchanged through SIP B2BUAs or other intermediaries.

REQ5: The identifier must not reveal any information related to any SIP device or domain identity, including IP Address, port, hostname, domain name, username, Address-of-Record, MAC address, IP address family, transport type, etc.

REQ6: The identifier must not reveal to the receiver of it that the Call-ID, tags, or any other SIP header or body portion have been changed by middleboxes, with as high a probability as possible.

REQ7: It must be possible to identity SIP traffic with an end-to-end session identifier from and to end devices that do not support this new identifier, such as by allowing an intermediary to inject an identifier into the session signaling.

REQ8: The identifier should be unique in time and space, similar to the Call-ID.

REQ9: The identifier should be constructed in such a way as to make it suitable for transmission in SIP, H.323, RSVP [3], and RTCP [4].

## 4. Session Identifier Use Cases

The Session Identifier is intended to uniquely identify a communication session end-to-end.  This document does not specify how the Session Identifier is to be used, but merely defines the identifier in such a way as to enable it to be used for situations encountered in real-world deployments of IP-based multimedia communication systems, including:

* End-to-end identification of a communication session

* Association of session signaling and media flows, made possible by including the session identifier in media-related messages (e.g., RSVP or RTCP)

* Identification of devices taking part in the same multipoint conference

* Tracking sessions transferred from one endpoint to another

* Facilitate the recording of SIP sessions and correlating those sessions

* Logging for the purposes of accounting, billing, debugging, communication tracking (such as for security purposes in case of theft of service), etc.

# 5. Related Work in other Standards Organizations

## 5.1. Coordination with the ITU-T

IP multimedia networks are often comprised of a mix of session protocols like SIP and H.323. A benefit of the Session Identifier is that it uniquely identifies a communication session end-to-end across session protocol boundaries. Therefore, the need for coordinated standardization activities across Standards Development Organizations (SDOs) is imperative.

To facilitate this, a parallel effort is underway in the ITU-T to introduce the Session Identifier for the H.323 protocol. The ITU-T SG16 has approved contribution C.552 [5] as a work item with the intent that it be a coordinated and synchronized effort between the ITU-T and the IETF.

## 5.2. Requirements within 3GPP

3GPP identified in their Release 9 the need for a Session Identifier for O&M purposes to correlate flows in an end-to-end communication session. TS24.229 (IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP)) [6] points to the fact that the Session Identifier can be used to correlate SIP messages belonging to the same session. In the case where signaling passes through SIP entities like B2BUAs, the end-to-end session identifier indicates that these dialogs belong to the same end-to-end SIP communication session.

# 6. Security Considerations

TBD

# 7. IANA Considerations

There are no IANA considerations associated with this document.

# 8. Acknowledgments

The authors would like to acknowledge Chris Pearce for his contribution and collaboration in developing this document.

This document was prepared using 2-Word-v2.0.template.dot.

# 9. References

## 9.1. Normative References

[1] Rosenberg, J., et al., "SIP: Session Initiation Protocol", RFC 3261, June 2002.

[2] Recommendation ITU-T H.323, "Packet-based multimedia communications systems", December 2009.

## 9.2. Informative References

[3] Braden, R., et al., "Resource ReSerVation Protocol (RSVP) -- Version 1 Functional Specification", RFC 2205, September 1997.

[4] Schulzrinne, H., et al., "RTP: A Transport Protocol for Real-Time Applications", RFC 3550, July 2003.

[5] International Telecommunications Union, "End-to-End Session Identifier for IP-based Multimedia Communication Systems", March 2011, ITU-T Contribution C.552, http://ftp3.itu.int/av-arch/avc-site/2009-2012/1103_Gen/SessionID.zip.

[6] 3GPP, "IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3", 3GPP TS 24.229 10.3.0, April 2011.

## Author's Addresses

Roland Jesske
Deutsche Telekom NP
64295 Darmstadt
Heinrich-Hertz-Str. 3-7
Germany

Phone: +49 6151 628 2766
Email: R.Jesske@telekom.de

Paul E. Jones
Cisco Systems, Inc.
7025 Kit Creek Rd.
Research Triangle Park, NC 27709
USA

Phone: +1 919 476 2048
Email: paulej@packetizer.com
IM: xmpp:paulej@packetizer.com

Hadriel Kaplan
Acme Packet
71 Third Ave.
Burlington, MA 01803, USA

Email: hkaplan@acmepacket.com

Laura Liess
Deutsche Telekom NP
64295 Darmstadt
Heinrich-Hertz-Str. 3-7
Germany

Phone: +49 6151 268 2761
Email: laura.liess.dt@gmail.com

Salvatore Loreto
Ericsson
Hirsalantie 11
Jorvas 02420
Finland

Email: salvatore.loreto@ericsson.com

James Polk
Cisco Systems, Inc.
3913 Treemont Circle
Colleyville, Texas,
USA

Phone: +1 817 271 3552
Email: jmpolk@cisco.com
IM: xmpp:jmpolk@cisco.com

Parthasarathi Ravindran
Cisco Systems, Inc.
Cessna Business Park,
Kadabeesanahalli Village, Varthur Hobli,
Sarjapur-Marathahalli Outer Ring Road
Bangalore, Karnataka 560103
India

E-Mail: partr@cisco.com

Gonzalo Salgueiro
Cisco Systems, Inc.
7025 Kit Creek Rd.
Research Triangle Park, NC 27709
USA

Phone: +1 919 392 3266
Email: gsalguei@cisco.com
IM: xmpp:gsalguei@cisco.com

# SP URN
draft-pfautz-service-provider-identifier-urn-00

Abstract

This document requests a service provider identifier URN namespace.

## Conventions used in this document

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 [RFC2119].

## Status of this Memo



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## 1. Background

A number of industry bodies have identified the need for a common global service provider identifier. In the IETF the DRINKS working group has sought an identifier for the owner of objects to be provisioned in registries for the exchange of Session Establishment Data and the ENUM WG and E2MD BOF discussed the need for a service provider identifier to associate with E.164 numbers. Outside of the IETF, the need for a service provider identifier has been discussed in ITU-T Study Group 2, in the i3 Forum, the GSMA, and ATIS. In most of these discussions a preference has been expressed for a numeric identifier that might be obtained by any type of entity as opposed to only certain types of entities, e.g., carriers with a particular national legal status. Although preference was also expressed for reuse of some existing identifier, if possible, as requirements have been elaborated no current identifier seems appropriate. Thus, this document requests registration of a service provider identifier URN namespace.

## 2. Requirements for Service Provider identifier

It is suggested that Service Provider Identifiers have the following characteristics:

- They SHOULD be globally unique
- They SHOULD be numeric, at least 8 digits long
- They SHOULD be fixed length
- they SHOULD be available to any type of entity
- Entities SHOULD be able to obtain mulitple identifiers.
- Some range of identifiers SHOULD be reserved for internal entity usage.

## 3. Namespace Considerations

URN values are to be assigned by IANA on a first come first served basis. The resources to be identified are service providers, e.g., (but not limited to) SIP service providers. Entities may obtain multiple assignments. A variety of services might be supported including exchange VoIP and other traffic types.

## 4. Community Considerations

Open assignment will allow all types of entities to exchange traffic as opposed to limiting the entities that may be represented as is the case with some other identifies (e.g., ITU-T M.1400 Carrier Codes). A fixed length digit string will be more easily processed by implementations that make use of prefixing as compared to Private Enterprise Numbers or ITADs, which are integer values.

## 5. URN Namespace Definition Template

Namespace ID:

to be assigned

Registration Information:

Version 1

Date: 2011-06-04

Declared registrant of the namespace:

Name: IETF

Contact: P. Pfautz

E-mail: ppfautz@att.com

Declaration of structure:

The identifier structure is as follows:

URN:<8>DIGIT

DIGIT=%x30-39

Relevant ancillary documentation:

Identifier uniqueness considerations:

Uniqueness is guaranteed as long as the assigned number is never reassigned.

Identifier persistence considerations:

TBD

Process of identifier assignment:

First come first served by IANA.

Process for identifier resolution:

None at this time.

Rules for Lexical Equivalence:

exact digit string match

Conformance with URN Syntax:

No special considerations.

Validation mechanism:

None specified.

Scope:

Global.

## 6. Security Considerations

Any security considerations would be a product of the applications making use of the new service provider identifiers.

## 7. IANA Considerations

TBD

## 8. Normative References

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

## Author's Address

Penn Pfautz
AT&T
200 S. Laurel Ave
Middletown, NJ 07748
USA

Email: ppfautz@att.com