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Reason header filed in Session Initiation Protocol (SIP) responses
draft-jesske-dispatch-reason-in-responses-03

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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[RFC3325] Jennings, C., Peterson, J., and M. Watson, "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks", RFC 3325, November 2002.

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

[RFC3398] Camarillo, G., Roach, A., Peterson, J., and L. Ong, "Integrated Services Digital Network (ISDN) User Part (ISUP) to Session Initiation Protocol (SIP) Mapping", RFC 3398, December 2002.

[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)".

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Alert-Info URNs for the Session Initiation Protocol (SIP)
draft-liess-dispatch-alert-info-urns-03

Abstract

The Session Initiation Protocol (SIP) supports the capability to provide a reference to a specific rendering to be used by the UA when the user is alerted. This is done using the Alert-Info header. However, the reference addresses only network resources with specific rendering properties. There is currently no support for predefined standard identifiers for describing the semantics of the alerting situation or the characteristics of the alerting signal, without being tied to a particular rendering. To overcome this limitation and support new applications, a new family of URNs for use in SIP Alert-Info header fields is defined in this specification.

Requirements Language

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].

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1. Introduction

1.1. Motivation

The Session Initiation Protocol (SIP) [RFC3261] allows for user agent servers (UAS) and proxies to provide the specific ringback or ring tone to the user agent (UA). In [RFC3261] this is done by including a URI reference in the Alert-Info header field, that points to the tone. The URI reference is most commonly the HTTP URI to the audio file. On the receipt of the Alert-Info header the user agent may fetch the referenced ringback or ring tone and play it to the user.

This mechanism does not ensure interoperability when there is no common understanding of the referenced content (different countries or vendors, hearing impaired) or when the user wants his own tones configured in the end device. If caller and callee are from different countries, the understanding of the tones may vary significantly. Hearing impaired users may not sense the specific tone if it is provided as an audio file. The tone per se is also not useful for automata.

There are currently interoperability issues around the use of the Alert-Info header field when not using an external ring file. For example, consider the PBX special ring tone for an external (to the PBX) caller. Different vendors use different approaches such as: Alert-Info: <file://ring.pcm>;alert=normal where ring.pcm is a dummy file or: Alert-Info: <file://normal.ring.pcm> or: Alert-Info: <sip:normal-ringtone@example.com>. As a result, Alert-Info currently only works when the same vendor provides proxy and UA, as only then is the same "fake" proprietary URI convention used.

Another limitation of the current solution is that the referenced tones are tied to particular rendering. It is not possible to provide semantic indications or names for rendering characteristics that signals the intent and allows the recipient to decide how to render the received information in an appropriate way.

To solve the described issues, this specification defines the new URN namespace 'alert' for the Alert-Info header that allows for programmatic user interface adaptation and for conversion of equivalent alerting tones in the Public Switched Telephone Network (PSTN) when the client is a gateway. The work to standardize an Alert-Info URN will increase SIP interoperability for this header field by replacing proprietary conventions used today.

Using the 'alert' namespace provides syntax for several different application spaces, e. g.:

- Names for service indications, such as call waiting or automatic callback, not tied to any particular rendering.

- Names for common ring tones generated by PBX phones for cases such as an internal enterprise caller, external caller, ringback after a transfer failure or expiration of a hold timer, etc.

- Names for country-specific ringback tones.

- Names for things with specific renderings that aren't purely audio. They might be static icons, video sequences, text, etc.

Some advantages of a URN rather than a URI reference to a downloadable resource:

- Do not need to download it or deal with security issues associated with dereferencing.

- No formatting or compatibility issues.

- No security risk of rendering something unexpected and undesirable.

- The tone can be stored locally in whatever format and at whatever quality level is appropriate, because it is specified "by name" rather than "by value".

- It is easier to make policy decisions about whether to use it or not.

- It facilitates translation for the hearing impaired.

The downside is that if the recipient does not understand the URN then it will only be able to render a default ringback or ring tone. To provide the general awareness about the Alert-Info URNs this document provides IANA template for registering the URNs and defines several typical identifiers.

1.2. Alert-Info Header Usage Change

This specification changes the usage of the SIP Alert-Info header defined in the [RFC3261]. The Alert-Info header can be used in INVITE requests and in provisional 1xx responses excepting the 100 response. Backward compatibility issues are not expected, devices that do not understand an Alert-Info URN ignore it.

1.3. Terminology

This specification uses a number of terms to refer to the roles played by participants in, and objects of, the SIP Alerting for User Devices. A "specifier" sends an "indication" or "identifier" (a URN in an Alert-Info header) to a "renderer" which then "renders" a "signal" or "rendering" based on the indication to a human user. A "category" is a characteristic whose "values" can be used to classify indications.

This specification uses the terms "ring tone" and "ringback tone". A "ring tone" or "calling signal" (terminology used in [ITU-T E.182]) is a signal generated by the callee’s end device, advising the callee about an incoming call. A "ringback tone" or "ringing tone" (terminology used in [ITU-T E.182]) is a signal advising the caller that a connection has been made and that a ring tone is being rendered to the callee.

2. Requirements

This section discusses the requirements for an identifier to transport the semantics of the signaling situation or the characteristics of the rendering.

REQ-1: The mechanism will allow user agents (UAs) and proxies to provide in the Alert-Info SIP-header an indication which describes the semantics of the signaling situation or the characteristics of the rendering and allows the recipient to decide how to render the received information to the user.

REQ-2: The mechanism will allow the identifier to be specified "by name" rather than "by value", to enable local policy decisions whether to use it or not.

REQ-3: Indications must be able to represent a wide variety of signals, which have many largely-orthogonal characteristics.

REQ-4: Indications include subsets which have distinctly different semantics, that is, they form disjoint "value spaces". For example, some indications should describe the semantics of the signaling situation whereas others should describe the audio characteristics of the signal. This implies that there is no single set of categories that can be used as independent coordinates of the value-space of indications.

REQ-5: The set of indications must be able to support extensibility by a wide variety of organizations that are not coordinated with each other. Extensions must be able to:

- add further values to any existing category

- add further categories that are orthogonal to existing categories

- semantically subdivide the meaning provided by any existing indication

- add further value spaces of indications whose semantics are not related to existing indications, and thus, whose categories differ from and do not interact with existing categories.

REQ-6: The mechanism will be flexible, so new identifiers can be defined in the future, when SIP-applications evolve. E. g. Alert-Info URNs could identify specific media by name, such as "Beethoven's Fifth", and the end device could render some small part of it as a ring tone.

REQ-7: An indication is transmitted from a specifier to a renderer, which must base its rendering decision only on the indication. In particular, there is no multi-message negotiation process or carrying of context from one indication to the next.

REQ-8: The mechanism will allow transmission in the Alert-Info header of SIP INVITE requests and provisional 1xx responses excepting the 100 responses.

REQ-9: The renderer may be customized in ways that limit the set of signals that it can render, or it may be provided with a set of signals that have uncommon semantics. (The canonical example is a UA for the hearing-impaired.) (By REQ-7, the renderer has no way of transmitting this fact to the specifier.)

REQ-10: If the specifier and the renderer have designs that are properly coordinated, the indications must be able to reliably carry all extensions that are supported in the coordinated designs. In any other situation, it is not required from the renderer to perform the best possible rendering.

REQ-11: In any situation, the renderer must be able to perform close to the best possible rendering that it could do even the specifier had specific knowledge of the renderer's capabilities.

REQ-12: The mechanism will allow interoperability for services as call waiting, forward, transfer-recall, auto-callback, hold-recall.

REQ-13: The mechanism will allow rendering common PBX ring tone types.

REQ-14: The mechanism will allow rendering specific country ringback tones.

REQ-15: The mechanism will allow rendering tones for emergency alerts. (Use cases and values definition are not subject of this specification.)

REQ-16: The mechanism will allow rendering using other means than tones, e.g. text or images.

REQ-17: The mechanism will allow TDM gateways to map ring/ringback tones from legacy protocols to SIP at the edge of a network, e.g. national ring tones as defined in TIA/EIA-41-D and 3GPP2 A.S0014. (Use cases and values definition are not subject of this specification.)

REQ-18: The mechanism must ensure that an UA receiving Alert-Info URNs or portions of an Alert-Info URN it does not understand, it can ignore them.

REQ-19: The mechanism will allow storage of the actual encoding locally rather than fetching it.

3. Use Cases

This section describes some use cases for which the Alert-Info URN mechanism is needed today.

3.1. PBX Ring Tones

This section defines some commonly encountered ring tones on PBX or business phones. They are as follows:

3.1.1. normal

This tone indicates that the default or normal ring tone should be rendered. This is essentially a no-operation Alert-Info URN and should be treated by the UA as if no Alert-Info URN is present. This is most useful when Alert-Info header field parameters are being used. For example, in [I-D.ietf-bliss-shared-appearances], an Alert-Info header field needs to be present containing the "appearance" parameter, but no special ring tone needs to be specified.

3.1.2. external

This tone is used to indicate that the caller is external to the enterprise or PBX system. This could be a call from the PSTN or from a SIP trunk.

3.1.3. internal

This tone is used to indicate that the caller is internal to the enterprise or PBX system. The call could have been originated from another user on this PBX or on another PBX within the enterprise.

3.1.4. priority

A PBX tone needs to indicate that a priority level alert should be applied for the type of alerting specified (e.g. internal alerting).

3.1.5. short

In this case the alerting type specified (e.g. internal alerting) should be rendered shorter than normal. In contact centers, this is sometimes referred to as "abbreviated ringing" or a "zip tone".

3.1.6. delayed

In this case the alerting type specified should be rendered after a short delay. In some bridged line/shared line appearance implementations, this is used so that the bridged line does not ring at exactly the same time as the main line, but is delayed a few seconds.

3.2. Service Tones

These tones are used to indicate specific PBX and public network telephony services.

3.2.1. call-waiting

The Call Waiting Service [TS24.615] permits a callee to be notified of an incoming call while the callee is engaged in an active or held call. Subsequently, the callee can either accept, reject, or ignore the incoming call. There is an interest on the caller side to be informed about the call waiting situation on the callee side. Having this information the caller can decide whether to continue waiting for callee to pickup or better to call some time later when it is estimated that the callee could have finished the ongoing conversation. To provide this information, the callee’s UAS (or proxy) aware of the call waiting condition can add the call-waiting indication to the Alert-Info header in the 180 Ringing response. As call-waiting information may be subject to the callee's privacy concerns, the exposure of this information SHALL be done only if explicitly required by the user.

3.2.2. forward

This feature is used in a 180 Ringing response when a call forwarding feature has been initiated on an INVITE. Many PBX system implement a forwarding "beep" followed by normal ringing to indicate this. Note that a 181 response can be used in place of this URN.

3.2.3. transfer-recall

This feature is used when a blind transfer [RFC5589] has been performed by a server on behalf of the transferor and fails. Instead of failing the call, the server calls back the transferor, giving them another chance to transfer or otherwise deal with the call. This service tone is used to distinguish this INVITE from any other normal incoming call.

3.2.4. auto-callback

This feature is used when a user has utilized a server to implement an automatic callback service. When the user is available, the server calls back the user and utilizes this service tone to distinguish this from any other normal incoming call.

3.2.5. hold-recall

This feature is used when a server implements a call hold timer on behalf of an endpoint. After a certain period of time of being on hold, the user who placed the call on hold is alerted to either retrieve the call or otherwise dispose of the call. This service tone is used to distinguish this case from any other normal incoming call.

3.3. Country-specific ringback tone indications for the public telephone network

In the PSTN, different tones are used in different countries. End users are accustomed to hear the callee's country ringback tone and would like to have this feature for SIP.

4. Namespace Registration Template

This section describes the registration template for the 'alert' URN namespace identifier (NID) according to the [RFC2141] and [RFC3406]

Namespace ID: alert

Registration Information:

Registration version: 1

Registration date: TBD

Declared registrant of the namespace:

Registering organization: IETF

Designated contact: Laura Liess

Designated contact email: l.liess@telekom.de

Declaration of syntactic structure:

The Namespace Specific String (NSS) for the "alert" URNs is called alert-identifier and has a hierarchical structure. The left-most label is called "alert-category" and is separated from the right-side of the alert-identifier, the alert-indication, by a colon. The general form is urn:alert:{alert-category}:{alert-indication}.

In this specification, following alert-categories identifiers are described: "service" , "priority" , "source" , "duration", "delay" and "locale". The alert-category set can be extended in the future.

The categories are orthogonal. Any Alert-Info URN defined in this specification is syntactically valid for ring and ringback tones and can be used in INVITE requests or in provisional 1xx responses excepting the 100 response.

The alert-indications are hierarchical identifiers, consisting of one label or a sequence of labels separated by periods. The set of allowable characters is the same as that for domain names [RFC1123]. Labels are case-insensitive, but MUST be specified in all lower-case.

The ABNF [RFC5234] for the Alert -Info URNs is shown below:

```
alert-URN       = "URN:alert:" alert-identifier
alert-identifier= alert-category ":" alert-indication
alert-category  = name
alert-indication= name *("." name)
name = let-dig [ *let-dig-hyp let-dig ]
let-dig-hyp     = let-dig / "-"
let-dig         = ALPHA / DIGIT
ALPHA           = %x41-5A / %x61-7A   ; A-Z / a-z
DIGIT           = %x30-39 ; 0-9
```

Relevant ancillary documentation: None

Community considerations: The alert URN is believed to be relevant to a large cross-section of Internet users, including both technical and non-technical users, on a variety of devices and with a variety of perception capabilities. The 'alert' URN will allow Internet users to receive more information and enable them to better make decisions about accepting an offered call, or get better feedback on the progress of a call they have made. User interfaces for the perception impaired users can better render the ringback indication based on the Alert-Info URN. The assignment of identifiers is described in Section 11. The Alert-Info URN does not prescribe a particular resolution mechanism, but it is assumed that a number of different entities could operate and offer such mechanisms.

Namespace considerations: There do not appear to be other URN namespaces that serve the same need of uniquely identifying 'alert' communication and information services.

Identifier uniqueness considerations: An Alert-Info URN identifies a logical service or tone, specified in the alert-indication registration (see Section 11). Resolution of the registered URN will return a particular instance of the alert identifier. Alert identifier URNs MUST be unique for each unique indication; this is guaranteed through the registration of each alert indication within this namespace, described in Section 11.

Identifier persistence considerations: The Alert-Info URN for the same indication is expected to be persistent, as long as it is registered with IANA.

Process of identifier assignment: The process of identifier assignment is described in Section 11.

Process for identifier resolution: 'Alert-Info URNs are statically resolved according to the IANA registry.

Rules for lexical equivalence: Alert-Info URNs are compared according to case-insensitive string equality.

Conformance with URN syntax: The BNF in the 'Declaration of syntactic structure' above constrains the syntax for this URN scheme.

Validation mechanism: Validation determines whether a given string is currently a validly-assigned URN [RFC3406]. Static validation is performed based on the currently registered Alert-Info URNs at IANA.

Scope: The scope for this URN is public and global.

5. Alert-Info URN Values Definitions

5.1. Alert-category Values Definitions

Following alert-category values are defined in this document:

- service

- source

- priority

- duration

- delay

- locale

5.2. Alert-indication Values Definitions

This section describes the Alert-Info URN indication values for the alert-categories defined in this document.

For each alert-category, a default indication is defined, which is essentially a no-operation Alert-Info URN and should be treated by the UA as if no Alert-Info URN for the respective category is present. Alert-Info URN default indications are most useful when

Alert-Info header field parameters are being used. For example, in [I-D.ietf-bliss-shared-appearances], an Alert-Info header field needs to be present containing the "appearance" parameter, but no special ringtone need be specified.

5.2.1. Alert-Info URN Indication Values for the alert-category 'service'

- normal (default)

- call-waiting

- forward

- recall.callback

- recall.hold

- recall.transfer

- private.<private-name>

Examples: urn:alert:service:call-waiting or urn:alert:service:recall.transfer.

5.2.2. Alert-Info URN Indication Values for the alert-category 'source'

- unclassified (default)

- internal

- external

- friend

- family

- private.<private-name>

Examples: urn:alert:source:external.

5.2.3. Alert-Info URN Indication Values for the alert-category 'priority'

- normal (default)

- low

- high

- private.<private-name>

Examples: urn:alert:priority:high.

5.2.4. Alert-Info URN Indication Values for the alert-category 'duration'

- normal (default)

- short

- long

- private.<private-name>

Examples: urn:alert:duration:short.

5.2.5. Alert-Info URN Indication Values for the alert-category 'delay'

- none (default)

- yes

- private.<private-name>

Examples: urn:alert:delay:yes .

5.2.6. Alert-Info URN Indication Values for the alert-category 'locale'

- default (default)

- country.<ISO 3166-1 country code>

- private.<private-name>

The ISO 3166-1 country code [ISO 3166-1] is used to inform the UA on the other side of the call that a country-specific rendering should be used. For example, to indicate ringback tones from South Africa, the following URN would be used: <urn:alert:locale:country.za>.

6. Extensibility Rules

6.1. General Extensibility Rules

TBD

6.2. Extensions Rules for Independent Organizations

The "private.<private-name>" syntax is for extensions specific to independent organizations. The "<private-name>" is used in the form of a "reverse FQDN" such as is used for Java package names. This gives a way of assigning unique names without the need for a new registry. The namespace for each alert category is independent. Those assigning new names must ensure they are in a position to assign names uniquely for the FQDN they choose. For example, a private company might want to define:
urn:alert:source:private.com.example.customer

Adding new categories and adding alert-indication values other than via the "private" mechanism is standards action.

7. Combinations of URNs

In some cases, more than one URN will need to be specified to fully define a particular tone. This is done by including multiple URNs separated by a comma. For example, an internal, priority call could be indicated by Alert-Info: <urn:alert:source:internal>, <urn:alert:priority:high>. A priority call waiting tone could be indicated by Alert-Info: <urn:alert:service:call-waiting>, <urn:alert:priority:high>.

The categories are orthogonal. Any Alert-Info URN defined in this specification is syntactically valid for ring and for ringback and can be used in an INVITE or in provisional 1xx responses excepting the 100 response. There can be at most one instance of each alert-category in an Alert-Info header. In principle any combination of Alert-Info URNs with different "alert-category" is valid and can be used for either ring or ringback, though some combinations may not make sense. The receiving UA should make the decision about what to render to the user and what device it is rendered on depending on the value of the Alert-Info URN and the kind of the received message (INVITE or provisional response). Typically, the same UA will do the rendering of a particular Alert-Info URN received in an INVITE differently from the rendering of the same Alert-Info URN received in a provisional response. The exact way in which the various categories are combined for rendering is left as an implementation issue. The implementation is free to ignore any or all received

Alert-Info URNs.

8. Priority Rules within Combinations of URNs

TBD

9. User Agent Behaviour

Upon receiving a SIP INVITE request or a SIP provisional response with an Alert-Info header that contains a single or multiple Alert-Info URNs, the User Agent (UA) attempts to match the received Alert-Info URNs with the known indications or indication combinations. The User Agent (UA) ignores the Alert-Info URNs for which no match is found and proceeds with the normal operation. If one or multiple URNs match(es) a known indication or a known indication combination, the User Agent (UA) renders the indication or the indication combination to the user accordingly. The User Agent (UA) is responsible for the non disturbing rendering if multiple indications and network resources are to be rendered simultaneously.

10. Proxy Behaviour

A SIP proxy MAY add a URN or multiple URNs to the Alert-Info header in a SIP request or a provisional 1xx response excepting 100 response when it needs to provide additional information about the call or about the provided service. A SIP Proxy SHOULD NOT add a mixture of Alert-Info URNs and URIs to the Alert-Info header that may cause disturbing rendering interference at the recipient's User Agent (UA).

Following example shows both the network audio resource referenced by the HTTP URI and the URN indication for the call-waiting service transported by the Alert-Info header in a 180 Ringing provisional response.

```
Alert-Info: <http://www.example.com/sound/moo.wav>,
 <urn:alert:service:call-waiting>
```

11. IANA Considerations

This section registers a new URN namespace identifier (NID) in accordance with RFC 3406 with the registration template provided in Section 4 .

11.1. New alert identifiers

Alert URN identifiers are identified by labels managed by IANA, according to the processes outlined in [RFC5226] in a new registry called "Alert URN Labels". Thus, creating a new Alert-Info URN identifier requires IANA action. The policy for adding a new alert category is ’Standards Action’. (This document defines the alert categories ’service’, ’source’, ’priority’, ’duration’, ’delay’ and ’locale’.) The policy for assigning labels to alert-indications and the rules to combine them may differ for each alert-category and MUST be defined by the document describing the corresponding alert category. The entries in the registration table have the following format:

alert-category/ alert-identifier	Reference	Description
foo	RFCxyz	Description of the ’foo’ alert-category
foo:bar	RFCabc	Description of the ’foo:bar’ alert-identifier

Each alert-category or alert-indication label MUST NOT exceed 27 characters.

11.2. Initial IANA Registration

11.2.1. The "service" alert-category and alert-identifiers

The following table contains the initial IANA registration for the "service" alert-category and alert-identifiers. The value of this indicator is set to a value different from "normal" if the caller or callee is informed that a specific telephony service which has been initiated.

alert-category/ alert-identifier	Reference	Description
service	RFC XXXX	Alert-category for "service" alert-identifiers.
service:normal	RFC XXXX	Normal ring /rinback rendering (default value).
service:call-waiting	RFC XXXX	Call waiting was initiated at the other side of the call.
service:forward	RFC XXXX	Call has been forwarded.
service:recall.calback	RFC XXXX	Recall due to callback.
service:recall.hold	RFC XXXX	Recall due to call hold.
service:recall.transfer	RFC XXXX	Recall due to callback.
service:private.<private-name>	RFC XXXX	Reserved for private extensions.

11.2.2. The "source" alert-category and alert-identifiers

The following table contains the initial IANA registration for the "source" alert-category and alert-identifiers. The value of this indicator provides information about the user at the other side of the call.

alert-category/ alert-identifier	Reference	Description
source	RFC XXXX	Alert-category for "source" alert-identifiers
source:unclassified	RFC XXXX	Unclassified ring /rinback rendering (default value)
source:internal	RFC XXXX	User at the other side of the call is internal to the enterprise or PBX system.
source:external	RFC XXXX	User at the other side of the call is internal to the enterprise or PBX system.
source:friend	RFC XXXX	User at the other side of the call is a friend.
source:family	RFC XXXX	User at the other side of the call is a family member.
source:private.<private-name>	RFC XXXX	Reserved for private extensions.

11.2.3. The "priority" alert-category and alert-identifiers

The following table contains the initial IANA registration for the "priority" alert-category and alert-identifiers. The value of this indicator provides information about the priority the alerted user should give to the call.

alert-category/ alert-identifier	Reference	Description
priority	RFC XXXX	Alert-category for "priority" alert-identifiers.
priority:normal	RFC XXXX	Normal ring /rinback rendering (default value).
priority:low	RFC XXXX	Low priority call.
priority:high	RFC XXXX	High priority call.
priority:private.<private-name>	RFC XXXX	Reserved for private extensions.

11.2.4. The "duration" alert-category and alert-identifiers

The following table contains the initial IANA registration for the "duration" alert-category and alert-identifiers. The value of this indicator provides information about the duration of the alerting signals compared to the default alerting signals.

alert-category/ alert-identifier	Reference	Description
duration	RFC XXXX	Alert-category for "duration" alert-identifiers
duration:normal	RFC XXXX	Normal ring /rinback rendering (default value)
duration:short	RFC XXXX	Shorter than normal
duration:long	RFC XXXX	Longer than normal
duration:private.<private-name>	RFC XXXX	Reserved for private extensions.

11.2.5. The "delay" alert-category and alert-identifiers

The following table contains the initial IANA registration for the "delay" alert-category and alert-identifiers. The value of this indicator provides information about the delay of the alerting signals.

alert-category/ alert-identifier	Reference	Description
delay	RFC XXXX	Alert-category for "delay" alert-identifiers
delay:none	RFC XXXX	Immediate alerting (default value)
delay:yes	RFC XXXX	Delayed alerting
delay:private.<private-name>	RFC XXXX	Reserved for private extensions.

11.2.6. The "locale" alert-category and alert-identifiers

The following table contains the initial IANA registration for the "locale" alert-category and alert-identifiers. The value of this indicator provides information about the location of the user at the other side of the call.

alert-category/ alert-identifier	Reference	Description
locale	RFC XXXX	Alert-category for "locale" alert-identifiers
locale:default	RFC XXXX	Alerting not location specific (default value)
locale:country.<ISO 3166-1 country code>	RFC XXXX	Country-specific alerting
locale:private.<private-name>	RFC XXXX	Reserved for private extensions.

12. Internationalization Considerations

The alert-identifier labels are protocol elements [RFC3536] and are not normally seen by users. Thus, the character set for these elements is restricted, as described in Section 11.

13. Security Considerations

As an identifier, the alert URN does not appear to raise any particular security issues. The indications described by the 'alert' URN are meant to be well-known, so privacy considerations do not apply to the URN.

Provision of the specific indications from callee to caller may raise privacy issues. Such provision SHALL always be explicitly authorised by the callee.

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