

SIP "cause" URI Parameter for Service Number Translation

Abstract

[RFC 4458](#) (regarding SIP URIs for applications) defines a "cause" URI parameter, which may appear in the Request-URI of a SIP request, that is used to indicate a reason why the request arrived to the User Agent Server (UAS) receiving the message. This document updates [RFC 4458](#) by creating a new predefined value for the "cause" URI parameter to cover service number translation for cases of retargeting due to specific service action leading to the translation of a called service access number. This document also provides guidance, which was missing in [RFC 4458](#), for using the "cause" URI parameter within the History-Info header field, since this use is mandatory in some IP networks' implementations.

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1. Introduction, Terminology, and Overview

[RFC4458] defines a mechanism to identify retargeting due to call forwarding supplementary services. The "cause" URI parameter in the target URI identifies the reason for retargeting and has defined values equivalent to the TDM (Time Division Multiplexing) Redirecting Reasons [[ITU-T_Q.763](#)]. The concept of "retargeting" is defined in [[RFC7044](#)].

In the Public Switched Telephone Network (PSTN) / Integrated Services Digital Network (ISDN), there is another kind of retargeting introduced by the Intelligent Network (IN) services based on a translation of the called number as mentioned in [[ITU-T_Q.1214](#)]. Indeed, IN aims to ease the introduction of new services (i.e., Universal Personal Telecommunication (UPT), Virtual Private Network (VPN), Freephone, etc.) based on greater flexibility and new

capabilities as described in [ITU-T_I.312_Q.1201]. For these IN services, ISDN User Part (ISUP) introduced the "called IN number" and the "original called IN number" parameters to capture the information of the requested service access number prior its translation [ITU-T_Q.763].

The term "service access number" is used in this specification to refer to the dialable number by which a specific service is reached. This special number is not a globally routable number; therefore, it needs to be translated into a routable SIP or tel URI to process the session establishment.

This specification proposes a solution to allow the identification of well-known services, such as premium or toll-free services that perform service access number translation, and to enable interworking with SIP signaling with the ISUP called IN number and original called IN number parameters.

The mechanism will allow a SIP network to insert and convey the service access number requested before its translation to the final destination.

In order to provide full call forwarding or access number translation services, usage of the "cause" URI parameter is only relevant within the History-Info header field defined in [RFC7044]. Because this relation has not been described in [RFC4458], this document provides guidance for using the "cause" URI parameter in conjunction with the History-Info header field.

This document also answers a need expressed by the Third Generation Partnership Project (3GPP) [TS.3GPP.24.229] to identify the service access number. The procedures it defines are intended for networks that use 3GPP-defined services. Their use is undefined for other networks.

2. Solution

A new value for the "cause" URI parameter of the 'sip:' or 'sips:' URI schemes is defined for the "Service number translation" use case. This value may be used in a 'sip:' or 'sips:' URI inserted in the Request-URI and in the History-Info header field [RFC7044] when the URI is issued from a retargeting or a service access number translation by a specific service similar to PSTN/ISDN IN services that is not a call forwarding service.

As defined in [RFC4458], the "cause" URI parameter must be encoded in the new target URI when generated by the service.

The ABNF grammar [RFC5234] for the cause-param and target-param parameters from [RFC4458] is summarized below (including updates described in [Err1409]). The Status-Code is defined in [RFC3261].

target-param = "target=" pvalue

cause-param = "cause=" Status-Code

The following value for this URI parameter is added to the existing ones:

Cause		Value
Service number translation		380

3. Guidelines

In order to help implementation of this solution for conveyance of the service access number, this document proposes guidance for usage of the "cause" URI parameter: guidance for the usage of the "cause" URI parameter in the request history information (in [Section 3.1](#)) and guidance for processing a service number translation service using the new "cause" URI parameter value (in [Section 3.2](#)).

3.1. Interaction with Request History Information

The History-Info header field defined in [RFC7044] specifies a means of providing the UAS and User Agent Client (UAC) with information about the retargeting of a request. This information includes the initial Request-URI and any retargeted URIs. This information is placed in History-Info headers as the request is retargeted and, upon reaching the UAS, is returned in certain responses. The History-Info header field enables many enhanced services by providing the information as to how and why a SIP request arrives at a specific application or user and to keep this information throughout the signaling path even when successive applications are involved.

When a proxy inserts a URI containing the "cause" URI parameter defined in [RFC4458] into the Request-URI of a forwarded request (per [RFC7044]), the proxy must also copy this new Request-URI within a History-Info header field entry into the forwarded request; thus, the URI in that entry includes the "cause" URI parameter. Therefore, even if the Request-URI is replaced as a result of rerouting by a downstream proxy, the History-Info header field will still contain these parameters, which can be of use to the UAS. Note that if a proxy does not support generation of the History-Info header field or

if a downstream proxy removes the History-Info header fields, an application will only have access to the "cause" URI parameter if the request is not subsequently retargeted (i.e., it will be contained only in the Request-URI in the incoming request). The implications of the solution are further discussed in [Section 3.2](#).

In order to be able to filter specific entries among the history information, header field parameters have been defined in [\[RFC7044\]](#). In particular, the "mp" and "rc" header field parameters have the following definitions:

- o When the new target was determined based on a mapping to a user other than the target user associated with the Request-URI being retargeted, the "mp" header field parameter is added. This allows the identification of retargets that are the result of an application decision on a user's behalf and also retargets that are the result of an internal decision made by an application.
- o The "rc" header field parameter is added when the new target represents a change in Request-URI, while the target user remains the same.

These header field parameters can be used in conjunction with the new "cause" URI parameter for certain applications, an example of which is provided in [Section 4](#).

When using the History-Info header field in conjunction with the "cause" URI parameter in a Request-URI, it is important to consider that the "cause" URI parameter is not the same parameter as the "cause" header field parameter included in the Reason header [\[RFC3326\]](#). The "cause" header field parameter of the Reason header field should be added to a History-Info entry only when the retargeting is due to a received SIP response.

3.2. Handling and Processing the Service Number Translation "cause" URI Parameter Value

At the Application Server:

When an application receiving a request that is addressed to a service access number changes the Request-URI into a routable number, it should insert within this new Request-URI a "cause" URI parameter value set to 380 "Service number translation". Following the process described in [\[RFC7044\]](#), the application server adds a new History-Info header field entry including the new Request-URI value including the "cause" URI parameter. It is

also possible for an application server to add a "target" URI parameter as defined in [RFC4458] with the initial value of the Request-URI received by the application server.

Note that if the new Request-URI is further replaced by a downstream proxy for any reason and if the History-Info header field is not supported, the information of the service access number initially requested would be lost. Thus, it is strongly recommended to support the History-Info header field all along the signaling path.

At the UAS:

When the UAS receiving the request wants to retrieve the service access number by which it has been reached, first it should look for the "cause" URI parameter value 380 in the History-Info header field. This History-Info entry should also contain an "mp" or "rc" header field parameter so that the UAS can find the requested service number in the History-Info entry having an index parameter value that matches this "mp" or "rc" header field parameter value. If, for any reason, there is no "mp" or "rc" header field parameter in the identified History-Info entry, the UAS can find the requested service number in the preceding History-Info entry.

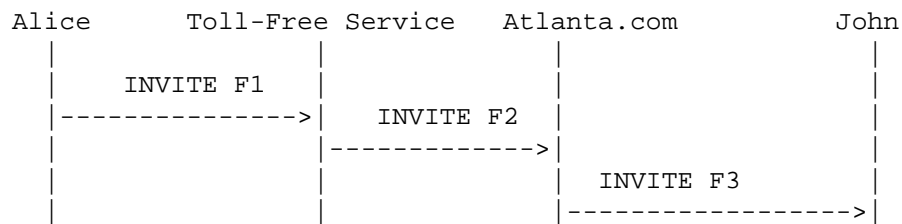
If the History-Info header is not supported or has been removed by a proxy for any reason, the UAS might be able to find the requested service access number before translation in either of the following ways, but there is no guarantee:

- o If the UAS is the direct target of the request coming from the application, the UAS ought to be able to find the service access number in the "target" URI parameter of the Request-URI if there is also a "cause" URI parameter set to 380 in this Request-URI.
- o If there is no "cause" URI parameter set to 380 in the Request-URI and there is no History-Info header field, the UAS will not be able to reliably retrieve the service access number before translation. Some existing implementations are known to extract the number from the To header field. While that approach may work in some situations, it will not work in the general case because the To header field value is sometimes changed by intermediaries, and such a change is not always detectable.

4. Example

In this section, an example is provided to illustrate the application of the new cause-param value.

In this example, Alice calls her bank customer care. John is the person at the call center that answers the call. John is in a call center that manages several toll-free services, and he needs to know for which service Alice is calling in order to provide the appropriate welcome speech.



* Rest of flow not shown *

Figure 1: Service Access Number Translation Example

Message Details

F1 INVITE [2001:db8:9::2] -> Toll-Free Service

In the initial request, the Request-URI contains the toll-free number dialed by Alice.

```

INVITE sip:+18005551002@example.com;user=phone SIP/2.0
Via: SIP/2.0/TCP [2001:db8:9::2]:5060;branch=z9hG4bK74bf
From: Alice <sip:+15551001@example.com;user=phone>;tag=9fxced76sl
To: <sip:+18005551002@example.com;user=phone>
Call-ID: c3x842276298220188511
CSeq: 1 INVITE
Max-Forwards: 70
Contact: <sip:alice@[2001:db8:9::2]>
Content-Type: application/sdp
Content-Length: <appropriate value>
  
```

[SDP Not Shown]

F2 INVITE Toll-Free Service -> Atlanta.com

The toll-free application receives the request and translates the service number into a routable number toward the call center. The Request-URI is changed, and, in the new Request-URI, a "cause" URI parameter set to 380 is added. As there was no History-Info header field in the received request, the application creates a History-Info header with two entries: one for the received Request-URI and one for the new Request-URI.

```
INVITE sip:+15555551002@atlanta.com;cause=380;user=phone SIP/2.0
Via: SIP/2.0/TCP [2001:db8:a::2];branch=z9hG4bK-ik8
Via: SIP/2.0/TCP [2001:db8:9::2]:5060;branch=z9hG4bK74bf
From: Alice <sip:+15551001@example.com;user=phone>;tag=9fxced76sl
To: <sip:+18005551002@example.com;user=phone>
Call-ID: c3x842276298220188511
CSeq: 1 INVITE
Max-Forwards: 69
Supported: histinfo
History-Info: <sip:+18005551002@example.com;user=phone>;index=1
History-Info: <sip:+15555551002@atlanta.com;cause=380;user=phone>;
              index=1.1;mp=1
Contact: <sip:alice@[2001:db8:9::2]>
Content-Type: application/sdp
Content-Length: <appropriate value>
```

[SDP Not Shown]

F3 INVITE Atlanta.com -> John

The call center proxy routes the received request to John's IP address by changing the Request-URI. When changing the Request-URI, the proxy adds a new entry in the History-Info header field.

```
INVITE sip:john@[2001:db8:b::2] SIP/2.0
Via: SIP/2.0/TCP [2001:db8:b::3]:5060;branch=z9hG4bKpxk7g
Via: SIP/2.0/TCP [2001:db8:a::2];branch=z9hG4bK-ik8
Via: SIP/2.0/TCP [2001:db8:9::2]:5060;branch=z9hG4bK74bf
From: Alice <sip:+15551001@example.com;user=phone>;tag=9fxced76sl
To: <sip:+18005551002@example.com;user=phone>
Call-ID: c3x842276298220188511
CSeq: 1 INVITE
Max-Forwards: 68
Supported: histinfo
History-Info: <sip:+18005551002@example.com;user=phone>;index=1
History-Info: <sip:+15555551002@atlanta.com;cause=380;user=phone>;
              index=1.1;mp=1
History-Info: <sip:john@[2001:db8:b::2]>;index=1.1.1;rc=1.1
Contact: <sip:alice@[2001:db8:9::2]>
Content-Type: application/sdp
Content-Length: <appropriate value>
```

[SDP Not Shown]

NOTE: Line breaks are for display purpose only

5. IANA Considerations

[RFC4458] defines a "cause" URI parameter specified as having predefined values. This document defines a new value for the "cause" URI parameter: 380.

IANA has modified the existing row for the "cause" URI parameter to add a reference to this document under the "SIP/SIPS URI Parameters" subregistry within the "Session Initiation Protocol (SIP) Parameters" registry:

Parameter Name	Predefined Values	Reference
-----	-----	-----
cause	Yes	[RFC4458][RFC8119]

6. Security Considerations

The security considerations in [RFC4458] apply.

If a privacy level of 'header' is requested in the Privacy header field as described in [RFC3323], the "cause" URI parameter must be removed from the Request-URI to maintain the network-provided privacy requested. Privacy of the parameters, when they form part of a URI within the History-Info header field, is covered in [RFC7044].

7. References

7.1. Normative References

- [RFC3261] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M., and E. Schooler, "SIP: Session Initiation Protocol", RFC 3261, DOI 10.17487/RFC3261, June 2002, <<http://www.rfc-editor.org/info/rfc3261>>.
- [RFC3323] Peterson, J., "A Privacy Mechanism for the Session Initiation Protocol (SIP)", RFC 3323, DOI 10.17487/RFC3323, November 2002, <<http://www.rfc-editor.org/info/rfc3323>>.
- [RFC3326] Schulzrinne, H., Oran, D., and G. Camarillo, "The Reason Header Field for the Session Initiation Protocol (SIP)", RFC 3326, DOI 10.17487/RFC3326, December 2002, <<http://www.rfc-editor.org/info/rfc3326>>.
- [RFC7044] Barnes, M., Audet, F., Schubert, S., van Elburg, J., and C. Holmberg, "An Extension to the Session Initiation Protocol (SIP) for Request History Information", RFC 7044, DOI 10.17487/RFC7044, February 2014, <<http://www.rfc-editor.org/info/rfc7044>>.
- [TS.3GPP.24.229] 3GPP, "IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3", 3GPP TS 24.229 13.6.0.0, January 2015.

7.2. Informative References

- [Err1409] RFC Errata, Erratum ID 1409, [RFC 4458](#).
- [ITU-T_I.312_Q.1201]
ITU-T, "Principles of Intelligent Network Architecture",
ITU-T Recommendation I312/Q.1201, October 1992.
- [ITU-T_Q.1214]
ITU-T, "Distributed Functional Plane For Intellignet
Network CS-1", ITU-T Recommendation Q.1214, October 1995.
- [ITU-T_Q.763]
ITU-T, "Signalling System No. 7 -- ISDN User Part formats
and codes", ITU-T Recommendation Q.763, December 1999.
- [RFC4458] Jennings, C., Audet, F., and J. Elwell, "Session
Initiation Protocol (SIP) URIs for Applications such as
Voicemail and Interactive Voice Response (IVR)", [RFC 4458](#),
DOI 10.17487/RFC4458, April 2006,
<<http://www.rfc-editor.org/info/rfc4458>>.
- [RFC5234] Crocker, D., Ed. and P. Overell, "Augmented BNF for Syntax
Specifications: ABNF", STD 68, [RFC 5234](#),
DOI 10.17487/RFC5234, January 2008,
<<http://www.rfc-editor.org/info/rfc5234>>.

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