**ATIS-1000099**

ATIS Standard on

**Robocall Call Blocking Notification**

**Alliance for Telecommunications Industry Solutions**

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**Abstract**

Abstract text here.

**Foreword**

The Alliance for Telecommunications Industry Solutions (ATIS) is a global standards development and technical planning organization that develops and promotes worldwide technical and operations standards for information, entertainment, and communications technologies. ATIS’ diverse membership includes key stakeholders from the Information and Communications Technologies (ICT) industry – wireless and wireline service providers, equipment manufacturers, broadband providers, software developers, VoIP providers, consumer electronics companies, public safety agencies, and internet service providers. ATIS is also a founding partner and the North American Organizational Partner of the Third Generation Partnership Project (3GPP), the global collaborative effort that has developed the Long-Term Evolution (LTE) and LTE-Advanced wireless specifications.

ATIS’ Packet Technologies and Systems Committee (PTSC) develops standards related to services, architectures, signaling, network interfaces, next generation carrier interconnect, cybersecurity, lawful intercept, and government emergency telecommunications service within next generation networks. As networks transition to all-IP, PTSC will evaluate the impact of this transition and develop solutions and recommendations where necessary to facilitate and reflect this evolution.

The SIP Forum is an IP communications industry association that engages in numerous activities that promote and advance SIP-based technology, such as the development of industry recommendations, the SIPit, SIPconnect-IT, and RTCWeb-it interoperability testing events, special workshops, educational seminars, and general promotion of SIP in the industry. The SIP Forum is also the producer of the annual SIP Network Operators Conference (SIPNOC), focused on the technical requirements of the service provider community. One of the Forum's notable technical activities is the development of the SIPconnect Technical Recommendation – a standards-based SIP trunking recommendation for direct IP peering and interoperability between IP Private Branch Exchanges (PBXs) and SIP-based service provider networks. Other important Forum initiatives include work in Video Relay Service (VRS) interoperability, security, Network-to-Network Interoperability (NNI), and SIP and IPv6.

Suggestions for improvement of this document are welcome. They should be sent to the Alliance for Telecommunications Industry Solutions, PTSC, 1200 G Street NW, Suite 500, Washington, DC 20005, and/or to the SIP Forum, 733 Turnpike Street, Suite 192, North Andover, MA, 01845.

The mandatory requirements are designated by the word *shall* and recommendations by the word *should*. Where both a mandatory requirement and a recommendation are specified for the same criterion, the recommendation represents a goal currently identifiable as having distinct compatibility or performance advantages. The word *may* denotes an optional capability that could augment the standard. The standard is fully functional without the incorporation of this optional capability.

The **ATIS/SIP Forum IP-NNI Task Force** under the **ATIS** **Packet Technologies and Systems Committee (PTSC)** and the **SIP Forum** **Technical Working Group (TWG)** was responsible for the development of this document.

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# Scope, Purpose, & Application

This document defines a mechanism providing real-time notification in the backward call direction (towards the calling party), that the associated call was blocked by the indicated voice service provider due to analytics-based call processing. It ensures that voice service providers can continue to use analytics to block calls suspected to be illegal, fraudulent or for other reasons undesirable, while providing immediate notice to callers.

To provide such notification, this specification defines a profile of the SIP 603 defined in RFC 3261, *SIP: Session Initiation Protocol*, response referred to herein as “603+”. A 603+ response is differentiated from a 603 response in two ways.

1. Its status line[[1]](#footnote-1) uses a unique reason phrase “Network Blocked”, rather than the Default Reason Phrase “Decline” specified in RFC 3261 [Ref 1].
2. It contains a SIP Reason Header defined in RFC 3326, *The Reason Header Field for the Session Initiation Protocol (SIP)*, encoded per this specification.

Any 603 response received without this syntax included should be treated as currently handled today.

This standard is primarily developed for and adoption by US voice service providers. Other countries may adopt these standards and they may be implemented through bilateral agreement with business partners in the US pursuant to their business agreement. This standard is not precluded from being used internationally.

# Normative References

The following standards contain provisions which, through reference in this text, constitute provisions of this Standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this Standard are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below.

[Ref 1] IETF RFC 3261, *SIP: Session Initiation Protocol*.[[2]](#footnote-2)

[Ref 2] IETF RFC 3326, *The Reason Header Field for the Session Initiation Protocol (SIP).*2

[Ref 3] IETF RFC 6432, *Carrying Q.850 Codes in Reason Header Fields in SIP (Session Initiation Protocol) Responses*.2

[Ref 4] IETF RFC 8606, *ISDN User Part (ISUP) Cause Location Parameter for the SIP Reason Header Field.*2

# Definitions, Acronyms, & Abbreviations

For a list of common communications terms and definitions, please visit the *ATIS Telecom Glossary*, which is located at < <https://glossary.atis.org/> >.

## Definitions

* **Analytics:** analysis of a call request to determine how likely it is to be fraudulent or undesirable for reasons not specific to, or likely to reveal the identity of, the intended recipient.

## Acronyms & Abbreviations

|  |  |
| --- | --- |
| ATIS | Alliance for Telecommunications Industry Solutions |
| AVP | Attribute-Value Pair |
| DNS | Domain Name System |
| SIP | Session Initiation Protocol |
| UA | User Agent |

# Blocking Call Processing

## Data Analytics Blocking

If a service provider blocks a call due to analytics, the service provider shall reply with a SIP 603 response encoded per this specification.

This capability defines a profile of SIP 603 [Ref 1], referred to as 603+. 603+ is differentiated from a 603 response in that it contains a format where it has a status line of 603 Network Blocked, with a Reason header containing an indication of the reason for the blocking and a version of this specification “analytics1”, and the contact information of the entity responsible for the call having been blocked. The contact information allows the calling party to learn why the call was blocked, and potentially to seek redress.

Any 603 received without this syntax included should be treated as currently handled today.

### Formal Specification of Reason Header Syntax for 603+

Reason headers used in 603+ responses shall comply with the syntax specified in RFC 3326 [Ref 2]; with the following restrictions per the profile of their usage defined in this specification.

The “protocol” parameter shall be either “ “Q.850”, as specified in RFC 6432, *Carrying Q.850 Codes in Reason Header Fields in SIP (Session Initiation Protocol) Responses*, or “SIP”.

Exactly one “protocol-cause” parameter shall be included. Its value is “21” if protocol parameter is set to “Q.850” or shall be “603” if protocol parameter is set to “SIP”.

Exactly one “reason-text” parameter shall be included. Its value of the “text” parameter shall be a quoted list of attribute-value pairs (AVPs). The semicolon character (“;”) shall be used to separate AVPs. The equals character (“=”) shall be used to separate attributes and values. Each AVP shall appear at most once. The ‘v’ attribute shall be included, and shall be the first AVP in the “reason-text” parameter. Supported AVPs are specified in Table 2.

The “url” parameter value shall be a valid HTTPS URL resolvable via the public DNS. The “tel” parameter value shall be a valid telephone number in global E.164 format. The “email” parameter value shall be a valid email address. The “text” value shall include at least one “url”, “tel”, or “email” parameter. The “id” parameter value shall be a string containing only alpha, digit, underscore, and/or dash characters and shall have a length of no more than 64 characters.

The “Reason” header value shall include exactly one “location” parameter, as specified in RFC 8606, *ISDN User Part (ISUP) Cause Location Parameter for the SIP Reason Header Field*. The “location” parameter shall have a value of “RLN” when blocking occurred in the network serving the called party. The “location” parameter shall have a value of “TN” when blocking occurred in a transit network. The “location” parameter shall have a value of “LN” when blocking occurred in the originating network. The “location” parameter shall have a value of “RPN” when blocking occurred in the private network serving the called party. The “location” parameter shall have a value of “LPN” when blocking occurred in the originating private network.

Table 4‑1: “Reason” header parameters

|  |  |  |
| --- | --- | --- |
| **Parameter** | **Mandatory** | **Value** |
| “protocol-cause” | Yes | “21”, “603” |
| “reason-text” | Yes | See Table 2 |
| “location” | Yes | “LN”, “TN”, “LPN”, “RPN”, or “RLN” |

Table 4‑2: “reason-text” value parameters attribute-value pairs (AVPs)

|  |  |  |
| --- | --- | --- |
| **Attribute** | **Mandatory** | **Value** |
| “v” | Yes | “analytics1” |
| “url” | If neither “tel” nor “email” are included | Valid HTTPS URL for the calling party to visit for redress |
| “tel” | If neither “url” nor “email” are included | Valid E.164 formatted telephone number for the calling party to call for redress |
| “email” | If neither “url” nor “tel” are included | Valid email address for the calling party to email for redress |
| “id” | No | Identifier used by the SP that blocked the call to facilitate redress (e.g., call identifier, blocking reason identifier, network segment identifier, etc.) |

### Examples of Reason Header Syntax for 603+

Example “Reason” headers are illustrated below:

Reason: Q.850;protocol-cause=21;reason-text="v=analytics1;url=https://example.com";location=LN

Reason: SIP;protocol-cause=603;reason-text="v=analytics1;url=https://example.com";location=LN

Reason: Q.850;protocol-cause=21;reason-text="v=analytics1;url=https://example.com;id=29016905-3bed-4c98-9423-03041160cc67";location=LN

Reason: SIP;protocol-cause=603;reason-text="v=analytics1;url=https://example.com;id=29016905-3bed-4c98-9423-03041160cc67";location=LN

Reason: Q.850;protocol-cause=21;reason-text="v=analytics1;email=support@example.com";location=LN

Reason: SIP;protocol-cause=603;reason-text="v=analytics1;email=support@example.com";location=LN

Reason: Q.850;protocol-cause=21;reason-text="v=analytics1;email=support@example.com;id=29016905-3bed-4c98-9423-03041160cc67";location=LN

Reason: SIP;protocol-cause=603;reason-text="v=analytics1;email=support@example.com;id=29016905-3bed-4c98-9423-03041160cc67";location=LN

Reason: Q.850;protocol-cause=21;reason-text="v=analytics1;tel=+12155551212";location=LN

Reason: SIP;protocol-cause=603;reason-text="v=analytics1;tel=+12155551212";location=LN

Reason: Q.850;protocol-cause=21;reason-text="v=analytics1;tel=+12155551212;id=29016905-3bed-4c98-9423-03041160cc67";location=LN

Reason: SIP;protocol-cause=603;reason-text="v=analytics1;tel=+12155551212;id=29016905-3bed-4c98-9423-03041160cc67";location=LN

Reason: Q.850;protocol-cause=21;reason-text="v=analytics1;url=https://example.com;email=support@example.com;tel=+12155551212";location=LN

Reason: SIP;protocol-cause=603;reason-text="v=analytics1;url=https://example.com;email=support@example.com;tel=+12155551212";location=LN

Reason: Q.850;protocol-cause=21;reason-text="v=analytics1;url=https://example.com;email=support@example.com;tel=+12155551212;id=29016905-3bed-4c98-9423-03041160cc67";location=LN

Reason: SIP;protocol-cause=603;reason-text="v=analytics1;url=https://example.com;email=support@example.com;tel=+12155551212;id=29016905-3bed-4c98-9423-03041160cc67";location=LN

### Transit Network Processing

The transit network shall transparently forward toward the Calling Party a SIP 603 response received. It shall not change the response code from 603 to a different value, or modify any part of the 603 response other than headers used to forward it (e.g., Via headers) except as required by its interconnect agreement or other contractual arrangements with the downstream network. It shall not retry the associated request. Analytic processing in a transit network MAY generate a 603+.

### Originating Network Processing

The originating network shall forward the SIP 603 response message toward the SIP UA that originated the call session. It shall not, except as required by contractual agreements with the entity responsible for the originating SIP UA or network (e.g., Enterprise) that must be traversed to reach the originating SIP UA, modify the 603 response.

If the status line of the 603 response identifies it as a “603+” response as defined by this specification, but its Reason Header does not adhere to the syntax in Clause 4.1.1, the Reason header shall be removed prior to forwarding the 603 response onward toward the originating SIP UA.

1. The first line of a SIP response message is called the “status line”. [↑](#footnote-ref-1)
2. This document is available from the Internet Engineering Task Force (IETF) at: < [http://www.ietf.org](http://www.ietf.org/) >. [↑](#footnote-ref-2)