ATIS-1000099

Robocall Call Blocking Notification

JOINT STANDARD

As a leading technology and solutions development organization, the Alliance for Telecommunications Industry Solutions (ATIS) brings together the top global ICT companies to advance the industry's most pressing business priorities. ATIS' nearly 200 member companies are currently working to address the All-IP transition, 5G, network functions virtualization, big data analytics, cloud services, device solutions, emergency services, M2M, cyber security, network evolution, quality of service, billing support, operations, and much more. These priorities follow a fast-track development lifecycle — from design and innovation through standards, specifications, requirements, business use cases, software toolkits, open source solutions, and interoperability testing.

ATIS is accredited by the American National Standards Institute (ANSI). The organization is the North American Organizational Partner for the 3rd Generation Partnership Project (3GPP), a founding Partner of the oneM2M global initiative, a member of the International Telecommunication Union (ITU), as well as a member of the Inter-American Telecommunication Commission (CITEL). For more information, visit www.atis.org.

The SIP Forum is a leading IP communications industry association that engages in numerous activities that promote and advance SIP-based technology, such as the development of industry recommendations; interoperability testing events and special workshops, educational activities, and general promotion of IP communications standards, services, and technology for service provider, enterprise, and governmental applications. The SIP Forum is also the producer of the annual SIPNOC conferences (for SIP Network Operators Conference), focused on the technical requirements of the service provider community. One of the Forum's technical activities is the development of the SIPconnect Technical Recommendation – a standards-based SIP trunking recommendation that provides detailed guidelines for direct IP peering and interoperability between IP PBXs and SIP-based service provider networks, and the SIPconnect Certification Testing Program, a unique certification testing program that includes a certification test suite and test platform, and an associated "SIPconnect Certified" logo program that provides an official "seal of certification" for companies products and services that have officially achieved conformance with the SIPconnect specification. Other important Forum initiatives include work in security, SIP and IPv6, and IP-based Network-to-Network Interconnection (IP-NNI). For more information about all SIP Forum initiatives, please visit:

< http://www.sipforum.org/ >

ATIS-1000099, Robocall Call Blocking Notification

Is an ATIS & SIP Forum Joint Standard developed by the **ATIS/SIP Forum IP-NNI Task Force** under the **ATIS Packet Technologies and Systems Committee (PTSC)** and the **Technical Working Group (TWG)** under the **SIP Forum**.

Published by
Alliance for Telecommunications Industry Solutions
1200 G Street, NW, Suite 500
Washington, DC 20005

SIP Forum LLC
733 Turnpike Street, Suite 192
North Andover, MA 01845

ATIS-1000099

ATIS Standard on

Robocall Call Blocking Notification

Alliance for Telecommunications Industry Solutions

Approved August 16, 2022

Abstract

This document defines a mechanism that provides 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.

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.

Table of Contents

Table of Tables

ATIS Standard on –

Robocall Call Blocking Notification

1 Scope, Purpose, & Application

This document defines a mechanism that provides 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 real-time notice to callers.

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

1. Its status line[1] uses a unique reason phrase, "Network Blocked", rather than the SIP 603 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 standard.

Any 603 response received without the syntax defined in the standard included should be treated as currently handled today.

This standard is primarily developed for, and to be adopted by, US voice service providers. It is not precluded from being used internationally. Other countries may adopt theis standard and it may be implemented through bilateral agreements with business partners in the US pursuant to their business agreements.

2 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]

[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]

3 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/ >.

[1] The first line of a SIP response message is called the "status line".

[2] This document is available from the Internet Engineering Task Force (IETF) at: < http://www.ietf.org >.

3.1 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.[3]
- **SIP 603 Response:** the current SIP 603 response as defined in RFC 3261 [Ref 1].
- **SIP 603+ Response:** a SIP 603 response that adheres to the encoding described in this standard.

3.2 Acronyms & Abbreviations

ATIS	Alliance for Telecommunications Industry Solutions
AVP	Attribute-Value Pair
DNS	Domain Name System
HTTPS	Hypertext Transfer Protocol Secure
ISUP	ISDN User Part
SIP	Session Initiation Protocol
UA	User Agent
URL	Uniform Resource Locator

4 Notification of Analytics-based Network Blocking

4.1 Analytics Blocking

When a service provider blocks a call due to analytics, the service provider shall reply with a SIP 603+ response encoded per the profile defined in this standard.

The profile of the SIP 603 response [Ref 1] defined in this standard is referred to as 603+.

A SIP 603+ response is different from a SIP 603 response in that it contains a format with a status line value of “603 - Network Blocked”, with a Reason header containing the reason for the blocking, a version of this specification, “analytics1”, and the contact information of the entity responsible for blocking the call. The contact information provides the calling party with information on where to go to find out why the call was blocked, and to potentially to seek redress.

Any 603 response received without the syntax defined in this standard should be treated as currently handled today.

4.1.1 Formal Specification of Reason Header Syntax for 603+ Response

Reason headers used in 603+ responses (see Table 4-1) shall comply with the syntax specified in RFC 3326 [Ref 2], with the following restrictions per the profile of their usage defined in this standard:

- 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” [Ref 3].

[3] ATIS-1000074 states, “Call Validation Treatment (CVT) – This is a logical function that could be an application server function or a third party application for applying call analytics and treatment techniques once the signature is positively or negatively verified.”

- Exactly one “cause” parameter shall be included. Its value is “21” if the protocol parameter is set to “Q.850”. The value of the “cause” parameter shall be “603” if the protocol parameter is set to “SIP”.
- Exactly one “text” parameter shall be included. The 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 “text” parameter. Supported AVPs are specified in Table 4-2.
- 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**
“cause”	Yes	“21” or “603”
“text”	Yes	See Table 4-2
“location”	Yes	“LN”, “TN”, “LPN”, “RPN”, or “RLN”

Table 4-2: “text” parameter 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 service provider that blocked the call to facilitate redress (e.g., call identifier, blocking reason identifier, network segment identifier, etc.)

4.1.2 Examples of Reason Header Syntax for 603+ Response

Example “Reason” headers with different parameters (“cause”, “text”, and “location”) and value parameters attribute-value pairs (AVPs) are illustrated below:

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

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

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

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

Reason: Q.850; cause=21; text="v=analytics1;email=support@example.com";location=RLN

Reason: SIP; cause=603; text="v=analytics1;email=support@example.com";location=RLN

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

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

Reason: Q.850; cause=21; text="v=analytics1;tel=+12155551212";location=RLN

Reason: SIP; cause=603; text="v=analytics1;tel=+12155551212";location=RLN

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

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

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

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

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

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

4.1.3 Transit Network Processing

A transit network shall transparently forward a received SIP 603+ response towards the calling party. It shall not change the response code from 603 to a different value, nor modify any part of the 603+ response other than headers used to forward it (e.g., Via headers) except as required by its interconnect agreements or other contractual arrangements with the downstream network. It shall not retry the associated request. Analytics processing in a transit network may result in the generation of a 603+ response.

4.1.4 Originating Network Processing

An originating network receiving a SIP 603+ response shall forward the SIP 603+ response towards the SIP User Agent (UA) that originated the call request. It shall not modify the 603 response, except as required by contractual agreements with the entity responsible for the originating SIP UA or the network (e.g., Enterprise) that must be traversed to reach the originating SIP UA,.

If an originating network receives a 603 response where the status line of the SIP 603 response identifies it as a 603+ response as defined by this standard, 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 towards the originating SIP UA.
