ATIS-1000099

Robocall Call Blocking Notification

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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.

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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 this 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/ >.

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[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 >.