**ATIS-0x0000x**

ATIS Standard on

**Analysis of Support of RFC 8197 (Unwanted) and RFC 8688 (Rejected) in VoIP Networks**

**Alliance for Telecommunications Industry Solutions**

Approved Month DD, YYYY

**Abstract**

Abstract text here.

**Foreword**

The Alliance for Telecommunications Industry Solutions (ATIS) serves the public through improved understanding between carriers, customers, and manufacturers. The [**COMMITTEE NAME**] Committee [**INSERT MISSION**]. [**INSERT SCOPE**].

The mandatory requirements are designated by the word *shall* and *must,* 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.

Suggestions for improvement of this document are welcome. They should be sent to the Alliance for Telecommunications Industry Solutions, [**COMMITTEE NAME**], 1200 G Street NW, Suite 500, Washington, DC 20005.

At the time of consensus on this document, [**COMMITTEE NAME**], which was responsible for its development, had the following leadership:

[**LEADERSHIP LIST**]

The **[SUBCOMMITTEE NAME]** Subcommittee was responsible for the development of this document.

**Revision History**

| **Date** | **Version** | **Description** | **Author** |
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# Scope

## Scope

This Technical Report provides an analysis of deploying these two RFCs in voice service provider networks, identifying questions of clarification, and making recommendations for deployment.

## Reason

On December 30, 2020 the FCC released the Fourth Report and Order in Docket 17-59 addressing notification of blocked calls. <https://docs.fcc.gov/public/attachments/FCC-20-187A1.pdf>

This includes the support for RFC 8197 - A SIP Response Code for Unwanted Calls, July 2017, IETF and RFC 8688 - A Session Initiation Protocol (SIP) Response Code for Rejected Calls, Published December 2019, IETF.

This Report and Order gives voice service providers until January 1, 2022, approximately 12 months after the adoption of this Order, to comply with our immediate notification requirements.

The Report and Order did not consider how the STIR/SHAKEN ecosystem of capabilities and tools are deployed in various voice service provider network deployments.

# 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.

ATIS-0x0000x, *Technical Report*.[[1]](#footnote-1)

ATIS-0x0000x.201x, *American National Standard*.

# 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

**AAA**: xxxx.

**Bbbb**: xxxx.

## Acronyms & Abbreviations

|  |  |
| --- | --- |
| ATIS | Alliance for Telecommunications Industry Solutions |

# Standards Landscape

Xxx

## Unwanted and Reject RFCs

### RFC 8197

RFC 8197 - A SIP Response Code for Unwanted Calls, July 2017, IETF

H. Schulzrinne, FCC

https://www.rfc-editor.org/pdfrfc/rfc8197.txt.pdf

This document defines the 607 (Unwanted) SIP response code, allowing called parties to indicate that the call or message was unwanted. SIP entities may use this information to adjust how future calls from this calling party are handled for the called party or more broadly.

### RFC 8688

RFC 8688 - A Session Initiation Protocol (SIP) Response Code for Rejected Calls, Published December 2019, IETF

E.W. Burger, Georgetown University

B. Nagda, Massachusetts Institute of Technology

https://www.rfc-editor.org/rfc/rfc8688.pdf

This document defines the 608 (Rejected) Session Initiation Protocol (SIP) response code. This response code enables calling parties to learn that an intermediary rejected their call attempt. No one will deliver, and thus answer, the call. As a 6xx code, the caller will be aware that future attempts to contact the same User Agent Server will likely fail. The initial use case driving the need for the 608 response code is when the intermediary is an analytics engine. In this case, the rejection is by a machine or other process. This contrasts with the 607 (Unwanted) SIP response code in which a human at the target User Agent Server indicates the user did not want the call. In some jurisdictions, this distinction is important. This document also defines the use of the Call-Info header field in 608 responses to enable rejected callers to contact entities that blocked their calls in error. This provides a remediation mechanism for legal callers that find their calls blocked.

### 4.1.2 Report and Order Further Considerations

Because SIP codes are not available on non-IP networks, ISUP code 21 is the appropriate code for calls blocked on a TDM network, therefore SIP 607 will interwork into ISUP CC 21 at a TDM gateway. Therefore, we require that terminating voice service providers that block calls on a TDM network return ISUP code 21.

To reduce ambiguity, we require that the cause location be “user” when using ISUP code 21 in this manner. Internet Engineering Task Force, Integrated Service Digital Network (ISDN) User Part (ISUP) to Session Initiation Protocol (SIP) Mapping (Dec. 2002), https://tools.ietf.org/html/rfc3398 (ISUP and SIP Code Mapping Specification). We recognize that ISUP code 21, even with the specified cause location, does not provide the same level of detail as either SIP code 607 or 608, and may be used in other contexts as well. As a result, receipt of ISUP code 21 does not guarantee that the call was blocked, but instead provides a signal that further investigation may be warranted.

IETF documentation currently recommends that ISUP code 21 be mapped to either SIP code 403 “Forbidden” or, where the cause location is “user,” SIP code 603 “Decline.” ISUP and SIP Code Mapping Specification. It is unlikely that SIP code 403 will be used where 607 or 608 is appropriate. We recognize, however, that, because the distinguishing factor is the cause location, it may be impossible for voice service providers to determine whether 603, 607, or 608 is the appropriate code when receiving cause code 21 with a cause location of “user.” For purposes of satisfying the rules we adopt today, we permit a voice service provider to use any of these codes it deems appropriate. Because the IETF recommends code 603, we encourage voice service providers to continue using this approach unless they have clear knowledge that 607 or 608 is the more appropriate code. As a result, when ISUP code 21 or SIP code 603 is returned, callers should investigate as they would if SIP code 607 or 608 were indicated.

## Related Standards

### ATIS-1000679.2015(R2020) Interworking between Session Initiation Protocol (SIP)and ISDN User Part

### ATIS-1000063 – SIP Forum TWG-6 JOINT ATIS/SIP FORUM TECHNICAL REPORT – IP NNI PROFILE

### 3GPP TS 24.229 IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3

### 3GPP TS 24.193

### 3GPP TS 24.195

### IETF RFC2026 ISUP to SIP Mapping

### I3 Forum

# Analysis

RFC 8197 and RFC 8688 calls out the distinction between SIP 607 (Unwanted) as a user (human) rejecting the call and SIP 608 (Rejected) as an intermediary engine's statistical rejection in general as the nominal behavior.

## General

The RFC’s and R&O define “in general as the nominal behavior”, as:

* 607 initiated at a UE based on human action, and
* 608 initiated at intermediary network element using statistical analysis and/or pre-provisioned data.

Question(s):

* So, is the assumption (at least initially) that deployments can only support the “nominal behavior”?

## Blocking Versus Reject

The R & O refers to IETF RFCs, where the architecture is Bob-> Proxy -> Alice, and did not take into account actual carrier deployments and where functions are deployed.

Depending on a voice service providers network architecture, either Blocking or Rejection can occur in multiple places in the network (e.g. UE, CVT, TAS, BAS, etc..).

607 is associated with a user (human) rejecting the call, but the user can either reject a call directly on the UE or pre-provision screening in network elements or 3rd part Analytic Engines (CVT).

There are also gray area cases between Blocking and Reject, such as users installing third party applications that intercept and hang up the call (these calls appear as successful but have short call durations) or receiving a label indicating that the call is spam and making the decision to either not answer the call or send the call to voicemail.

Questions:

1. If the user creates a profile in the CVT whether it’s a Block/Allow list, block list or based on other rules, is this a user unwanted or CVT reject?
	1. Since the end user did not receive the call nor had the option to reject the call, the terminating or intermediary provider would respond with SIP Code 608, regardless of the fact that the customer opted into a third-party fraud prevention service.
2. If the user provisions supplementary services in the TAS or Business AS to mark calls as blocked or redirected to voice mail, are they rejected or unwanted, when does 607 or 608 apply?
	1. Typically calls redirected to voicemail have SIP codes of 200 OK, resulting in neither a 607 nor a 608 being returned to the originating provider. In these instances, SIP progress code 120 “ -“Rated”[[2]](#footnote-2) (or similar concept) should be developed at IETF, incorporated by IP providers, and returned to originating providers to cover gaps in observability provided by the absence of Reject/Unwanted responses in scenarios where the call ultimately did not connect with the intended target due to programmatic labeling and associated decision making. Without a progress code, any decision made by a CVT to label or otherwise treat the call with normal call termination procedures will go unnoticed by the originator. The ISUP equivalent of a 120 in-progress SIP code is not defined today. The proposed behavior is to follow RFC 3398 for 183 session progress. This means the legacy/TDM network element will not have knowledge of call ratings, and is therefore incumbent upon the terminating providers IP gateway element, such as the SBC (assuming one exists) to process ‘Rated’ progress messaging.
3. If a user receives a display from their CVT (e.g., SPAM, Fraud, Telemarketer, etc..), and the user rejects the call, is that rejected or unwanted, when does 607 or 608 apply?
	1. Some third-party fraud-prevention apps, such as Nomorobo, answer the suspected fraudulent calls and proceeds to hang up on behalf of the consumer. In these instances, originating providers will receive 200 OK from the terminating provider with no insight into the calls being marked as spam and that the call was blocked by a network as opposed to the end-user.[[3]](#footnote-3) This is another use case where ‘120 Rated’ concept would be useful for originating provider to have awareness of potential bad actors on their network, as well as providing a remediation contact to use as applicable.

## RFC xxxx – Unwanted (607)

Questions:

1. What is a UE to do with a 607? Display what?
2. Though 3GPP TS 24.229 supports 607, under what conditions will a UE send a 607?

## RFC 8688 - Rejected (608)

The RFC illustrates a call flow with announcement at the originating carrier as being optional.

Questions:

1. Regarding the use of SIP 608, the RFC is written for any intermediary to implement but the FCC report only appears to grant this privilege to terminating carriers. Are gateway and intermediate carriers permitted to implement SIP 608 using our own analytics, or are we restricted to only passing it through from downstream?
	1. Intermediaries today block calls they suspect fraud by responding with a SIP Code 503. This SIP Code prompts the originating provider to attempt the call with another partner. Intermediaries should leverage SIP Code 608 when blocking calls specifically when fraud is suspected, excluding business as usual purposes. The intermediary or terminating provider issuing SIP Code 608 should provide jcard information for originating providers to redress any inappropriate call blocking behavior on behalf of their customers.
2. It is unclear whether support of jcard in Call-Info is optional or mandatory?
	1. Passing jcard information should be mandatory.
3. If there was an announcement played, what would it be?
	1. All announcements should be up to individual carriers given the wide variance of device capabilities. When display options are not available, an audio announcement should be played by the terminating provider.
4. Feature Caps is supported by mobile UEs, but 608 needs to be added to 3GPP TS 24.229, so how is it expected to be delivered to a UE?
	1. Until support have been added in TS 24.229, and after for non-capable devices, the terminating service provider should have play an announcement explaining the call was rejected. There should be no standardization of alerting the customer, and should be left up to the individual originating service providers.
5. Once 608 is included in 3GPP TS 24.229, how is the jcard information validated and displayed at the UE?
	1. The validation of jcard information is up to the terminating provider who also signs the rejection with a certificate, allowing for traceability of the source of the rejection. Display would be up to the device software.
6. For enterprise, there is no standard to update to support 608, never mind support of jcard and having it displaying it. So why are we doing this?
	1. Enterprises often have sophisticated implementations into telephony networks that include SIP trunks and robust API access. SIP Codes 607 and 608 will be beneficial for these enterprises to understand when their calls are being inappropriately blocked. This information can be passed to enterprises today without the need of additional standards. In cases where an enterprise may not be aware of how to treat a 607 or 608, they simply treat that code as any other 6XX SIP Code in use today.

# Conclusion and Recommendations

**Annex A**

(normative/informative)

# Annex Title

Xxx

*For an Informative Annex or Bibliography, the following boilerplate paragraph should be used:*

At the time of publication, the editions indicated were valid. All standards are subject to revision, and users of this document are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below.

1. This document is available from ORGANIZATION at <website>. [↑](#footnote-ref-1)
2. https://tools.ietf.org/html/draft-penar-sipcore-ratingprovided-01 [↑](#footnote-ref-2)
3. https://nomorobo.zendesk.com/hc/en-us/articles/200536477-How-does-it-work-on-Landlines- [↑](#footnote-ref-3)