NENA NG9-1-1 Call Processing Metrics Standard

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1 Executive Overview

The intent of this document is to define normalized NG9-1-1 call processing metrics for computing useful statistics so that independent implementations can derive the same comparable measurements.

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2 Document Conventions

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2.1 Document Terminology

This section defines keywords, as they should be interpreted in NENA documents. The form of emphasis (UPPER CASE) shall be consistent and exclusive throughout the document. Any of these words used in lower case and not emphasized do not have special significance beyond normal usage.

1. MUST, SHALL, REQUIRED: These terms mean that the definition is a normative (absolute) requirement of the specification.
2. MUST NOT: This phrase, or the phrase "SHALL NOT", means that the definition is an absolute prohibition of the specification.
3. SHOULD: This word, or the adjective "RECOMMENDED", means that there may exist valid reasons in particular circumstances to ignore a particular item, but the full implications must be understood and carefully weighed before choosing a different course.
4. SHOULD NOT: This phrase, or the phrase "NOT RECOMMENDED" means that there may exist valid reasons in particular circumstances when the particular behavior is acceptable or even useful, but the full implications should be understood and the case carefully weighed before implementing any behavior described with this label.
5. MAY: This word, or the adjective "OPTIONAL", means that an item is truly optional. One vendor may choose to include the item because a particular marketplace requires it or because the vendor feels that it enhances the product while another vendor may omit the same item. An implementation which does not include a particular option "must" be prepared to interoperate with another implementation which does include the option, though perhaps with reduced functionality. In the same vein an implementation which does include a particular option "must" be prepared to interoperate with another implementation which does not include the option (except, of course, for the feature the option provides.)

These definitions are based on IETF RFC 2119 [2].

2.2 Describing Interfaces for XML and JSON Objects

This document defines one or more interfaces, often called "Application Programming Interfaces" (APIs) that provide a standardized way to communicate with a Functional Element. The interface consists of data objects and the operations that apply to those data

objects. In this document, the operations and data objects are documented as a formal definition of an interface or schema and a set of "business rules" which are also required in the software or database environment to comply with the standard and achieve interoperability. The business rules are specified with normative statements using keywords that are defined in the Document Terminology section above.

Describing REST/JSON Interfaces

This document uses "OpenAPI" Version 3.1 as the formal description of a REST/JSON interface. This description is manifested in a file ("yaml") that is not contained directly in this document. The file is maintained in a NENA GitHub repository. In the interface description, a hyperlink to the yaml file is provided. The yaml file is the normative description of the interface and was reviewed with the text of this document.

In the description of the interface in the text, a table often appears that describes the parameters to the interface and the data objects the interface exchanges. These tables are informative: the yaml file is the normative documentation of the interface. If there is a discrepancy between the table and the yaml file, the yaml controls. In the tables, a column is headed "Required". Three values may occur in this column. "y" means the parameter or member is required to be present. It will appear in the "required" section of the yaml file. "n" means that the parameter or member is optional. It will not appear in the "required" section of the yaml file. "c" means that the parameter or member is conditional. This value alerts the reader that a business rule is specified that controls the presence of the parameter or member. It will not appear in the "required" section of the yaml file.

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2.4 Reason for Issue/Reissue

NENA reserves the right to modify this document. Upon revision, the reason(s) will be provided in the table below.

Document Number	Approval Date	Reason For Issue/Reissue
NENA-STA-019.1-2018	07/02/2018	Initial Document
NENA-STA-019.2-2022	September 21, 2022	Updated to align with NENA-STA-010.3-2021 [3] Added metrics from NENA-STA-024.1-202Y [7]

3 Call Processing Metrics

3.1 Introduction

Call processing metrics are measurements between events in the call processing chain and are used to drive reporting, analysis, and real-time monitoring. This document concentrates on operations and management metrics specific to NG9-1-1, but some are equivalent to SIP session metrics defined in RFC 6076 [4]. Instances where the metrics defined herein overlap with those in RFC 6076 [4] are identified. In addition to the SIP messages used in RFC 6076 [4] to derive metrics, this document identifies related LogEvents [3] to achieve the same. RFC 6076 [4] includes computed metrics such as computed ratios and percentages. Computations involving metrics are out of scope for this document. Accordingly, all computed metrics from RFC 6076 [4] have been explicitly ignored. Any calculation that looks at metrics applied to multiple calls to compute things like mean/average/worst case are beyond the scope of this document, but any such computations/reports MUST be based on metrics defined in this document.

3.2 Call Related Definitions

Please note that the following definitions only apply to emergency calls.

3.2.1 Call

3.2.1.1 Definition

A generic term referring to any request for emergency assistance, regardless of the media used to make that request. This term may appear in conjunction with specific media, such as "voice Call", "video Call", "text Call", or "data-only Call" when the specific media is of importance. The term "non-human-initiated Call" refers to an emergency call that is initiated automatically, carries data, does not establish a two-way interactive media session, and typically does not involve a human at the "initiating" end.

3.2.1.2 Determination Using LogEvents

A Call can be identified by the logging of a CallStartLogEvent from the view of the Functional Element (FE) logging the event.

3.2.1.3 Determination Using SIP Messages

The first INVITE the FE receives for a particular NENA Call Tracking Identifier

3.2.2 Answered Call

3.2.2.1 Definition

A Call (excluding non-human-initiated Calls) that has been answered by an Agent and two-way communication has been established, irrespective of whether the call was auto-answered for the Agent or if an auto-greeting message was played. For a non-human-initiated Call, that Call may be accepted by an automaton.

3.2.2.2 Determination Using LogEvents

The logging of a CallStateChangeLogEvent containing callAnswered and agencyAgentId.

3.2.2.3 Determination Using SIP Messages

The reception or transmission of the ACK message of the 200 OK message to the initial call's INVITE.

3.2.3 Attempted Call

3.2.3.1 Definition

A call presented to an FE (such as Call Handling Functional Element) regardless of whether successful completion status was achieved.

3.2.3.2 Determination Using LogEvents

The logging of the CallStartLogEvent.

3.2.3.3 Determination Using SIP Messages

The reception of an INVITE message.

3.2.4 Prematurely Disconnected Call

3.2.4.1 Definition

An Answered Call that terminated before the parties have finished their conversation[1].

3.2.4.2 Determination Using LogEvents

Impossible to determine at this time.

3.2.4.3 Determination Using SIP Messages

Impossible to determine.

[1] Currently there is no LogEvent or SIP header that carry this information. It is feasible that it can be added in the future.

3.2.5 Diverted Call

3.2.5.1 Definition

A call that was rerouted due to the nominal destination's unavailability or inability to accept.

3.2.5.2 Determination Using LogEvents

- For the nominal PSAP, diversion notifications are currently not logged.
- For the diverted-to PSAP, the logging of a CallSignalingMessageLogEvent containing the SIP INVITE message where the History-Info header specifies the call has been redirected.

3.2.5.3 Determination Using SIP Messages

- For the nominal PSAP, the reception of either MESSAGE or NOTIFY message which specifies that call has been redirected.
- For the diverted-to PSAP, the reception of a SIP INVITE message where History-Info header specifies call has been redirected.

3.2.6 Abandoned Call

3.2.6.1 Definition

A call placed to 9-1-1 in which the caller disconnects before the call can be answered by the Public Safety Answering Point (PSAP).

3.2.6.2 Determination Using LogEvents

Logging of a CallSignalingMessageLogEvent containing the SIP CANCEL message and/or a CallStateChangeLogEvent containing callCancel.

3.2.6.3 Determination Using SIP Messages

The reception of a SIP CANCEL message and/or reception of a SIP NOTIFY message with the AbandonedCall event package.

3.2.7 Misrouted Call

3.2.7.1 Definition

A call routed to a PSAP that should not have received it due to a provisioning error (for example in the ECRF [Emergency Call Routing Function], in the PRF [Policy Routing Function], or the LIS [Location Information Server]) or other misconfigurations.

3.2.7.2 Determination Using LogEvents

The logging of a DiscrepancyReportLogEvent where "type" contains DiscrepancyReportRequest and the Discrepancy Report specifies the call has been misrouted.

3.2.7.3 Determination Using SIP Messages

Impossible to determine.

3.3 Call-Related Metrics

For a PSAP, call-related metrics are measured when the request hits the Call Handling Element but for other services (such as the Next Generation Core Services [NGCS]), they can be measured at another FE (such as a BCF [Border Control Function] or ESRP [Emergency Service Routing Proxy]) except as otherwise noted.

3.3.1 Call Network Transit (in an ESInet)

3.3.1.1 Definition

The difference in time between a call's ingress into an ESInet and the time the call is processed at the egress from the same ESInet.

3.3.1.2 Determination Using LogEvents

The difference in time between the CallStartLogEvent by the ingress FE and the CallStartLogEvent by the egress FE.

3.3.1.3 Determination Using SIP Messages

The difference in time between the reception of the INVITE or MESSAGE for a call by the ingress FE and the transmission of the INVITE or MESSAGE transaction for the same call by the egress FE.

3.3.2 Inter-Network Transit

3.3.2.1 Definition

The difference in time between a call's ingress into the ESInet and the time the call is processed at the ingress of the next downstream network.

3.3.2.2 Determination Using LogEvents

The difference in time between the CallStartLogEvent by the ingress FE of one network and the CallStartLogEvent by the ingress FE of the next downstream network.

3.3.2.3 Determination Using SIP Messages

The difference in time between the reception of the INVITE or MESSAGE request by the ingress FE and the reception of the INVITE or MESSAGE transaction for the same call by the ingress FE of the next downstream network.

3.3.3 Session Duration

3.3.3.1 Definition

The difference in time between the start of a session (INVITE) and the end of the same session (BYE or final response error code) for a unique SIP Call-ID. For a PSAP, this is measured at the Call Handling FE and, for an NGCS, this would be measured at the ingress BCF.

3.3.3.2 Determination Using LogEvents

The difference in time between SessionStartLogEvent and SessionEndLogEvent for a particular session with the same callIdSip. For a PSAP, this is measured at the Call Handling FE and, for an NGCS, this would be measured at the ingress BCF.

3.3.3.3 Determination Using SIP Messages

The difference in time between INVITE message and the final session message (BYE or error code) for a particular session with the same SIP Call-ID. For a PSAP, this is measured at the Call Handling FE and, for an NGCS, this would be measured at the ingress BCF.

3.3.4 Successful Session Request Delay (SSRD)

3.3.4.1 Definition

The difference in time from the session establishment request to the notification that the session is proceeding per RFC 6076 [4]. For example, the difference in time between the INVITE message (request) and the 180 RINGING message (response) associated with the same call.

3.3.4.2 Determination Using LogEvents

The difference in time between the SessionStartLogEvent and the CallSignalingMessageLogEvent that contains the non-100 provisional response.

3.3.4.3 Determination Using SIP Messages

The difference in time between the INVITE and a non-100 provisional response message associated with the same session.

3.3.5 Session Disconnect Delay (SDD)

3.3.5.1 Definition

The difference in time between a request to terminate a session and its response received per RFC 6076 [4].

3.3.5.2 Determination Using LogEvents

The difference in time between the SessionEndLogEvent and the CallSignalingMessageLogEvent that contains the response.

3.3.5.3 Determination Using SIP Messages

The difference in time between the BYE message and the final response to that BYE.

3.3.6 Call Answered Delay

3.3.6.1 Definition

The difference in time from a Call establishment request to the Call being established. This would include early media if any early media is exchanged.

3.3.6.2 Determination Using LogEvents

The difference in time between CallStartLogEvent and the CallStateChangeLogEvent containing callAnswered.

3.3.6.3 Determination Using SIP Messages

The difference in time between the INVITE of the initial session signaling the Call and the 200 OK response of the session answered by an Agent**.**

3.3.7 Session Answered Delay

3.3.7.1 Definition

The difference in time from a session establishment request to the session being established for a unique SIP Call-ID. This would include early media if any early media is exchanged.

3.3.7.2 Determination Using LogEvents

The difference in time between SessionStartLogEvent and the SessionStateChangeLogEvent containing callAnswered for the same callIdSip.

3.3.7.3 Determination Using SIP Messages

The difference in time between the INVITE and its 200 OK response for the same SIP Call-ID**.**

3.3.8 Call Failed Delay

3.3.8.1 Definition

The difference in time from the Call establishment request to the issuance of an error message.

3.3.8.2 Determination Using LogEvents

The difference in time between CallStartLogEvent and the CallSignalingMessageLogEvent that contains the error.

3.3.8.3 Determination Using SIP Messages

The difference in time between the INVITE or MESSAGE request and the error response sent, or timeout for the request. Please note that an abandoned call due to CANCEL falls within this definition as it will fail with a 587 Request Terminated response.

3.3.9 Session Failed Delay

3.3.9.1 Definition

The difference in time from the session establishment request to the issuance of an error message for a unique SIP Call-ID.

3.3.9.2 Determination Using LogEvents

The difference in time between SessionStartLogEvent and the CallSignalingMessageLogEvent that contains the error for the same callIdSip.

3.3.9.3 Determination Using SIP Messages

The difference in time between the INVITE or MESSAGE request and the error response sent, or timeout for the request for the same SIP Call-ID. Please note that an abandoned call due to CANCEL falls within this definition as it will fail with a 587 Request Terminated response.

3.3.10 Call Alerting Delay

3.3.10.1 Definition

The delay between when the Call enters the alerting state (such as ringing or other notification) on an end-point device and the Call being answered.

3.3.10.2 Determination Using LogEvents

The difference in time between a CallStateChangeLogEvent containing callAlerting and a CallStateChangeLogEvent containing callAnswered.

3.3.10.3 Determination Using SIP Messages

Because an implementation can answer a session in order to play an announcement, it is therefore impossible to determine when a Call is answered by an Agent using SIP messages.

3.3.11 Session Alerting Delay

3.3.11.1 Definition

The delay between when the Session enters the alerting state (such as ringing or other notification) on an end-point device and the Session being answered for a unique SIP-Call ID.

3.3.11.2 Determination Using LogEvents

The difference in time between a SessionStateChangeLogEvent containing callAlerting and a SessionStateChangeLogEvent containing callAnswered for the same callIdSip.

3.3.11.3 Determination Using SIP Messages

The difference in time between a 180 Ringing, 182 Queued or 183 SessionProgress response and a 200 OK for a particular session.

3.3.12 Time to Invite Third-Party Delay

3.3.12.1 Definition

The difference in time between the time a Call is being answered and the time the same Call is offered to another party. This metric is only applicable to FEs which implement the Bridging Function.

3.3.12.2 Determination Using LogEvents

The difference in time between the CallStateChangeLogEvent containing callAnswered and the CallStartLogEvent for the call to the third party.

3.3.12.3 Determination Using SIP Messages

The difference in time between the 200 OK response sent to the calling party and the INVITE message sent to the third party to join the session.

3.3.13 Location Dereference Query Response Delay

3.3.13.1 Definition

The difference in time between when a location dereference query is generated, and the response is received.

3.3.13.2 Determination Using LogEvents

The difference in time between the LocationQueryLogEvent and its applicable LocationResponseLogEvents.

3.3.13.3 Determination Using SIP Messages

The difference in time between a SIP SUBSCRIBE request and its first NOTIFY response.

3.3.13.4 Determination using i3 HELD Messages

The difference in time between a HELD (HTTP [hyper-text transfer protocol] Enabled Location Delivery) query request and its response.

3.3.14 Location Inter-Notification Delay

3.3.14.1 Definition

The time between two consecutive location update notifications for a specific subscription.

3.3.14.2 Determination Using LogEvents

The difference in time between one LocationResponseLogEvent and the next LocationResponseLogEvent for the same subscription.

3.3.14.3 Determination Using SIP Messages

The difference in time between one SIP NOTIFY and the next SIP NOTIFY message for the same subscription.

3.3.15 LoST Dereference Query/Response Delay

3.3.15.1 Definition

Time between when a LoST (Location to Service Translation) query is generated, and the response is received.

3.3.15.2 Determination Using LogEvents

The difference in time between the LoSTQueryLogEvent and its applicable LoSTResponseLogEvent.

3.3.15.3 Determination Using LoST Messages

The difference in time between the LoST query message and its applicable LoST response message.

3.3.16 Hold Time

3.3.16.1 Definition

The time between when a Call is placed on hold and when that Call changes state.

3.3.16.2 Determination Using LogEvents

The difference in time between the callHold CallStateChangeLogEvent and the subsequent CallStateChangeLogEvent.

3.3.16.3 Determination Using SIP Messages

Due to the different possibilities as to how the hold functionality can be implemented (e.g., transfer to a Music-on-Hold media server, one-way mute at the position, etc.), determining Hold Time using SIP messages is left undefined.

3.3.17 Park Time

3.3.17.1 Definition

The time between when a Call is placed in a parked state and when that Call changes state.

3.3.17.2 Determination Using LogEvents

The difference in time between the callPark CallStateChangeLogEvent and the subsequent CallStateChangeLogEvent.

3.3.17.3 Determination Using SIP Messages

Due to the different possibilities as to how the park functionality can be implemented, determining Park Delay using SIP messages is left undefined.

3.3.18 Call Queued Delay

3.3.18.1 Definition

The time between when a Call is placed in a queue waiting to be offered to an agent and when that Call changes state.

3.3.18.2 Determination Using LogEvents

The difference in time between the callQueued CallStateChangeLogEvent and the subsequent CallStateChangeLogEvent.

3.3.18.3 Determination Using SIP Messages

Due to the different possibilities as to how the queue functionality can be implemented, determining Call Queued Delay using SIP messages is left undefined.

3.3.19 Announcement Duration

3.3.19.1 Definition

The time between the beginning and the end of an announcement.

3.3.19.2 Determination Using LogEvents

The difference in time between the AnnouncementStartLogEvent and the AnnouncementEndLogEvent.

3.3.19.3 Determination Using SIP Messages

Due to the different possibilities in announcements that can be implemented, determining Announcement Duration using SIP messages is left undefined.

3.3.20 Total Call Duration

3.3.20.1 Definition

The time between the beginning and the end of a Call for a particular NENA Call Identifier (urn:emergency:uid:callid:).

3.3.20.2 Determination Using LogEvents

The difference in time between the CallStartLogEvent and the associated CallEndLogEvent for a particular NENA Call Identifier.

3.3.20.3 Determination Using SIP Messages

The difference in time between the INVITE message and the final disposition message (BYE or final response error code) for a particular NENA Call Identifier.

3.3.21 Call Media Quality Metrics

Reports on several quality metrics of the media for a Call.

3.3.21.1 Definition

Contains the media quality statistics reported in the MediaEndLogEvent. It includes, among other items, the jitter, delay, and packet loss statistics for the Call.

3.3.21.2 Determination Using LogEvents

Statistics reported in the MediaQualityStats element of the MediaEndLogEvent.

3.3.21.3 Determination Using SIP Messages

Statistics in sender and receiver reports that come with a Real Time Control Protocol (RTCP) BYE message (as defined in RFC 3550 [5], Section 6) at the end of a media exchange.

3.3.22 Route Determination Time

3.3.22.1 Definition

The time between when the Call entered the ESRP and when the route is determined for that Call.

3.3.22.2 Determination Using LogEvents

The difference in time between the CallStartLogEvent or CallProcessLogEvent and the RouteLogEvent.

3.3.22.3 Determination Using SIP Messages

The difference in time between when the INVITE is received by the ESRP and when the INVITE is transmitted to the next hop downstream.

3.3.23 Message Session Relay Protocol (MSRP) Automated Response Message Delay

3.3.23.1 Definition

The time between the reception of a session initiation message from the caller and the reception of an automated response message from the PSAP.

3.3.23.2 Determination Using LogEvents

The time between the first MSRP message from the caller logged in a NonRtpMediaMessageLogEvent and the first automated MSRP message from the PSAP logged in a subsequent NonRtpMediaMessageLogEvent.

3.3.23.3 Determination Using SIP/MSRP Messages

The difference in time between the INVITE message of a MSRP session requested by the caller and the first automated MSRP SEND from the PSAP.

3.3.24 MSRP Response Message Delay

3.3.24.1 Definition

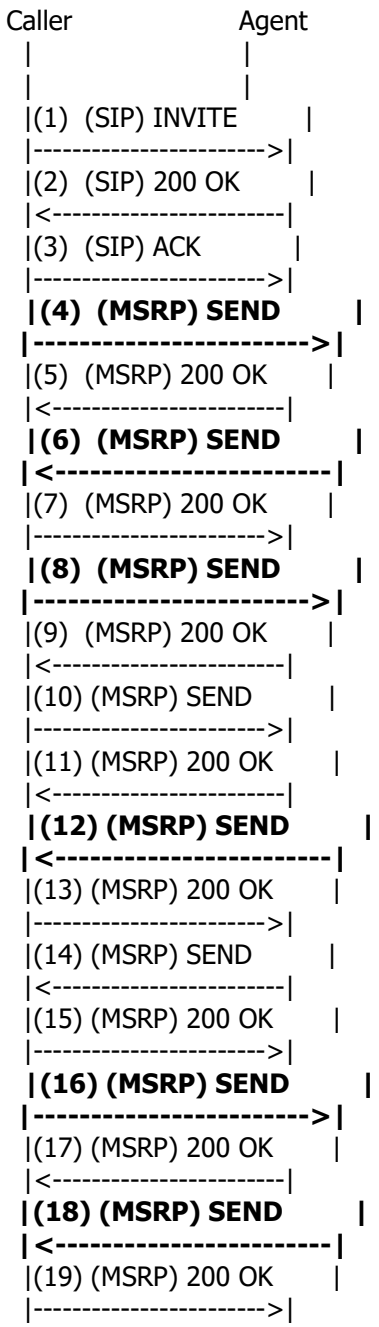
The time between the reception of a MSRP message from the caller and the response message from the agent.

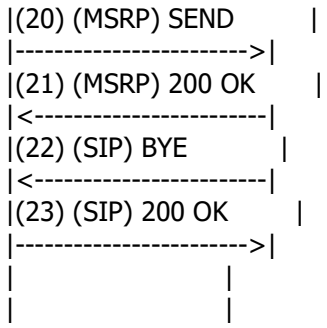
3.3.24.2 Determination Using LogEvents

The time between the first MSRP message from the caller logged in a NonRtpMediaMessageLogEvent and the first MSRP message from the Agent logged in a subsequent NonRtpMediaMessageLogEvent.

3.3.24.3 Determination Using SIP/MSRP Messages

The difference in time between the first MSRP SEND message from the caller (either after a session has been established or after a MSRP SEND message from the agent) and the first subsequent MSRP SEND message from the agent. For example, in the following SIP/MSRP call, MSRP Response Message Delay is the difference in time between messages (4) and (6), (8) and (12) and (16) and (18).





3.3.25 Additional Data Query Response Delay

3.3.25.1 Definition

The difference in time between when an Additional Data query (including a dereference) is generated, and the response is received.

3.3.25.2 Determination Using LogEvents

The difference in time between the AdditionalDataQueryLogEvent and its applicable AdditionalDataResponseLogEvent.

3.3.25.3 Determination Using HTTP Messages

The difference in time between a HTTP GET or POST request and its response.

3.3.26 EIDO Dereference Factory Query Response Delay

3.3.26.1 Definition

The difference in time between when an EIDO Dereference Factory query is generated, and the response is received.

3.3.26.2 Determination Using LogEvents

The difference in time between the EidoDereferenceFactoryQueryLogEvent and its applicable EidoDereferenceFactoryQueryResponseLogEvent.

3.3.26.3 Determination Using HTTP Messages

The difference in time between a HTTP GET request and its response.

3.3.27 Subscription Request Response Delay

3.3.27.1 Definition

The difference in time between when a subscription request is generated, and the response is received.

3.3.27.2 Determination Using LogEvents

The difference in time between the SubscriptionRequestedLogEvent and its applicable SubscriptionRequestedResponseLogEvent.

3.3.27.3 Determination Using EIDO Conveyance Messages

The difference in time between a "subscribe" request and its "subscribeResponse" as per section WebSocket-based Incident Data Subscription in NENA-STA-024 [7].

3.3.27.4 Determination Using SIP Messages

The difference in time between a SIP SUBSCRIBE request and its first NOTIFY response.

3.3.28 Subscription Duration

3.3.28.1 Definition

The difference in time between the start of a subscription and the end of the same subscription for a unique subscription identifier.

3.3.28.2 Determination Using LogEvents

The difference in time between the SubscriptionRequestedLogEvent and SubscriptionTerminatedLogEvent with the same subscriptionId.

3.3.28.3 Determination Using EIDO Conveyance Messages

The difference in time between a "subscribe" request and, "terminate" notification with the same subscriptionId or renewal subscribe request rejected with a response of 481 "Subscription does not exist" (in the case the subscription was terminated unbeknownst to the subscriber) as per section WebSocket-based Incident Data Subscription in NENA-STA-024 [7].

3.3.28.4 Determination Using SIP Messages

The difference in time between a SIP SUBSCRIBE request and, a NOTIFY with the same SIP Call-Id and a "Subscription-State" value of "terminated" or renewal SUBSCRIBE request rejected with a response of 481 "Subscription does not exist" (in the case the subscription was terminated unbeknownst to the subscriber).

3.3.29 WebSocket Duration

3.3.29.1 Definition

The difference in time between the start of a websocket and its termination.

3.3.29.2 Determination Using LogEvents

The difference in time between the WebSocketEstablishedLogEvent and its applicable WebSocketTerminatedLogEvent as per section WebSocket-based Incident Data Subscription in NENA-STA-024 [7].

3.3.29.3 Determination Using HTTP/Web Socket Messages

The difference in time between when the HTTP 101 Switching Protocols response is exchanged which indicates the websocket is established and when Close frames are exchanged.

3.3.30 ALI Query Response Delay

3.3.30.1 Definition

The difference in time between when an ALI query is generated, and the response is received.

3.3.30.2 Determination Using LogEvents

The difference in time between the AliLocationQueryLogEvent and its applicable AliLocationResponseLogEvent.

3.4 Agent Related Metrics

In this section determinations only pertain to difference in time between AgentStateChangeLogEvent LogEvents.

3.4.1 Agent Availability Metric

3.4.1.1 Definition

Measured as the time an agent enters a primary agent state (Available or Not Available) until the time the agent transitions to another primary agent state.

3.4.1.2 Determination Using LogEvents

Difference in elapsed time between entering a primaryAgentState and the time of transition to the next primaryAgentState PrimaryAgentStateValuesCode value.

3.4.2 Agent Secondary State Metric

3.4.2.1 Definition

Measured as the time an agent enters a secondary agent state (e.g., LoggedOut, Break, Waiting, Active, Hold, Reserved) until the time the agent transitions to another secondary agent state.

3.4.2.2 Determination Using LogEvents

Difference in elapsed time between entering a SecondaryAgentState and the time of transition to the next SecondaryAgentState value.

3.5 Logging Service REST/JSON Web Service

This standard defines version 1.1 of the Logging Service as specified in loggingservice.yaml, authoritative version of which can be found at https://github.com/NENA911/Logging-Service/tree/V1.1. This new version specifies the following additions:

3.5.1 AnnouncementStartLogEvent

Used by an element to log the beginning of the playing of an (multimedia) announcement to the caller, such as an automatic answer greeting or an interactive media response announcement.

Name	Required	Description
clientAssignedIdentifier	N	An identifier assigned by the client
logEventType	Y	MUST be "AnnouncementStartLogEvent"
timestamp	Y	The timestamp when the announcement started playing
elementId	Y	Element identifier of the element that logged the event
agencyId	Y	AgencyId of the agency that logged the event
agencyAgentId	C	See note 1
agencyPositionId	N	Identifier for the position that is handling the call.
callId	C	See note 2
incidentId	C	See note 3
callIdSIP	C	See note 4 CallIdSIP will be deprecated in a future version of NENA-STA-010 [3].
sessionId	C	See note 4
ipAddressPort	C	See note 5
extension	N, occurs 0 or more times	Optional private extension parameters

Name	Required	Description
announcementType	Y	Values limited to those in the AnnouncementTypes registry
announcementTag	N	For local use, can be used to identify the specific announcement played, for example, the name of the VoiceXML announcement script

3.5.2 AnnouncementEndLogEvent

Used by an element to log the end of the playing of an (multimedia) announcement to the caller, such as an automatic answer greeting or an interactive media response announcement.

Name	Required	Description
clientAssignedIdentifier	N	An identifier assigned by the client
logEventType	Y	MUST be "AnnouncementEndLogEvent"
timestamp	Y	The timestamp when the announcement ended playing
elementId	Y	Element identifier of the element that logged the event
agencyId	Y	AgencyId of the agency that logged the event
agencyAgentId	C	See note 1
agencyPositionId	N	Identifier for the position that is handling the call.
callId	C	See note 2
incidentId	C	See note 3
callIdSIP	C	See note 4 CallIdSIP will be deprecated in a future version of NENA-STA-010 [3].
sessionId	C	See note 4
ipAddressPort	C	See note 5
extension	N, occurs 0 or more times	Optional private extension parameters

Name	Required	Description
announcementType	Y	Values limited to those in the AnnouncementTypes registry
announcementTag	N	For local use, can be used to identify the specific announcement played, for example, the name of the VoiceXML announcement script

3.5.3 SessionStartLogEvent

Each element that is call stateful logs the beginning of its processing of a SIP Session with SessionStartLogEvent. This allows differentiation between the start of a Call versus the start of a session that establishes a Call.

Name	Required	Description
clientAssignedIdentifier	N	An identifier assigned by the client
logEventType	Y	MUST be "SessionStartLogEvent"
timestamp	Y	The timestamp when the INVITE was received or sent by the element logging the event
elementId	Y	Element identifier of the element that logged the event
agencyId	Y	AgencyId of the agency that logged the event
agencyAgentId	C	See note 1
agencyPositionId	N	Identifier for the position that is handling the call.
callId	C	See note 2
incidentId	C	See note 3
callIdSIP	C	See note 4 CallIdSIP will be deprecated in a future version of NENA-STA-010 [3].
sessionId	C	See note 4
ipAddressPort	C	See note 5
extension	N, occurs 0 or more times	Optional private extension parameters

Name	Required	Description
direction	Y	MUST be one of "incoming" or "outgoing" where "incoming" means a session was received and "outgoing" means a session placed by the element
standardPrimaryCallType	N	Values limited to those in the LogEvent CallTypes registry
standardSecondaryCallType	N	Values limited to those in the LogEvent CallTypes registry
localCallType	N	Any value defined locally
localUse	N	Limited to 128 bytes
to	N	Target of session
from	N	Originator of session

3.5.4 SessionEndLogEvent

Each element that is call stateful logs the end of its processing of a SIP Session with SessionEndLogEvent. This allows differentiation between the end of a Call versus the end of a session that establishes a Call.

Name	Required	Description
clientAssignedIdentifier	N	An identifier assigned by the client
logEventType	Y	MUST be "SessionEndLogEvent"
timestamp	Y	The timestamp when the BYE or equivalents to these messages, or the final status code was received or sent by the element logging the event
elementId	Y	Element identifier of the element that logged the event
agencyId	Y	AgencyId of the agency that logged the event
agencyAgentId	C	See note 1
agencyPositionId	N	Identifier for the position that is handling the call.
callId	C	See note 2
incidentId	C	See note 3

Name	Required	Description
callIdSIP	C	See note 4 CallIdSIP will be deprecated in a future version of NENA-STA-010 [3].
sessionId	C	See note 4
ipAddressPort	C	See note 5
extension	N, occurs 0 or more times	Optional private extension parameters
direction	Y	MUST be one of "incoming" or "outgoing" where "incoming" means a session was received and "outgoing" means a session placed by the element
standardPrimaryCallType	N	Values limited to those in the LogEvent CallTypes registry
standardSecondaryCallType	N	Values limited to those in the LogEvent CallTypes registry
localCallType	N	Any value defined locally
localUse	N	Limited to 128 bytes
to	N	Target of session
from	N	Originator of session

3.5.5 SessionStateChangeLogEvent

Used by an element to log a SIP session state change, such as logging an "answered" event by a device.

Name	Required	Description
clientAssignedIdentifier	N	An identifier assigned by the client
logEventType	Y	Must be "SessionStateChangeLogEvent"
timestamp	Y	The timestamp when the state changed
elementId	Y	Element identifier of the element that logged the event
agencyId	Y	AgencyId of the agency that logged the event
agencyAgentId	C	See note 1

Name	Required	Description
agencyPositionId	N	Identifier for the position that is handling the call.
callId	C	See note 2
incidentId	C	See note 3
callIdSIP	C	See note 4 CallIdSIP will be deprecated in a future version of NENA-STA-010 [3].
sessionId	C	See note 4
ipAddressPort	C	See note 5
extension	N, occurs 0 or more times	Optional private extension parameters
state	Y	Values limited to those in the CallStates registry
direction	Y	MUST be one of two values: "incoming", meaning the element logging the state change received a message or other notice that changed the state; and "outgoing", meaning this element caused the state change.
legCallId	C	See note 6
targetId	C	See note 7
changeReason	N	Contains the reason why the state changed

3.5.6 Notes

Note 1: REQUIRED if the log record is traceable to an agent.

Note 2: For this particular Log Event, MUST be the Call Identifier associated with the call.

Note 3: For this particular Log Event, MUST be the Incident Tracking Identifier associated with the call.

Note 4: For this particular Log Event, MUST be the SIP Call-ID of the SIP session.

Note 5: For this particular Log Event, MUST be the IP address and port of the remote SIP session party

Note 6: For state changes that involve another "leg" of a call, such as AddParty, a "legCallId" member MUST contain the call id of that leg. If the leg is a SIP leg, this Id is the SIP Call-Id of the leg otherwise it may be another identifier for that call.

Note 7: If the target involved in the state change is not the element identified in the header field, the identifier of the target whose state changed MUST be included in a "targetId" member. If the target is a SIP device, this must be the SIP URI of the target.

4 IANA Actions

4.1 AnnouncementTypes Registry

IANA is requested to add the following values to the "AnnouncementTypes" registry in the "Emergency" area.

Name	Purpose	Reference
AutoAnswerGreeting	Indicates an announcement played automatically after a call is answered by an agent. Typically recorded with the agent's voice, this type of recorded greeting is used to standardize the answering of calls.	This document
NoAgentsAvailableAnnouncement	Indicates an announcement indicating no agents are currently available to take the call and the call will be answered by the next available agent.	This document
StandardAnnouncement	Indicates an announcement played to all calls regardless of the availability of agents to take the call.	This document

4.2 LogEvent Registry

IANA is requested to add the following values to the "LogEvent" registry in the "Emergency" area.

Name	**Purpose**	**Reference**
AnnouncementStartLogEvent	Used by an element to log the beginning of the playing of a (multimedia) announcement to the caller, such as an automatic answer greeting or an interactive media response announcement.	This document
AnnouncementEndLogEvent	Used by an element to log the end of the playing of a (multimedia) announcement to the caller, such as an automatic answer greeting or an interactive media response announcement.	This document

Name	Purpose	Reference
SessionStartLogEvent	Used by an element to log the beginning of a SIP session	This document
SessionEndLogEvent	Used by an element to log the end of a SIP session	This document
SessionStateChangeLogEvent	Used by an element to log a session state change to one of the states listed in the CallStates registry.	This document

5 Impacts and Considerations

5.1 Operations Impacts Summary

Agencies may use this standard as a reference in preparation of procurement documents for an NG9-1-1 system and in reviewing agencies' current systems or current policies to see if they conform. This standard may change how statistics are collected and reported. Reporting systems for NG9-1-1 may require operations personnel to identify differences between their current measurement systems and the metrics contained in this standard.

5.2 Technical Impacts Summary

System designs and/or measurements may change based on the adoption of the metrics defined in this standard. The metrics defined in this document are based on NG9-1-1 events and are therefore applicable to NG9-1-1 deployments only.

5.3 Security Impacts

There are no known security impacts from implementing this standard.

5.4 Recommendation for Additional Development Work

Metrics for incident processing and agent state transitions will need to be developed. In addition, consideration for NGCS SIP transactions (for example between Outbound Call Interface [OCIF] and Secure Telephone Identity Authentication Service [STI-AS]) when all of these transactions are identified.

5.5 Anticipated Timeline

This document provides specifications for systems that need to generate statistics. This document is not implementable on its own but is expected to be used immediately upon publication and on an ongoing basis for NG9-1-1.

5.6 Cost Factors

Some costs to the vendors for implementing the metrics in this document are expected. While implementations typically carry a cost, specific costs cannot be determined.

5.7 Cost Recovery Considerations

Not applicable.

5.8 Additional Impacts (non-cost related)

Having standardized statistics will improve the ability to make comparisons between systems and agencies in the processing of 9-1-1 calls. Definition of standardized call processing metrics will influence 9-1-1 Management Information Systems (MIS) to fully utilize all available call processing metrics. This document does not specify performance metrics and is not intended to replace documents that do specify performance metrics such as NFPA 1221 [6] or NENA STA-010 [3].

6 Abbreviations, Terms, and Definitions

See the NENA Knowledge Base for a Glossary of terms and abbreviations used in NENA documents. Abbreviations and terms used in this document are listed below with their definitions.

Term or Abbreviation (Expansion)	Definition / Description
Abandoned Call	An emergency Call in which the caller disconnects before the Call can be answered by the PSAP (Public Safety Answering Point).
Answered Call	A Call (excluding non-human-initiated Calls) that has been answered by an Agent and two-way communication has been established, irrespective of whether the Call was auto-answered for the Agent or if an auto-greeting message was played. For a non-human-initiated Call, that Call may be accepted by an automaton.
Attempted Call	A Call presented to a FE (such as a Call Handling Functional Element) regardless of whether successful completion status was achieved.
BCF (Border Control Function)	Provides a secure entry into the ESInet for emergency calls presented to the network. The BCF incorporates firewall, admission control, and may include anchoring of session and media as well as other security mechanisms to prevent deliberate or malicious attacks on PSAPs or other entities connected to the ESInet.

Term or Abbreviation (Expansion)	Definition / Description
Call	A generic term referring to any request for public safety assistance, regardless of the media used to make that request. This term may appear in conjunction with specific media, such as "voice Call", "video Call", "text Call", or "data-only Call" when the specific media is of importance. The term "non-interactive Call" refers to an emergency Call that is initiated automatically, carries data, does not establish a two-way interactive media session, and typically does not involve a human at the "initiating" end.
Diverted Call	A Call that was rerouted due to the nominal destination's unavailability or inability to accept. Calls may be diverted for conditions that are scheduled (e.g., maintenance, hours the PSAP is not staffed, etc.), or for events that cannot be scheduled (e.g., equipment or network failure, disasters, etc.)
ECRF (Emergency Call Routing Function)	A functional element in an NGCS (Next Generation Core Services) which is a LoST protocol server where location information (either civic address or geo-coordinates) and a Service URN serve as input to a mapping function that returns a URI used to route an emergency call toward the appropriate PSAP for the caller's location or towards a responder agency. - External ECRF: An ECRF instance that resides outside of an NGCS instance. - Internal ECRF: An ECRF instance that resides within and is only accessible from an NGCS instance.
ESInet (Emergency Services IP Network)	An ESInet is a managed IP network that is used for emergency services communications, and which can be shared by all public safety agencies. It provides the IP transport infrastructure upon which independent application platforms and core services can be deployed, including, but not restricted to, those necessary for providing NG9-1-1 services. ESInets may be constructed from a mix of dedicated and shared facilities. ESInets may be interconnected at local, regional, state, federal, national, and international levels to form an IP-based internetwork (network of networks). The term ESInet designates the network, not the services that ride on the network. See NG9-1-1 Core Services.
ESRP (Emergency Service Routing Proxy)	An i3 functional element which is a SIP proxy server that selects the next-hop routing within the ESInet, based on location and policy. There is an ESRP on the edge of the ESInet. There is

Term or Abbreviation (Expansion)	Definition / Description
	usually an ESRP at the entrance to an NG9-1-1 PSAP. There may be one or more intermediate ESRPs between them.
FE (Functional Element)	An abstract building block that consists of a set of interfaces and operations on those interfaces to accomplish a task. Mapping between functional elements and physical implementations may be one-to-one, one-to-many, or many-to-one.
HELD (HTTP Enabled Location Delivery)	A protocol that can be used to acquire Location Information (LI) from a LIS (Location Information Server) within an access network as defined in IETF RFC 5985.
IETF (Internet Engineering Task Force)	Lead standard-setting authority for Internet protocols.
LoST (Location-to-Service Translation) Protocol	A protocol that takes location information and a Service URN and returns a URI. Used generally for location-based call routing. In NG9-1-1, used as the protocol for the ECRF and LVF.
LIS (Location Information Server)	A Location Information Server (LIS) is a functional element that provides locations of endpoints. A LIS can provide Location-by-Reference, or Location-by-Value, and, if the latter, in geodetic or civic forms. A LIS can be queried by an endpoint for its own location, or by another entity for the location of an endpoint. In either case, the LIS receives a unique identifier that represents the endpoint, for example an IP address, circuit-ID, or Media Access Control (MAC) address, and returns the location (value or reference) associated with that identifier. The LIS is also the entity that provides the dereferencing service, exchanging a location reference for a location value.
LogEvent	A data structure defined in NENA-STA-010 that is used to convey Call processing and Incident processing event information to the Logging Service.
Misrouted Call	A Call routed to a PSAP that should not have received it due to a provisioning error (for example in the ECRF, in the PRF, or the LIS) or other misconfigurations.
NG9-1-1 Call Processing	The sequence of steps performed by operations personnel and NG9-1-1 systems in the handling of a NG9-1-1 Call.
*NGCS (*Next Generation 9-1-1 (NG9-1-1*) Core Services)*	The set of services needed to process a 9-1-1 call on an ESInet. It includes, but is not limited to, the ESRP, ECRF, LVF, BCF, Bridge, Policy Store, Logging Services, and typical IP services

Term or Abbreviation (Expansion)	Definition / Description
	such as DNS and DHCP. The term NG9-1-1 Core Services includes the services and not the network on which they operate. See Emergency Services IP Network.
Prematurely Disconnected Call	An Answered Call that terminated before the parties have finished their conversation.
PSAP (Public Safety Answering Point)	PSAP (Public Safety Answering Point) is a physical or virtual entity where 9-1-1 calls are delivered by the 9-1-1 Service Provider.
SIP (Session Initiation Protocol)	A protocol specified by the IETF (RFC 3261) that defines a method for establishing multimedia sessions over the Internet. Used as the call signaling protocol in VoIP, NENA i2 and NENA i3.

7 Recommended Reading and References

[1] National Emergency Number Association. *NENA Master Glossary of 9-1-1 Terminology*. NENA-ADM-000.24-2021. Arlington, VA: NENA, approved June 22, 2021.

[2] Internet Engineering Task Force. *Key words for use in RFCs to Indicate Requirement Levels*. S. Bradner. RFC 2119, March 1997.

[3] National Emergency Number Association, *NENA i3 Standard for Next Generation 9-1-1*, NENA-STA-010.3b-2021. Arlington, VA: NENA, approved October 7, 2021.

[4] Internet Engineering Task Force. *Basic Telephony SIP End-to-End Performance Metrics*. D. Malas and A. Morton. RFC 6076, January 2011.

[5] Internet Engineering Task Force. *RTP: A Transport Protocol for Real-Time Applications.* H. Schulzrinne, S. Casner, R. Rederick, and V. Jacobson. RFC 3550, July 2003.

[6] National Fire Protection Association. *Standard for the Installation, Maintenance, and Use of Emergency Services Communications Systems.* NFPA 1221. Quincy, MA: NFPA, 2019.

[7] National Emergency Number Association. *NENA Standard for the Conveyance of Emergency Incident Data Objects (EIDOs) between Next Generation (NG9-1-1) Systems and Applications*, NENA-STA-024.1-202Y (forthcoming).

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