A Hitchhikers Guide to the Session Initiation Protocol (SIP)
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Status of this Memo

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Abstract

   The Session Initiation Protocol (SIP) is the subject of numerous
   specifications that have been produced by the IETF. It can be
   difficult to locate the right document, or even to determine the set
   of Request for Comments (RFC) about SIP. Don't Panic! This
   specification serves as a guide to the SIP RFC series. It lists the
   specifications under the SIP umbrella, briefly summarizes each, and
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1. **Introduction**

   The Session Initiation Protocol (SIP) [1] is the subject of numerous
   specifications that have been produced by the IETF. It can be
   difficult to locate the right document, or even to determine the set
   of Request for Comments (RFC) about SIP. Don't Panic! This
   specification serves as a guide to the SIP RFC series. It lists the
   specifications under the SIP umbrella. For each specification, a
   paragraph or so description is included that summarizes the purpose
   of the specification. Each specification also includes a letter that
   designates its category in the standards track [2]. These values
   are:

   S: Standards Track (Proposed Standard, Draft Standard, or Standard)

   E: Experimental

   B: Best Current Practice

   I: Informational

   The specifications are grouped together by topic. Typically, SIP
   extensions fit naturally into topic areas, and implementations
   interested in a particular topic often implement many or all of the
   specifications in that area.

   This document itself is not an update to RFC 3261 or an extension to
   SIP. It is an informational document, meant to guide newcomers and
   implementors to the SIP suite of specifications.

2. **Scope of this Document**

   It is very difficult to enumerate the set of SIP specifications.
   This is because there are many protocols that are intimately related
   to SIP and used by nearly all SIP implementations, but are not
   formally SIP extensions. As such, this document formally defines a
   "SIP specification" as any specification that defines an extension to
   SIP itself, where an extension is a mechanism that changes or updates
   in some way a behavior specified in RFC 3261. This is in contrast to
   the "SIP family of specifications", which represent the set of
   specifications that define protocols that are integral parts of any
   SIP deployment, but are not SIP extensions per se. The SIP family of
   specifications includes the following specifications and their
   respective extensions:

RFC 3550: Real Time Transport Protocol (RTP) RTP [4] is the
   specification that started it all. It is the first in a long line
   of IETF specifications related to multimedia communications. Its
   initial version, RFC 1889 [3] was first specified in January of
   1996. RTP is used to carry multimedia traffic, including voice,
   video and text, and is set up by SIP. There are countless
   extensions and payload formats (possibly as many as 42) defined
   for RTP.

RFC 2327: The Session Description Protocol (SDP) RFC 2327 [5] defines
   the payload carried in SIP messages for describing a multimedia
   session. Closely related to SDP is the Offer/Answer mechanism,
   defined in RFC 3264 [6]. This mechanism defines how SDP is
   exchanged between a set of peers to agree on the makeup of a
   session. SDP itself has numerous extensions, including
   Interactive Connectivity Establishment (ICE) [7], which is the
   primary mechanism for NAT traversal for RTP streams set up by SIP.

RFC 3320: Signaling Compression (Sigcomp) RFC 3320 [8] defines a
   mechanism for compressing SIP messages over low bandwidth links.
   Sigcomp is not specific to SIP, though it was designed explicitly
   for SIP. Sigcomp has several extensions defined for improved
   compression.

RFC 3761: Telephone Number Mapping (ENUM) RFC 3761 [9] defines a
   mechanism to look up a phone number in DNS and obtain a URI, such
   as a SIP URI. ENUM defines numerous enum services, which
   represent particular types of information that can be obtained
   from the DNS related to that phone number.

RFC 3966: The tel URI scheme RFC 3966 [10] defines the tel URI
   scheme, used to identify phone numbers. The tel URI is carried
   primarily in SIP messages, and has numerous extensions defined for
   it.

RFC 3863: The Presence Information Data Format (PIDF) PIDF [11]
   defines an XML-based format for representing presence information.
   It is carried in SIP NOTIFY requests. Numerous extensions have
   been defined to PIDF for conveying additional pieces of presence
   information.

RFC XXXX: The Message Session Relay Protocol (MSRP) MSRP [12]
   provides a way to carry instant messages between agents. MSRP is
   to instant messages as RTP is to voice and video. It is a
   transport that is set up by SIP.

   RFC XXXX: Simple Traversal of UDP Through NAT (STUN) STUN [13]
      defines a basic toolkit for facilitating NAT traversal for
      protocols such as SIP. Extensions have been defined to STUN for
      functions such as packet relaying.

   RFC XXXX: XML Configuration Access Protocol (XCAP) XCAP [14] provides
      a means for clients to read and write XML-based application data
      in the network. It was originally conceived for managing buddy
      lists and presence authorization lists.

   RFC 3219: Telephony Routing over IP (TRIP) TRIP [15] defines a
      mechanism for exchanging SIP routes between administrative
      domains. Its derived from BGP.

   Despite the importance of the SIP family of specifications, this
   document concerns itself entirely with defining the set of
   specifications that make up SIP itself. Excluded from this list are
   requirements, architectures, registry definitions, non-normative
   frameworks, and processes. Best Current Practices are included when
   they are effectively standard mechanisms for accomplishing a task.

   Also excluded are definitions of MIME objects that are used by SIP,
   such as the Authenticated Identity Body (AIB) [16] and various XML
   documents and extensions used by SIP. Though they are used by SIP,
   they are not extensions to SIP. [[OPEN ISSUE: Is this arbitrary?
   Maybe they should be included if they are specific to SIP.]]

   The SIP change process [17] defines two types of extensions to SIP.
   These are normal extensions and the so-called P-headers, which are
   meant to be used in areas of limited applicability. P-headers cannot
   be defined in the standards track. For the most part, P-headers are
   not included in the listing here, with the exception of those which
   have seen general usage despite their P-header status.

3. Core SIP Specifications

   The core SIP specifications represent essential functionality for
   almost any implementation.

   RFC 3261: The Session Initiation Protocol (S) This is the core SIP
      protocol itself. RFC 3261 is an update to RFC 2543 [18]. It is
      the president of the galaxy as far as the suite of SIP
      specifications is concerned.

   RFC 3263: Locating SIP Servers (S) RFC 3263 [19] provides DNS
      procedures for taking a SIP URI, and determining a SIP server that
      is associated with that SIP URI. RFC 3263 is essential for any
      implementation using SIP with DNS. RFC 3263 makes use of both DNS

SRV records [20] and NAPTR records [21].

RFC 3265: SIP-Specific Event Notification RFC 3265 defines the
    SUBSCRIBE and NOTIFY methods. These two methods provide a general
    event notification framework for SIP. To actually use the
    framework, extensions need to be defined for specific event
    packages. An event package defines a schema for the event data,
    and describes other aspects of event processing specific to that
    schema. An RFC 3265 implementation is required when any event
    package is used.

RFC 3323: A Privacy Mechanism for the Session Initiation Protocol
    (SIP) (S) RFC 3323 [23] defines the Privacy header field, used by
    clients to request anonymity for their requests. Though it
    defines numerous privacy services, the only one broadly used is
    the one that supports privacy of the P-Asserted-ID header field
    [24].

RFC 3325: Private Extensions to SIP for Asserted Identity within
    Trusted Networks (I) Though its P-header status implies that it has
    limited applicability, RFC 3325 [24], which defines the
    P-Asserted-ID header field has been widely deployed. It is used
    as the basic mechanism for providing secure caller ID services.

RFC 3327: SIP Extension Header Field for Registering Non-Adjacent
    Contacts (S) RFC 3327 [25] defines the Path header field. This field
    is inserted by proxies between a client and their registrar. It
    allows inbound requests towards that client to traverse these
    proxies prior to being delivered to the user agent. It is
    essential in any SIP deployment that has edge proxies, which are
    proxies between the client and the home proxy or SIP registrar.
    It is also instrumental in the SIP NAT traversal specifications.

RFC 3581: An Extension to SIP for Symmetric Response Routing RFC 3581
    [26] defines the rport parameter of the Via header. It is an
    essential piece of getting SIP through NAT. NAT traversal for SIP
    is considered a core part of the specifications.

RFC 4320: Actions Addressing Issues Identified with the Non-INVITE
    Transaction in SIP (S) RFC 4320 [27] formally updates RFC 3261, and
    modifies some of the behaviors associated with non-INVITE
    transactions. These address some problems found in timeout and
    failure cases.

RFC XXXX: Enhancements for Authenticated Identity Management in SIP
    (S) RFC XXXX [28] defines a mechanism for providing a
    cryptographically verifiable identity of the calling party in a
    SIP request. Also known as "SIP Identity", this mechanism

provides an alternative to RFC 3325. It has seen little
deployment so far, but its importance as a key construct for
almost also anti-spam techniques makes it a core part of the SIP
specifications.

RFC XXXX: Obtaining and Using Globally Routable User Agent
Identifiers (GRUU) in SIP (S) RFC XXXX [29] defines a mechanism for
directing requests towards a specific UA instance. GRUU is
essential for features like transfer and provides another piece of
the SIP NAT traversal story.

RFC XXXX: Managing Client Initiated Connections through SIP (S) RFC
XXXX [30], also known as SIP outbound, defines important changes
to the SIP registration mechanism which enable delivery of SIP
messages towards a UA when it is behind a NAT. This specification
is the cornerstone of the SIP NAT traversal strategy.

4. Public Switched Telephone Network (PSTN) Interworking

Numerous extensions and usages of SIP related to interoperability and
communications with or through the PSTN.

RFC 2848: The PINT Service Protocol (P) This is one of the earliest
extensions to SIP. It defines procedures for using SIP to invoke
services that actually execute on the PSTN. Its main application
is for third party call control, allowing an IP host to set up a
call between two PSTN endpoints. PINT has a relatively narrow
focus and has not seen widespread deployment.

RFC 3910: The SPIRITS Protocol (P) Continuing the trend of naming
PSTN related extensions with alcohol references, SPIRITS [32]
defines the inverse of PINT. It allows a switch in the PSTN to
ask an IP element about how to proceed with call waiting. It was
developed primarily to support Internet Call Waiting (ICW).
Perhaps the next specification will be called the PGGB.

RFC 3372: SIP for Telephones (SIP-T): Context and Architectures
(I) SIP-T [33] defines a mechanism for using SIP between pairs of
PSTN gateways. Its essential idea is to tunnel ISUP signaling
between the gateways in the body of SIP messages. SIP-T motivated
the development of INFO [39]. SIP-T has seen widespread
implementation.

RFC 3398: ISUP to SIP Mapping (P) RFC 3398 [34] defines how to do
protocol mapping from the SS7 ISDN User Part (ISUP) signaling to
SIP. It is widely used in SS7 to SIP gateways and is part of the
SIP-T framework.

RFC 3578: Mapping of ISUP Overlap Signaling to SIP RFC 3578 [35]
   defines a mechanism to map overlap dialing into SIP. This
   specification is widely regarded as the ugliest SIP specification,
   as the introduction to the specification itself advises that it
   has many problems. Overlap signaling (the practice of sending
   digits into the network as dialed instead of waiting for complete
   collectoin of the called party number) is largely incompatible
   with SIP at some fairly fundamental levels. That said, RFC 3578
   is mostly harmless and has seen some usage.

RFC 3960: Early Media and Ringtone Generation in SIP (I) RFC 3960
   defines some guidelines for handling early media - the practice of
   sending media from the called party towards the caller - prior to
   acceptance of the call. Early media is generated only from the
   PSTN.

5. **General Purpose Infrastructure Extensions**

   These extensions are general purpose enhancements to SIP that can
   serve a wide variety of uses. However, they are not as widely used
   or as essential as the core specifications.

   RFC 3262: Reliability of Provisional Responses in SIP (S) SIP defines
      two types of responses to a request - final and provisional.
      Provisional responses are numbered from 100 to 199. In SIP, these
      responses are not sent reliably. This choice was made in RFC 2543
      since the messages were meant to just be truly informational, and
      rendered to the user. However, subsequent work on PSTN
      interworking demonstrated a need to map provisional responses to
      PSTN messages that needed to be sent reliably. RFC 3262 [37] was
      developed to allow reliability of provisional responses. The
      specification defines the PRACK method, used for indicating that a
      provisional response was received. Though it provides a generic
      capability for SIP, RFC 3262 implementations have been most common
      in PSTN interworking devices. However, PRACK brings a great deal
      of complication for relatively small benefit. As such, it has
      seen only mild levels of deployment.

   RFC 3311: The SIP UPDATE Method (S) RFC 3311 [38] defines the UPDATE
      method for SIP. This method is meant as a means for updating
      session information prior to the completion of the initial INVITE
      transaction. It was developed primarily to support RFC 3312 [68].

   RFC 2976: The INFO Method (S) RFC 2976 [39] was defined as an
      extension to RFC 2543. It defines a method, INFO, used to
      transport mid-dialog information that has no impact on SIP itself.
      Its driving application was the transport of PSTN related

information when using SIP between a pair of gateways. Though
originally conceived for broader use, it only found standardized
usage with SIP-T [33]. It has been used to support numerous
proprietary and non-interoperable extensions due to its poorly
defined scope.

RFC 3326: The Reason header field for SIP (S) RFC 3326 [40] defines
the Reason header field. It is used in requests, such as BYE, to
indicate the reason that the request is being sent.

RFC 3608: SIP Extension Header Field for Service Route Discovery
During Registration RFC 3608 [41] allows a client to determine, from
a REGISTER response, a path of proxies to use in requests it sends
outside of a dialog. In many respects, it is the inverse of the
Path header field, but has seen less usage since default outbound
proxies have been sufficient in many deployments.

RFC 3840: Indicating User Agent Capabilities in SIP (S) RFC 3840
defines a mechanism for carrying capability information about a
user agent in REGISTER requests and in dialog-forming requests
like INVITE. It has found use with conferencing (the isfocus
parameter declares that a user agent is a conference server) and
with applications like push-to-talk.

RFC 3841: Caller Preferences for SIP (S) RFC 3841 [43] defines a set
of headers that a client can include in a request to control the
way in which the request is routed downstream. It allows a client
to direct a request towards a UA with specific capabilities.

RFC 4028: Session Timers in SIP (S) RFC 4028 [44] defines a keepalive
mechanism for SIP signaling. It is primarily meant to provide a
way to cleanup old state in proxies that are holding call state
for calls from failed endpoints which were never terminated
normally. Despite its name, the session timer is not a mechanism
for detecting a network failure mid-call. Session timers
introduces a fair bit of complexity for relatively little gain,
and has thus seen little deployment.

RFC 4168: SCTP as a Transport for SIP RFC 4168 [45] defines how to
carry SIP messages over the Stream Control Transmission Protocol
(SCTP). SCTP has seen very limited usage for SIP transport.

RFC 4244: An Extension to SIP for Request History Information (S) RFC
4244 [46] defines the History-Info, which indicates information on
how a call came to be routed to a particular destination. Its
primary application was in support of voicemail services.

6. Minor Extensions

   These SIP extensions don't fit easily into a single specific use
   case. They have somewhat general applicability, but they solve a
   relatively small problem or provide an optimization.

   RFC XXXX: Suppression of the SIP REFER Implicit Subscription (S) RFC
      XXXX [47] defines an enhancement to REFER. REFER normally creates
      an implicit subscription to the target of the REFER. This
      subscription is used to pass back updates on the progress of the
      referral. This extension allows that implicit subscription to be
      bypassed as an optimization.

   RFC XXXX: Request Authorization through Dialog Identification in SIP
   (S) RFC XXXX [48] provides a mechanism that allows a UAS to authorize
      a request because the requestor proves it knows a dialog that is
      in progress with the UAS. The specification is useful in
      conjunction with the SIP application interaction framework [85].

   RFC XXXX: Conveying Feature Tags with the REFER Method in SIP (S) RFC
      XXXX [49] defines a mechanism for carrying RFC 3840 feature tags
      in REFER. It is useful for informing the target of the REFER
      about the characteristics of the REFER target.

   RFC XXXX: Requesting Answer Modes for SIP (S) RFC XXXX [50] defines
      an extension for indicating to the called party whether or not the
      phone should ring and/or be answered immediately. This is useful
      for push-to-talk and for diagnostic applications.

   RFC XXXX: Rejecting Anonymous Requests in SIP (S) RFC XXXX [51]
      defines a mechanism for a called party to indicate to the calling
      party that a call was rejected since the caller was anonymous.
      This is needed for implementation of the Anonymous Call Rejection
      (ACR) feature in SIP.

   RFC XXXX: Referring to Multiple Resources in SIP (S) RFC XXXX [52]
      allows a UA sending a REFER to ask the recipient of the REFER to
      generate multiple SIP requests, not just one. This is useful for
      conferencing, where a client would like to ask a conference server
      to eject multiple users.

7. Call Control Primitives

   Numerous SIP extensions provide a toolkit of dialog and call
   management techniques. These techniques have been combined together
   to build many SIP-based services.

RFC 3515: The REFER Method (S) REFER [53] defines a mechanism for
   asking a user agent to send a SIP request. Its a form of SIP
   remote control, and is the primary tool used for call transfer in
   SIP.

RFC 3725: Best Current Practices for Third Party Call Control (3pcc)
(B) RFC 3725 [54] defines a number of different call flows that allow
   one SIP entity, called the controller, to create SIP sessions
   amongst other SIP user agents.

RFC 3891: The SIP Replaces Header RFC 3891 [55] defines a mechanism
   that allows a new dialog to replace an existing dialog. It is
   useful for certain advanced transfer services.

RFC 3892: The SIP Referred-By Mechanism (S) RFC 3892 [56] defines the
   Referred-By header field. It is used in requests triggered by
   REFER, and provides the identity of the referring party to the
   referred-to party.

RFC 3911: The SIP Join Header Field (S) RFC 3911 [57] defines the
   Join header field. When sent in an INVITE, it causes the
   recipient to join the resulting dialog into a conference with
   another dialog in progress.

RFC 4117: Transcoding Services Invocation in SIP Using Third Party
Call Control (I) RFC 4117 [58] defines how to use 3pcc for the
   purposes of invoking transcoding services for a call.

8. Event Packages

   RFC 3265 defines a basic framework for event notification in SIP. It
   introduces the notion of an event package, which is a collection of
   related state and event information. Much of the state and events in
   SIP systems have event packages, allowing other entities to learn
   about changes in that state.

   RFC 3903: SIP Extension for Event State Publication (S) RFC 3903 [59]
      defines the PUBLISH method. It is not an event package, but is
      used by all event packages as a mechanism for pushing an event
      into the system.

   RFC 3680: A SIP Event Package for Registrations (S) RFC 3680 [60]
      defines an event package for finding out about changes in
      registration state.

   RFC 3842: A Message Summary and Message Waiting Indication Event
   Package for SIP (S) RFC 3842 [73] defines a way for a user agent to
      find out about voicemails and other messages that are waiting for
      it. Its primary purpose is to enable the voicemail waiting lamp
      on most business telephones.

   RFC 3856: A Presence Event Package for SIP (S) RFC 3856 [62] defines
      an event package for indicating user presence through SIP.

   RFC 3857: A Watcher Information Event Template Package for SIP
      (S) RFC 3857 [63], also known as winfo, provides a mechanism for a
      user agent to find out what subscriptions are in place for a
      particular event package. Its primary usage is with presence, but
      it can be used with any event package.

   RFC 4235: An INVITE Initiated Dialog Event Package for SIP (S) RFC
      4235 [64] defines an event package for learning the state of the
      dialogs in progress at a user agent.

   RFC XXXX: A SIP Event Package for Conference State (S) RFC XXXX [65]
      defines a mechanism for learning about changes in conference
      state, including group membership.

   RFC XXXX: A SIP Event Package for Keypress Stimulus (KPML) (S) RFC
      XXXX [66] defines a way for an application in the network to
      subscribe to the set of keypresses made on the keypad of a
      traditional telephone.

   RFC XXXX: SIP Event Package for Voice Quality Reporting (S) RFC XXXX
      [67] defines a SIP event package that enables the collection and
      reporting of metrics that measure the quality for Voice over
      Internet Protocol (VoIP) sessions.

9. Quality of Service

   Several specifications concern themselves with the interactions of
   SIP with network Quality of Service (QoS) mechanisms.

   RFC 3312:Integration of Resource Management and SIP (S) RFC 3312
      [68], updated by RFC 4032 [69] defines a way to make sure that
      the phone of the called party doesn't ring until a QoS reservation
      has been installed in the network. It does so by defining a
      general preconditions framework, which defines conditions that
      must be true in order for a SIP session to proceed

RFC 3313: Private SIP Extensions for Media Authorization (I) RFC 3313
   [70] defines a P-header that provides a mechanism for passing an
   authorization token between SIP and a network QoS reservation
   protocol like RSVP. Its purpose is to make sure network QoS is
   only granted if a client has made a SIP call through the same
   providers network. This specification is sometimes referred to as
   the SIP walled garden specification by the truly paranoid androids
   in the SIP community. This is because it requires coupling of
   signaling and the underlying IP network.

10. Operations and Management

   Several specifications have been defined to support operations and
   management of SIP systems. These include mechanisms for
   configuration and network diagnostics.

   RFC XXXX: Diagnostic Responses for SIP Hop Limit Errors (S) RFC XXXX
      defines a mechanism for including diagnostic information in a 483
      response. This response is sent when the hop-count of a SIP
      request was exceeded.

   RFC XXXX: A Framework for SIP User Agent Profile Delivery (S) RFC
      XXXX [71] defines a mechanism that allows a SIP user agent to
      bootstrap its configuration from the network, and receive updates
      to its configuration should it change. This is considered an
      essential piece of deploying a usable SIP network.

   RFC XXXX: SIP Event Package for Voice Quality Reporting (S) RFC XXXX
      [67] defines a SIP event package that enables the collection and
      reporting of metrics that measure the quality for Voice over
      Internet Protocol (VoIP) sessions.

11. SIP Compression

   Sigcomp [8] was defined to allow compression of SIP messages over low
   bandwidth links. Sigcomp is not formally part of SIP. However,
   usage of Sigcomp with SIP has required extensions to SIP.

   RFC 3486: Compressing SIP (S) RFC 3486 [72] defines a SIP URI
      parameter that can be used to indicate that a SIP server supports
      Sigcomp.

12. SIP Service URIs

   Several extensions define well-known services that can be invoked by

constructing requests with the specific structures for the Request
URI, resulting in specific behaviors at the UAS.

RFC 3087: Control of Service Context using Request URI (I) RFC 3087
   [74] introduced the context of using Request URIs, encoded
   appropriately, to invoke services.

RFC XXXX: A SIP Event Notification Extension for Resource Lists
   (S) RFC XXXX [75] defines a resource called a Resource List
   Server. A client can send a subscribe to this server. The server
   will generate a series of subscriptions, and compile the resulting
   information and send it back to the subscriber. The set of
   resources that the RLS will subscribe to is a property of the
   request URI in the SUBSCRIBE request.

RFC XXXX: Subscriptions To Request-Contained Resource Lists in SIP
(S) RFC XXXX [76] allows a client to subscribe to a resource called a
   Resource List Server. This server will generate a series of
   subscriptions, and compile the resulting information and send it
   back to the subscriber. For this specification, the list of
   things to subscribe to is in the body of the SUBSCRIBE request.

RFC XXXX: Multiple-Recipient MESSAGE Requests in SIP (S) RFC XXXX
   [77] is similar to [76]. However, instead of subscribing to the
   resource, a MESSAGE request is sent to the resource, and it will
   send a copy to each recipient.

RFC XXXX: Conference Establishment Using Request-Contained Lists in
SIP (S) RFC XXXX [78] is similar to [76]. However, instead of
   subscribing to the resource, an INVITE request is sent to the
   resource, and it will act as a conference focus and generate an
   invitation to each recipient in the list.

13. Security Mechanisms

   Several extensions provide additional security features to SIP.

   RFC 3853: S/MIME AES Requirement for SIP RFC 3853 [79] is a brief
      specification that updates the cryptography mechanisms used in SIP
      S/MIME. However, SIP S/MIME has seen very little deployment.

   RFC 3329: Security Mechanism Agreement for SIP (S) RFC 3329 [80]
      defines a mechanism to prevent bid-down attacks in conjunction
      with SIP authentication. The mechanism has seen very limited
      deployment. It was defined as part of the 3gpp IMS specification
      suite, and is needed only when there are a multiplicity of
      security mechanisms deployed at a particular server. In practice,

   this has not been the case.

   RFC XXXX: End-to-Middle Security in SIP (S) RFC XXXX [81] defines
      mechanisms for encrypting content from user agents to specific
      network intermediaries.

14. Instant Messaging and Presence

   SIP provides extensions for instant messaging and presence.

   RFC 3428: SIP Extension for Instant Messaging RFC 3428 [82] defines
      the MESSAGE method, used for sending a page mode instant message.

   RFC 3856: A Presence Event Package for SIP (S) RFC 3856 [62] defines
      an event package for indicating user presence through SIP.

   RFC 3857: A Watcher Information Event Template Package for SIP
      (S) RFC 3857 [63], also known as winfo, provides a mechanism for a
      user agent to find out what subscriptions are in place for a
      particular event package. Its primary usage is with presence, but
      it can be used with any event package.

15. Emergency Services

   Emergency services here covers both emergency calling (for example,
   911 in the United States), and pre-emption services, which allow
   authorized individuals to gain access to network resources in time of
   emergency.

   RFC 4411: Extending the SIP Reason Header for Preemption Events
      (S) RFC 4411 [83] defines an extension to the Reason header,
      allowing a UA to know that its dialog was torn down because a
      higher priority session came through.

   RFC 4412: Communications Resource Priority for SIP (S) RFC 4412 [84]
      defines a new header field, Resource-Priority, that allows a
      session to get priority treatment from the network.

16. Security Considerations

   This specification is an overview of existing specifications, and
   does not introduce any security considerations on its own.

17. IANA Considerations

   None.

18. Informative References

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