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Requirements for an End-to-End Session Identification in IP-Based Multimedia Communication Networks
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Abstract

This document specifies the requirements for an end-to-end session identifier in IP-based multimedia communication networks. This identifier would enable endpoints, intermediate devices, and management and monitoring systems to identify a session end-to-end across multiple SIP devices, hops, and administrative domains.

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1. Introduction

IP-based multimedia communication systems like SIP [1] and H.323 [2] have the concept of a "call identifier" that is globally unique. The identifier is intended to represents an end-to-end communication session from the originating device to the terminating device. Such an identifier is useful for troubleshooting, billing, session tracking, and so forth.

Unfortunately, there are a number of factors that contribute to the fact that the current call identifiers defined in SIP and H.323 are not suitable for end-to-end session identification. Perhaps most significant is the fact that the syntax for the call identifier in SIP and H.323 is different between the two protocols. This important fact makes it impossible for call identifiers to be exchanged end-to-end when a network utilizes one or more session protocols.

Another reason why the current call identifiers are not suitable to identify the session end-to-end is that in real-world deployments devices like Back-to-Back User Agents often change the values as the session signaling passes through. This is true even when a single session protocol is employed and not a byproduct of protocol interworking.

Lastly, identifiers that might have been used to identify a session end-to-end fail to meet that need when sessions are manipulated through supplementary service interactions. For example, when a session is transferred or if a PBX joins or merges two communication sessions together locally, the end-to-end properties of currently-defined identifiers are lost.

2. Conventions used in this document

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119 [3] when they appear in ALL CAPS. These words may also appear in this document in lower case as plain English words, absent their normative meanings.

3. Terminology

SIP defines additional terms used in this document that are specific to the SIP domain such as "proxy"; "registrar"; "redirect server"; "user agent server" or "UAS"; "user agent client" or "UAC"; "user agent" (UA); "back-to-back user agent" or "B2BUA"; "dialog"; "transaction"; "server transaction".

In this document, the word "session" refers to a "communication session" that may exist between two SIP user agents and that may pass through one or more intermediary devices, including B2BUAs or SIP proxies. A communication session consists of one of more SIP transactions. A very simple communication session may be a single out-of-dialog SIP transaction (e.g., MESSAGE/200). In the case of a SIP dialog, the SIP message exchange includes a dialog creating INVITE transaction, and zero of more subsequent SIP transactions within the same dialog (e.g., ACK, re-INVITE, BYE). Due to the existence of middle boxes, there may be multiple related SIP transactions or dialogs chained together along the path from one end of the communication session to the other. For example:

```
A               Middlebox               B
SIP trans1 -----------------> SIP trans 2
SIP dialog 1 <--------------- SIP dialog 2
```

Figure 1 - Communication Session through a Middlebox

In such cases, the two SIP transactions or dialogs are part of a single communication session.

The term "end-to-end" in this document means the communication session from the point of origin, passing through any number of intermediaries, to the ultimate point of termination. It is recognized that legacy devices may not support the "end-to-end" session identifier, though an identifier might be created by an intermediary when it is absent from the session signaling.

4. Session Identifier Use Cases

4.1. End-to-end identification of a communication session

For SIP messaging that either does not involve SIP servers or only involves SIP proxies, the Call-ID header value sufficiently identifies each SIP message within a transaction or dialog. This is not the case when either B2BUAs or SBCs are in the signaling path between UAs. Therefore, we need the ability to identify each communication session through a single SIP header-value regardless of which type of SIP servers are in the signaling path between UAs. For transactions that create a dialog, each message within the same dialog MUST use the same identifier.

Derived Requirements: All Requirements in Section 4

4.2. Protocol Interworking

A communication session might originate in an H.323 endpoint and pass through a Session Border Controller before ultimately reaching a terminating SIP user agent. Likewise, a call might originate on a SIP user agent and terminate on an H.323 endpoint. It MUST be possible to identify such sessions end-to-end across the plurality of devices, networks, or administrative domains.

It is expected that the ITU-T will define protocol elements for H.323 to make the end-to-end signaling possible.

Derived Requirements: REQ5, REQ7

4.3. Traffic Monitoring

UA A and UA B communicate using SIP messaging with a SIP B2BUA acting as a middlebox which belongs to a SIP service provider. For privacy reasons, the B2BUA changes the SIP headers that reveal information

related to the SIP users, device or domain identity. The service provider uses an external device to monitor and log all SIP traffic coming to and from the B2BUA. In the case of failures reported by the customer or when security issue arise (e.g. theft of service), the service provider has to analyze the logs from the past several days or weeks and correlates those messages which were messages for a single end-to-end SIP session.

For this scenario, we must consider three particular use cases:

a) The UAs A and B support the end-to-end Session Identifier.

   Derived Requirements: REQ1, REQ3, REQ4, REQ6.

b) Only the UA A supports the end-to-end Session Identifier, the UA B does not.

   Derived Requirements: REQ1, REQ3, REQ4, REQ5, REQ6.

c) UA A and UA B do not support the end-to-end Session Identifier.

   Derived Requirements: REQ1, REQ3, REQ4, REQ5, REQ6

4.4. Tracking transferred sessions

It is difficult to track which SIP messages where involved in the same call across transactions, especially when invoking supplementary services such as call transfer or call join. There exists a need for the ability to track communications sessions as they are transferred, one side at a time, until completion of the session (i.e., until a BYE is sent).

Derived Requirements: REQ1, REQ2, REQ9

4.5. Session Signal Logging

An after-the-fact search of SIP messages to determine which messages were part of the same transaction or call is difficult when B2BUAs and SBCs are involved in the signaling between UAs. Mapping more than one Call-ID together can be challenging because all of the values in SIP headers on one side of the B2BUA or SBC will likely be different than those on the other side. If multiple B2BUAs and/or SBCs are in the signaling path, more than two sets of header values will exist, creating more of a challenge. Creating a common header value through all SIP entities will greatly reduce any challenge for the purposes of debugging, communication tracking (such as for security purposes in case of theft of service), etc.

Derived Requirements: REQ1, REQ3, REQ5, REQ6

4.6. Identifier Syntax

A syntax that is too restrictive (e.g., one that allows special characters or a very long identifier) would make it difficult to encode the identifier in other protocols. Therefore, the syntax of the identifier should be reasonably restrictive.

Derived Requirements: REQ8

4.7. 3PCC Use Case

Third party call control refers to the ability of an entity to create a call in which communication is actually between two or more parties. For example, a B2BUA acting as a third party controller could establish a call between two SIP UA's using 3PCC procedures as described in section 4.1 of RFC 3725 [7], the flow for which is reproduced below.

```
              A              Controller               B
              |(1) INVITE no SDP  |                   |
              |<------------------|                   |
              |(2) 200 offer1     |                   |
              |------------------>|                   |
              |                   |(3) INVITE offer1  |
              |                   |------------------>|
              |                   |(4) 200 OK answer1 |
              |                   |<------------------|
              |                   |(5) ACK            |
              |                   |------------------>|
              |(6) ACK answer1    |                   |
              |<------------------|                   |
              |(7) RTP            |                   |
              |.......................................|
```

Figure 2 - Session-ID 3PCC Scenario

Such a flow must result in a single session identifier being used for the communication session between UA A and UA B. This use case does not extend to three SIP UAs.

Derived Requirements: REQ9

5. Requirements for the End-to-End Session Identifier

The following requirements are derived from the use cases and additional constraints regarding the construction of the identifier.

REQ1: It must be possible for an administrator or an external device which monitors the SIP-traffic to use the identifier to identify

those dialogs, transactions and messages which were at some point in time components of a single end-to-end SIP session (e.g., parts of the same call).

REQ2: It must be possible to correlate two end-to-end sessions when a session is transferred or if two different sessions are joined together via an intermediary (e.g., a PBX).

REQ3: The solution must require that the identifier, if present, pass unchanged through SIP B2BUAs or other intermediaries.

REQ4: The identifier must not reveal any information related to any SIP user, device or domain identity. This includes any IP Address, port, hostname, domain name, username, Address-of-Record, MAC address, IP address family, transport type, subscriber ID, Call-ID, tags, or other SIP header or body parts.

REQ5: It must be possible to identity SIP traffic with an end-to-end session identifier from and to end devices that do not support this new identifier, such as by allowing an intermediary to inject an identifier into the session signaling.

REQ6: The identifier should be unique in time and space, similar to the Call-ID.

REQ7: The identifier should be constructed in such a way as to make it suitable for transmission in SIP and H.323.

REQ8: The identifier should use a restricted syntax and length so as to allow the identifier to be used in other protocols.

REQ9: It must be possible to correlate two end-to-end sessions when the sessions are created by a third party controller using 3PCC procedures shown in Figure 1 of RFC 3725 [7].

6. Related Work in other Standards Organizations

6.1. Coordination with the ITU-T

IP multimedia networks are often comprised of a mix of session protocols like SIP and H.323. A benefit of the Session Identifier is that it uniquely identifies a communication session end-to-end across session protocol boundaries. Therefore, the need for coordinated standardization activities across Standards Development Organizations (SDOs) is imperative.

To facilitate this, a parallel effort is underway in the ITU-T to introduce the Session Identifier for the H.323 protocol. The ITU-T SG16 has approved contribution C.552 [5] as a work item with the

intent that it be a coordinated and synchronized effort between the ITU-T and the IETF.

6.2. Requirements within 3GPP

3GPP identified in their Release 9 the need for a Session Identifier for O&M purposes to correlate flows in an end-to-end communication session. TS24.229 (IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP)) [6] points to the fact that the Session Identifier can be used to correlate SIP messages belonging to the same session. In the case where signaling passes through SIP entities like B2BUAs, the end-to-end session identifier indicates that these dialogs belong to the same end-to-end SIP communication session.

7. Security Considerations

An end-to-end identifier, if not properly constructed, could provide information that would allow one to identify the individual, device, or domain initiating or terminating a communication session. In adherence with REQ4, the solution produced in accordance with these requirements MUST NOT provide any information that allow one to identify a person, device, or domain. This means that information elements such as the MAC address or IP address MUST NOT be used when constructing the end-to-end session identifier.

8. IANA Considerations

There are no IANA considerations associated with this document.

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10. Contributors

Two other people originally participated as co-authors and provided substantial contributions to this document, namely Roland Jesske, Parthasarathi Ravindran.

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