

Benchmarking Methodology Working
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C. Davids
Illinois Institute of Technology
V. Gurbani
Bell Laboratories, Alcatel-Lucent
S. Poretsky
Allot Communications
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**Terminology for Benchmarking Session Initiation Protocol (SIP)
Networking Devices
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Abstract

This document provides a terminology for benchmarking the SIP performance of networking devices. The term performance in this context means the capacity of the device- or system-under-test to process SIP messages. Terms are included for test components, test setup parameters, and performance benchmark metrics for black-box benchmarking of SIP networking devices. The performance benchmark metrics are obtained for the SIP signaling plane only. The terms are intended for use in a companion methodology document for characterizing the performance of a SIP networking device under a variety of conditions. The intent of the two documents is to enable a comparison of the capacity of SIP networking devices. Test setup parameters and a methodology document are necessary because SIP allows a wide range of configuration and operational conditions that can influence performance benchmark measurements. A standard terminology and methodology will ensure that benchmarks have consistent definition and were obtained following the same procedures.

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1. Terminology

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 [BCP 14](#), [RFC2119](#) [[RFC2119](#)]. [RFC 2119](#) defines the use of these key words to help make the intent of standards track documents as clear as possible. While this document uses these keywords, this document is not a standards track document. The term Throughput is defined in [RFC2544](#) [[RFC2544](#)].

For the sake of clarity and continuity, this document adopts the template for definitions set out in [Section 2 of RFC 1242](#) [[RFC1242](#)].

The terms Device Under Test (DUT) and System Under Test (SUT) are defined in the following BMWG documents:

Device Under Test (DUT) (c.f., [Section 3.1.1 RFC 2285](#) [[RFC2285](#)]).
System Under Test (SUT) (c.f., [Section 3.1.2, RFC 2285](#) [[RFC2285](#)]).

Many commonly used SIP terms in this document are defined in [RFC 3261](#) [[RFC3261](#)]. For convenience the most important of these are reproduced below. Use of these terms in this document is consistent with their corresponding definition in [[RFC3261](#)].

- o Call Stateful: A proxy is call stateful if it retains state for a dialog from the initiating INVITE to the terminating BYE request. A call stateful proxy is always transaction stateful, but the converse is not necessarily true.
- o Stateful Proxy: A logical entity that maintains the client and server transaction state machines defined by this specification during the processing of a request, also known as a transaction stateful proxy. The behavior of a stateful proxy is further defined in [Section 16 of RFC 3261](#) [[RFC3261](#)]. A transaction stateful proxy is not the same as a call stateful proxy.
- o Stateless Proxy: A logical entity that does not maintain the client or server transaction state machines defined in this specification when it processes requests. A stateless proxy forwards every request it receives downstream and every response it receives upstream.
- o Back-to-back User Agent: A back-to-back user agent (B2BUA) is a logical entity that receives a request and processes it as a user agent server (UAS). In order to determine how the request should be answered, it acts as a user agent client (UAC) and generates requests. Unlike a proxy server, it maintains dialog state and must participate in all requests sent on the dialogs it has established. Since it is a concatenation of a UAC and a UAS, no explicit definitions are needed for its behavior.

- o Loop: A request that arrives at a proxy, is forwarded, and later arrives back at the same proxy. When it arrives the second time, its Request-URI is identical to the first time, and other header fields that affect proxy operation are unchanged, so that the proxy will make the same processing decision on the request it made the first time. Looped requests are errors, and the procedures for detecting them and handling them are described by the SIP protocol[RFC3261] and also by [RFC 5393](#)

2. Introduction

Service Providers and IT Organizations deliver Voice Over IP (VoIP) and Multimedia network services based on the IETF Session Initiation Protocol (SIP) [[RFC3261](#)]. SIP is a signaling protocol originally intended to be used to dynamically establish, disconnect and modify streams of media between end users. As it has evolved it has been adopted for use in a growing number of services and applications. Many of these result in the creation of a media session, but some do not. Examples of this latter group include text messaging and subscription services. The set of benchmarking terms provided in this document is intended for use with any SIP-enabled device performing SIP functions in the interior of the network, whether or not these result in the creation of media sessions. The performance of end-user devices is outside the scope of this document.

A number of networking devices have been developed to support SIP-based VoIP services. These include SIP Servers, Session Border Controllers (SBC), Back-to-back User Agents (B2BUA), and SIP-Aware Stateful Firewalls. These devices contain a mix of voice and IP functions whose performance may be reported using metrics defined by the equipment manufacturer or vendor. The Service Provider or IT Organization seeking to compare the performance of such devices will not be able to do so using these vendor-specific metrics, whose conditions of test and algorithms for collection are often unspecified. SIP functional elements and the devices that include them can be configured many different ways and can be organized into various topologies. These configuration and topological choices impact the value of any chosen signaling benchmark. Unless these conditions-of-test are defined, a true comparison of performance metrics will not be possible. Some SIP-enabled network devices terminate or relay media as well as signaling. The processing of media by the device impacts the signaling performance. As a result, the conditions-of-test must include information as to whether or not the device under test processes media and if the device does process media, a description of the media handled and the manner in which it is handled. This document and its companion methodology document [[I-D.ietf-bmwg-sip-bench-meth](#)] provide a set of black-box benchmarks

for describing and comparing the performance of devices that incorporate the SIP User Agent Client and Server functions and that operate in the network's core.

The definition of SIP performance benchmarks necessarily includes definitions of Test Setup Parameters and a test methodology. These enable the Tester to perform benchmarking tests on different devices and to achieve comparable results. This document provides a common set of definitions for Test Components, Test Setup Parameters, and Benchmarks. All the benchmarks defined are black-box measurements of the SIP signaling plane. The Test Setup Parameters and Benchmarks defined in this document are intended for use with the companion Methodology document. Benchmarks of internal DUT characteristics (also known as white-box benchmarks) such as Session Attempt Arrival Rate, which is measured at the DUT, are described in [Appendix A](#) to allow additional characterization of DUT behavior with different distribution models.

2.1. Scope

The scope of this work item is summarized as follows:

- o This terminology document describes SIP signaling performance benchmarks for black-box measurements of SIP networking devices. Stress and debug scenarios are not addressed in this work item.
- o The DUT must be an [RFC 3261](#) capable network equipment. This may be a Registrar, Redirect Server, Stateless Proxy or Stateful Proxy. A DUT MAY also include a B2BUA, SBC functionality. The DUT MAY be a multi-port SIP-to-switched network gateway implemented as a SIP UAC or UAS.
- o The DUT MAY include an internal SIP Application Level Gateway (ALG), firewall, and/or a Network Address Translator (NAT). This is referred to as the "SIP Aware Stateful Firewall."
- o The DUT or SUT MUST NOT be end user equipment, such as personal digital assistant, a computer-based client, or a user terminal.
- o The Tester acts as multiple "Emulated Agents" (EA) that initiate (or respond to) SIP messages as session endpoints and source (or receive) associated media for established connections.
- o SIP Signaling in presence of Media
 - * The media performance is not benchmarked in this work item.
 - * It is RECOMMENDED that SIP signaling plane benchmarks be performed with media present, but this is optional.
 - * The SIP INVITE requests MUST include the SDP body.
 - * The type of DUT dictates whether the associated media streams traverse the DUT or SUT. Both scenarios are within the scope of this work item.
 - * SIP is frequently used to create media streams; the signaling plane and media plane are treated as orthogonal to each other in this document. While many devices support the creation of

media streams, benchmarks that measure the performance of these streams are outside the scope of this document and its companion methodology document [[I-D.ietf-bmwg-sip-bench-meth](#)]. Tests may be performed with or without the creation of media streams. The presence or absence of media streams MUST be noted as a condition of the test as the performance of SIP devices may vary accordingly. Even if the media is used during benchmarking, only the SIP performance will be benchmarked, not the media performance or quality.

- o Both INVITE and non-INVITE scenarios (such as Instant Messages or IM) are addressed in this document. However, benchmarking SIP presence is not a part of this work item.
- o Different transport mechanisms -- such as UDP, TCP, SCTP, or TLS -- may be used. The specific transport mechanism MUST be noted as a condition of the test as the performance of SIP devices may vary accordingly.
- o Looping and forking options are also considered since they impact processing at SIP proxies.
- o REGISTER and INVITE requests may be challenged or remain unchallenged for authentication purpose. Whether or not the REGISTER and INVITE requests are challenged is a condition of test which will be recorded along with other such parameters which may impact the SIP performance of the device or system under test.
- o Re-INVITE requests are not considered in scope of this work item since the benchmarks for INVITES are based on the dialog created by the INVITE and not on the transactions that take place within that dialog.
- o Only session establishment is considered for the performance benchmarks. Session disconnect is not considered in the scope of this work item. This is because our goal is to determine the maximum throughput of the device or system under test, that is the number of simultaneous SIP sessions that the device or system can support. It is true that there are BYE requests being created during the test process. These transactions do contribute to the load on the device or system under test and thus are accounted for in the metric we derive. We do not seek a separate metric for the number of BYE transactions a device or system can support.
- o SIP Overload [[I-D.ietf-soc-overload-design](#)] is within the scope of this work item. We test to failure and then can continue to observe and record the behavior of the system after failures are recorded. The cause of failure is not within the scope of this work. We note the failure and may continue to test until a different failure or condition is encountered. Considerations on how to handle overload are deferred to work progressing in the SOC working group [[I-D.ietf-soc-overload-control](#)]. Vendors are, of course, free to implement their specific overload control behavior as the expected test outcome if it is different from the IETF recommendations. However, such behavior MUST be documented and

interpreted appropriately across multiple vendor implementations. This will make it more meaningful to compare the performance of different SIP overload implementations.

- o IMS-specific scenarios are not considered, but test cases can be applied with 3GPP-specific SIP signaling and the P-CSCF as a DUT.

2.2. Benchmarking Models

This section shows ten models to be used when benchmarking SIP performance of a networking device. Figure 1 shows the configuration needed to benchmark the tester itself. This model will be used to establish the limitations of the test apparatus.

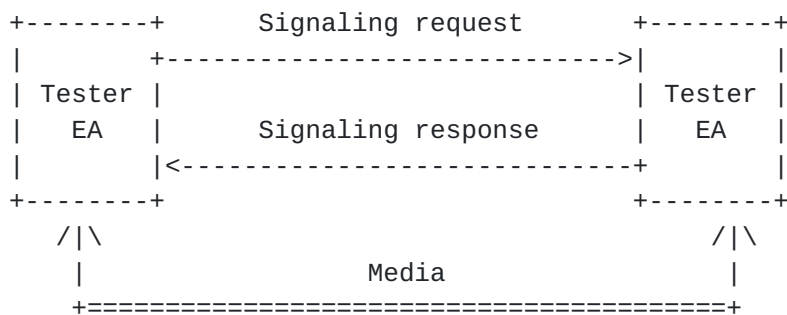


Figure 1: Baseline performance of the Emulated Agent without a DUT present

Figure 2 shows the DUT playing the role of a user agent client (UAC), initiating requests and absorbing responses. This model can be used to baseline the performance of the DUT acting as an UAC without associated media.

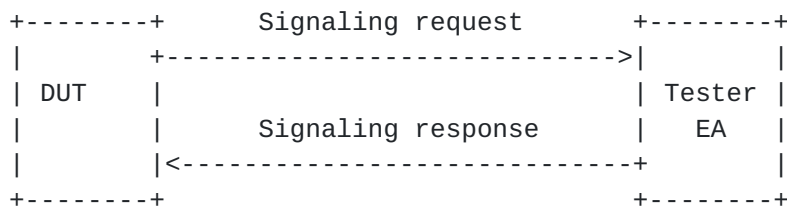


Figure 2: Baseline performance for DUT acting as a user agent client without associated media

Figure 3 shows the DUT plays the role of a user agent server (UAS), absorbing the requests and sending responses. This model can be used as a baseline performance for the DUT acting as a UAS without

associated media.

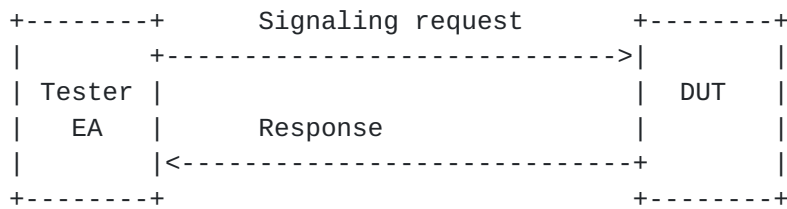


Figure 3: Baseline performance for DUT acting as a user agent server without associated media

Figure 4 shows the DUT plays the role of a user agent client (UAC), initiating requests and absorbing responses. This model can be used as a baseline performance for the DUT acting as a UAC with associated media.

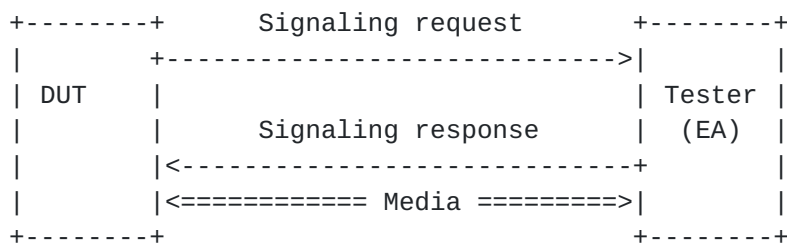


Figure 4: Baseline performance for DUT acting as a user agent client with associated media

Figure 5 shows the DUT plays the role of a user agent server (UAS), absorbing the requests and sending responses. This model can be used as a baseline performance for the DUT acting as a UAS with associated media.

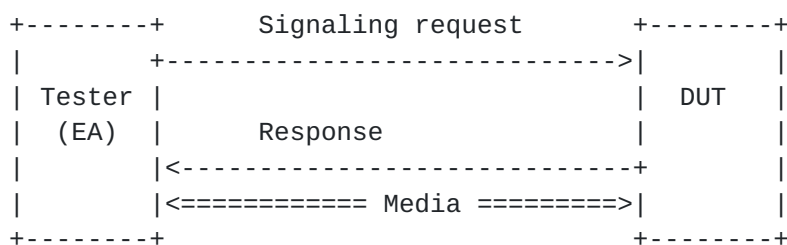


Figure 5: Baseline performance for DUT acting as a user agent server

with associated media

Figure 6 shows that the Tester acts as the initiating and responding EA as the DUT/SUT forwards Session Attempts.

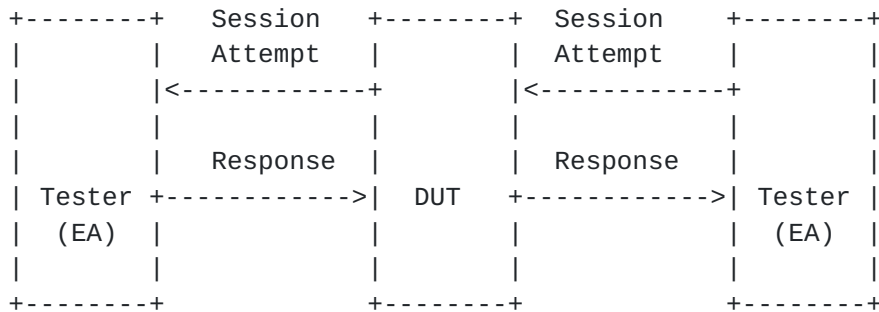


Figure 6: DUT/SUT performance benchmark for session establishment without media

Figure 7 is used when performing those same benchmarks with Associated Media traversing the DUT/SUT.

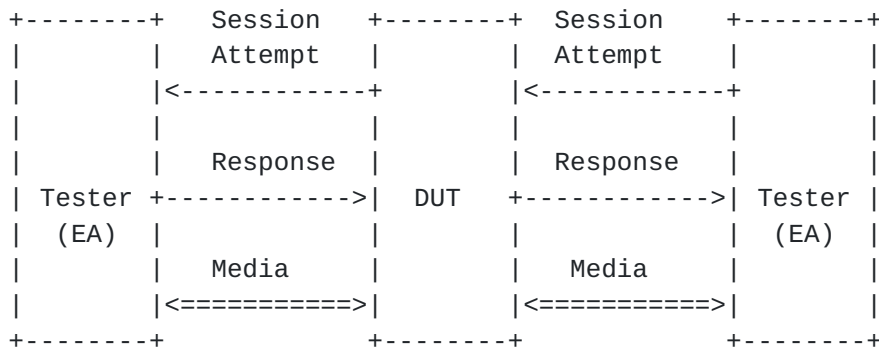


Figure 7: DUT/SUT performance benchmark for session establishment with media traversing the DUT

Figure 8 is to be used when performing those same benchmarks with Associated Media, but the media does not traverse the DUT/SUT. Again, the benchmarking of the media is not within the scope of this work item. The SIP control signaling is benchmarked in the presence of Associated Media to determine if the SDP body of the signaling and the handling of media impacts the performance of the DUT/SUT.

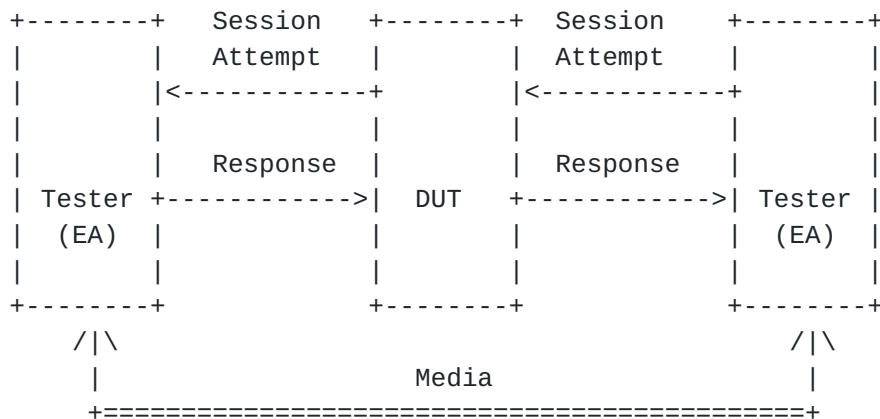


Figure 8: DUT/SUT performance benchmark for session establishment with media external to the DUT

Figure 9 is used when performing benchmarks that require one or more intermediaries to be in the signaling path. The intent is to gather benchmarking statistics with a series of DUTs in place. In this topology, the media is delivered end-to-end and does not traverse the DUT.

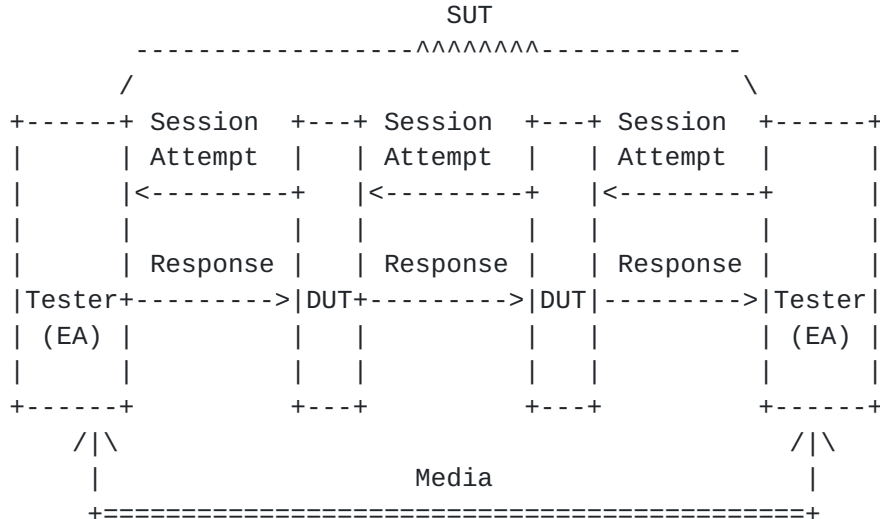


Figure 9: DUT/SUT performance benchmark for session establishment with multiple DUTs and end-to-end media

Figure 10 is used when performing benchmarks that require one or more intermediaries to be in the signaling path. The intent is to gather benchmarking statistics with a series of DUTs in place. In this topology, the media is delivered hop-by-hop through each DUT.

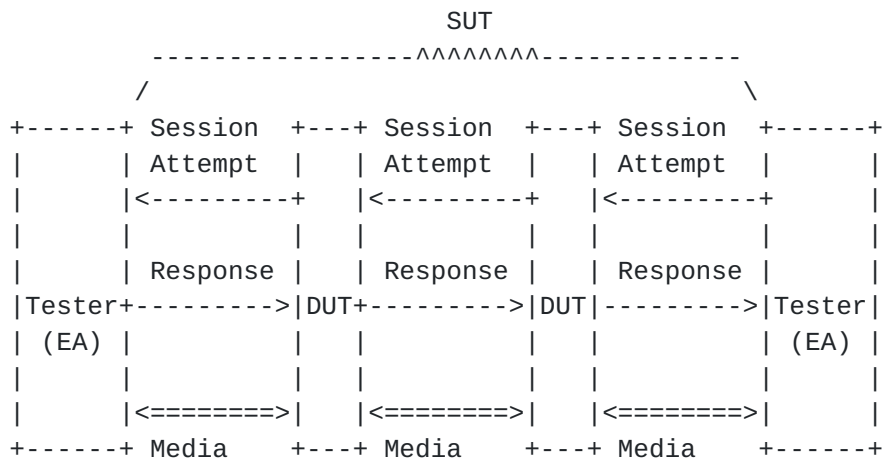


Figure 10: DUT/SUT performance benchmark for session establishment with multiple DUTs and hop-by-hop media

Figure 11 illustrates the SIP signaling for an Established Session. The Tester acts as the EAs and initiates a Session Attempt with the DUT/SUT. When the Emulated Agent (EA) receives a 200 OK from the DUT/SUT that session is considered to be an Established Session. The illustration indicates three states of the session bring created by the EA - Attempting, Established, and Disconnecting. Sessions can be one of two type: Invite-Initiated Session (IS) or Non-Invite Initiated Session (NS). Failure for the DUT/SUT to successfully respond within the Establishment Threshold Time is considered a Session Attempt Failure. SIP Invite messages MUST include the SDP body to specify the Associated Media. Use of Associated Media, to be sourced from the EA, is optional. When Associated Media is used, it may traverse the DUT/SUT depending upon the type of DUT/SUT. The Associated Media is shown in Figure 11 as "Media" connected to media ports M1 and M2 on the EA. After the EA sends a BYE, the session disconnects. Performance test cases for session disconnects are not considered in this work item (the BYE request is shown for completeness.)

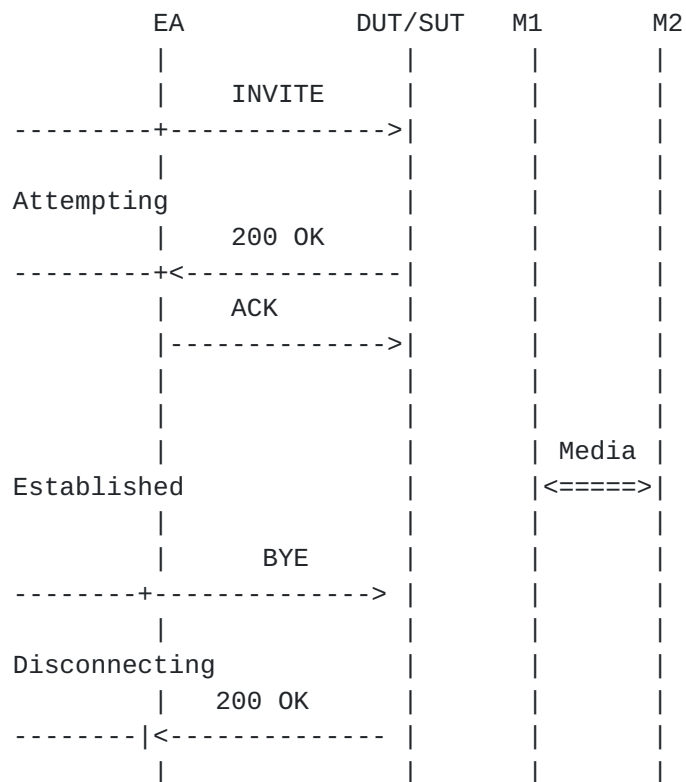


Figure 11: Invite-initiated Session States

3. Term Definitions

3.1. Protocol Components

3.1.1. Session

Definition:

The combination of signaling and media messages and processes that support a SIP-based service.

Discussion:

SIP messages are used to create and manage services for end users. Often, these services include the creation of media streams that are defined in the SDP body of a SIP message and carried in RTP protocol data units. However, SIP messages can also be used to create Instant Message services and subscription services, and such services are not associated with media streams. SIP reserves the term "session" to describe services that are analogous to telephone calls on a circuit switched network. SIP reserves the term "dialog" to refer to a signaling-only relationship between User Agent peers. SIP reserves the term "transaction" to refer to

the brief communication between a client and a server that lasts only until the final response to the SIP request. None of these terms describes the entity whose performance we want to benchmark. For example, the MESSAGE request does not create a dialog and can be sent either within or outside of a dialog. It is not associated with media, but it resembles a phone call in its dependence on human rather than machine initiated responses. The SUBSCRIBE method does create a dialog between the originating end-user and the subscription service. It, too, is not associated with a media session.

In light of the above observations we have extended the term "session" to include SIP-based services that are not initiated by INVITE requests and that do not have associated media. In this extended definition, a session always has a signaling component and may also have a media component. Thus, a session can be defined as signaling-only or a combination of signaling and media. We define the term "Associated Media", see [Section 3.1.4](#), to describe the situation in which media is associated with a SIP dialog. The terminology "Invite-initiated Session" (IS) [Section 3.1.8](#) and "Non-invite-Initiated Session" (NS) [Section 3.1.9](#) are used to distinguish between these two types of session. An Invite-initiated Session is a session as defined in SIP. The performance of a device or system that supports Invite-initiated Sessions that do not create media sessions, "Invite-initiated Sessions without Associated Media", can be measured and is of interest for comparison and as a limiting case. The REGISTER request can be considered to be a "Non-invite-initiated Session without Associated Media." A separate set of benchmarks is provided for REGISTER requests since most implementations of SIP-based services require this request and since a registrar may be a device under test.

A Session in the context of this document, can be considered to be a vector with three components:

1. A component in the signaling plane (SIP messages), sess.sig;
2. A media component in the media plane (RTP and SRTP streams for example), sess.med (which may be null);
3. A control component in the media plane (RTCP messages for example), sess.medc (which may be null).

An IS is expected to have non-null sess.sig and sess.med components. The use of control protocols in the media component is media dependent, thus the expected presence or absence of sess.medc is media dependent and test-case dependent. An NS is expected to have a non-null sess.sig component, but null sess.med and sess.medc components.

Packets in the Signaling Plane and Media Plane will be handled by different processes within the DUT. They will take different paths within a SUT. These different processes and paths may produce variations in performance. The terminology and benchmarks defined in this document and the methodology for their use are designed to enable us to compare performance of the DUT/SUT with reference to the type of SIP-supported application it is handling.

Note that one or more sessions can simultaneously exist between any participants. This can be the case, for example, when the EA sets up both an IM and a voice call through the DUT/SUT. These sessions are represented as an array `session[x]`.

Sessions will be represented as a vector array with three components, as follows:

`session->`

`session[x].sig`, the signaling component

`session[x].medc[y]`, the media control component (e.g. RTCP)

`session[x].med[y]`, an array of associated media streams (e.g. RTP, SRTP, RTSP, MSRP). This media component may consist of zero or more media streams.

Figure 12 models the vectors of the session.

Measurement Units:

N/A.

Issues:

None.

See Also:

Media Plane

Signaling Plane

Associated Media

Invite-initiated Session (IS)

Non-invite-initiated Session (NS)

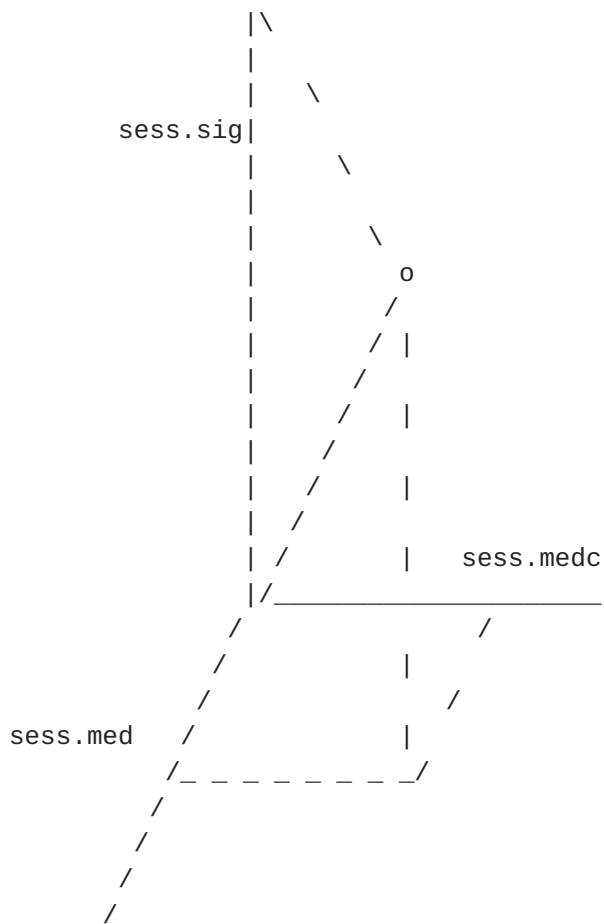


Figure 12: Session components

3.1.2. Signaling Plane

Definition:

The plane in which SIP messages [[RFC3261](#)] are exchanged between SIP Agents [[RFC3261](#)].

Discussion:

SIP messages are used to establish sessions in several ways: directly between two User Agents [[RFC3261](#)], through a Proxy Server [[RFC3261](#)], or through a series of Proxy Servers. The Session Description Protocol is included in the Signaling Plane. (SDP). The Signaling Plane for a single Session is represented by session.sig.

Measurement Units:

N/A.

Issues:

None.

See Also:

Media Plane

EAs

3.1.3. Media Plane

Definition:

The data plane in which one or more media streams and their associated media control protocols are exchanged between User Agents after a media connection has been created by the exchange of signaling messages in the Signaling Plane.

Discussion:

Media may also be known as the "bearer channel". The Media Plane MUST include the media control protocol, if one is used, and the media stream(s). Examples of media are audio and video. The media streams are described in the SDP of the Signaling Plane. The media for a single Session is represented by session.med. The media control protocol for a single media description is represented by session.medc.

Measurement Units:

N/A.

Issues:

None.

See Also:

Signaling Plane

3.1.4. Associated Media

Definition:

Media that corresponds to an 'm' line in the SDP payload of the Signaling Plane.

Discussion:

Any media protocol MAY be used.

For any session's signaling component, `session.sig`, there may be zero, one, or multiple associated media streams. When there are multiple media streams, these are represented by a vector array `session.med[y]`. When there are multiple media streams there will be multiple media control protocol descriptions as well. They are represented by a vector array `session.medc[y]`.

Measurement Units:

N/A.

Issues:

None.

3.1.5. Overload

Definition:

Overload is defined as the state where a SIP server does not have sufficient resources to process all incoming SIP messages [[I-D.ietf-soc-overload-design](#)].

Discussion:

The distinction between an overload condition and other failure scenarios is outside the scope of black box testing and of this document. Under overload conditions, all or a percentage of Session Attempts will fail due to lack of resources. In black box testing the cause of the failure is not explored. The fact that a failure occurred for whatever reason, will trigger the tester to reduce the offered load, as described in the companion methodology document, [[I-D.ietf-bmwg-sip-bench-meth](#)]. SIP server resources may include CPU processing capacity, network bandwidth, input/output queues, or disk resources. Any combination of resources may be fully utilized when a SIP server (the DUT/SUT) is in the overload condition. For proxy-only type of devices, it is expected that the proxy will be driven into overload based on the delivery rate of signaling requests.

For UA-type of network devices such as gateways, it is expected that the UA will be driven into overload based on the volume of media streams it is processing.

Measurement Units:

N/A.

Issues:

The issue of overload in SIP networks is currently a topic of discussion in the SIPPING WG. The normal response to an overload stimulus -- sending a 503 response -- is considered inadequate and new response codes and behaviors may be specified in the future. From the perspective of this document, all these responses will be considered to be failures. There is thus no dependency between this document and the ongoing work on the treatment of overload failure.

3.1.6. Session Attempt**Definition:**

A SIP request sent by the EA that has not received a final response.

Discussion:

The attempted session may be Invite Initiated or Non-invite Initiated. When counting the number of session attempts we include all INVITEs that are rejected for lack of authentication information. The EA needs to record the total number of session attempts including those attempts that are routinely rejected by a proxy that requires the UA to authenticate itself. The EA is provisioned to deliver a specific number of session attempts per second. But the EA must also count the actual number of session attempts per given tie interval.

Measurement Units:

N/A.

Issues:

None.

See Also:

Session
Session Attempt Rate
Invite-initiated Session
Non-Invite initiated Session

3.1.7. Established Session**Definition:**

A SIP session for which the EA acting as the UE/UA has received a 200 OK message.

Discussion:

An Established Session MAY be Invite Initiated or Non-Invite Initiated.

Measurement Units:

N/A.

Issues:

None.

See Also:

Invite-initiated Session
Session Attempting State
Session Disconnecting State

3.1.8. Invite-initiated Session (IS)**Definition:**

A Session that is created by an exchange of messages in the Signaling Plane, the first of which is a SIP INVITE request.

Discussion:

When an IS becomes an Established Session its signaling component is identified by the SIP dialog parameter values, Call-ID, To-tag, and From-tag ([RFC3261](#) [[RFC3261](#)]). An IS may have zero, one or multiple Associated Media descriptions in the SDP body. The inclusion of media is test case dependent. An IS is successfully established if the following two conditions are met:

1. Sess.sig is established by the end of Establishment Threshold Time (c.f. [Section 3.3.3](#)), and
2. If a media session is described in the SDP body of the signaling message, then the media session is established by the end of Establishment Threshold Time (c.f. [Section 3.3.3](#)).
An SBC or B2BUA may receive media from a calling or called party before a signaling dialog is established and certainly before a confirmed dialog is established. The EA can be built in such a way that it does not send early media or it needs to include a parameter that indicates when it will send media. This parameter must be included in the list of test setup parameters in Section 5.1 of [[I-D.ietf-bmwg-sip-bench-meth](#)]

Measurement Units:

N/A.

Issues:

None.

See Also:

Session

Non-Invite initiated Session

Associated Media

[3.1.9.](#) Non-INVITE-initiated Session (NS)

Definition:

A session that is created by an exchange of SIP messages in the Signaling Plane the first of which is not a SIP INVITE message.

Discussion:

An NS is successfully established if the Session Attempt via a non- INVITE request results in the EA receiving a 2xx reply before the expiration of the Establishment Threshold timer (c.f., [Section 3.3.3](#)). An example of a NS is a session created by the SUBSCRIBE request.

Measurement Units:

N/A.

Issues:

None.

See Also:

Session

Invite-initiated Session

[3.1.10.](#) Session Attempt Failure

Definition:

A session attempt that does not result in an Established Session.

Discussion:

The session attempt failure may be indicated by the following observations at the EA:

1. Receipt of a SIP 4xx, 5xx, or 6xx class response to a Session Attempt.
2. The lack of any received SIP response to a Session Attempt within the Establishment Threshold Time (c.f. [Section 3.3.3](#)).

Measurement Units:

N/A.

Issues:

None.

See Also:

Session Attempt

3.1.11. Standing Sessions Count

Definition:

The number of Sessions currently established on the DUT/SUT at any instant.

Discussion:

The number of Standing Sessions is influenced by the Session Duration and the Session Attempt Rate. Benchmarks MUST be reported with the maximum and average Standing Sessions for the DUT/SUT for the duration of the test. In order to determine the maximum and average Standing Sessions on the DUT/SUT for the duration of the test it is necessary to make periodic measurements of the number of Standing Sessions on the DUT/SUT. The recommended value for the measurement period is 1 second. Since we cannot directly poll the DUT/SUT, we take the number of standing sessions on the DUT/SUT to be the number of distinct calls as measured by the number of distinct Call-IDs that the EA is processing at the time of measurement. The EA must make that count available for viewing and recording.

Measurement Units:

Number of sessions

Issues:

None.

See Also:

Session Duration

Session Attempt Rate

Session Attempt Rate

Emulated Agent

3.2. Test Components

3.2.1. Emulated Agent

Definition:

A device in the test topology that initiates/responds to SIP messages as one or more session endpoints and, wherever applicable, sources/receives Associated Media for Established Sessions.

Discussion:

The EA functions in the Signaling and Media Planes. The Tester may act as multiple EAs.

Measurement Units:

N/A

Issues:

None.

See Also:

Media Plane
Signaling Plane
Established Session
Associated Media

3.2.2. Signaling Server

Definition:

Device in the test topology that acts to create sessions between EAs. This device is either a DUT or a component of a SUT.

Discussion:

The DUT MUST be an [RFC 3261](#) capable network equipment such as a Registrar, Redirect Server, User Agent Server, Stateless Proxy, or Stateful Proxy. A DUT MAY also include B2BUA or SBC.

Measurement Units:

NA

Issues:

None.

See Also:

Signaling Plane

3.2.3. SIP-Aware Stateful Firewall

Definition:

Device in the test topology that provides protection against various types of security threats to which the Signaling and Media Planes of the EAs and Signaling Server are vulnerable.

Discussion:

Threats may include Denial-of-Service, theft of service and misuse of service. The SIP-Aware Stateful Firewall MAY be an internal component or function of the Session Server. The SIP-Aware Stateful Firewall MAY be a standalone device. If it is a standalone device it MUST be paired with a Signaling Server. If it is a standalone device it MUST be benchmarked as part of a SUT. SIP-Aware Stateful Firewalls MAY include Network Address Translation (NAT) functionality. Ideally, the inclusion of the SIP-Aware Stateful Firewall in the SUT does not lower the measured values of the performance benchmarks.

Measurement Units:

N/A

Issues:

None.

See Also:**3.2.4. SIP Transport Protocol****Definition:**

The protocol used for transport of the Signaling Plane messages.

Discussion:

Performance benchmarks may vary for the same SIP networking device depending upon whether TCP, UDP, TLS, SCTP, or another transport layer protocol is used. For this reason it MAY be necessary to measure the SIP Performance Benchmarks using these various transport protocols. Performance Benchmarks MUST report the SIP Transport Protocol used to obtain the benchmark results.

Measurement Units:

TCP,UDP, SCTP, TLS over TCP, TLS over UDP, or TLS over SCTP

Issues:

None.

See Also:

[3.3. Test Setup Parameters](#)

[3.3.1. Session Attempt Rate](#)

Definition:

Configuration of the EA for the number of sessions per second that the EA attempts to establish using the services of the DUT/SUT.

Discussion:

The Session Attempt Rate is the number of sessions per second that the EA sends toward the DUT/SUT. Some of the sessions attempted may not result in a session being established. A session in this case may be either an IS or an NS.

Measurement Units:

Session attempts per second

Issues:

None.

See Also:

Session

Session Attempt

[3.3.2. IS Media Attempt Rate](#)

Definition:

Configuration on the EA for the rate, measured in sessions per second, at which the EA attempts to establish INVITE-initiated sessions with Associated Media, using the services of the DUT/SUT.

Discussion:

An IS is not required to include a media description. The IS Media Attempt Rate defines the number of media sessions we are trying to create, not the number of media sessions that are actually created. Some attempts might not result in successful sessions established on the DUT.

Measurement Units:

session attempts per second (saps)

Issues:

None.

See Also:

IS

3.3.3. Establishment Threshold Time

Definition:

Configuration of the EA for representing the amount of time that an EA will wait before declaring a Session Attempt Failure.

Discussion:

This time duration is test dependent.

It is RECOMMENDED that the Establishment Threshold Time value be set to Timer B (for ISs) or Timer F (for NSs) as specified in [RFC 3261](#), Table 4 [[RFC3261](#)]. Following the default value of T1 (500ms) specified in the table and a constant multiplier of 64 gives a value of 32 seconds for this timer (i.e., $500\text{ms} * 64 = 32\text{s}$).

Measurement Units:

seconds

Issues:

None.

See Also:

session establishment failure

3.3.4. Session Duration

Definition:

Configuration of the EA that represents the amount of time that the SIP dialog is intended to exist between the two EAs associated with the test.

Discussion:

The time at which the BYE is sent will control the Session Duration

Normally the Session Duration will be the same as the Media Session Hold Time. However, it is possible that the dialog established between the two EAs can support different media sessions at different points in time. Providing both parameters allows the testing agency to explore this possibility.

Measurement Units:
seconds

Issues:
None.

See Also:
Media Session Hold Time

3.3.5. Media Packet Size

Definition:
Configuration on the EA for a fixed size of packets used for media streams.

Discussion:
For a single benchmark test, all sessions use the same size packet for media streams. The size of packets can cause variation in performance benchmark measurements.

Measurement Units:
bytes

Issues:
None.

See Also:

3.3.6. Media Offered Load

Definition:
Configuration of the EA for the constant rate of Associated Media traffic offered by the EA to the DUT/SUT for one or more Established Sessions of type IS.

Discussion:
The Media Offered Load to be used for a test MUST be reported with three components:

1. per Associated Media stream;
2. per IS;
3. aggregate.

For a single benchmark test, all sessions use the same Media Offered Load per Media Stream. There may be multiple Associated Media streams per IS. The aggregate is the sum of all Associated Media for all IS.

Measurement Units:

packets per second (pps)

Issues:

None.

See Also:

Established Session
Invite Initiated Session
Associated Media

3.3.7. Media Session Hold Time**Definition:**

Parameter configured at the EA, that represents the amount of time that the Associated Media for an Established Session of type IS will last.

Discussion:

The Associated Media streams may be bi-directional or uni-directional as indicated in the test methodology. Normally the Media Session Hold Time will be the same as the Session Duration. However, it is possible that the dialog established between the two EAs can support different media sessions at different points in time. Providing both parameters allows the testing agency to explore this possibility.

Measurement Units:

seconds

Issues:

None.

See Also:

Associated Media
Established Session
Invite-initiated Session (IS)

3.3.8. Loop Detection Option**Definition:**

An option that causes a Proxy to check for loops in the routing of a SIP request before forwarding the request.

Discussion:

This is an optional process that a SIP proxy may employ; the process is described under Proxy Behavior in [RFC 3261](#) [[RFC3261](#)] in [Section 16.3](#) Request Validation and that section also contains suggestions as to how the option could be implemented. Any procedure to detect loops will use processor cycles and hence could impact the performance of a proxy.

Measurement Units:

NA

Issues:

None.

See Also:**[3.3.9. Forking Option](#)****Definition:**

An option that enables a Proxy to fork requests to more than one destination.

Discussion:

This is an process that a SIP proxy may employ to find the UAS. The option is described under Proxy Behavior in [RFC 3261](#) in [Section 16.1](#). A proxy that uses forking must maintain state information and this will use processor cycles and memory. Thus the use of this option could impact the performance of a proxy and different implementations could produce different impacts. SIP supports serial or parallel forking. When performing a test, the type of forking mode MUST be indicated.

Measurement Units:

The number of endpoints that will receive the forked invitation. A value of 1 indicates that the request is destined to only one endpoint, a value of 2 indicates that the request is forked to two endpoints, and so on. This is an integer value ranging between 1 and N inclusive, where N is the maximum number of endpoints to which the invitation is sent.
Type of forking used, namely parallel or serial.

Issues:

None.

See Also:

3.4. Benchmarks

3.4.1. Registration Rate

Definition:

The maximum number of registrations that can be successfully completed by the DUT/SUT in a given time period without registration failures in that time period.

Discussion:

This benchmark is obtained with zero failure in which 100% of the registrations attempted by the EA are successfully completed by the DUT/SUT. The registration rate provisioned on the Emulated Agent is raised and lowered as described in the algorithm in the companion methodology draft [[I-D.ietf-bmwg-sip-bench-meth](#)] until a traffic load consisting of registrations at the given attempt rate over the sustained period of time identified by T in the algorithm completes without failure.

Measurement Units:

registrations per second (rps)

Issues:

None.

See Also:

3.4.2. Session Establishment Rate

Definition:

The maximum number of sessions that can be successfully completed by the DUT/SUT in a given time period without session establishment failures in that time period.

Discussion:

This benchmark is obtained with zero failure in which 100% of the sessions attempted by the Emulated Agent are successfully completed by the DUT/SUT. The session attempt rate provisioned on the EA is raised and lowered as described in the algorithm in the accompanying methodology document, until a traffic load at the given attempt rate over the sustained period of time identified by T in the algorithm completes without any failed session attempts. Sessions may be IS or NS or a mix of both and will be defined in the particular test.

Measurement Units:

sessions per second (sps)

Issues:

None.

See Also:

Invite-initiated Sessions
Non-INVITE initiated Sessions
Session Attempt Rate

3.4.3. Session Capacity**Definition:**

The maximum value of Standing Sessions Count achieved by the DUT/SUT during a time period T in which the EA is sending session establishment messages at the Session Establishment Rate.

Discussion:

Sessions may be IS or NS. If they are IS they can be with or without media. When benchmarking Session Capacity for sessions with media it is required that these sessions be permanently established (i.e., they remain active for the duration of the test.) This can be achieved by causing the EA not to send a BYE for the duration of the testing. In the signaling plane, this requirement means that the dialog lasts as long as the test lasts. When media is present, the Media Session Hold Time MUST be set to infinity so that sessions remain established for the duration of the test. If the DUT/SUT is dialog-stateful, then we expect its performance will be impacted by setting Media Session Hold Time to infinity, since the DUT/SUT will need to allocate resources to process and store the state information. The report of the Session Capacity must include the Session Establishment Rate at which it was measured.

Measurement Units:

sessions

Issues:

None.

See Also:

Established Session

Session Attempt Rate
Session Attempt Failure

3.4.4. Session Overload Capacity

Definition:

The maximum number of Established Sessions that can exist simultaneously on the DUT/SUT until it stops responding to Session Attempts.

Discussion:

Session Overload Capacity is measured after the Session Capacity is measured. The Session Overload Capacity is greater than or equal to the Session Capacity. When benchmarking Session Overload Capacity, continue to offer Session Attempts to the DUT/SUT after the first Session Attempt Failure occurs and measure Established Sessions until no there is no SIP message response for the duration of the Establishment Threshold. Note that the Session Establishment Performance is expected to decrease after the first Session Attempt Failure occurs.

Units:

Sessions

Issues:

None.

See Also:

Overload
Session Capacity
Session Attempt Failure

3.4.5. Session Establishment Performance

Definition:

The percent of Session Attempts that become Established Sessions over the duration of a benchmarking test.

Discussion:

Session Establishment Performance is a benchmark to indicate session establishment success for the duration of a test. The duration for measuring this benchmark is to be specified in the Methodology. The Session Duration SHOULD be configured to infinity so that sessions remain established for the entire test duration.

Session Establishment Performance is calculated as shown in the following equation:

$$\text{Session Establishment Performance} = \frac{\text{Total Established Sessions}}{\text{Total Session Attempts}}$$

Session Establishment Performance may be monitored real-time during a benchmarking test. However, the reporting benchmark MUST be based on the total measurements for the test duration.

Measurement Units:

Percent (%)

Issues:

None.

See Also:

Established Session

Session Attempt

3.4.6. Session Attempt Delay

Definition:

The average time measured at the EA for a Session Attempt to result in an Established Session.

Discussion:

Time is measured from when the EA sends the first INVITE for the call-ID in the case of an IS. Time is measured from when the EA sends the first non-INVITE message in the case of an NS. Session Attempt Delay MUST be measured for every established session to calculate the average. Session Attempt Delay MUST be measured at the Session Establishment Rate.

Measurement Units:

Seconds

Issues:

None.

See Also:

Session Establishment Rate

3.4.7. IM Rate

Definition:

Maximum number of IM messages completed by the DUT/SUT.

Discussion:

For a UAS, the definition of success is the receipt of an IM request and the subsequent sending of a final response.

For a UAC, the definition of success is the sending of an IM request and the receipt of a final response to it. For a proxy, the definition of success is as follows:

- A. the number of IM requests it receives from the upstream client MUST be equal to the number of IM requests it sent to the downstream server; and
- B. the number of IM responses it receives from the downstream server MUST be equal to the number of IM requests sent to the downstream server; and
- C. the number of IM responses it sends to the upstream client MUST be equal to the number of IM requests it received from the upstream client.

Measurement Units:

IM messages per second

Issues:

None.

See Also:

4. IANA Considerations

This document requires no IANA considerations.

5. Security Considerations

Documents of this type do not directly affect the security of Internet or corporate networks as long as benchmarking is not performed on devices or systems connected to production networks. Security threats and how to counter these in SIP and the media layer is discussed in [RFC3261](#) [[RFC3261](#)], [RFC 3550](#) [[RFC3550](#)], [RFC3711](#) [[RFC3711](#)] and various other drafts. This document attempts to formalize a set of common terminology for benchmarking SIP networks. Packets with unintended and/or unauthorized DSCP or IP precedence values may present security issues. Determining the security consequences of such packets is out of scope for this document.

6. Acknowledgments

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[Appendix A.](#) White Box Benchmarking Terminology

Session Attempt Arrival Rate

Definition:

The number of Session Attempts received at the DUT/SUT over a specified time period.

Discussion:

Sessions Attempts are indicated by the arrival of SIP INVITES OR SUBSCRIBE NOTIFY messages. Session Attempts Arrival Rate distribution can be any model selected by the user of this document. It is important when comparing benchmarks of different devices that same distribution model was used. Common distributions are expected to be Uniform and Poisson.

Measurement Units:

Session attempts/sec

Issues:

None.

See Also:

Session Attempt

Authors' Addresses

Carol Davids
Illinois Institute of Technology
201 East Loop Road
Wheaton, IL 60187
USA

Phone: +1 630 682 6024

Email: davids@iit.edu

Vijay K. Gurbani
Bell Laboratories, Alcatel-Lucent
1960 Lucent Lane
Rm 9C-533
Naperville, IL 60566
USA

Phone: +1 630 224 0216
Email: vkg@bell-labs.com

Scott Poretsky
Allot Communications
300 TradeCenter, Suite 4680
Woburn, MA 08101
USA

Phone: +1 508 309 2179
Email: sporetsky@allot.com

