Huawei Cloud-based SBC
Performance Testing and Function Validation

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1.0 Executive Summary

Huawei Technologies engaged Miercom to perform an independent performance assessment of its Cloud-based SBC – a virtualized implementation of its SE2900 Session Border Controller designed for cloud-based deployment. The SBC software ran in an OpenStack environment on a server platform consisting of Intel-based server blades.

The Huawei cloud-based SBC is designed for various voice and video over IP and real-time communication services for fixed line, mobile and over-the-top (OTT) in cloud environments. With the support of signaling and media plane separation, stateless design and service-oriented micro-service architecture, the Huawei cloud-based SBC provides carrier-class reliability, smart interoperability and leading performance.

All of the SBC topologies tested were conventional configurations, set up via Huawei's CloudOpera CSM management interface. The goal was to independently verify the SBC's performance in various SIP-call-handling, vocoder, encryption and security-threat environments. In addition, testing determined the virtual system's cloud resilience.

Key Findings and Conclusions

Huawei's virtualized, software-based and cloud-based SBC exhibited impressive performance and resilience. Among the most noteworthy findings:

- Testing found the SBC supports: 1,200 registrations per second (rps); handles registration storms delivered at up to 6,000 rps; and sustains 800 TLS-based rps.
- Rates of 300 calls per second (cps) for SIP-over-UDP calls, with 14 messages per call side, and 500 cps for SIP-over-TLS calls, with 7 messages per call side, were observed. Each call was composed of 1-to-1 caller-callee session.
- We confirmed the SBC SE2900 supports 45,200 concurrent SIP sessions with 4.39 MOS for G.711 media; 12,000 sessions with media over encrypted SRTP with 4.39 MOS; and 4,720 concurrent G.711-G.729 transcoded sessions, with 4.39 MOS for G.711 media and 4.09 MOS for G.729 media.
- Enhanced features and functions were verified: The SBC could handle a heavy call load of 40,000 concurrent sessions using Enhanced Voice Services (EVS) codec with 4.32 MOS; SIP header manipulation was performed on up to 40 cps; and dynamic bi-directional IPv4-IPv6 translation was sustained in real-time for 400 concurrent sessions.
- We found the SBC can automatically add (scale out) and reduce (scale in) processing modules as load varies, along with various other cloud-resilience features. Additionally, tests confirmed that the SBC successfully withstands Denial-of-Service attacks against its SIP signaling interface.

The test results confirm that Huawei's Cloud-based SBC delivers one the best performing SBCs we have tested to date. We observed impressive and solid performance from its Virtual Network Function running on a conventional Intel server. With broad call handling, internetworking, transcoding, auto-scaling and DoS-attack protection, we are pleased to award the Huawei Cloud-based SBC the ***Miercom Performance Verified*** certification.

Robert Smithers

CEO

Miercom

2.0 About the Product Tested

The Huawei Cloud-based SBC SE2900 is a software version of Huawei's SE2900 SBC appliance. In the test bed, the Linux-based SBC package was tested on OpenStack, a leading cloud-computing operating environment, running on a COTS enclosure with four server blades.

Each server blade featured two Intel Xeon E5 2658 V4, 2.3-GHz, 14-core CPUs and 256 GB of memory.

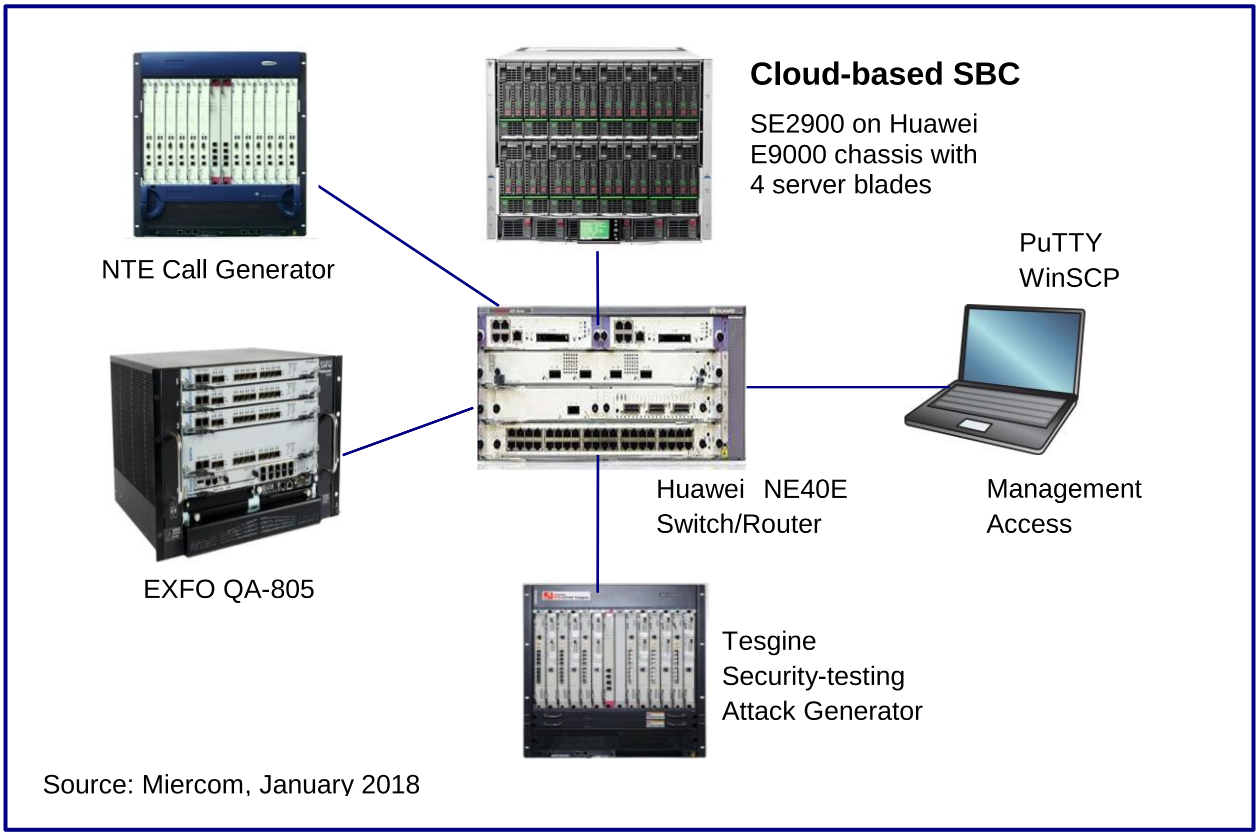
The SBC package tested is a “cloud orchestration,” which means all the discrete software modules – called Virtual Network Functions, or VNF – share some resources, according to the policy settings.

3.0 How We Did It

A straightforward test-bed network was assembled for this study (see below diagram). The Linux-based SE2900 virtual SBC software package, as previously noted, ran on four Huawei-manufactured server boards, configured in a Huawei multislot E9000 chassis.

Each of the four server boards contained two Intel Xeon CPUs – both 2.3-GHz, Intel Xeon E4-2658 v4 CPUs – and 256 GB of memory. The virtualization environment used was OpenStack, a public-domain operating system from the OpenStack Foundation.

Figure 1: Logical Configuration of the SBC Test Bed



In addition to the system under test (the SE2900 Cloud-based SBC), several other key components were used in the test bed:

NTE (Network Traffic Emulator) Call-Load Generator: A traffic-load generator and test tool developed by Huawei Technologies and used by many carriers and service providers to performance-test access and core-network telecom equipment. The NTE code version used was V3 R5C30.

EXFO QA-805: A powerful platform for testing VoIP and IP Multimedia Systems (IMS) networks and telecom systems, from Quebec, Canada-based EXFO (www.exfo.com). The QA-805 can emulate over 5 million subscribers/registrants, 8 million data-signaling sessions and 1.25 million RTP media sessions. The EXFO delivered SIP calls and registration requests to the SBC and, as the originator and terminating endpoint of all calls, the EXFO tracked completed calls, any dropped calls, and also calculated the MOS (mean opinion score) quality ratings of calls. EXFO version 9.7 code was run in the tests.

Tesgine: A Huawei-developed security and performance test tool, based on the ATCA (Advanced Telecom Computing Architecture) framework. The Tesgine can deliver malformed packets and messages as part of security testing. The Tesgine generated and launched DoS (Denial-of-Service) flood-attack packet streams against the SBC. Tesgine version 2.0 was used.

NE40E Universal Service Router (USR): Connecting everything in the test bed was a Huawei NE40E Universal Service Router (USR). The router, supporting many Gigabit/s Ethernet ports, was configured with separate IP subnetworks for SIP signaling and media traffic to and from the SBC, and management access to all the devices.

PuTTY: Software for assessing SSH (Secure Shell) and Telnet connections with a device supporting SSH and Telnet connections.

WinSCP: SFTP (Secure File Transfer Protocol) client software that supports SCP (Secure Copy Protocol); enables secure SSH file transfers between hosts over a network; includes mechanisms for authentication and data integrity. Version 5.1.5 was used.

4.0 SIP Registration and Call Handling

Various performance tests were performed to determine registration and call rates, both over unsecured UDP (User Datagram Protocol) and over TLS (transport-layer security); registration storm handling, handling of G.711 media over RTP (unsecured) and over Secure RTP (SRTP); and G.711-G.729 transcoding.

Support for 1,200 SIP Registrations per Second

What's measured

Whether the SBC supports at least 1,200 SIP registrations per second delivered via regular UDP.

Test set-up

Two SCU (Session Control Units) instances were configured within the SBC. This test involved SIP registration requests delivered to the SBC at a brisk rate of 1,200 registrations per second (rps). There is no media sent as part of the SIP registration process.

Results

The SBC accepted and successfully processed 100 percent of registration requests delivered at 1,200 rps.

Summary of Confirmed Results

Rate of registration delivery
1,200 rps

Handling of SIP Registration Storm

What's measured

The ability of the SBC to accommodate a registration storm – a very high rate of registration received over a short period of time, such as after a re-powering of the data center after a massive power outage.

Test set-up

SIP-over-UDP registrations were delivered by the NTE test equipment at a very high rate of 6,000 rps. The load was maintained until all 200,000 user registration requests were successfully processed.

Results

The SBC SE2900 was able to successfully process all 200,000 registrations in less than three minutes, from a registration storm delivered at 6,000 rps.

Summary of Confirmed Results

Concurrent registrations sustained	Registration delivery rate	Time to register all 200,000 users
200,000	**6,000 rps**	**< 3 minutes**

Support for Secure SIP-over-TLS Registrations

What's measured

The SBC's handling of secure SIP registrations, delivered via encrypted and secure TLS (Transport Layer Security) connections.

Test set-up

This test involved SIP registration requests delivered to the SBC via secure TLS connections, at a rate of 800 rps.

Results

The SBC accepted and successfully processed 100 percent of the TLS-based registration requests delivered at 800 rps.

Summary of Confirmed Results

Rate of TLS-based registration delivery
800 rps

SIP-over-UDP Call Rate

What's measured

The ability of the SBC to process SIP calls set up over UDP connections. No media was sent during this test.

Test set-up

First, 100,000 SIP users were registered with the SBC SE2900. Then SIP-over-UDP calls were delivered, using 14-message SIP call set-up. A 30-second Call Hold Time (CHT) was used for all calls.

Results

The SBC SE2900 was able to successfully process and sustain 300 SIP-over-UDP calls per second (cps).

Summary of Confirmed Results

SIP-over-UDP calls per second	Call Hold Time
300 cps	**30 seconds**

Secure SIP-over-TLS Call Rate

What's measured

The ability of the SBC to process SIP calls set up over secure TLS connections. No media was sent during this test, just the encrypted SIP call setup via TLS.

Test set-up

First, 64,000 SIP users were registered with the SBC SE2900. Then SIP-over-TLS calls were delivered, using 7-message SIP call set-up. A 30-second CHT was used for all calls.

Results

The SBC SE2900 was able to successfully process and sustain 500 secure SIP-over-TLS cps.

Summary of Confirmed Results

Secure SIP-over-TLS calls per second	Call Hold Time
500 cps	**30 seconds**

SIP Calls with G.711 Media

What's measured

The SBC's ability to concurrently sustain 45,200 SIP calls using traditional G.711 media over Real-time Transport Protocol (RTP).

Test set-up

With 100,000 end users registered with the SBC, full 14-message SIP-over-UDP calls were delivered to the SBC at a rate of 113 cps. The CHT for all calls was 200 seconds. The media for all calls was G.711 encoded.

Results

The SBC accepted and successfully processed 100 percent of the SIP calls with full G.711-encoded media, delivered at 113 cps. The SBC SE2900 readily accepted and sustained 45,200 SIP sessions. The average Mean Opinion Score (MOS) rating of the calls media was 4.39.

Summary of Confirmed Results

Concurrent SIP sessions with G.711 media sustained	Rate of call delivery	Call Hold Time	Average Call MOS Score
45,200	**113 cps**	**200 seconds**	**4.39**

SIP Calls with G.711 Media over SRTP

What's measured

The SBC's ability to concurrently sustain 12,000 sessions with G.711 media transported over Secure RTP (SRTP).

Test set-up

With 20,000 end users registered with the SBC, SIP-over-UDP calls were delivered to the SBC at a rate of 100 cps. The CHT for all calls was 60 seconds. The media for all calls was G.711 encoded, and the media streams were carried over encrypted SRTP.

Results

The SBC accepted and successfully processed 100 percent of the SIP calls delivered at 100 cps, with full G.711-encoded media, carried over encrypted SRTP. The SBC SE2900 readily accepted and sustained 12,000 sessions, equating to 6,000 complete SIP calls. The average MOS rating of the calls media was an excellent 4.39.

Summary of Confirmed Results

Concurrent SIP sessions with G.711 media over SRTP	Rate of call delivery	Call Hold Time	Average Call MOS Score
12,000 sessions	**100 cps**	**60 seconds**	**4.39**

Support for G.711-G.729 Transcoding, Two VPUs

What's measured

The SBC's ability to scale up the number of concurrently supported transcoded (G.711-G.729) SIP calls by configuring two VPUs.

Test set-up

With two VPUs (voice and video processing units) configured and 100,000 end users registered, SIP calls were delivered to the SBC at a rate of 10 cps. The CHT for all calls was 236 seconds.

Results

The SBC SE2900 accepted and successfully processed 100 percent of the SIP calls requiring G.711-G.729 transcoding, delivered at 10 cps, concurrently sustaining 4,720 transcoded SIP sessions. The average MOS rating of the G.711 calls was 4.39; the average MOS for G.729 calls was 4.09.

Summary of Confirmed Results

Total Transcoded SIP sessions sustained	Rate of call delivery	Call Hold Time	Average Call MOS Score
4,720	**10 cps**	**236 seconds**	**4.39 (G.711)** **4.09 (G.729)**

5.0 Enhanced Features and Functions

The Huawei SBC was tested with real SIP VoIP calls delivered by the EXFO call generator. Call set-up and media streams consisted of two sessions each – that is, the calling and called SIP users each exchanged signaling with the SBC and the SBC processed the media streams in both directions during each call.

Support for Enhanced Voice Services Codec

What's measured

The SBC's support for the Enhanced Voice Services (EVS) codec. EVS is a form of AMR-WB (adaptive multi-rate wideband), a sophisticated voice-encoding technique standardized as EVS by the 3GPP (3rd Generation Partnership Project), which delivers high voice quality in up to 20 kHz of audio bandwidth.

Test set-up

The test environment exercised the SBC's support for the EVS codec with a heavy call load of up to 40,000 EVS sessions, delivered at a rate of 100 calls per second. In this test 40,000 EVC sessions were sustained with a CHT of 200 seconds.

Results

The SBC successfully processed and sustained 40,000 EVS sessions with excellent MOS quality of 4.32.

Summary of Confirmed Results

EVS Sessions sustained	Call delivery rate	Average MOS	Call Hold Time
40,000	100 cps	4.32	200 sec

SIP Header Manipulation

What's measured

The ability of the SBC to make changes to the SIP header when processing SIP call requests.

Test set-up

The SIP-header change was the addition by the SBC of a user-agent header in the SIP call **INVITE**, performed by the SBC on every call request. Calls were delivered to the SBC at a rate of 40 cps. The CHT was 30 seconds.

Results

The SBC successfully processed and manipulated each Invite call header, and then successfully completed and sustained 2,400 header-manipulated sessions.

Summary of Confirmed Results

Header-manipulated sessions sustained	Call delivery rate	Call Hold Time
2,400 sessions	**40 cps**	**30 seconds**

SIP Calls with IPv4-IPv6 Translation

What's measured

There has been some deployment of IP version 6 (IPv6), primarily internally within some organizations. Most of the rest of the world, however, continues to use the legacy IP version 4 (IPv4). When VoIP callers communicate from the IPv4 world into an organization running over IPv6, these calls need to have their IP transport translated in real-time between IPv4 and IPv6. This testing confirmed the ability of the SBC to perform this dynamic translation.

Test set-up

In this test environment callers delivered call requests over IPv6, and the callees (called SIP end users) responded over IPv4. The NTE call generator delivered call requests over IPv6 to the SBC at a rate of 40 cps. Media was not sent in this test. The calls were then closed (by the caller) after five seconds.

Results

The SBC successfully set-up, sustained and then tore down up to 400 concurrent sessions, with the SBC translating the transport of all signaling between IPv6 callers and IPv4 callees.

Summary of Confirmed Results

IPv4-IPv6 sessions	Call Hold Time	Call delivery rate
400	**5 seconds**	**40 cps**

6.0 Cloud Resilience

The following test series focused on the ability of the Huawei SBC SE2900 to automatically adjust to widely varying call loads, depending on its configuration, and its support of resilience and automatic re-configuration. The Huawei SBC environment runs on the Cloud Operating System and uses its CSM management application and graphical user interface (GUI) for configuration and management.

The Cloud-based SBC is designed for deployment in cloud environments, where applications run on Virtual Machines (VMs) in a virtualized operating system on typically Intel-based servers. Those deploying SBCs in cloud environments want the communications application to expand to handle large call loads (called “scale out”), and contract as appropriate with light loads, so that cloud computing resources can be reallocated as needed.

Automatic SCU Scale Out

What’s measured

The ability of the SBC to automatically expand and replicate call-handling software modules as the call load demands. The main software module involved is the Session Control Unit, or SCU, which handles call-set-up and control signaling.

Test set-up

Using the CSM interface, the CPU usage ceiling for the SCU (Session Control Unit) VM (virtual machine) was set to 30 percent. If usage exceeds that, a third, originally two, SCU VM is supposed to be automatically created, and then load-balance calls across the three SCU VMs. With 100,000 SIP end-users registered, SIP-over-UDP call requests were delivered at a rate of 200 cps to the SBC. These were all full, 14-message SIP-call sequences. The CHT was 50 seconds. There was no media sent as part of this test, just signaling.

Results

When call load applied more than 30 percent CPU usage on the SCU Virtual Machine, a third SCU VM was created, and call load was then balanced between the three SCUs.

Summary of Confirmed Results

Call Delivery Rate	SIP registrations	SCU CPU-usage load set for scale out	Did scale-out occur automatically as expected when CPU-usage threshold was exceeded?
200 cps	**100,000**	**30 percent**	**Yes**

Automatic SCU Scale In

What's measured

The capability of the SBC to automatically constrict call-handling software modules as call load warrants. The main software module involved is the Session Control Unit, or SCU, which handles call-set-up and control signaling.

Test set-up

Using the CSM interface, the CPU usage ceiling for the SCU (Session Control Unit) VM (virtual machine) was set to 40 percent. If usage drops below that, the third SCU VM is supposed to be automatically dismantled, and the resources returned to the resource pool. With 100,000 SIP end-users registered, SIP-over-UDP call requests were delivered at a rate of 200 cps to the SBC. These were all full, 14-message SIP-call sequences. The CHT was 50 seconds. There was no media sent as part of this test, just signaling.

Results

When call load dropped to less than 40 percent CPU usage, the third SCU Virtual Machine was automatically dismantled, and resources were returned to the resource pool. No calls were dropped during this process.

Summary of Confirmed Results

Call Delivery Rate	SIP registrations	SCU CPU usage load set for scale in	Did scale-in occur automatically as expected when CPU-usage fell below threshold?	Calls lost or dropped during scale-in process
200 cps	100,000	40 percent	Yes	0

Automatic HRU Scale Out

What's measured

The ability of the SBC to automatically expand and replicate call-media handling software modules as load demands. The main software module involved is the High-speed Routing Unit, or HRU, which handles media processing.

Test set-up

Three HRU VMs were initially configured. The CPU-usage ceiling was set to 40 percent. If usage exceeds that for more than 10 seconds, a new HRU VM is supposed to be automatically created. With 100,000 SIP registrants, SIP-over-UDP calls with G.729 media were delivered at a rate of 100 cps, with a 90-second CHT.

Results

When call load exceeded 40 percent CPU usage over the three HRU Virtual Machines for 10 seconds, a new HRU VM was automatically created and media calls were load balanced.

Summary of Confirmed Results

G.729 Call Delivery Rate	CPU load set for HRU VM scale out	Did automatic scale-out occur as expected?	Calls dropped or lost	Average MOS call quality
100 cps	**40 percent**	**Yes**	**0**	**4.09**

Automatic HRU Scale In

What's measured

The ability of the SBC to automatically reduce call-media handling software modules as load permits. The main software module involved is the High-speed Routing Unit, or HRU, which handles media processing.

Test set-up

Four HRU VMs are initially configured and operational. The CPU-usage ceiling was set to 40 percent. If usage drops below that for more than 30 seconds, one HRU VM is supposed to be dismantled and the resources reassigned back to the resource pool. With 100,000 SIP registrants, SIP-over-UDP calls with G.729 media were delivered at a rate of 100 cps, with a 90-second CHT.

Results

When call load dropped below 40 percent CPU usage for 30 seconds, one HRU VM was dismantled, leaving just three operating HRU Virtual Machines.

Summary of Confirmed Results

G.729 Call Delivery Rate	CPU load set for HRU VM scale in	Did automatic scale-in occur as expected?	Calls dropped or lost	Average MOS call quality
100 cps	**40 percent**	**Yes**	**0**	**4.09**

Live Migration of VPU (Voice and Video Processing Unit) Virtual Machine

What's measured

The ability to dynamically migrate key modules of the SBC between physically separated server resources. In this test we migrated a VPU (Voice and Video Processing Unit) module from the server in Slot 12 to Slot 13.

Test set-up

The VPU was configured and running on the server in Slot 12. There were no transcoding calls running. Calls are placed and actively running. The VPU module is then dynamically migrated over to the server in Slot 13.

Results

The VPU VM was successfully migrated over from Slot 12 to Slot 13. We noted, too, that if the resources during migration are not sufficient, the SBC will direct the administrator through Error messages to: "Keep the original affinity rules."

Summary of Confirmed Results

Could VPU module be dynamically migrated over from server Slot 12 to Slot 13?	Does system warn administrator if resources are insufficient for migration?
Yes	**Yes**

Live Migration of SCU (Session Control Unit) Virtual Machine

What's measured

The ability to dynamically and automatically relocate key modules of the SBC between physically separated server resources. In this test we migrated an SCU Virtual Machine from the server in Slot 4 to Slot 5.

Test set-up

The SCU was configured and running on the server in Slot 4. There are no calls running. Slot 4 is then powered off.

Results

All the modules on Slot 4, including the SCU, showed an Error condition when Slot 4 was powered off. With no further action, the SCU on Slot 4 was automatically migrated to Slot 5. A check confirmed the automatic migration of the SCU from Slot 4 to Slot 5.

Summary of Confirmed Results

Did SCU module automatically migrate over from server Slot 4 to Slot 5?	Did all modules on Slot 4 automatically migrate to Slot 5 after power loss on Slot 4?
Yes	**Yes**

7.0 Denial-of-Service (DoS) Tests and Results

The Tesgine tester was used to launch DoS (Denial of Service) attacks directly to the SIP signaling interface of the SBC SE2900 on the virtualized server.

The DoS attacks were delivered at high volumes, while high volumes of calls were being handled by the SBC. Media was not sent as part of the background calls. The objective of the tests was to see if any of the DoS attacks impacted calls in progress.

SIP INVITE Flood DoS Attack

What's measured

The impact on the SBC of a SIP INVITE Flood Denial-of-Service attack directed at the SIP signaling interface of the SBC (Port 5060). The DoS attack consisted of SIP INVITE requests.

Test set-up

While the SBC is processing 150 cps, without media, and 200 rps, the Tesgine test system is used to generate and deliver to the SIP-signaling Port 5060 a SIP INVITE flood of 15,000 packets per second (pps) of SIP INVITE requests. CHT was 30 seconds. No media was sent as part of the call load.

Results

There were no calls dropped as a result of the SIP INVITE Flood DoS attack. Alarm generated by SBC saying attackers' IP blacklisted.

Summary of Confirmed Results

DoS attack	Directed at	Attack volume	Registrations being handled	Calls being handled	Calls dropped
SIP INVITE Flood	**SIP signaling Port 5060**	**15,000 pps**	**200 rps**	**150 cps (no media)**	**0**

SIP Registration Flood DoS Attack

What's measured

The impact on the SBC of a SIP Registration Flood Denial-of-Service attack directed at the SIP signaling interface of the SBC (Port 5060). The DoS attack consisted of SIP Registration requests.

Test set-up

While the SBC is processing 150 cps, without media, and 200 rps, the Tesgine test system is used to generate and deliver to the SIP-signaling Port 5060 a SIP Registration flood of 15,000 pps. CHT was 30 seconds. No media was sent as part of the call load.

Results

There were no calls dropped as a result of the SIP Registration Flood DoS attack. Alarm generated by SBC about undesirable registration attempts.

Summary of Confirmed Results

DoS attack	Directed at	Attack volume	Registrations being handled	Calls being handled	Calls dropped
SIP Registration Flood	**SIP signaling Port 5060**	**15,000 pps**	**200 rps**	**150 cps (no media)**	**0**

About Miercom

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