Premier Services Evaluation Program	**Performance Evaluation Report** Internet Security Systems Proventia G400, Proventia G2000 Date: 17 January 2006

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Executive Summary

Internet Security Systems (ISS) contracted ICSA Labs to perform performance testing of their Proventia G-series Appliances (G400 and G2000) with the goal of establishing the level of Voice over Internet Protocol (VoIP) traffic the devices could pass while still maintaining acceptable Voice Quality (VQ).

The product evaluation for this engagement focused on security products developed by ISS for sale as VoIP-enabled Intrusion Prevention Systems. The ISS Proventia G400 and G2000 series appliances are intended to provide traditional and VoIP specific Network Intrusion Prevention services for the converged networks where they are deployed without adversely affecting the quality of the Voice traffic they are processing.

ICSA Labs analysts gathered data from technical documentation provided by ISS, and testing of their Proventia G400 and G2000 appliances, (herein collectively named G400 or G2000 respectively, or the Device Under Test (DUT)). Testing was conducted in the ICSA Labs testing facilities located in Mechanicsburg, PA, at lab facilities located at the ISS headquarters in Atlanta, GA and at CT Labs facilities provided by Empirix, located in Rocklin, CA. ICSA Labs Analysts then reduced and evaluated the resultant. Findings contained within this report embody the results of that reduction and analysis.

Summary of Findings and Observations

In pursuit of the goal stated above, ICSA Labs evaluated a number of testing solutions from both commercial product vendors, and “home-grown” options based either on code developed in-house, or from open sources, or a combination of some or all of the above. It is important to understand that while VoIP is not a new technology, the uses to which it is currently being put (widespread use by mainstream consumer and commercial telephony users), and the volumes of calls being generated are a new phenomenon. In general, commercial (and even open source) test solutions tend to lag somewhat behind the leading edge of technology. In this particular case, ICSA Labs was not able to locate a single, integrated test solution which supported all of the desired test scenarios, VoIP protocols or volumes of calls and which produced reliable, dependable statistics. The Hammer family of products from Empirix represented the best fit available to us at the time of testing, even though the Hammer would not allow us to test all of the additional VoIP protocols supported by the ISS Proventia Appliances.

Even using the Empirix Hammer equipment, and using expert assistance from Empirix and their subsidiary, CT Labs, it proved to be extremely difficult to generate the desired volume of calls, to perform voice quality tests and once testing was complete, complex and difficult to analyze and understand the resultant data. Due to these complexities and difficulties, it is therefore not practical to distill the volume of resultant test data contained in the Call Volume and Voice Quality sections of this report to a simplified summary.

In general, the G2000 exhibited superior performance to the G400 as expected and as demonstrated by the tests conducted. Additionally, the G2000 appeared to be more robust in its ability to sustain loads at the upper end of its capabilities, probably due the multi-processor architecture of the G2000 appliance.

Details of the tests conducted, the test environment and methodology are contained in the body of this report, along with the results.

Evaluated Product Components

ICSA Labs Premier Services requires that vendors submit all necessary hardware, software, product documentation and in some cases supporting infrastructure for continuous deployment and evaluation at ICSA Labs. These components, collectively known as Device(s) Under Test (DUT), remain at ICSA Labs for the period of the testing contract. This section details the components of the Devices submitted by ISS for evaluation, as well as any relevant components supplied by ICSA Labs.

Hardware

ISS supplied Proventia G400 and G2000 appliances. Both appliances are 2U rack-mountable servers with the following significant characteristics:

Model:	G400	G2000
Processor type and quantity:	1 Intel 2.4Ghz Xeon	2 Intel 3.6Ghz Xeon
Memory:	2 Gb	3 Gb
Storage:	2x 37Gb drives	2x 37Gb drives
Network Interfaces:	8: 4 on 133Mhz PCI Bus, 4 on 100Mhz PCI Bus	8: 4 on 133Mhz PCI Bus, 4 on 100Mhz PCI Bus
Power Supplies:	Dual 500 watt	Dual 700 watt

ICSA Labs supplied administrative workstations, the network infrastructure used for initial testing, packet sniffers and protocol analyzers, as well as several instances of test equipment and/or software initially used for testing

Empirix initially supplied Hammer NXT-IP test equipment and FX-IP software. Later, testing was conducted at Empirix facilities (CT Labs, Rocklin, California), where Empirix provided Hammer FX-IP systems (used for Voice Quality measurement) and Hammer NXT- IP equipment (used for load testing).

Software

The Proventia Appliances were loaded with base firmware version 1.0_2005.0212_10.08.50, latest firmware update 2005-05-20 07:15:12 version 1.1, and with the initial security content update version of 1.45. In the course of testing the following upgrades and patches were applied:

- ISS supplied a patched version of libCrm.so to address errors detected in the Local Management Interface (LMI).
- The security content was updated to version 1.57 for final testing. Security content updates may contain integrated product revisions.
- Immediately prior to the final round of tests conducted, ISS loaded an update to the systems advertised to improve SIP processing efficiency.

Documentation

ICSA Labs used the following documents and resources in the course of the evaluation. ISS supplied these documents in either printed hardcopy or electronic forms.

- ISS Proventia G400/G 2000 Intrusion Prevention Appliance Quick Start Guide, December 7, 2004 publication date
- ISS Proventia G400/G2000 Appliances User Guide, March 27, 2005 publication date
- ISS SiteProtector Best Practices Guide Version 2.0 Service Pack 5
- XForce Help Files
- pam.chm – Additional Protocol Analysis Module (PAM) documentation

Evaluated Product Configuration

Both G-series appliances were configured identically for the testing conducted. In both cases, the following options were configured:

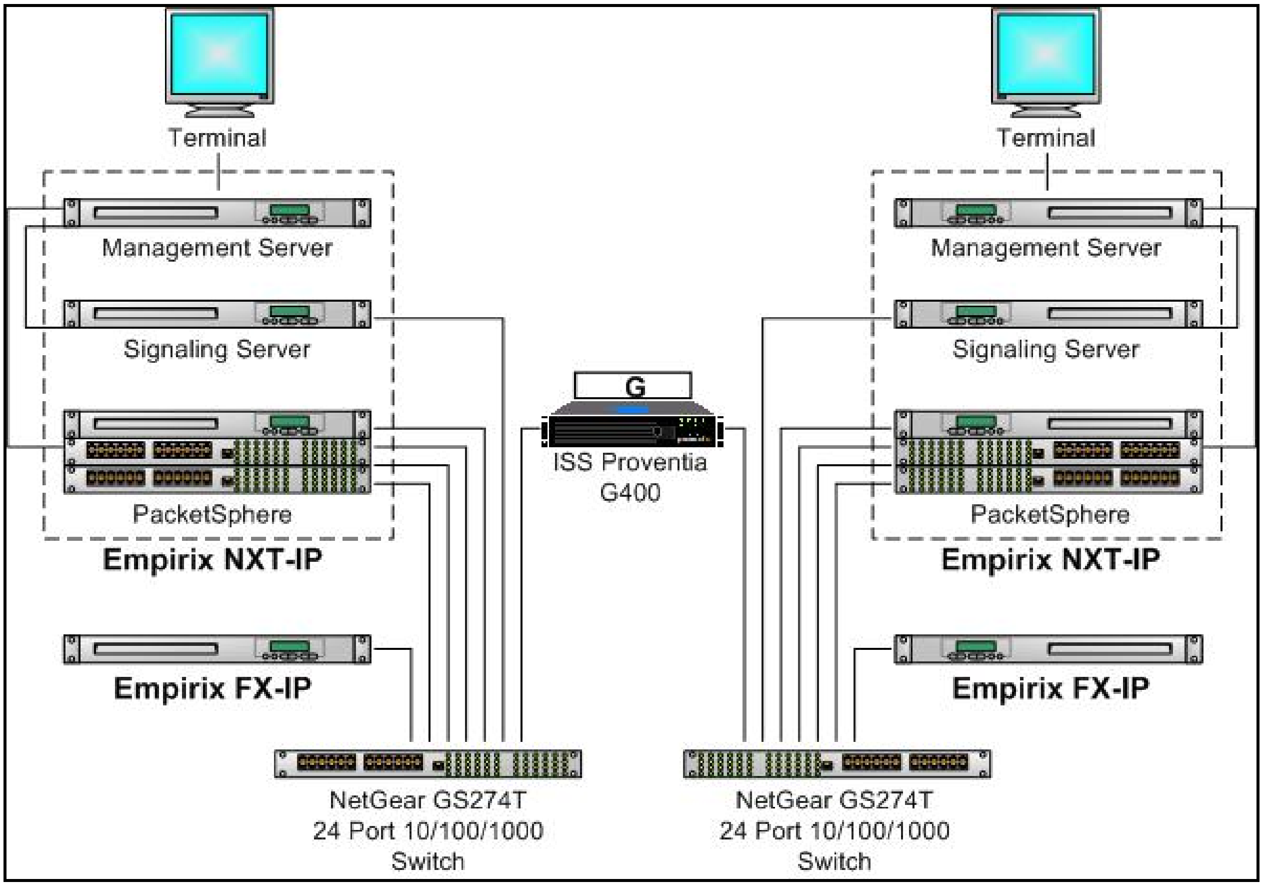
Attack Checks	All Enabled
Attack Blocks	All Disabled
Audit Checks	All Disabled
Evidence Logging	All Disabled

To reduce the load on the DUTs, the Local Management Interface (LMI) was not used while tests were running. Status of the DUTs, to include CPU utilization, was monitored via a serial console using utilities and facilities of the host operating system (the "top" and other utilities, "tailing" the unix system log file).

Test Methodology

ICSA Labs designs a test plan for each Premier Services project to simulate a realistic deployment of the Evaluated Product in a typical end user environment.

In this case, we attempted construction of realistic deployment scenarios with several configurations in support of a number of commercial and open source test tools with varying levels of success before settling on the network topology depicted in the diagram below, and the methods described in the following paragraphs:



The Proventia Appliances were positioned to simulate deployment in a high-volume converged network, specifically a network carrying Voice over Internet Protocol (VoIP) traffic. The network and test scenario was designed to cause increasing levels of VoIP traffic to flow through the DUT (Proventia G400 pictured in the diagram above) while conducting Voice Quality (VQ) and other measurements of the network traffic.

The essential items of test equipment used in these tests were the pairs of Hammer FX-IP and NXT-IP test tools provided by Empirix. Both of these pairs of devices are specifically designed to conduct Telephony-specific tests.

In the tests conducted by ICSA Labs, we used the Hammer NXT-IP to generate an increasingly large volume of calls to the test environment, varying the volume of calls upward during the specific instances of the test, while simultaneously using the Hammer FX-IP to transmit a small but fixed number of calls while measuring the Voice Quality (VQ) of those calls. In other words, the FX-IP equipment generated and transmitted a comparatively small (less than 200) volume of calls and conducted voice quality measurements while the NXT-IP produced the variable portion of the call load.

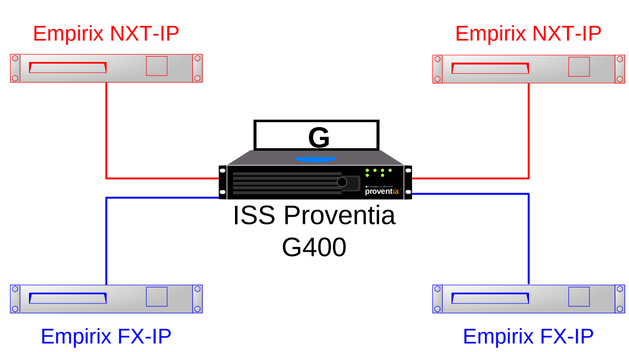
The critical assumption behind this choice of deployment was that the quality of all calls passing through the DUT would be relatively uniform. In particular, even though through the course of testing we varied the call setup protocol (SIP or H.323) and the CODEC (various) transmitted by the NXT-IP equipment, each FX-IP device transmitted at a constant rate (96 calls), using SIP to set up the calls, and with G.711 as the CODEC. ICSA Labs elected to use this static configuration to minimize the setup and transition time from test run to test run, and to maximize the reliability of the resultant voice quality measurements.

The diagram above documents the specific physical network topology of the test environment, including the manufacturer and model of the switches used to connect the DUT to the test infrastructure. All network cabling was category 5e throughout, and the data rate between the DUT and the test equipment was 1 gigabit / second, full duplex, as reported by the Netgear switches.

Prior to conducting tests with the DUT inline, we conducted baseline tests, first using a crossover cable, and eventually with the Proventia G400 and G2000 devices inline but in "passthrough mode". Passthrough Mode is a feature of the Proventia Appliances which, under certain specific (and configurable) conditions such as loss of power, causes hardware relays on the network interfaces to close, bypassing the device, and electrically connecting the two network cables directly. The purpose of this baseline test was to establish the baseline performance characteristics of the test infrastructure itself.

The logical relationship between the network test devices and the DUT is depicted below. To clarify, the pair of Hammer NXT-IP devices initiated and terminated calls between themselves, and the pair of Hammer FX-IP devices initiated and terminated calls between themselves. There was no cross-talk between the NXT-IP equipment and the FX-IP equipment.

Call Volume



Call Quality

As previously described, the FX-IP was configured and used to measure Voice Quality, specifically to compute the Perceptual Evaluation of Speech Quality (PESQ) scores.

PESQ is an internationally recognized and standardized (ITU Standard P.862) automatable measurement algorithm which produces Voice Quality measurement scores on the ITU (International Telecommunication Union) Mean Opinion Score (MOS) P.800.1 scale. MOS provides a method produce objective VQ evaluation, by arithmetically averaging a large number of human evaluations of the call quality of voice samples, originally subjectively judged on a scale from 1 to 5, using the following standardized assignments of values:

Subjective Judgment	Assigned Value
Excellent	5
Good	4
Fair	3
Poor	2
Bad	1

Historically, evaluation of Voice Quality was performed using several human evaluators and reference recordings of tones and voices. After training the evaluators, the reference tones are transmitted through the DUT and the evaluators are asked to assign a rating of the VQ on the scale above. Using large numbers of evaluators, large numbers of playbacks and statistical methods, this method produced objective VQ evaluations, albeit at the costs of time, difficulty and tying up human resources.

The automated PESQ computation yields objective, repeatable, and comparable results based on the same scale, but the computation is conducted without the human components, instead substituting a perceptual model. Besides being inherently repeatable due to being automated, another advantage offered by automated PESQ computations is the ability to compute dependable results based on smaller numbers and shorter samples. PESQ scores in this report range from 1.0 to 4.5. Additional technical details on VQ and PESQ measurements relevant to these tests are available from Empirix at www.empirix.com.

It is important to understand that even under optimal conditions, MOS scores on modern Telephone networks do not come close to 4.5, much less to 5.0. The easiest way to explain this fact is to understand that even on the traditional PSTN (Public Switched Telephone Network), carriers use digital encoding for the backbone. The CODECs used for this purpose (and for VoIP) all make compromises of one sort or another to reduce the amount of data transmitted. With the compromise comes a loss of Voice Quality. So, while theoretically, PESQ scores range up to 4.5, in practice the measured scores are lower.

For the purposes of this evaluation, ISS set a threshold of 2.5 as the low water mark for acceptable voice quality. The overall goal of the testing was to measure the rate of calls carried by the Device Under Test as the calculated Voice Quality (PESQ) score degraded and crossed the 2.5 threshold.

Samples, Frame Rates, CODECs, Packet Sizes and other variables; Understanding the Big Picture of Internet Telephony.

It is critical to understand the inter-relationships of several factors in Digital (Internet) Telephony, specifically sample sizes and CODECs, and their potential influences.

Traditional Telephony transmits sounds in their original analog forms. Modern telephone systems, as mentioned earlier, encode these analog sounds into digital form for transmission, and then decode them back into analog form for the listener.

In the process of encoding, the continuous stream of analog sound is broken into discreet samples of a given length. The choice of the sample size, typically measured in milliseconds (1/1000 of a second), directly drives the number of samples per second. For example, a sample of size of 20 milliseconds means that 50 samples occur per second (1000 milliseconds per second divided by 20 milliseconds per sample yields 50 samples per second).

The rate of samples per second is determined by the choice of the sample size, and is one of two major influences on the final frame rate at the end of the conversion process. The other significant influence is an option known as “Silence Suppression” or “Voice Activity Detection”. By way of explanation, consider a normal telephone conversation between two parties. At least some of the time, neither party is speaking, so there is no data (or precisely, only silence) to be sampled, encoded and transmitted. Silence Suppression or Voice Activity Detection are optional features to detect the case where there is no sound to be transmitted, and to reduce the number of final packets transmitted by not encoding and transmitting packets which contain only silence. In practice, this option is desirable in many cases because it reduces the packet rate, maximizing the utilization of finite transmission resources (primarily bandwidth). For purposes of testing, we chose to disable this option, thus forcing a worse case situation which maximizes the packet rate.

Processing an analog stream by converting it into samples, encoding it for transmission, and converting it back into analog form on the receiving end is done using a particular method called a CODEC (COder-DEcoder). Simply put, CODECs are standardized methods to put the analog stream into digital form, and then back into analog form. Some CODECs are “lossy”, that is, there is some (ideally below the human threshold of perception) loss of information (quality) in the process. This loss is accepted as a conscious compromise to sacrifice some information for a gain of some sort, typically in terms of computational intensity in encoding and decoding, resultant packet size, etc. Other CODECs are lossless, meaning that all of the information encoded is recoverable when decoded. Additionally, some CODECs have built-in data compression, which may further reduce the size of the transmitted packet, but at the cost of increased computational intensity.

In summary, the choice of the sample size and the CODEC greatly influence the rate of packets per second, and the size of those packets for a given number of simultaneous calls. Frequently, from the point of view of an infrastructure device such as the ISS Proventia Appliance, the values of these variables are no more controllable than any of the many other aspects of the public internet (lack of guaranteed bandwidth, lack of fixed and /or acceptable propagation delay, best-effort as opposed to guaranteed packet delivery, dynamic routing, etc.) which can influence the quality of voice calls placed.

In addition to the actual encoding, transmission and decoding of the voices, there is a call setup component. A number of protocols exist and serve this purpose, and based on the product documentation from ISS, the Proventia series supports and decodes:

- SIP (Session Initiation Protocol, IETF RFC 3261)
- H,323 (an ITU VoIP protocol, including sub-protocols H.225 and H.245)
- MGCP (Media Gateway Control Protocol, documented by several IETF RFCs beginning with RFC 2705, and standardized by several ITU documents)
- SCCP (Skinny Client Control Protocol, defined by Cisco)
- STUN (Simple Traversal of UDP over NATs).

At the time these tests were conducted, the FP-IP and NXT-IP equipment from Empirix only supported the SIP, H.323 and MGCP protocols. In order to reduce testing complexity, ICSA Labs and ISS agreed to limit the testing to pure SIP and pure H.323 only. Additionally, each individual set of tests with SIP or H.323 were conducted with a uniform mix of CODECs.

From the point of view of the caller or the call recipient, any intervening infrastructure device is probably beyond his control, and all intervening devices contribute to the overall call experience. From the point of view of the administrator of an infrastructure device, the nature of the VoIP traffic passing through his device may also be beyond his control. Given this lack of control, it is particularly important that each and every infrastructure device potentially involved in the transmission of VoIP influence the traffic which they carry in the least possible way. In other words, these devices should be designed to have as close as possible to no impact on Voice Quality.

The stated goal of the testing conducted for this project was to establish the rate of calls that the Device Under Test could sustain while maintaining PESQ scores of 2.5 or greater, yet one must also consider the question of some of the other aspects of telephonic quality, specifically availability (the ability to establish a connection) and reliability (dropped calls, wrong numbers, etc.). This question has both practical application and implications in the testing environment.

From the testing point of view, the Empirix test equipment will only tolerate a certain level of errors (dropped calls, etc.) before aborting the test in progress. If the call quality has not degraded, or has not yet degraded below the desired threshold of a 2.5 PESQ score, but the packet loss characteristics are causing a high number of failed calls, then the test will simply abort.

In the practical sense, this is also significant. Consider an example application of the Device Under Test being deployed in front of a high volume call center, perhaps one handling emergency service calls. In that case, it might be perfectly reasonable to trade some degradation of call quality for higher call volume capacity, so long as calls go through and the quality of those calls has not degraded past the point of intelligibility. The failure of a single call in this environment, where lives may be at risk, even if the voice quality of those calls which succeed approaches perfection, is probably not acceptable.

Tabulated Call Volume Results and Discussion

Both of the protocol sections below (H.323 and SIP) contain a summary table documenting the CODECs and sample rates tested, along with the resultant packet sizes generated by the test equipment. Subsequent tables in each section detail the maximum observed call volume and corresponding CPU utilization percentage as observed on the tested appliance console.

In all cases, call volume was elevated until one or more of the following limiting conditions were reached, or event occurred:

- Onset of failed calls reported by either of the sets of test equipment
- 100% CPU utilization on the DUT (as reported by the top utility running on the system serial console)
- Network Transmit or Receive buffer overflows occurred (reported in the system log file)
- Severe packet loss reported by either of the sets of test equipment
- Notable manually observed or computed degradation of voice quality
- The Hammer NXT-IP reached documented call volume performance limits

Footnotes are provided throughout to indicate the limits reached.

Selected H.323 Call Results

As discussed earlier, choice of the sample size and CODEC influence the size of the data payloads in the resultant RTP datagrams. The Empirix test equipment has differing traffic generation capacities depending on these (and other) choices, based on limiting factors in the appliance hardware. Detailed call volume data is reported in individual tables, along with the other relevant parameters.

H.323 CODEC/Payload Size Iterations

Table 1 H.323 Codec/Payload Size

PROTOCOL	CODEC	SIZE OF PAYLOAD (BYTES)	G400 TESTED	G2000 TESTED
H.323	G.726 40 KBPS (10 MS)	50	✓	✓
H.323	G.726 40 KBPS (20 MS)	100	✓	✓
H.323	G.726 40 KBPS (30 MS)	150	✓	✓
H.323	G.726 32 KBPS (10 MS)	40	✓	✓

Table 2 H.323 G.726 40 KBPS MAX CALL RESULTS

PROTOCOL	CODEC	SIZE OF PAYLOAD (BYTES)	G400 CPU	G400 CALL VOLUME	G2000 CALL VOLUME	G2000 CPU
H.323	G.726 40 KBPS (10 MS)	50	99.9 %	2,785	6,000	65.2%
H.323	G.726 40 KBPS (20 MS)	100	99.9 %	3,081	4,783	99.9%
H.323	G.726 40 KBPS (30 MS)	150	99.9%	4,123	5,849	99.9%

Table 3 H.323 G.726 32 KBPS MAX CALL RESULTS

PROTOCOL	CODEC	SIZE OF PAYLOAD (BYTES)	G400 CPU	G400 CALL VOLUME	G2000 CALL VOLUME	G2000 CPU
H.323	**G.726 32 KBPS (10 MS)**	40	99.9 %	3,485	6,000	62.45 %

Selected SIP Call Results

As with the H.323 tests and results above, we noted variances in the performance of the Devices Under Test with SIP testing as well. Detailed call volume data is reported in individual tables.

SIP CODEC/Payload Size Iterations

Table 4 SIP Codec/Payload Size

PROTOCOL	CODEC	PAYLOAD OF SIZE (BYTES)	G400 TESTED	G2000 TESTED
SIP	**G.726 16 KBPS (20 MS)**	40	✓	✓
SIP	**G.726 16 KBPS (30 MS)**	60	✓	✓
SIP	**G.729 A (10 MS)**	10	✓	✓
SIP	**G.729 A (20 MS)**	20	✓	✓
SIP	**G.729A (40 MS)**	40	✓	✓

Table 5 SIP G.726 16 KBPS MAX CALL RESULTS

PROTOCOL	CODEC	SIZE OF PAYLOAD (BYTES)	G400 CPU	G400 CALL VOLUME	G2000 CALL VOLUME	G2000 CPU
SIP	**G.726 16 KBPS (20 MS)**	40	99.9 %	2,885	10,288[1]	70.2 %
SIP	**G.726 16 KBPS (30 MS)**	60	99.9%	2,345	10,400	73.4 %

Table 6 SIP G.729a MAX CALL RESULTS

PROTOCOL	CODEC	SIZE OF PAYLOAD (BYTES)	G400 CPU	G400 CALL VOLUME	G2000 CALL VOLUME	G2000 CPU
SIP	**G.729A (10 MS)**	10	99.9 %	12,288	12,288[2]	72.5%
SIP	**G.729A (20 MS)**	20	99.9 %	2,150	9,000[3]	70.5 %
SIP	**G.729A (40 MS)**	40	99.9%	2,288	9,000	82.5%

[1] 0 – 1% PACKET LOSS OBSERVED THROUGH TEST EXECUTION
[2] LIMITATION OF TEST EQUIPMENT REACHED, UNABLE TO TEST PAST THIS CALL VOLUME ON THIS PARTICULAR CODEC/SAMPLING RATE
[3] LOST CALLS OBSERVED ON NXT-IP ON 2ND ITERATION OF TEST CYCLE, PACKET LOSS >9.4 % WHEN CALLS WERE INCREMENTED DURING TEST CYCLE

Graphical Call Volume / Voice Quality Results and Discussion

As discussed previously in this document, ICSA Labs used Hammer FX-IP and NXT-IP test equipment to impose call load and measure voice quality. Simultaneously, we used the NXT-IP to introduce variable and increasing loads using the H.323 and SIP protocols, and various CODECs and sample rates, and the FX-IP to measure voice quality.

While conducting Voice Quality measurements, each FX-IP was configured to establish and maintain 96 simultaneous SIP calls using the G.711 u-law CODEC with a 20 millisecond sample size, and to measure the Voice Quality by computing PESQ values. The configuration of the FX-IP equipment remained static throughout all VQ testing. The basis and justification for this fixed configuration is discussed below.

It is important to understand that the current versions of the Empirix Hammer FX-IP and NXT-IP software, while quite capable, do have some limitations.

The specific parameters of the calls generated by the NXT-IP (for example, the number, type and duration of simultaneous calls) can be configured with a great degree of precision, and while the equipment provides graphical utilities to monitor tests in progress, the number of calls in progress at a particular instant in time is not displayed, recorded or logged directly by the test equipment. Individual call detail records are recorded, so it is possible, though time consuming and difficult, to calculate the volume of calls at a particular point in time by post-processing the output data, and to manually correlate the resulting volume statistics with written notes describing the test being performed. Doing so allowed us to synthesize a record of the call volume correlated with the time, and the protocol / CODEC combination being tested.

Correspondingly, the FX-IP as it was deployed at CT Labs during testing, was not capable of calculating and recording Voice Quality data in such a way as to facilitate correlation with other externally generated data, specifically call volume.

The FX-IP measures voice quality by transmitting a reference waveform through the Device Under Test, and recording the waveform as received. The PESQ is then computed by comparing the resultant sample with the original waveform, and the result is recorded along with a timestamp. Unfortunately, processing resources on the test equipment are finite. Priority is given to transmitting, receiving and recording of the samples, so that it is possible that the PESQ scoring will be queued, and not processed in or near real time. While the resultant scores PESQ are accurate, and refer to the call in question, the time recorded refers to the current time when the computation of the PESQ completed, not when the sample was collected. Because of this, we manipulated the configuration of the FX-IP to minimize the resultant time skew, specifically by reducing the volume of calls it sent to a fixed rate of 96 simultaneous calls, and by choosing the protocol and CODEC to minimize FX-IP CPU utilization. Even when this is done, the accuracy of the time stamps on the PESQ computations is not exact, and makes correlating these PESQ values with call volumes more challenging.

To satisfy the goal of testing for this project, it was necessary to correlate the PESQ scores computed and logged by the FX-IP equipment with the call volume, protocol and CODEC data synthesized as described above. ICSA Labs calculated the call volumes for the tests run, and correlated them with the PESQ scored calculated and recorded.

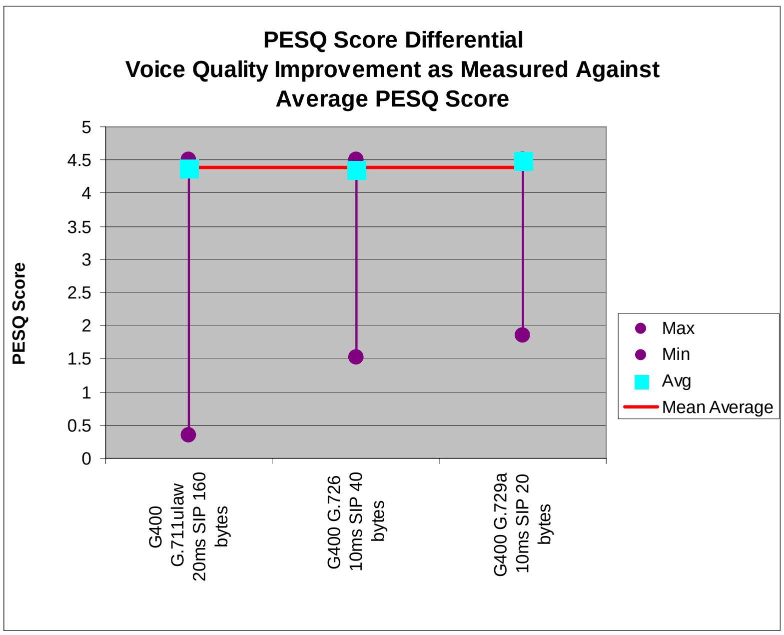
Due to the factors described above, it was not always possible for us to correlate these quantities with a comfortable degree of accuracy, or in some cases at all. Consequently, we have chosen in those cases to omit those results from this report.

The format chosen by ISS to present these results depicts the mean average PESQ score for the CODECs tested, and also plots the individual maximum and minimum PESQ scores recorded for each CODEC. It is important to understand that the PESQ is measured multiple times by the test equipment during each test run. The single lowest data point (PESQ score) recorded during a test will cause the minimum in the chart to plot in some cases significantly lower than the mean average, and may fall below the value of 2.5, which was the targeted lower bound for the testing.

Finally, in the course of testing we discovered and documented some other limitations of the Empirix test equipment. Specifically, while it was possible to configure the NXT-IP to send calls using SIP or H.323 and the G.726 CODEC, and to observe the tests in progress, the equipment did not consistently log those calls. Call volume data reported in the previous section of this report for these combinations was the result of manual observation and data correlation while the tests were in progress. Correlation of call volume and VQ in most of these cases was not possible.

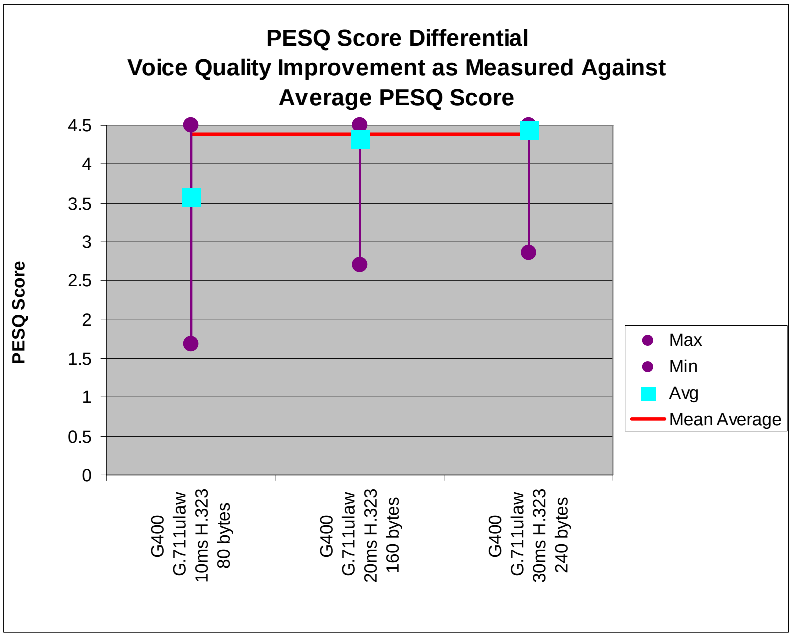
G400 SIP PESQ Score Differential

Table 7 G400 SIP PESQ Score Differential



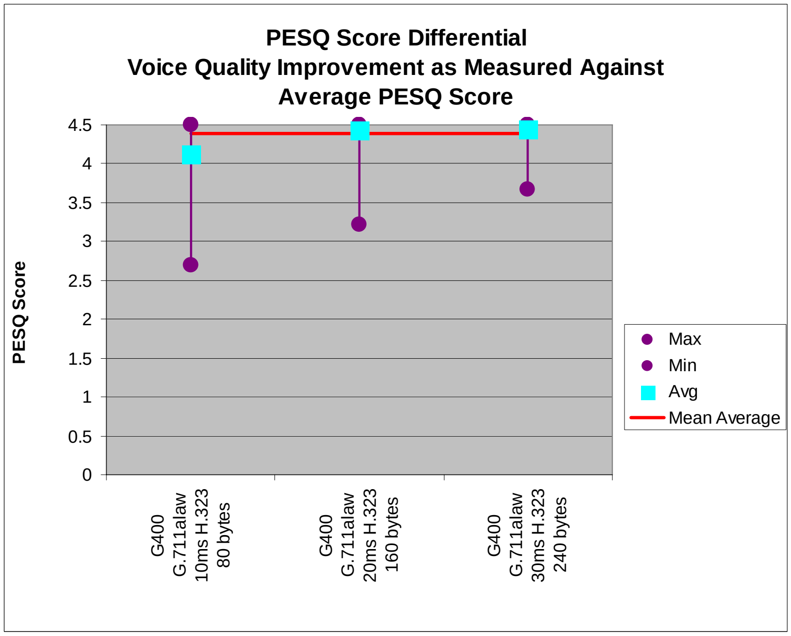
G400 H.323 PESQ Score Differential I

Table 8 G400 H.323 PESQ Score Differential I



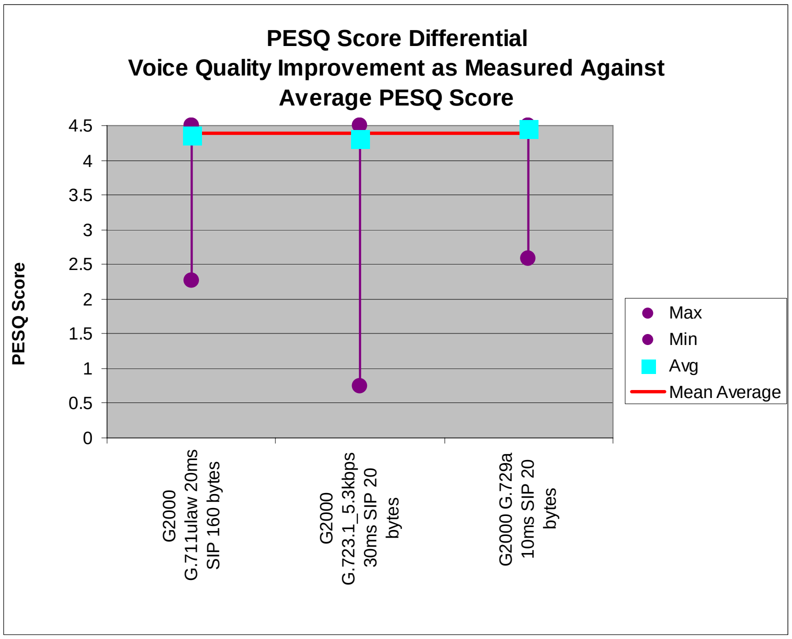
G400 H.323 PESQ Score Differential II

Table 9 G400 H.323 PESQ Score Differential II



G2000 SIP PESQ Score Differential

Table 10 G2000 SIP PESQ Score Differential



G2000 H.323 PESQ Score Differential

Table 11 G2000 H.323 PESQ Score Differential