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Voice over IP Testing - A Practical Guide

RADCOM White Paper

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Communications Category
One of two Runners Up
2001

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1. Introduction

Voice over IP networks are complex! They represent the converging worlds of tele- and data communications, and therefore present myriad implementation and testing challenges:

- Integration to traditional telecom infrastructure
- *Integration to billing systems*
- *Many add-on services*
- *Large variety of protocols*
- *Quality is an issue*
- *Network specialists are expensive and scarce*
- *Reliability is a must*
- *Multiple High Quality Services: voice, fax, video, unified messaging, call centers, etc.*

This white paper presents a typical VoIP architecture and then suggests a framework for testing VoIP networks. The test strategy is presented as well as a detailed discussion of the actual testing required for each network element. Finally, a list of Voice over IP specifications is provided as an appendix as well as a list of acronyms. The main objective of this paper is to provide insight into the intricacies of architecting Voice over IP networks of carrier grade quality. It is intended for network design and test engineers.

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2. VoIP Architecture

A typical VoIP network includes the following components:

- *Media gateways*
- *Signalling gateways*
- *Gatekeepers*
- *Class 5 switches*
- *SS7 network*
- *Network management system*
- *Billing systems*

All of these network elements communicate with each other using a plethora of protocols, as can be seen in Figure 1. A detailed list of protocols and specifications can be seen in Appendix I.

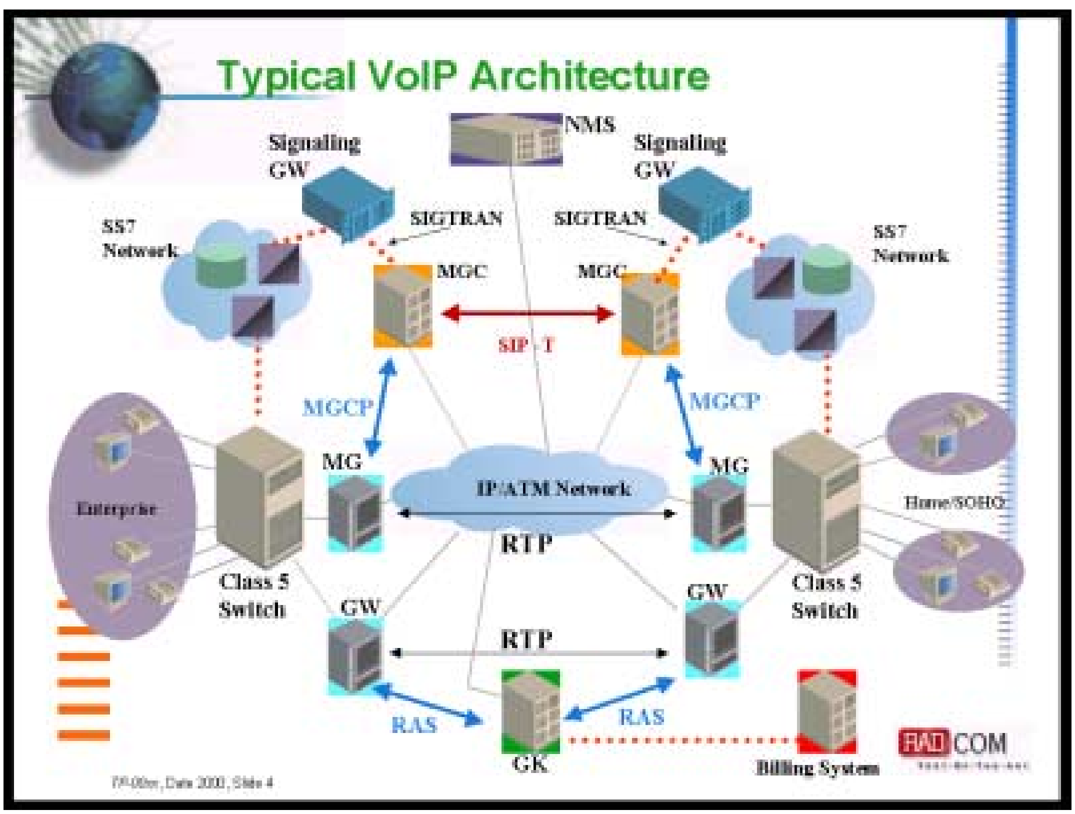


Figure 1 - Typical VoIP architecture

3. Test Strategy

Testing VoIP networks is a tri-fold task:

- *Functionality verification*
- *Standards compliance*
- *Performance verification*

A successful pre-deployment testing strategy must address each of these three facets:

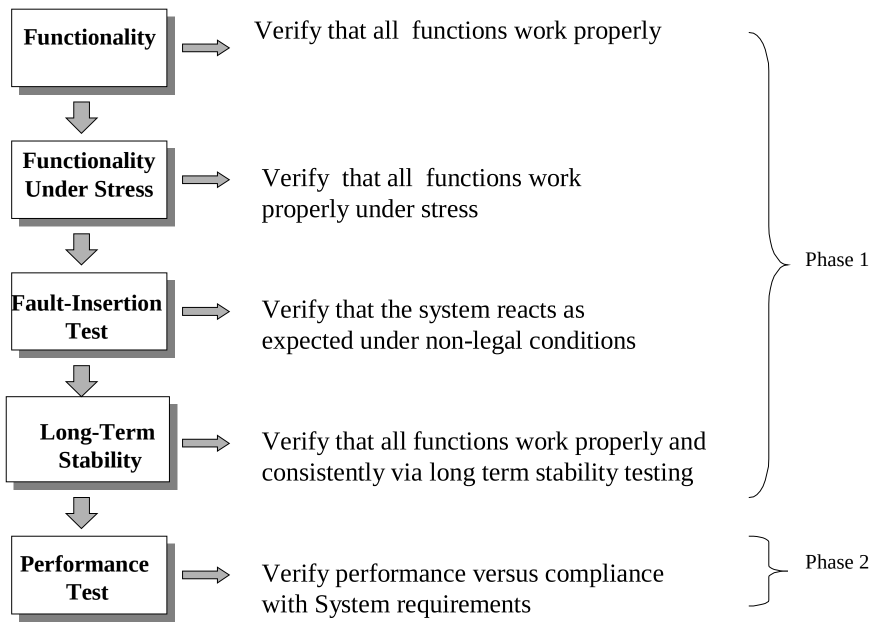


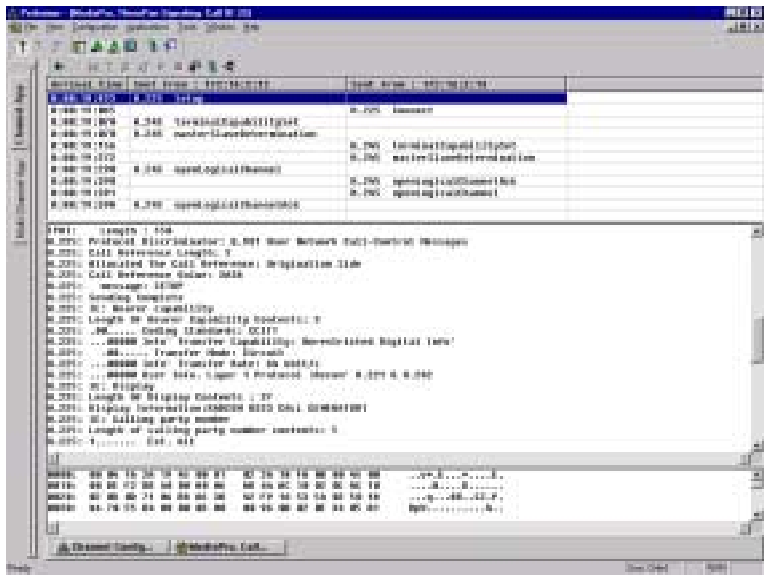
Figure 2 - Test Strategy

Changes such as software or hardware version upgrades can cause degradation in functionality, quality and performance. Therefore, it is very important to repeat this test cycle after every change made to the VoIP network.

4. VoIP Testing

Following are VoIP network components that must be tested prior to deployment:

- *Gateway (GW) and Media Gateway (MG)*
- *Gatekeeper (GK) and Media Gateway Controller (MGC)*
- *Signaling Gateway*
- *Interactive Voice Response (IVR) and Voice Mails*
- *Billing and Prepaid system*
- *Network Management System (NMS)*



Ideally, these tests should be performed in a lab environment so as to minimize deployment, troubleshooting, operational and maintenance costs. When functional tests fail there is no way of avoiding the “dive” into the detailed protocol implementation to verify the conformance of the VoIP devices. This requires detailed decoding capabilities of all VoIP protocols. H.323 protocols use the ASN.1 notation while protocols such as SIP and Megaco use plain ASCII messages. Figure 3 shows the signalling decodes of a VoIP call and Appendix I includes a complete list of all VoIP protocols and their specifications.

Effective pre-deployment testing follows a well-defined methodology that addresses the variety of issues that can impact the network’s adherence to specifications in a real world environment. Special consideration should be given to the expected behavior of the VoIP network. This includes parameters such as the number of anticipated users and the estimated amount of traffic per user. Existing network infrastructure should also be taken into account – what type of network is used: Frame Relay, ATM, VSAT, xDSL, WLL etc. The expected network performance including parameters such as latency, packet loss and available bandwidth is also of significant importance. The test engineer should also consider implementation specific parameters such as the compression methods that will be used, the packet structure of the packetized voice and more.

The Poisson statistical model, a generally accepted tool to predict end user behavior, should be incorporated in the pre-deployment test plan. Using this model and based on the assumption that the average call duration is 180 sec, the VoIP network specifications can be defined using the following parameters:

1. **Blocking** - defined as the percentage of calls that get a busy signal because all lines are in use. This can be calculated as,

$$\text{Blocking} = \frac{\text{Required Grade Of Service}}{100}$$

   Or in other words,

$$\text{Blocking} = \frac{\text{Number of failed call attempts}}{\text{Total number of call attempts}}$$

2. **Busy Hour Traffic** -This is the amount of call traffic handled by a group of phone lines during the busiest hour of the busiest day for your system. Busy Hour Traffic is defined in units of Erlangs or CCS. It can be typically calculated as,

   B.H.T= (Number of anticipated end users) * 0.05

3. **Centi-Call Seconds (CCS)** – This is a unit of Busy Hour Traffic commonly used for traffic measurement. 36 CCS equals 1 Erlang of traffic.
4. **Erlang** – This is a unit of Busy Hour Traffic and represents the continuous use of a single line for one hour. For example, 30 calls of 2 minutes holding time each would equal 1 Erlang of traffic. On a typical Voice over IP network the end user traffic is between 0.01 Er and 0.15 Er. For detailed Erlang calculations you may refer to http://www.erlang.com/calculator/.

When designing a Voice over IP network it is important to avoid bottlenecks in the design. A T1 can usually support up to 18 Erlang with a Grade of Service of 5%. An E1, on the other hand, can support up to 24.8 Erlang with a Grade of Service of 5%. From these requirements one can calculate the number of customers a typical link can support. For a T1,

$$N(T1) = \frac{18 Erlang}{0.05 Erlang} = 360$$ customers.

And for an E1,

$$N(E1) = \frac{24.8 Erlang}{0.05 Erlang} = 496$$ customers.

Simultaneous calls can be made according to number of trunks i.e. 24/23/30 (for T1-CAS/T1-PRI/E1-PRI respectively), but the limitation will be derived from two other factors:

- *Compression method*
- *Guaranteed bandwidth*

After the Voice over IP network has been proven for functionality, a series of stress tests should be conducted. It is important to have a consistent definition of stress. The recommended criteria for a stressed network dictate the configuration of the test devices and are as follows:

A. Pre-define number of calls per session and 100 setup calls per second.
B. Create Jitter, Packet-loss, Packet out of sequence and Latency in Uniform mode.
C. The VAD and the silence suppression mechanism should be activated.
D. The RTP packets should consist of 1 frame per packet and 3 frames per packet.

The foregoing reflects general requirements involved in VoIP network testing. The following will address specific tests of the various components:

- *Gateway testing*
- *Gatekeeper testing*
- *IVR testing*
- *Billing system testing*
- *Network management system testing*

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5. Gateway Testing

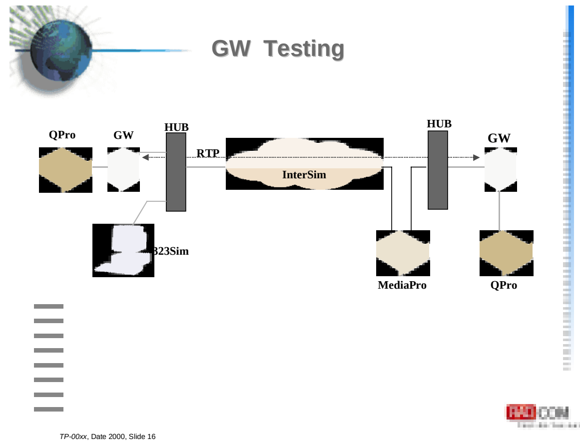

Figure 3 – Gateway testing

Testing a gateway gets to the heart of the convergence VoIP network – the connection between the packet side and the circuit side. One has to test the functionality of the gateway and its capability to operate under stress. Signalling performance is measured as the Grade of Service (GoS) and media performance is measured as Quality of Service (QoS). The tests include the generation of a large volume of calls from the circuit side and analysis of the signalling and media performance of these calls on the packet side. A second stage includes the generation of a large volume of calls from the packet side and analysis of the performance of these calls on the circuit side. Finally, it is recommended that the complete system be tested using an end-to-end test scheme, like the one displayed in Figure 3. Two gateways are connected through an Internet cloud passing calls that are generated on the circuit side. This is the most ubiquitous configuration in current VoIP networks. The scenario includes performance measurement on both the circuit side and the packet side to provide a complete picture of the capability of the network under test.

The tests should include a variety of aspects:

- *Compression and De-compression*
- *Bandwidth utilization*
- *Silence suppression and VAD*
- *DTMF detection and Generation*
- *Jitter suppression and Echo cancellation*
- *Fall-back to PSTN mechanism*
- *Alternative re-routing mechanism*
- *IVR for 2-Stage Dialing*

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Moreover, testing and evaluating the Voice Quality is extremely important. The algorithm most commonly used for these purposes was developed by British Telecom and it is called **PAMS** (Perceptual Analysis Measurement System). A speech signal is generated on one side of the network and the degraded signal is captured at the other side. A quality prediction is made on the received signal based on a mathematical comparison to a stored reference file. The PAMS algorithm implements a model of the human hearing and transforms the speech signal to a three-domain representation – time, frequency and amplitude. It is important to be able to perform this test from the circuit network to the packet network and from the packet network to the circuit network.

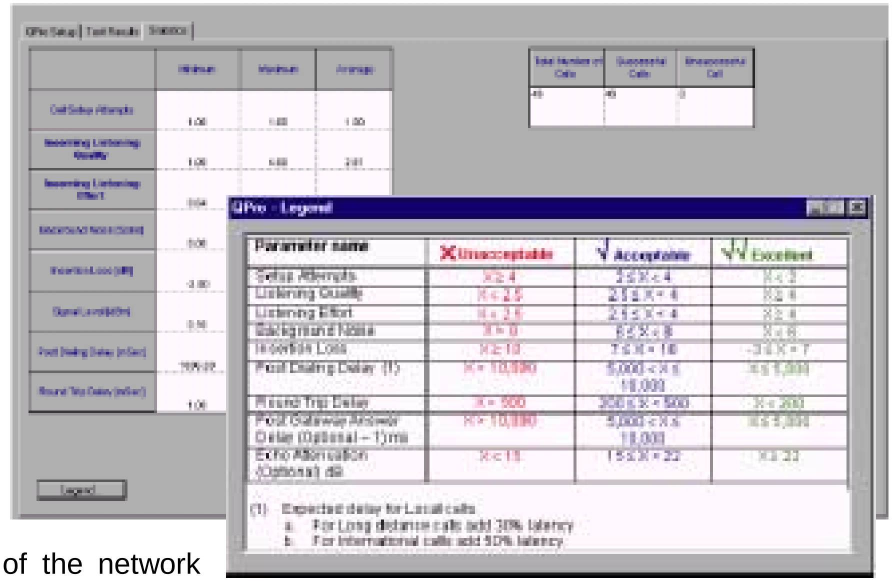

Figure 4 – PAMS provides objective MOS results

Finally, in a real converged network voice and data are not the only types of traffic. Fax is also very common on VoIP networks. When considering fax transmissions the most important thing to test is the packet loss recovery mechanism. This includes the T.38 redundant packet transmission, the TCP retransmission sliding window mechanism and the FEC (Forward Error Correction). Furthermore, the switching mechanism between fax and voice needs to be tested. All of these tests can be performed by sending fax traffic through a simulated packet network with a variety of different network conditions emulating the loss of packets and measuring the quality of the fax received.

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6. Gatekeeper Testing

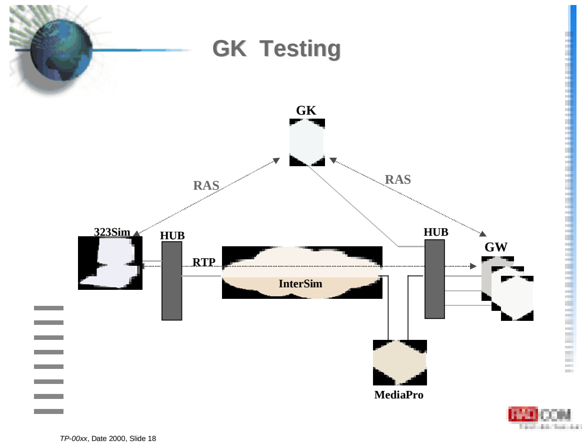

Figure 5 - Gatekeeper testing

The **Gatekeeper** is the traffic controller of the Voice over IP network. It determines the call routing scheme and its correct operation under stressful network conditions is crucial for providing a carrier grade solution (an acceptable Grade of Service). The first thing to test on a **Gatekeeper** is its **Registration** mechanism – to ensure that it can register VoIP elements. Privacy and security are an important aspect of any network and are of particular concern on a VoIP network. Therefore, it is also important to test the **Admission** and **Authorization** mechanism on the **Gatekeeper**.

The Gatekeeper communicates with both the VoIP terminals and the Gateway, and the language it uses is H.225 and more specifically RAS (Registration, Admission, Status). To properly test the compliancy of the Gatekeeper's implementation of RAS, emulation of a VoIP terminal performing RAS negotiation with the Gatekeeper under a stressed network is required.

Once the Gatekeeper accepts a terminal, it can make calls and use the Routing Directory Service that the Gatekeeper provides. This routing can be done in two ways – least cost routing or best cost routing. Least cost routing means that the least costly route will be selected. Best cost routing means that the best BPS (Bit Per Second) route will be selected. In other words, the Gatekeeper will choose a route that provides the best combination of performance and cost. Some Gatekeepers support RSVP (Resource ReSerVation Protocol) and can assign a route to a call based on the resources available toward the receiving end.

Gatekeepers have two modes of operation - direct mode and routed mode. The routed mode is more commonly used. When the gatekeeper performs address translation, the gatekeeper provides endpoints with the transport address for the call signaling channel destination. In the direct mode, the gatekeeper provides the endpoints with the address of the destination endpoint and directs them to the call-signaling channel so that all messages can be exchanged directly between the two endpoints without gatekeeper involvement. The Gatekeeper test procedure should include tests for both modes of call control routing.

The Gatekeeper can also control bandwidth allocation. Through H.225.0 signaling, the gatekeeper is able to limit the bandwidth of the call to less than what was requested as well as reject calls from a terminal if it determines that there is insufficient bandwidth available on the network to support the call. The testing scenario should include several tests with calls generated asking for a bandwidth that is just below the allocated bandwidth and just above it to verify the operation of the bandwidth allocation mechanism on the Gatekeeper. This should be performed with a variety of bandwidth settings on the Gatekeeper.

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7. IVR Testing

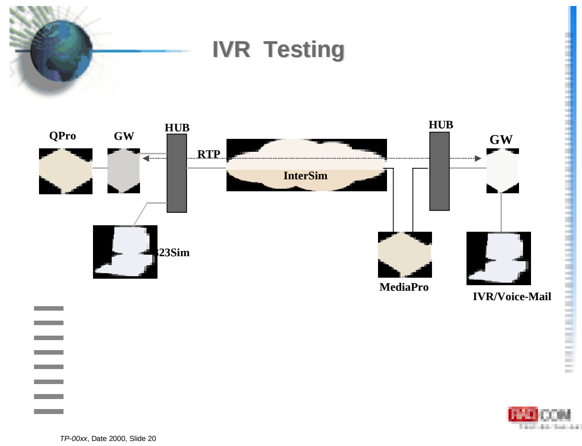

Figure 6 - IVR testing

IVR (Interactive Voice Response) is an integral part of any business phone system. Practically every call center implements some sort of an IVR system because it reduces operational and human resource costs. For VoIP systems to be used in a business environment they must support IVR, which also means that they have to be tested to ensure their correct operation in real world applications. Both functionality and performance under stress need to be tested. IVR systems use DTMF (Dual Tone Multi Frequency) tones to transfer user requests to the system. DTMF tones are the same tones used for tone dialing. The DTMF tones are sums of two sine wave tones at the following frequencies:

	1209 Hz	1336 Hz	1477 Hz
697 Hz	1	2 ABC	3 DEF
770 Hz	4 GHI	5 JKL	6 MNO
852 Hz	7 PQRS	8 TUV	9 WXYZ
941 Hz	*	0	#

Figure 7 - DTMF frequencies

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Testing the capability of VoIP networks to deal with IVR systems must include a DTMF integrity test that passes all combinations of DTMF tones on the VoIP network and verifies the correct transmission over the packet network. But verifying correct transmission alone is not sufficient, careful attention should be given to ensure that the transmission would remain correct even when the network is under stress traffic.

Of paramount importance to IVR systems is the ability to record the user's voice. Voice mail is the most common application. Testing this capability of the IVR system requires the ability to play back the voice mail and measure voice quality on the recorded audio stream.

Voice recognition is another mechanism of IVR systems and it should be tested to ensure its functionality and reliability under stressed network conditions.

Finally, all of the above mentioned tests must be conducted under rather severe network conditions_since Latency, jitter, packet loss and out of sequence packets are common occurrences in a real world packet network.

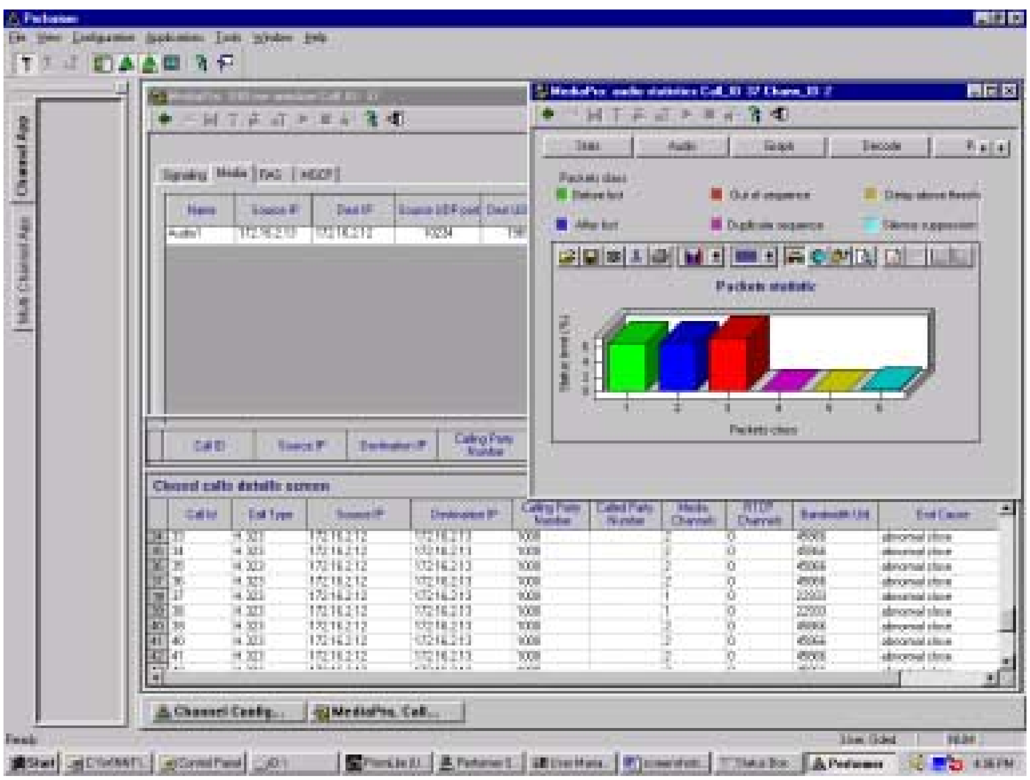

Figure 8 – VoIP call analysis and packet statistics

8. Billing & Pre-paid Testing

Billing systems are arguably the most mission critical part of the Voice over IP network. If they fail, the service provider's bottom line can be adversely affected. It is crucial to ensure CDR (Call Detail Record) integrity when the network is operational – which means 24*7*365. CDR integrity consists of the correct transmission and measurement of the following parameters:

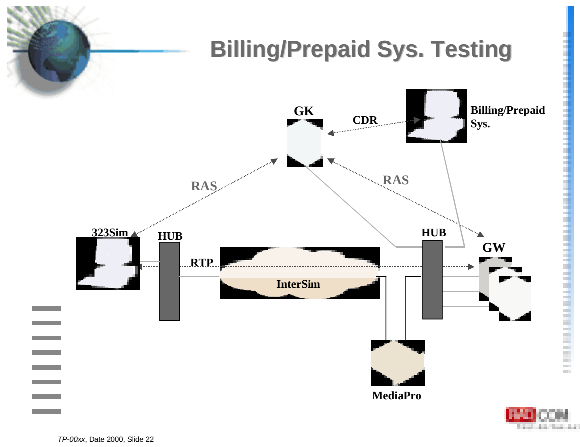

Figure 9 - Billing/Prepaid system testing

- *CLID (Calling Line Identification)*
- *Call duration*
- *Called ID*
- *PIN (Personal Identification Number)*

When the network is used for both voice and data traffic, the billing system should also be able to measure bandwidth used by the customer, as well as the Quality of Service provided.

Prepaid calling cards allow mobile users to place inexpensive phone calls. This service employs a combination of an IVR system and the billing system and, as such, should also be tested for functionality.

The billing system is automatically connected to the charging system – automatically charging a customer's account (service provider account or credit card account) upon usage of the network. This is another aspect of the billing system that needs to be verified to ensure that there is no lost revenue.

Once again, it is important to perform all of these tests under stressed network conditions.

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9. NMS Testing

The Network Management System will typically have connections to the Gateway and the Gatekeeper of the Voice over IP network. It will aggregate and report on network alarms such as over utilization of the assigned bandwidth, bottlenecks and network degradation situations. This is usually done in two ways:

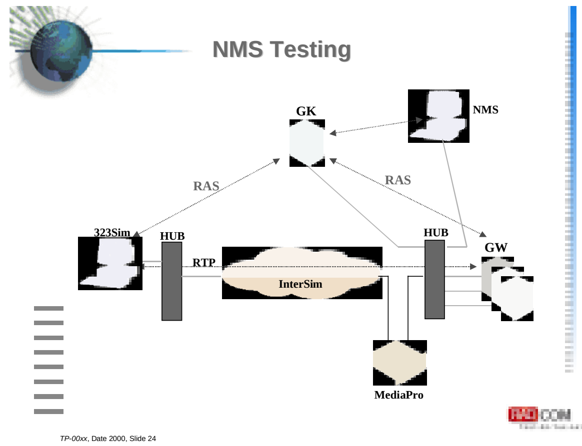

Figure 10 - NMS testing

- *Proactive and preventive – a status report will be generated every pre-configured period of time.*
- *Breakdown maintenance – alarms will be sent when a specific failure has occurred.*

The testing should include alarms verification when specific failures occur. This can be accomplished by emulating the types of errors that might occur in the real world:

- *Jitter exceeds a certain threshold – a typical number would be 5 mSec.*
- *Packet loss percentage exceeds a certain threshold – a typical number would be 5%.*
- *Bandwidth exceeds a certain threshold – a typical number would be 30% of the pipe's bandwidth.*

10. Conclusions

Since VoIP enables provisioning of enhanced telephony services, many service providers and infrastructure vendors are aggressively focusing on this technology. Service providers eye global expansion as a means of achieving economies of scale and increasing their subscriber base. Toward that end, many are engaged in building POPs on international markets and/or entering partnerships with local players. However, in order to attract and maintain customers, VoIP networks must deliver a successful combination of functionality, performance and quality. This paper offers a guideline to pre-deployment testing methodology that will help ensure consistent and reliable delivery of the carrier-grade customer experience demanded by mission-critical applications.

11. Appendix I - List of Specifications

Protocol	Description	Spec.	URL
H.323 including H.225, RAS, H.245, H.248, H.261, H.263		ITU specs	Can be downloaded from the ITU web site if you are a member of the ITU forum at http://www.itu.int/search/index.html just search for the name of the spec.
IPDC	Internet Protocol Device Control	draft-taylor-ipdc-00.txt	http://www.alternic.org/drafts/drafts-t-u/draft-taylor-ipdc-00.txt
MGCP/SGCP	Media Gateway Control Protocol	RFC 2705	http://www.ietf.org/rfc/rfc2705.txt?number=2705
Megaco	MEdia GAteway COntrol	RFC 3015	http://www.ietf.org/rfc/rfc3015.txt
SDP	Session Description Protocol	RFC 2327	http://www.ietf.org/rfc/rfc2327.txt?number=2327
SIP	Session Initiation Protocol	RFC 2543	http://www.ietf.org/rfc/rfc2543.txt?number=2543
RTP	Real Time Protocol	RFC 1889	http://www.ietf.org/rfc/rfc1889.txt?number=1889
RTCP	Real Time Control Protocol	RFC 1889	http://www.ietf.org/rfc/rfc1889.txt?number=1889
RSTP	Real Time Streaming Protocol	RFC 2326	http://www.ietf.org/rfc/rfc2326.txt?number=2326
RSVP	Resource ReSerVation Protocol	RFC 2205	http://www.ietf.org/rfc/rfc2205.txt?number=2205

12. Appendix II – Glossary

Acronym . . .	***Stands for . . .***
ASN.1	Abstract Syntax Notation 1 - An international standard for classifying data structures. There are 27 data types with tag values starting with 1; for example, Boolean (1), integer (2), and bit string (3). ASN.1 is widely used in ground and cellular telecommunications as well as aviation. ASN.1 uses additional rules to lay out the physical data, the primary set being the Basic Encoding Rules (BERs), which are often considered synonymous with ASN.1. Distinguished Encoding Rules (DER) are used for encrypted applications, and Canonical Encoding Rules (CER) is a DER derivative that is not widely used. Packed Encoding Rules (PER) result in the fewest number of bytes.
CAS	Channel Associated Signalling
CCS	Centi Call Seconds - A unit of measurement equal to 100 seconds of conversation. One hour = 36 CCS.
CLID	Calling Line IDentification
db	Decibel - The unit that measures loudness or strength of a signal. dBs are a relative measurement derived from an initial reference level and a final observed level. A whisper is about 20 dB, a normal conversation about 60 dB, a noisy factory 90 dB and loud thunder 110 dB. 120 dB is the threshold of pain.
dBm	Decibels referenced to 1mW
DTMF	Dual Tone Multi Frequency (DTMF, or "touch-tone") is a method used by the telephone system to communicate the keys pressed when dialing. Pressing a key on the phone’s keypad generates two simultaneous tones, one for the row and one for the column. These are decoded by the exchange to determine which key was pressed.
Frame	A fixed length block of data for transmission. It is comprised of a number of packets or blocks.
FXO	Foreign Exchange Office

GoS	Grade of Service - The probability of a call being blocked or delayed more than a specified interval, expressed as a decimal fraction. Grade of service may be applied to the busy hour or to some other specified period or set of traffic conditions. Grade of service may be viewed independently from the perspective of incoming versus outgoing calls, and is not necessarily equal in each direction.
H.245	The H.245 control channel is responsible for control messages governing operation of the H.323 terminal.
H.323	This standard defines a set of call control channel set up and CODEC Specifications for transmitting real time voice and video over networks that don't offer guaranteed service or high quality of service. H.323 is comprised of a number of standards.
IE	Information Element – a field within a signalling message.
IP	Internet protocol - The IP part of the TCP/IP protocol, which routes a message across networks. Each entry on the Internet has a unique IP address for purposes of routing.
IPDC	(**I**nternet **P**rotocol **D**evice **C**ontrol) A protocol for controlling media gateways developed by the Technical Advisory Committee, which was convened by Level 3 and others. It analyzes incoming data signals, in band control signals and tones and sets up and controls the appropriate gateways. It also handles management and reporting.
ISP	Internet Service Provider
ITSP	Internet Telephony Service Provider
IVR	(**I**nteractive **V**oice **R**esponse) An automated telephone answering system that responds with a voice menu and allows the user to make choices and enter information via the keypad. IVR systems are widely used in call centers as well as a replacement for human switchboard operators. The system may also integrate database access and fax response.
Jitter	The Jitter of an audio stream is defined as the variation (calculated as standard deviation) of the inter arrival times of the audio RTP packets. For each pair of successive RTP packets the difference in arrival time at

	the receiver is divided by the difference in the transmission time at the transmitter. These ratios are accumulated for the whole audio stream and the standard deviation of these values provides the jitter of the stream.
Kbps	Kilo bits per second.
KHz	KiloHertz
LIM	Line Interface Module
Mbps	Million bits per second
Megaco	(**ME**dia **GA**teway **CO**ntrol) An IP telephony protocol that is a combination of the MGCP and IPDC protocols. It is simpler than H.323
MGCP	Media Gateway Control Protocol. Used for controlling telephony gateways from external call control elements called media gateway controllers or call agents.
MOS	Mean Opinion Score – a method for measuring voice quality. Provides a means of evaluating the subjective performance of voice and/or video transmission equipment using procedures as set out in ITU-T P.800
Packet	A frame or block of data used for transmission over communication channels.
PAMS	Perceptual Analysis Measurement System
PDD	Post Dialing Delay - The time between punching in the last digit of a telephone number and receiving a ring or busy signal.
PGAD	Post Gateway Answer Delay
Port	A communications connection to the PC or to a device
QoS	Quality of Service - The ability to define a level of performance in a data communications system.
RTCP	Real time control protocol, used for control of RTP.
RTP	Real Time protocol, used by RSVP to establish communication between user and network.
RTP	Real time protocol, IETF specification for audio and video signal management.

Silence Suppression	Transmission where silence during the voice conversation is filled with other transmission such as data, video etc.
SIP	Session Initiation Protocol, an application layer control simple signaling protocol for VoIP implementations.
SSRC	A unique identifier of the audio stream, part of the RTP header.
UDP	User datagram protocol, the transport layer above IP.
VoD	Voice over Data
VoIP	Voice over Internet Protocol

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13. About the Author

Mr. Oded Agam is a frequent contributor of tutorial and industry-trend articles published by various prestigious trade journals including Telephony, Tele.com, Telecom Business, and Communications News.

His expertise covers a broad range of data- and telecommunications technologies including Voice over Data, ATM, Frame Relay, TCP/IP, Ethernet, WDM, and Wireless.

Mr. Agam's experience includes over ten years in computer networking which began as a Captain in the Israeli Navy. Following military service, Mr. Agam's prior positions as Engineering Manager and Director of Technical Services led to his current post as Vice President of RADCOM, a leading provider of network test and quality management solutions.

Mr. Agam holds a B.S. In Electrical Engineering from the Technion (IIT), an M.S. in Electrical Engineering from Tel Aviv University, and an Executive MBA from the Stern School of Business at NYU.