Filling the VoID in VoIP Testing

An Ameritec White Paper

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Filling the VoID in VoIP Testing

With the telecommunications industry's quick move to adopt Perceptual Speech Quality Measurement (PSQM) as the "standard" for voice quality testing of VoP networks and systems, a testing void was created due to a possible lack of understanding of PSQM's intent and limitations. Although it is potentially useful, it is evident that PSQM does not completely characterize those factors which contribute to poor voice quality. As indicated in the following excerpts, VoP developers and network operators need more data than PSQM can provide to characterize their systems.

The following excepts are taken from the article "***Measuring Voice Quality: Listening by the Numbers***" by ***David Willis***, in the ***May 1999*** issue of ***Network Computing***:

"**It's What's Inside That Counts** As flawed as they are, automated measurements such as PSQM fill a real need—but it's not in external test equipment. Instead, we need embedded, real-time performance measurement inside next-generation voice-over-packet products. Imagine an RMON standard for voice services, with internal probes generating PSQM-like scores between critical points in the network and issuing alerts when the quality falls below a service-level threshold. The alerts might trigger an automatic failover to the circuit-switched network."

"Yet PSQM isn't a complete solution for use in voice-over-packet networks. By the author's own admission" (KPN Research Holland), "P.861's approach doesn't account for several key factors that may critically affect perception –such as cell and packet loss, the clipping effect of bad voice-activity-detection mechanisms or the impact of bit errors. These problems are commonly found in voice-over-frame relay, ATM, and IP networks."

"Recognizing the failure of the ITU-T specs, Ameritec has taken a radically different approach with its Voice Over Packet application test suite on its existing (and highly useful) call generators. The software measures dropouts, round-trip delay, and signaling errors, in addition to its normal call-loading capabilities." "… it tends to produce quantifiable, reproducible, and comparable output."

Ameritec has realized and examined this need and has created a test suite that fills this void in VoP testing by providing delay, jitter, clipping, and dropout (packet loss) measurement capabilities.

Quantitative Analysis for Voice over Packet Testing – The Ameritec Solution

Introduction

Voice over Packet (VoP) network testing should be concerned with the three Q's:

- Voice **Q**uality
- Conversation **Q**uality
- Service **Q**uality

The first part of this paper covers Voice, Conversation, and Service Quality issues. Voice and Conversation quality covers latency, jitter, dropouts, and clipping. Service Quality includes a brief introduction to RSVP, DiffServ and MPLS protocols, of which, the latter two protocols will likely become the main protocols found in future Quality of Service (QoS) domains. The last portion of the paper covers network testing, problem identification, fault isolation, and network optimization techniques.

VoP network testing differs from standard packet network testing in that packet networks were not originally designed for time-sensitive real-time voice applications. Lower bit-rate codecs and non-linear compression techniques used in packet networks degrade the quality of the source signal. While codecs can be measured using artificial qualitative testing techniques such as the PSQM or Perceptual Analysis/Measurement System (PAMS), component delays and group delay variances that have a negative effect on the ebb and flow of a conversation can only be truly measured quantitatively. Lost packets, out-of-sequence packets, and signal clipping affect signal clarity and are more easily detected using quantitative measuring techniques.

Quantitative measurements make it easier to identify specific network components or network paths that need optimization or replacement because quantitative measurements are category specific, can be reproduced, and can be used for comparisons with previous measurements. Impairments that affect qualitative results, such as lost packets, jitter, or noise, cannot be isolated in mean scoring approaches. A low score simply indicates that something is bad. Was the problem caused by lost packets, quantizing noise, circuit noise, or jitter? Qualitative scoring, which ignores delays, may identify a problem but offers no solution. While qualitative and quantitative measurements are both artificial means of substituting for the human element, qualitative measurements do not isolate factors that may cause a low score.

Voice testing of packet networks should be concerned with both the clarity of the reproduced voice signal and the rhythm of the conversation, as well as service issues related to guaranteed performance and call setup and teardown. While clarity may be tested using qualitative methods, it does not test the rhythm and feel of the conversation and is not suited for fault isolation. The ability to quantize timed results to within 5 msec, to monitor the complete conversation, and to generate multiple calls with realistic traffic profiles, are the reasons Ameritec has chosen quantitative testing techniques to set the VoP network testing standards.

The next few paragraphs demonstrate the strengths and weaknesses of purely qualitative voice measurements for Voice over Packet Testing on complete networks. The PSQM algorithm is used as an example.

Voice and Conversational Quality

Qualitative Scoring using PSQM: Delays and Jitter and their effect on Scoring

PSQM, defined in ITU-T P.861, is an algorithm primarily designed for evaluating and optimizing codecs by comparing a preconditioned source signal with a coded output signal. The audio input is preconditioned per ITU-T P.830 to simulate how a human voice would sound speaking into a telephone handset. ITU-T P.50 defines the characteristics and phrases to be used in the source signal. The algorithm measures voice quality as it would be perceived at the human ear and is based on the Mean Objective Score (MOS). The MOS is a subjective measurement, defined in ITU-T P.800, that uses live participants to rate voice quality. MOS scoring is scaled from 1 to 5, with five being best; while PSQM scoring is scaled from 0 to 6.5, with 0 being best.

The PSQM algorithm is time alignment sensitive, that is, the algorithm compensates for latency by subtracting the known delay from the delayed samples. If the delay is not known, the delay can be raised by trial and error to find the delay that results in the best PSQM score.

The PSQM algorithm is best suited to measuring the effects of digital distortions caused by non-linear voice compressions, quantizing noise, transcoding errors, and random bursty errors such as packet loss. Today, however, PSQM testing is not limited to evaluating and optimizing codecs. PSQM is being used to check the one-way listening-only voice quality of complete networks that may even include PSTN tail circuits. Packet networks, however, were not designed for time-sensitive real-time voice applications. Network delays that degrade conversation quality are common, but due to the PSQM time alignment techniques, these delays do not affect scoring. Secondary effects of delays, however, may affect scoring. Examples of secondary effects are:

- delays that cause uncancelled Public Switched Telephone Network (PSTN) echo components to be noticeable, such as those caused by doubletalk or borderline return loss noise hovering around the normally acceptable -30 dBm level.
- congestion delays that manifest themselves as lost or out-of-sequence packets.

DELAYS

Latency and delay variations (jitter) are factored out of the PSQM algorithm and have a major effect on listening quality of actual human ears, thus limiting PSQM's ability to adequately test a network or complete system. Delays that cause a conversation to have gaps or sound choppy have little or no effect on PSQM scores. This phenomenon proved itself during Ameritec field testing of IP networks with long delays. In fact, the PSQM algorithm did its job; it compensated for the delays very well, giving the impression that the perceived voice quality was better than it really was.

One-way delays greater than 150 msec begin to cause the listener to subconsciously hesitate before responding; the greater the delay, the longer the hesitation. This makes the mood of the conversation sound cold. Response hesitations may become more pronounced if the delays cause an echo component to be introduced into the conversation. Different path delays through a network may cause individuals to interrupt one another (doubletalk). Echo cancellers often have trouble compensating for this doubletalk.

Delays greater than 400 msec may result in lost or out-of-sequence packets, which, like echo, affect both the sound reproduction and the conversation flow. Often in conversations, small return utterances that acknowledge a point are expected. If these utterances occur at the wrong time or are dropped, confusion results. Acknowledgement examples in English, are mm hmm, yes, OK, etc.; in Japanese, Hai; in Chinese, Dui; etc. These acknowledgements are short but necessary elements in many conversations.

Typical Delays

The following list describes the various entities that add delays in a VoP or VoIP network.

PSTN: PSTN delays are typically 30 msec and 'seldom' more than 150 msec (G.114). Delays from a wireless network, however, may be higher, and, if a satellite is in the loop, the one-way delays can add an additional 250 msec.

Internet Delays: Internet delays are totally unpredictable and are based on best-effort routing. Delays of 50 msec to 400 msec are typical. Delays greater than 400 msec with high dropout rates are not, however, unusual.

Packet Network: Delays in packet networks occur as a result of quantizing, buffering, queuing, switching, and routing. Propagation delays are very short and have little effect. The following is a list of typical areas where delays occur in a Packet Network listed either by function or device:

Packet Capture Delay: Packet capture delay occurs at the Access PSTN gateway and is the time that it takes to process the entire packet before forwarding it to the router. Processing a packet consists of: voice activation detection, silence suppression, look-ahead buffering, quantizing, encoding, and comfort noise generation. Packet capture delay varies with the vocoders used. The following table shows the typical delays associated with various vocoders:

Vocoder	Rate	Delay
G.711	64K	< 1 msec
G.721	32K	< 1 msec
G.723	6.4K or 5.3K	67-70 msec
G.728 CELP	16K	2 msec
G.728 ACELP	8K	20 msec
G.729	8K	20-25 msec

Switching, Router: Switching and router delays are related to the time it takes to check the header, check the routing table, and to route the packet to correct output port. Newer hardware-based routing is faster than older software-based routing.

Queuing Delays: Queuing delays are the delays resulting from packets waiting in router buffers while the routers perform the switching and routing functions on other packets. 'Best effort' routers usually route items to the destination using a lookup table with the shortest distance being the typical route. This creates congestion on some paths while others have little activity. In best effort networks, most of the congestion and delays take place at the core routers and are variable. In DiffServ or Multi-Protocol Label Switching (MPLS) QoS-based networks the delays occur at the edge routers, where the header

setup or labeling takes place. These delays are usually 20-30 msec in duration. Using these protocols, the individual core router delays are reduced to approximately 5 msec each.

In QoS networks, path management protocols such as MPLS should optimize throughput. By virtue of using multiple paths, however, these protocols can introduce jitter.

Jitter Buffers: Jitter buffers at the egress or exiting Gateways have delays that range from 0 (off) to 255 msec (maximum). Some jitter buffers are self optimizing while others can be optimized by the network administrator. In networks whose total delays are less than 100 msec, jitter buffers can be set as low as 20 to 40 msec. Networks with longer delays need larger buffers settings. Ameritec Call Generators, that measure delays and jitter to within 5 msec, can be used both to optimize buffer sizes and to verify the effectiveness of path management protocols.

Miscellaneous Delays: Equipment such as Standalone Firewalls and Proxy Servers can also add upwards of 150 msec of delay to the network because they must receive, decode, examine, validate, encode, and transmit every packet.

Packet Length: Optimizing packet lengths, like optimizing buffer size, involve tradeoffs. Generally, shorter packets create less processing delays at router buffers and egress gateways. A higher proportion of packet size, however, is dedicated to header information, thereby lowering the effective bandwidth of the network.

The following table summarizes typical delays in an uncongested network:

Device / Function	Typical Delay
Access Gateway (incoming line speed dependent)	0-8 msec
Compression / Coding (vocoder type dependent)	0-70 msec
Interprocess Delays (between coding & queuing)	10 msec
Best Effort Routers (no congestion)	20-25 msec each
DiffServ or MPLS Edge Routers	20-30 msec
DiffServ or MPLS Core Routers	5 msec each
Jitter Buffers at Egress Gateways	10 – 80 msec
Decompression / Decoding	< 10 msec
Egress Gateway (outgoing line speed dependent)	0-8 msec

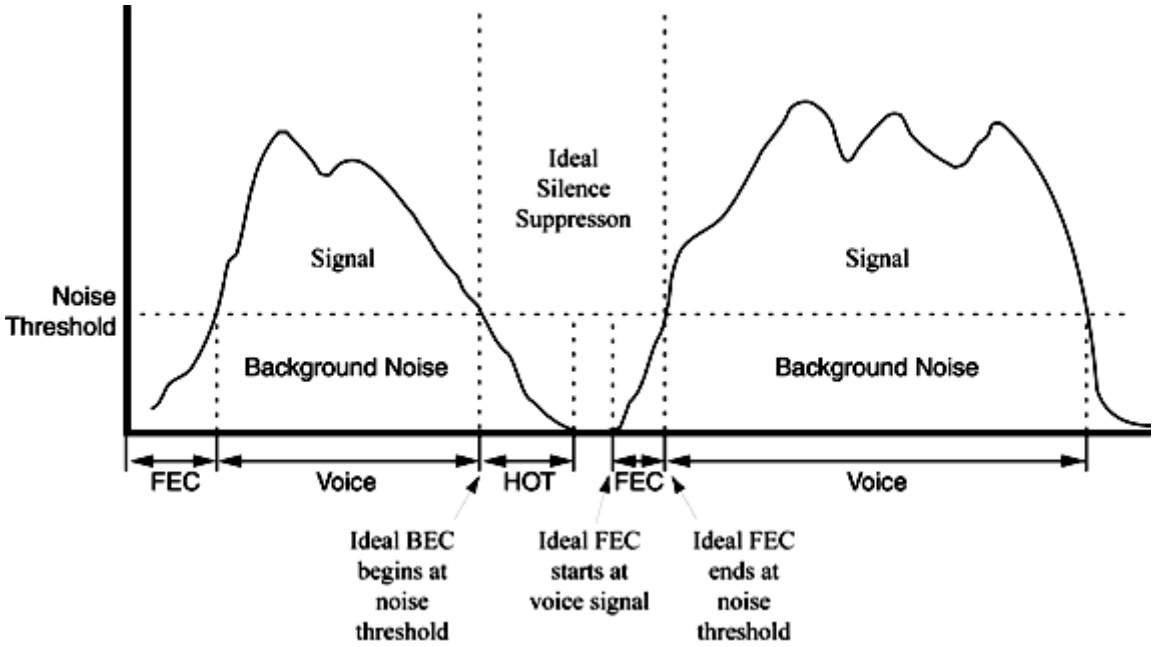
Relieving Network Congestion: Sound Suppression and Comfort Noise

Most voice conversations are half-duplex in nature. One person is usually listening, while the other is talking. To reduce network congestion, it's only necessary to send packets when someone is talking. Between listening and normal conversation lulls, voice traffic can be reduced in half by not sending packets during this quiet time. This is accomplished by Voice Activation Detectors (VAD) at network gateways. When voice is detected, the gateway sends packets; when it stops, the gateway stops sending packets (silence suppression).

When the packets are reassembled on the other end, the quiet time is replaced by a comfortable noise level generated by a Comfort Noise Generator (CNG) in the outgoing gateway. Comfort noise is generated to prevent the absence of sound from being perceived as a disconnect. Using sampling techniques, newer CNGs try to emulate the actual background noise in both level and frequency.

Voice Activation Detection, Comfort Noise, and Clipping

The time between the start of a voice signal and the detection of that signal by the Voice Activation Detector is known as Front-End-Clipping (FEC). The transition from the back end of a voice signal to the detection of silence is called HoldOver Time (HOT). If the VAD compensates for HOT too early, Back End Clipping (BEC) will occur.



Excessive FEC makes words difficult to understand while too little BEC or excessive HOT creates unnecessary packets, adding to congestion. If the true background noise, generated by too little BEC or excessive HOT, does not sound the same as the comfort noise, it can cause the conversation to sound choppy and unconnected. The comfort noise should match the actual background noise in level and frequency and should be dynamic. If the BEC occurs too soon, the transition from voice to comfort noise will be too abrupt to sound natural. Excessive FEC and BEC affect both the sound and conversation quality. Too little BEC affects the conversation quality. During Ameritec field testing, varying FEC and BEC on a Voice Activation Detector had no effect on the PSQM scores, thus PSQM alone did not detect this impairment.

The Ameritec VoP Call Generator Analysis tests allow you to program up to five FEC and five BEC categories for various time frames. During testing, it counts the number of times that FEC or BEC readings fall into each category.

Dropped Packets

Since voice packets are time sensitive, there is only a narrow window of time to resequence and align the packets into intelligible speech. Dropped or lost packets which may be caused by congestion, delays, or error retransmissions may cause holes in the conversation or cause a conversation to be garbled. Individual packet losses may have an annoying effect on the conversation while bursty losses can be quite intolerable. The conversation effect varies with the vocoders used. A single packet loss has a greater effect when low-bit rate vocoders are used than it would if a G.711 codec was used.

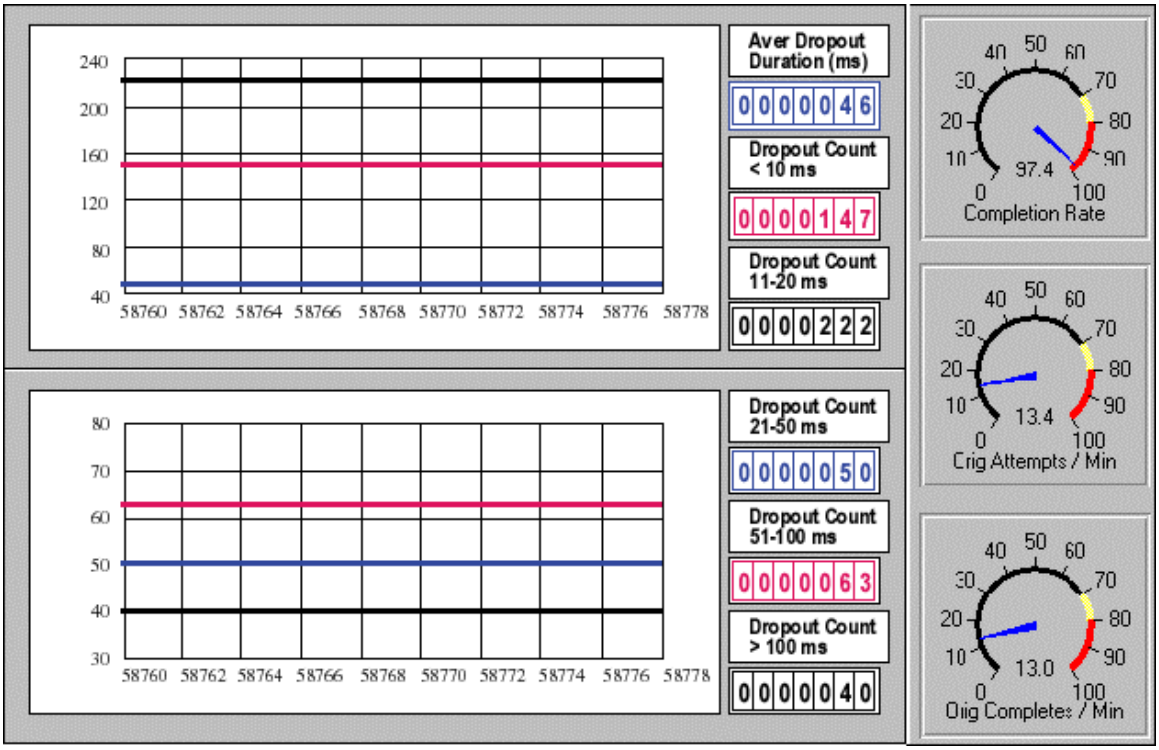
Note: Some G.711 codecs compensate for packet loss using a Packet Loss Compensation (PLC) algorithm. PLC samples the speech before and after the loss and tries to fill in the gap. Lower rate vocoders do not have this capability.

Packet size also has a direct effect on speech quality. Larger packets decrease network congestion because the overall header-to-content ratio is smaller. However, when a large packet is lost, it causes a more noticeable gap in the speech element. Larger packets also require more buffer space, adding to delays. The percentage of overhead associated with narrower packets, however, has an adverse effect, in that it can increase circuit congestion, thus offsetting the benefits of shorter delays and increased fidelity.

PSQM scoring does not weight packet loss sufficiently. Individual packet losses as high as 5% have little effect on the scoring. A 5% packet loss can sound choppy and, if bursty in nature, can be very irritating. Ideally, packet loss should be < 1%, with 3% acceptable. A modified algorithm, PSQM+, has been created that places a more realistic weighting on lost packets.

Better yet, Ameritec Call Generators measure voice dropouts as small as 5msec. This lost voice is a direct result of lost packets. Each voice dropout is sorted into one of five statistical bins in accordance with its time duration. The bins are programmable and can be set to record single and multiple packet losses that can be based on the length of a typical packet. Network buffers, in particular, Jitter Buffers, can be optimized using the FeatureCall™ Test Case Traffic Profiler to accurately profile network traffic and to monitor both the dropped packets and jitter. Optimization is accomplished by comparing the results before and after buffer adjustment. The Traffic Profiler can also be used to verify that buffers, which automatically adjust for traffic, are working properly by varying the profiles used.

Real -Time Test Results using FeatureCall Graphics and Statistics Tables



FeatureCall graphs can be programmed to monitor up to six categories for trends. Details of up to 32 columns of user selectable data can be viewed on the statistics tables. The statistics tables provide data on a per call basis by line / channel assignment.

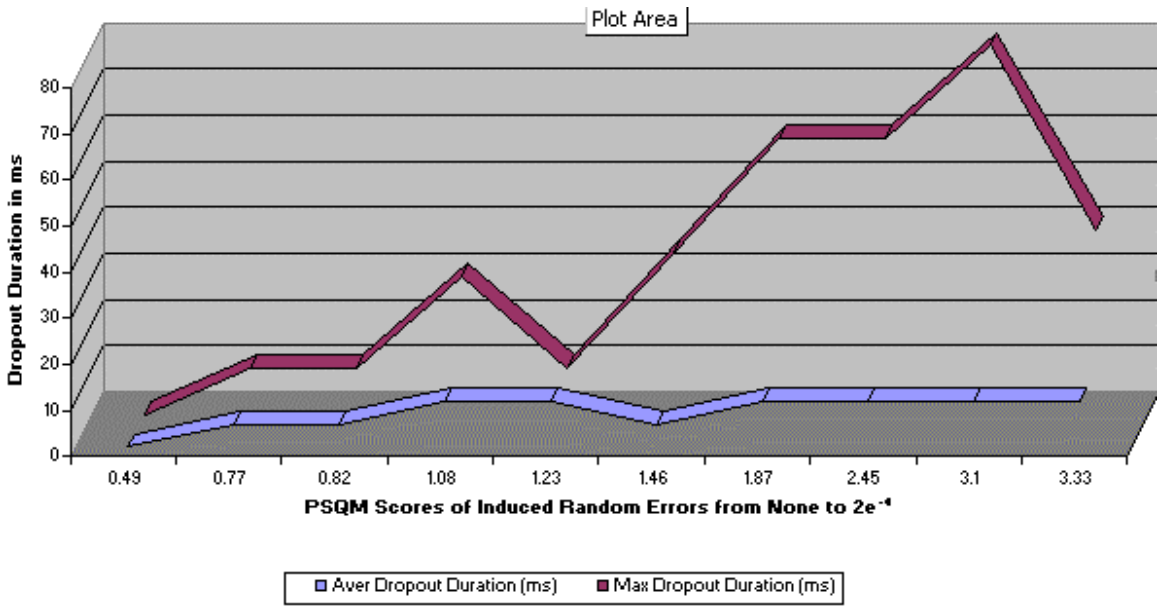
Unpaired Orig VOP II Drop Out Test

Ameritec
12 Sep 00 09:22:05am

CHAN	Aver Dropout Duration (ms)	Max Dropout Duration (ms)	Dropout Count < 10 msec	Dropout Count 11-20 msec	Dropout Count 21-50 msec	Dropout Count 51-100 msec	Dropout Count > 100 msec
A1	32.0	142	15	20	2	3	1
•				•			•
•				•			•
•				•			•
A13	63.0	224	5	10	7	8	4
A14	57.0	157	7	11	6	6	5
TOTL	46.0	224	147	222	050	63	40

Statistics are stored in the tabular format shown above and can be easily imported into spreadsheets, databases, and word processors for more sophisticated reporting and graphics, such as scatter graphs and bar-charts. Tabular formats also makes it easy to combine Call Generator results with other test results. For example, the following graph, generated in Microsoft Excel®, shows the effect of an increasing amount of random network errors on PSQM scores and dropouts.

® Microsoft Excel is a registered trademark of Microsoft Corporation.



Signal Level

Another factor that affects the perceived quality of speech is the level of loudness and fading. Many signal level problems can be traced to PSTN or wireless tail circuits, however, the multiplicity of carriers and equipment vendors that supply A/D converters, low bit-rate codecs, voice-activity-detectors, echo cancellers, and fax modems are impacting the voice levels. Newer equipment built to ITU-T G.169 recommendations should minimize this problem in the future.

Service Quality

Identifying Voice Packets

In open networks, it's very difficult to distinguish between voice and data packets. In fact, legacy frame relay networks have no service type headers and many IP or ATM network gateways do not identify the service type and will not prioritize voice packets. Protocols designed to alleviate this weakness are not widely implemented and may be incompatible with older gateways.

With more users and businesses going online everyday, coupled with the demand for more and better multimedia service, Internet traffic is doubling three times a year. This phenomenon increases the need for more bandwidth. Congestion in public networks, especially at peak periods, will be a problem for some time to come. In the meantime, the industry must learn to make the best use of what's available, by delivering high-priority voice traffic with fidelity, low latency, and few dropouts. Some routers can be programmed to identify voice/fax calls by looking for the gateways 'UDP port number' and give these packets higher priority. Most of the near term improvements in VoP quality will come in private networks using VoFR, VoIP, or VoATM because delays can be controlled and newer QoS protocols can be consistently applied. Bandwidth can also be increased incrementally as the need increases. Voice over the Internet, on the other hand, will have quality problems for a long time to come.

Quality of Service Protocols, An Introduction

Quality of Service (QoS) protocols basically are sets of rules that tell a network how to identify and route individual packets or groups of packets based on class of service requirements. Protocols exist at various OSI layers, often allowing more than one set of rules to operate simultaneously. For the purpose of this paper, we are going to ignore gateway control protocols and concentrate on QoS related protocols.

Quality of Service protocols must provide a measure of predictability and control above and beyond the 'best effort' methods used by the legacy Ethernet, legacy Frame Relay networks, and the Internet. There are four basic approaches to achieving the goals of service predictability and control.

- bandwidth allocation
- class prioritization
- path management
- flow control and queue management

QoS protocols use one or more of these techniques in varying degrees of complexity to achieve their goals. The following is a brief description of each technique:

Bandwidth Allocation: Allots specific amounts of bandwidth for different traffic types.

Class Prioritization: Assigns specific priorities to various service classes with drop precedences for dropping packets during times of congestion. Priorities are enforced by aggregate bandwidth allocation or on a 'packet-by-packet' basis.

Path Management: 'Best effort' routers usually route items to the destination using a lookup table with the shortest distance being the typical route regardless of traffic on that path. Path management techniques attempt to optimize path flow through the network so that all paths through the network core have a similar amount of traffic.

Flow Control and Queuing Management: A method of scaling back the network traffic so that the network doesn't become congested by limiting ingress traffic and buffering egress traffic as needed to maintain acceptable delays, jitter, and flow rate. Queuing Management is an often-used subset of Flow Control. Queue management occurs in core routers and consists of reviewing packet markers and establishing multiple priority queues. Two commonly accepted methods of queuing management are Weighted Random Early Detection (WRED) and Weighted Fair Queuing (WFQ).

Introduction to RSVP, DiffServ, and MPLS

Generally, QoS protocols such as Resource Reservation Protocol (RSVP), Multi-Protocol Label Switching (MPLS), and DiffServ (DS) will need to be incorporated, must be interoperable, and interact at domain boundaries. The next few paragraphs provide an overview of three QoS protocols: RSVP, MPLS, and DS. The former, RSVP, has many current installations but is extremely complex and has a lot of overhead. MPLS and DS, because of their simplicity and interoperability are or will be implemented in many of the newer routers. RSVP is also strictly a network protocol, whereas MPLS and DS are an intergral part of the data transfer.

RSVP: RSVP protocols, such as Integrated Services (IntServ) operate in similar fashion to ATM QoS, that is, they allocate bandwidth based on perceived requirements. RSVP does not carry data. RSVP is a

network protocol, designed to setup and teardown a call and works in parallel with data flow protocols such as UDP or TCP. RSVP operates by receiving service requests and then establishing and reserving a guaranteed path for that request. The request contains information such as minimum and maximum bandwidth, delay, and jitter requirements. RSVP sends out reservation requests to the first router in the proposed path. If the router can meet the requirements, it passes that request forward to the next router in the path until a virtual circuit is established or until a router returns an error message indicating the resources are not available. Once all the routers have accepted the request, the path is either guaranteed for the duration of the data transfer or call (highest priority) or the router will attempt to provide the service and maintain the circuit as long as the network is uncongested.

The main drawbacks to RSVP are its complexity and overhead. It has to allocate a guaranteed bandwidth based on a range and has long setup and teardown times. During the transfer, the reservations must be periodically updated. If the parameters cannot be met, the call or transfer cannot take place. Once a call / transfer is completed by the sender or receiver, RSVP must also teardown the call, so that the resources can be reallocated.

Differentiated Services (DiffServ or DS): DS relies on prioritizing, policing, and managing techniques. DS classifies traffic by marking packets with service level indicators using the first six bits of the IP Type of Service (ToS) header. These six bits are known as the DiffServ Code Points (DSCP). Routing decisions are made on a per packet basis rather than a per session basis. DS enforces service agreements between domains, manages traffic by dropping traffic when necessary and, when implemented, uses shaping polices to smooth out bursty transmissions. The major keys to the DS protocol model are the Class of Service (CoS) levels, the 'per hop' behavior method of routing, and queuing management mechanisms in the core routers.

The 'per hop' method of routing is on a packet-by-packet basis at each router, thus eliminating bandwidth allocation and simplifying the work of core routers. The DS edge routers handle the overhead associated with traffic conditioning. This allows core routers, where most of the congestion occurs, to simplify their processes based on the negotiated service level rating in the DSCP header and queue management.

DS has three basic 'per hop' service levels. The service levels in order of priority are Expedited Forwarding, Assured Service, and Best Effort. Contracted services use Expedited Forwarding and Assured Service. Expedited Forwarding guarantees bandwidth, while Assured Service allocates four classes of bandwidth. Each Assured Service bandwidth class has three drop precedences. Best effort uses whatever bandwidth remains after contracted services have used their allotment and are the first to be dropped.

DS uses both WRED and WFQ Queue management in the core routers. WRED randomly discards packets in a TCP network that is starting to become congested. This process causes the TCP to slow down the traffic rate. Since WRED is weighted, higher priority traffic is protected.

WFQ gives higher preference to low volume traffic. This prevents high-volume traffic from consuming large chunks of bandwidth and over shadowing the low-volume traffic. WFQ only operates on packets of equal priority, that is high-priority large bandwidth packets would still have priority over low-volume low-priority data.

MPLS: MPLS relies on path determination and flow manipulation. MPLS, which resides between Layer 2 and Layer 3, attaches a 20-bit label to each packet at the ingress boundaries of the network and removes it at the egress boundary. The 20-bit label includes a 3-bit CoS field. MPLS routes the labeled packets

across the least congested portions of the network rather than across the shortest distance. MPLS thereby optimizes the use of each network path, maximizing throughput and response times. MPLS operates by attaching routing labels to each packet at the access routers and forwarding the packet to the next router. The receiving router updates the routing label based on lookup tables and forwards the packet to the next router. This process continues until the packet reaches the egress router, where the labeling is stripped. MPLS also supports tunneling techniques. Tunneling allows the ingress router to define the complete path through the network similar to RSVP. MPLS labeling does not interfere with other protocols, such as DS, allowing networks to simultaneously employ multiple QoS techniques.

VoIP, VoFR, and VoATM Interoperability

Work is being performed by the Frame Relay Forum vendors to incorporate QoS techniques for Voice over Frame Relay as well as interoperability. Connection-oriented ATM already has QoS capabilities built in. ATM work is being done in the area of interoperability with IP QoS protocols. Until interoperability with networks as well as Application Programming Interfaces (APIs) are defined and incorporated across the board, QoS problems will continue when crossing network boundaries, causing high-priority packets to revert to 'best effort' behavior. Testing needs to occur both within a network and across various domains to assure VoP acceptance. For the near term, successful implementations may be limited to Private Virtual Networks where bandwidth, delays, and QoS mechanisms can be controlled.

Guaranteeing Service Levels

Since congestion will be a problem for some time, and the increase in voice calls continues, the need for better network management continues to grow. At first glance, it would seem that as the new protocols evolve and become an inherent part of networks, that QoS would improve and, with improvement, that there would be a decreased need for extensive testing.

The QoS evolution, however, will increase the need for testing. Very simply, a service provider guarantees and bills for a class of service, and thus must have a means of verifying that that quality of service can be delivered or customers can be lost. A good example of this phenomenon is the 'churn' or customer turnover experienced by the wireless industry. The question is, then, how can a service provider verify that its customers are getting the quality for which they are paying? How can they identify a problem before the customer sees it?

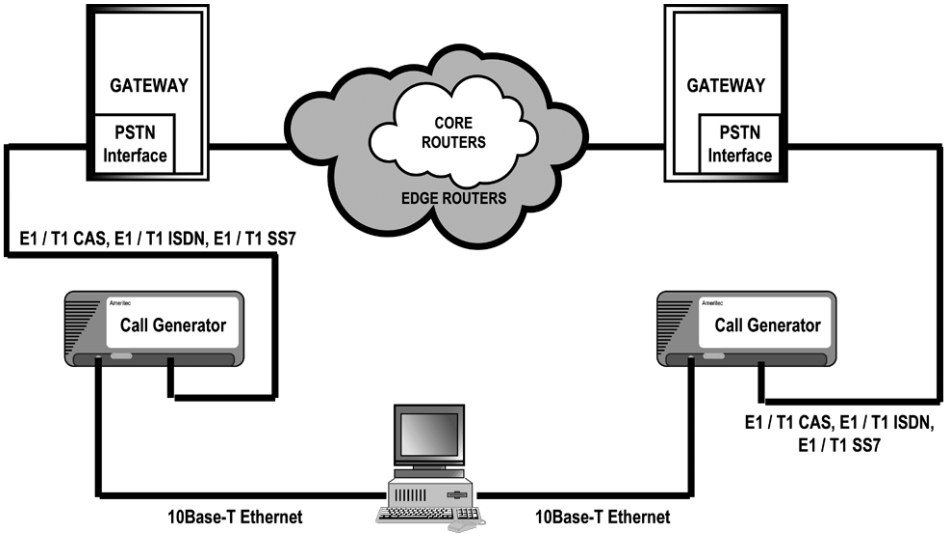
The easiest way to ascertain that high priority packets are getting through is to originate and terminate multiple VoP calls at various gateway locations throughout the network and record the results.

QoS Problem Identification and Isolation

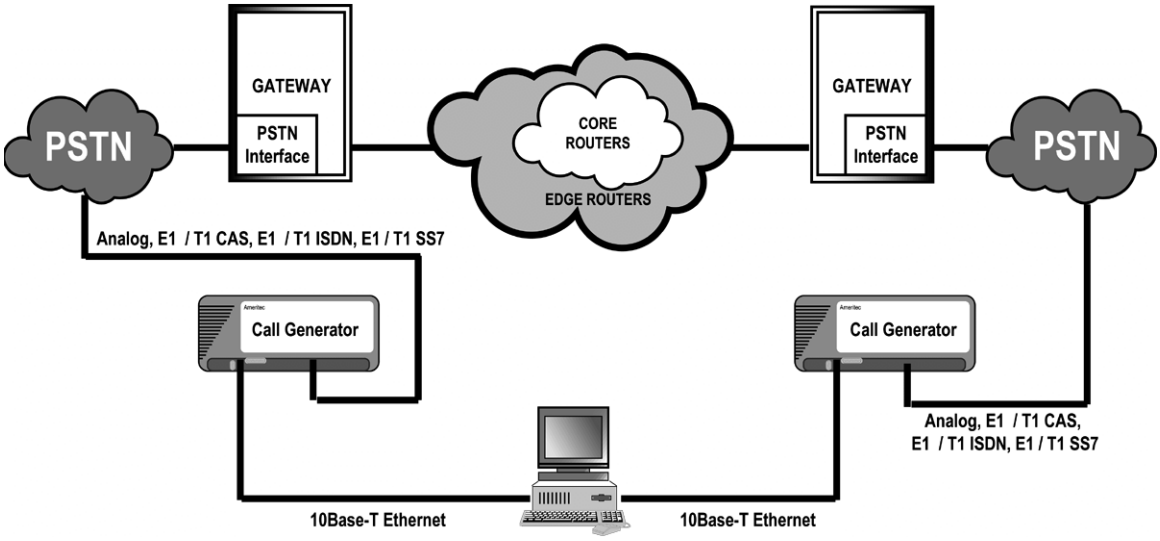
Using Ameritec's GPS synchronized bulk call generator with the VoP Test Suite is an excellent way to verify that high priority DS VoIP calls are getting through with a minimal amount of latency and dropouts. You can check the balancing of MPLS networks by comparing the simultaneous results of different calls being made on different network paths. If the average delays and dropouts vary greatly from call-to-call,

then the edge routers are not maximizing the throughput by balancing the network paths. Additionally, Analysis Tests that measure dropouts also measure jitter and front- and back-end clipping (FEC & BEC). If FEC or BEC is too short, extra packets are sent. If FEC or BEC is too long, or if unacceptable jitter occurs, the voice and conversational quality degrades.

Once network problems are identified, the path containing the problem(s) may be narrowed down by originating and terminating more calls in suspected areas, or by using a tracer program. Once identified, suspected components can be tested, online or offline, using a router tester, digital network tester, or protocol analyzer. The first figure that follows shows Call Generators connecting directly to network gateways. The second shows the Call Generators connecting to the gateways via the PSTN.



Call Generators as a Problem Identification Tool using a Direct Connect to Gateways

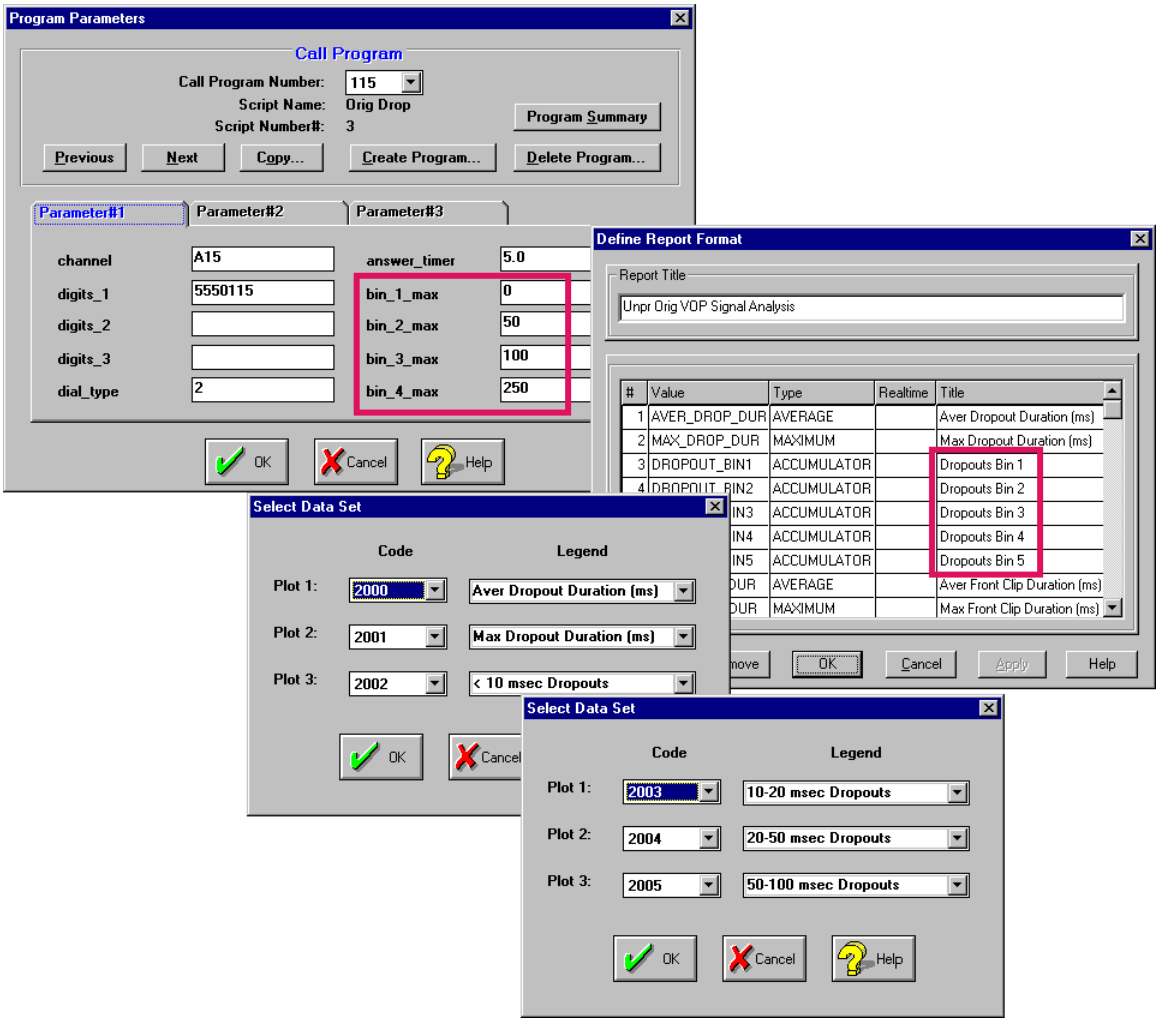


Call Generators as a Problem Identification Tool Connecting via the PSTN

Fault Isolation and Network Optimization

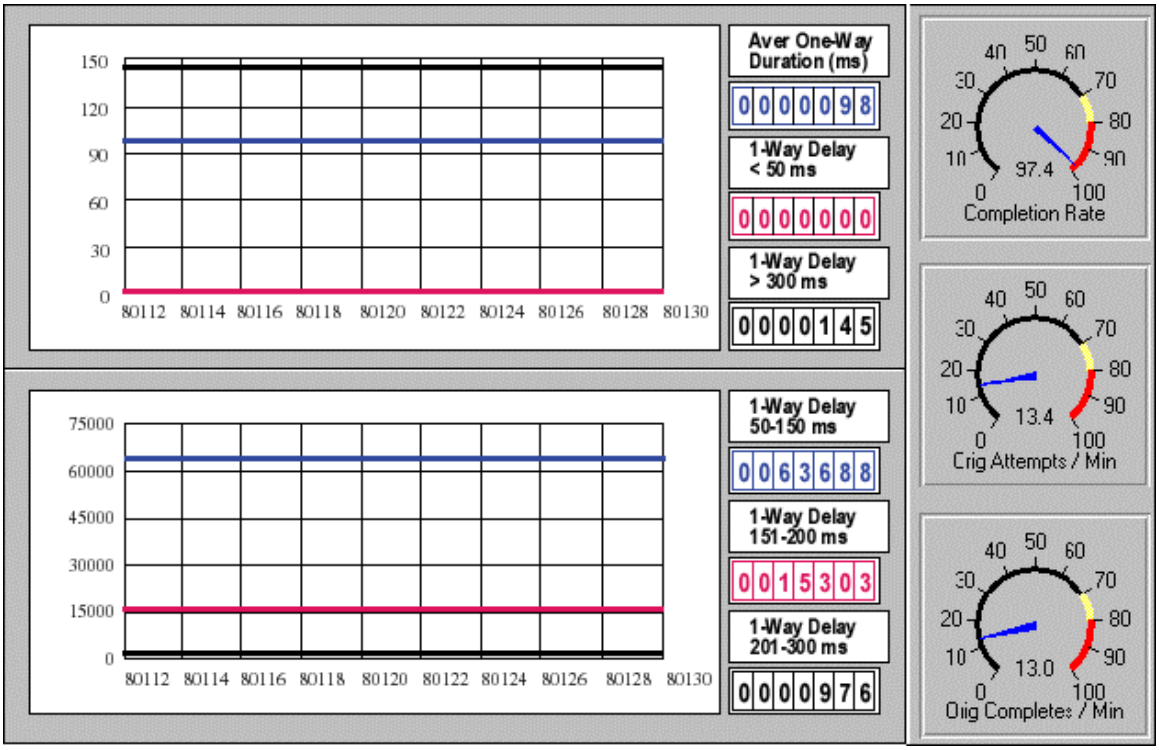
The following paragraphs list the Ameritec field tests and the quantized data collected by each test. Not only can categorical results be used to help determine the health of a network, they can also assist in isolating impaired equipment or network paths that are causing problems.

The following graphics show some of the user-friendly tools available for modifying the reporting capabilities to meet dynamic testing requirements. The bin parameters and titles shown in the examples below can be easily changed using the FeatureCall™ Call Program setup and the ScriptMate® Define Report Format windows. The Select Data Set windows are used to select up to six presentation categories for FeatureCall Graphs.



One-Way or Round-Trip Delay Tests

One-Way Delay Tests measure single direction (half-duplex) delays. The originate side of the call either sends or receives a pulse from the terminate side. This flexibility allows measurements to be made in either direction. Round-Trip delays measure the full duplex delay. It measures the time it takes a pulse to travel from the terminate side of the call to the originate side and then back to the terminate side.



The graphs can be a useful tool in determining the ranges where most of the delays are occurring. The above graph represents a network test at maximum call rates showing an average delay of 98 msec. Based on the above results, another set of call programs can be run with more narrowly defined range settings, for example, from 60-80msec, 81-100 msec, 101-150 msec, 151 to 220 msec, and >220 msec. This new setup, with tighter granularity, will provide a further breakdown of the delay distribution. Comparable results are objective if the average delay remains at a similar level. Once objective measurements have been made, then traffic profiling can be applied to check the results with varying call rates. Comparing results to realistic objectives can be a useful tool for optimizing networks and identifying potential and real problems. The following list highlights some of the causes of network delays:

Little or no Delay with Profiled Traffic: Network well optimized. Conversation should be smooth.

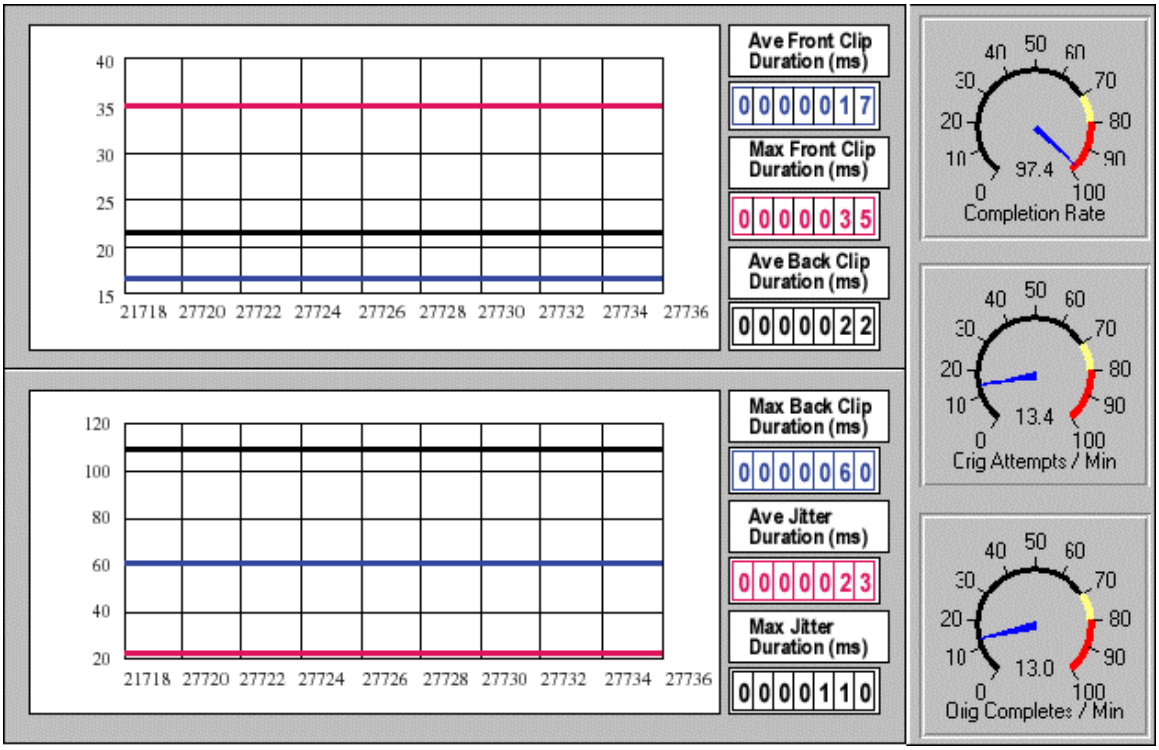
Excessive Delay with Profiled Traffic: Can be caused by one or more of the following:

- Queuing Congestion, gatekeeper not balancing routing.
- Traffic may have outgrown network design.
- Network is using software-based routing, which is inherently slower than hardware-based routing.
- Packet size assignments too short.
- Network buffer capacity inadequate or not optimized.

- One or more VADs too sensitive, causing background noise to be sent as packets.
- Background noise is so high that the VADs can't distinguish between signal and silence. Packets are simultaneously sent on both the speaker and listener's end. If the noise is high on all network calls then the problem is probably in the network. If the noise is intermittent, it could be PSTN generated. If the Network is clean, check incoming trunks. They could be receiving the noise from a common source.

VOP Analysis Test

Ameritec's Analysis tests measure Dropouts, Jitter, and Front- and Back-End clipping and tracks statistics for 23 VOP specific categories in addition to normal call setup and teardown categories. Unlike delay tests, the jitter and clipping tests are based on a square wave with a user selectable 25%, 50%, or 75% duty cycle as the reference for measuring variances. Dropout testing in the Analysis Test uses the same 25%, 50%, or 75% duty cycle while the standalone Drop Out Test uses a 100% duty cycle.



The sample graph above displays the results of the average and maximum clipping and jitter. If more definition is required, Front- or Back-End clipping, like dropout and delay tests, can be graphically monitored in five user-programmable categories. The details can also be viewed using the related statistics page.

If the average and maximum jitter duration is too high, a one-way delay test with narrowly defined ranges can be used to optimize jitter buffers. For example, one-way delay test parameters can be set for continuous testing in a single call with delay categories counting the number of hits. The categories with

the majority of the hits can then be further broken down into five separate ranges and the test rerun. When the majority of hits fall into one or two narrowly defined ranges, the buffers are optimized.

The following list highlights the problems that can be identified using the Analysis test as well as the possible causes:

Excessive FEC: VAD not sensitive enough or background noise too high.

Excessive BEC: Overcompensation for holdover time or background noise too high.

Excessive simultaneous BEC and FEC: Background noise too high.

Little or No FEC or BEC: VAD so sensitive that it interprets background noise as a signal with the following results:

- eliminates silence suppression
- sends unneeded packets
- can cause dropouts by creating or adding to network congestion

Little or No BEC, FEC OK:

- Extra packets on network; adds to congestion.
- Potential noticeable difference between actual background noise and the simulated background noise created by the Comfort Noise Generator

Excessive Jitter: Jitter is the result of sequential packets arriving with variable delays and can be caused by:

- jitter buffers too small.
- network path jitter buffers not equally optimized.
- components in one leg have longer or shorter delays than comparable components in other network legs.
- gatekeeper not balancing packet routing.
- routers not optimized or paths contain a mix of software and hardware-based routers.
- newly installed router learning optimal paths for each address.

Dropped Packet Test

The Dropped Packet Test measures the number of dropped packets that fall into specific categories or bins based on the length of the packet loss. The dropped packed test can be run within the Analysis Test or as a separate test. In the Analysis Test a one second square wave with a 50% duty cycle is the default test. In the standalone Dropped Packet Test a 100% duty cycle is used as a baseline. Category thresholds are set in milliseconds. The duration of the dropout test and category thresholds are programmable. Delay measurements should be performed if packet loss starts occurring in non-congested networks.

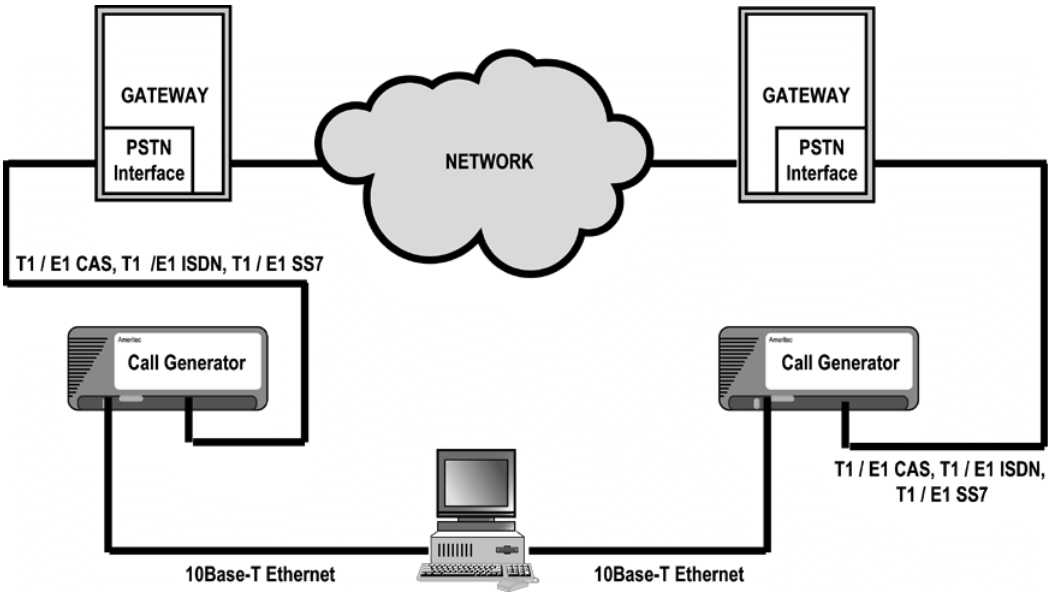
Dropped or Lost Packets: usually occur as a result of excessive network latency in one or more paths.

- Transmission Control Protocol (TCP): retransmits errored packets, causing extra delays that can result in lost or out-of-sequence packets.
- Universal Datagram Protocol (UDP): no retransmission. Errored packets are just dropped.

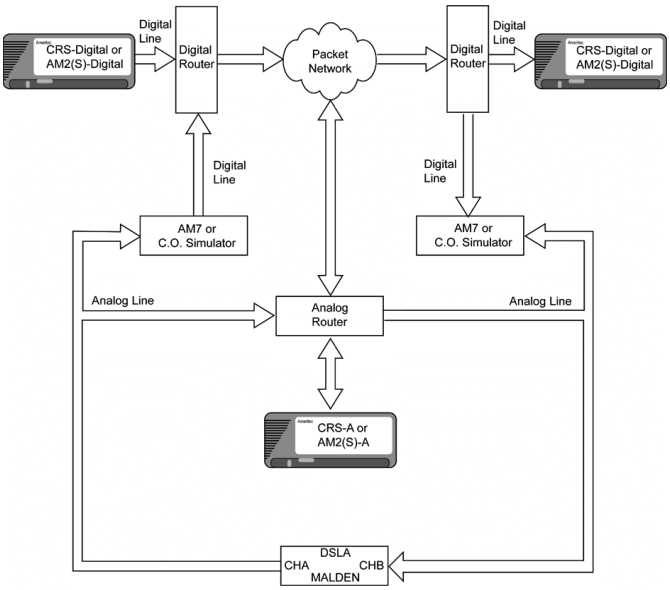
- Network buffers not optimized. If the buffers are too small, packets can be lost; if they are too large, longer delays occur, which can also lead to lost packets.
- The primary cause of lost packets are network delays caused by congestion. See the subtitle **Excessive Delays** under the ***One-Way*** or ***Round-Trip Delay Tests*** for details.

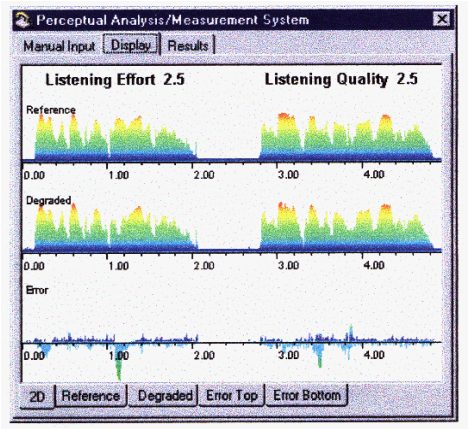
The Ameritec Solution

Ameritec Bulk Call Generators quantitative tests should be the initial tests performed by any user or developer of VoIP technology because they perform tests with the tightest tolerances, not one call at a time but using realistically profiled traffic. They can be used for new network qualification or as a problem identification tool in an existing network. The following figure shows Call Generators connected to network Gateways at the PSTN ports.



Bulk call testing can be supplemented with PSQM/PAM testers. The following diagram shows the Ameritec Bulk Call Generator operating in parallel with a Malden Digital Speech Level Analyzer (DSLA). This setup simultaneously provides qualitative and quantitative scores using the Perceptual Analysis Measurement System (PAMS). PAMS, a British Telecom standard, is recommended because it scores both the Listening Quality and Listening Effort. The Malden DSLA was selected because it provides a graphic comparison of the source signal and the degraded signal, as well as the error delta. The error delta can be viewed in two or three dimensions. The third dimension relates the error degradations to frequency. PAMS follows the 0 to 5 MOS score with an accuracy of one-half point. An experienced technician using the DSLA can better isolate problem types using the two- and three-dimensional graphics then could be accomplished with a scoring-only qualitative system. The following diagram shows a typical application using a DSLA in conjunction with a bulk call generator. The picture to the right is a DSLA PAMS graphic test result for both Listening Effort and Listening Quality measurements.





Conclusion

In conclusion, the primary factors for choosing the Ameritec VoP/VoIP Call Generators for VoIP equipment are:

Voice Quality testing using MOS type scoring does a poor job of evaluating the conversational quality of a call and does not assist in isolating problems; it merely provides a rating of the voice reproduction, while ignoring network latency and clipping. Latency, dropouts, jitter, and clipping can best be measured quantitatively, where tests can be performed repetitively and problem areas identified.

Since VoIP networks are now based on varying degrees of the 'best effort' model, VoIP manufacturers and service providers are concerned with QoS, as their customers are expecting these networks and equipment to perform at levels comparable to the PSTN. Delays and dropped packets become the real criteria for judging a network's credibility. Delay and dropout testing at simulated loads is one of the easiest methods to verify that the protocols are doing their job.

Whether testing best effort networks or QoS-based networks, Ameritec VoIP Call Generators can be used to analyze performance and identify network problems. Although voice quality testing can be supplemented with MOS type qualitative testing, the real 'quality' of a network can be better tested using quantitative analysis because quantitative testing is:

- ✓ category specific.
- ✓ reproducible.
- ✓ can be used for comparison and analysis.
- ✓ can assist in fault isolation.

Thank you for considering Ameritec.

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