HP J1981C

VQT Quality Tester

Agilent Technologies

Performing Pre-VoIP Network Assessments

Application Note 1402

Agilent Technologies

Issues with VoIP Network Performance

Voice is more than just an IP network application. It is a fundamental business and consumer service that has for a century been delivered on a daily basis with predictable quality. When VoIP technology is deployed for voice services on an enterprise or commercial network, users will both expect and need voice service quality that matches that of the Public Switched Telephone Network (PSTN).

Voice, being a real-time media application, requires special QoS considerations that are not needed by data. Being time-sensitive, voice media has a very low tolerance for delay, and an even lower tolerance for delay variance or jitter. Voice applications can tolerate data loss more than data applications, but because voice most often utilizes UDP rather than TCP for layer 4 transport, a lost packet means lost data; there are no re-transmissions. As a result, voice actually has a lower tolerance for packet loss.

It is well-known that the IP network performance parameters that impact voice are packet loss, delay, and jitter. But the type and degree of impact that these parameters have on voice quality is lesser known. This is because there are many other VoIP processes that impact voice, and these various processes, together with IP network performance, influence each other in complex ways to affect overall voice service quality.

The two key parameters of voice service quality most affected by IP network performance and VoIP processing are voice clarity (also known as speech quality) and voice delay. Knowing how an IP network will perform in terms of these important end user service parameters, and in terms of the underlying factors of packet loss and jitter, is very valuable prior to making critical decisions and investments regarding a VoIP deployment. This is the purpose of a pre-VoIP network assessment.

Why Perform a Pre-VoIP Network Assessment?

A key driver behind VoIP network deployments is the lesser capital costs than a traditional circuit-switched network deployment. Nonetheless, a VoIP deployment is still an expensive investment. In order to make this investment pay off, it is vital that the appropriate IP network architectures and configurations, and VoIP systems, be put in place to deliver appropriate levels of voice service quality. This means that up-front design and purchasing decisions will be critical in the success and payoff of a VoIP strategy.

Such decisions begin with assessing the IP network for VoIP performance, prior to VoIP deployment. This will provide valuable guidance in determining what needs to be done to the IP network, and what VoIP systems and architectures need to be deployed. This will enable an organization to put in place first the appropriate IP network architectures, configurations, and possibly QoS mechanisms, to guarantee voice service performance. It will also enable the organization to select the optimal VoIP systems and architectures needed to take advantage of the particular IP network in place in delivering voice services.

How to Perform a Pre-VoIP Network Assessment

There is only one way to ensure a pre-VoIP network assessment provides the guidance needed and is not a waste of time: perform a complete and comprehensive assessment to ensure that nothing is overlooked. It is not enough to simply measure IP packet loss, delay, and jitter. Knowing these performance parameters will not provide an adequate indication of how well a voice service will perform. One must also know how these parameters affect *voice clarity* and *delay*.

Voice clarity depends on many factors in addition to packet loss and jitter, and the various factors influence one another. A certain degree of packet loss can have varying affects on clarity. It would be unwise, for example, to invest in excessive QoS technology to overcome a perceived packet loss problem, if the packet loss does not affect voice quality.

Voice delay includes much more than just IP packet transmission delay. Much delay is added by VoIP gateway processes such as codecs and jitter buffers. Packet delay will simply add to this. Furthermore, high packet jitter can add to delay by increasing a gateway's jitter buffer size requirements. ***Thus, it is vital to know what the end user delay experience will be, and this can only be accomplished with active voice delay measurements.***

Therefore, an adequate pre-VoIP network assessment must benchmark a network's performance in terms of voice clarity and delay, as well as packet loss and jitter.

The Agilent Voice Quality Tester (VQT) and IP Telephony Analyzer (IPTA) provide the means for performing a complete and comprehensive pre-VoIP network assessment. The VQT and IPTA can be used on an IP network prior to any VoIP systems or capabilities being deployed. No VoIP phones, gateways, gatekeepers, soft switches, or call agents are required.

The Agilent Advisor can provide the VQT and IPTA in a single product. An Advisor with a VQT SIP/H.323 license and with a IPTA license, will perform VQT testing over its 10/100 Ethernet NIC, and IPTA testing over its data network interface module (e.g., Fast Ethernet). With three or more Advisors, a complete assessment can be performed. Alternatively, the VQT Portable Analyzer (J1981B) with a SIP/H.323 license can be used for VQT testing, and the Agilent Network Analyzer (J6800A) can be used for IPTA testing.

The assessment performed by the VQT and IPTA generates actual VoIP traffic, and measures actual end user parameters in addition to IP network performance parameters. ***The value of this technique is that VoIP network performance is actually measured, not estimated or predicted, and with actual VoIP traffic, not with "packet emulation".*** Also, actual end user voice delay is measured, rather than just IP packet delay. This provides a complete and comprehensive assessment of VoIP network performance, ensuring that critical end user parameters are known prior to designing and deploying VoIP services.

In addition, the Agilent IP Telephony Reporter (J5422A) provides graphs of time-correlated VQT and IPTA measurement results. Voice clarity and delay measurements, trended over time and correlated with packet loss and jitter measurements, provide valuable insight into the performance of the IP network in delivering VoIP services.

Core Techniques

Most basic pre-VoIP assessments can be performed using some core techniques, described as follows:

Process Overview

Measure voice clarity and delay between each site at which VoIP will be deployed. If trended measurement results fall within the thresholds of acceptability, refer to packet loss and jitter measurement results for acceptable baseline values. These baseline values should be maintained for acceptable service quality when VoIP services have been deployed. However, when actual VoIP services have been deployed, clarity and delay testing should be repeated to certify the deployment.

If measurement results indicate potential quality problems, refer to packet loss and jitter measurement results for indications of possible causes. The IPTA can measure loss and jitter at virtually any point in the network, thereby locating problem sources.

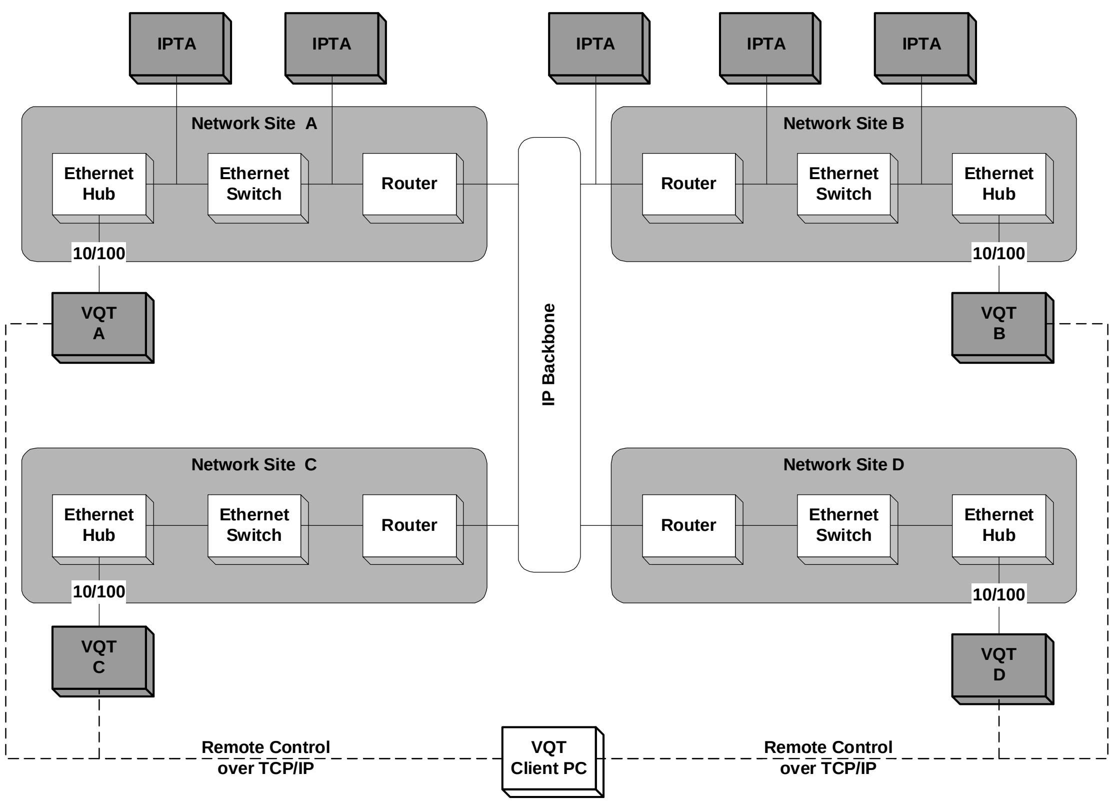

Figure 1

Connect the VQTs

Begin the assessment by connecting two or more VQTs to the IP network, as shown in **Figure 1**. The 10/100 Ethernet interface of each VQT, used for placing SIP or H.323 calls and measuring voice quality, should be connected to the IP network at VoIP access points. These will typically be LAN hubs or switches at which VoIP media gateways or IP phones would be connected.

Multiple VQTs can be used and remotely controlled via a single PC. This enables convenient synchronized testing between VQTs at different network sites. Alternatively, each VQT (Portable Analyzer or Advisor) can be locally controlled with the on-board keyboard, monitor, and mouse.

Each VQT has at least one 10/100 Ethernet port, and may have two. These are used both for remotely controlling the VQT (if that option is used) and for placing VoIP calls. The VQT's VoIP interface may use the same or different 10/100 Ethernet port as the remote control interface. In **Figure 1**, it is the VoIP interface that is shown connected to the Ethernet hubs at each site.

It will be necessary to first configure the IP network properties for each VQT VoIP interface, to assign a host name and to use assigned IP addresses, or to use DHCP. The assigned host name or IP address for each VQT will be used as the calling address for VoIP calls, in place of a telephone number.

Establish a VoIP Call Between Two VQTs

Configure the Port Setup of each VQT for SIP; do not specify the use of a proxy server. Establish a VoIP call between two VQTs; for example, from VQT A to VQT B. Use the VQT's assigned host name or IP address as the address to call. Each VQT acts as a SIP user agent. No external proxy server or gateway is needed.

Measure Voice Clarity

Once the call is established, run a series of clarity measurements using PESQ, PSQM+, or PAMS. The VQT's clarity trending capabilities should be used to capture the performance of the network over time and with different levels of network usage.

Assess the performance of the network in delivering voice clarity using both non-compression and compression voice encoding. This will help to determine if bandwidth-saving voice compression technology can be utilized in an actual VoIP deployment, while staying within service quality objectives. Perform clarity measurements using both G.711 (non-compression) and G.729 (compression) codecs on the VQT. Ideally, results using both codecs will indicate acceptable service quality. But if results using G.729 are significantly worse than results using G.711, then deployment plans should include the use of G.711 as a default codec, or should include additional IP QoS technology for VoIP.

If clarity trending results indicate poor voice clarity at a certain time of day, or at all times of day, then perform individual clarity measurements for further analysis. Use the VQT's clarity measurement graphs to determine what is contributing to poor clarity. The error surface graphs in PESQ and PAMS, and the frame disturbance graphs in PESQ and PSQM+, can provide indications of codec distortion, packet loss, uncompensated delay jitter, time-clipping, and level-clipping.

Measure VoIP Packet Loss and Jitter

For more precise determination of causes of poor voice clarity, or to baseline the performance parameters of the IP network under conditions of acceptable service quality, use the Agilent IP Telephony Analyzer (IPTA) to measure IP network performance for VoIP on various network links.

Referring to **Figure 1**, connect the IPTA to different network links between two VQTs engaged in a VoIP call. The IPTA is a software application on the Agilent Advisor and Network Analyzer. The Advisor and Network Analyzer support many different data network interfaces, including Fast Ethernet, Gigabit Ethernet, ATM, and frame relay; so that the IPTA can measure VoIP network performance at virtually any link. At the least, one IPTA can be used to measure VoIP network performance at each link, one link at a time. Preferably, multiple IPTAs can be used to measure VoIP network performance at each link simultaneously during a VQT call.

With one or more IPTAs connected and running, establish a VoIP call between VQT A and VQT B, and perform a clarity measurement. Each IPTA will measure and report the VoIP packet loss and jitter in each direction, on a packet-by-packet basis, throughout the call. In addition, the decoded voice from the VQT call can be played out in real-time on the IPTA, for subjective listening and quality assessment at each network link.

If and when the VQTs report poor clarity measurement results, use the IPTA results to determine if packet loss or jitter were high, and if so, on which links. The IPTA can also be used to decode the RTP packets, including packets before and after lost packets and packets experiencing high jitter, to determine possible causes. The IPTA on an Agilent Advisor or Network Analyzer provides OSI layers 1-7 protocol decodes and network analysis, so that in addition to RTP packet decodes, one can measure the network utilization of each link during an impairment, determine what types of traffic are on a link, and thoroughly troubleshoot network impairments.

If packet loss is low on a VQT destination link, but the VQT clarity measurements indicate packet loss is a problem, it could be that late packets are being dropped by the VQT jitter buffer. This may indicate a packet jitter problem. Determine if the IPTA's jitter measurement results are higher than the VQT's jitter buffer setting. If so, increase the VQT's jitter buffer and repeat the clarity measurements. If clarity improves, than VoIP packet jitter is most likely the problem.

Measure Voice Delay

Round-trip voice delay measurements are valuable because they more accurately characterize a user's experience with regard to delay. A telephone user perceives round-trip voice delay, not one-way voice delay. That is, the delay for a speaker's voice to reach a listener's ear is perceptible to *neither* speaker nor listener. However, the delay between a speaker saying something, and then hearing the other person's response, *is* perceptible. Round-trip voice delay is the primary cause of disruptive conversational pauses. A standard objective for round-trip voice delay is 300 milliseconds.

Round-trip voice delay is measured between two VQTs by applying a port loopback on one VQT, and measuring round-trip delay from the other VQT. The port loopback on the VQT's 10/100 Ethernet interface is applied directly within the RTP/UDP/IP stack, so the added delay of this loopback is less than 10 milliseconds.

Refer to the Agilent Application Note 5988-6164EN "Troubleshooting Voice Delay on IP Telephony Networks" for more techniques on measuring voice delay with the VQT .

With a VoIP call already established between two VQTs, run a series of voice delay measurements. The VQT's delay trending capabilities (using *"n"* or "continuous" repetitions in the delay measurement) should be used to capture the performance of the network over time and with different levels of network usage.

Assess the performance of the network in terms of voice delay using both non-compression and compression voice encoding. This will help to determine if bandwidth-saving voice compression technology can be utilized in an actual VoIP deployment without exceeding a delay budget. Perform delay measurements using both G.711 (non-compression) and G.729 (compression) codecs on the VQT. G.729 will add more delay, and it is important to know if it can be used prior to a VoIP deployment. If round-trip delay using G.729 exceeds 300 milliseconds, then deployment plans should include the use of G.711 as a default codec, or should include additional IP QoS technology for VoIP.

If the measured delay is too high (e.g., greater than 300 milliseconds), adjust the VQT jitter buffer to a lower setting and repeat delay measurements and perform clarity measurements. A lower setting on jitter buffer is not necessarily the solution to high delay problems, but this technique will illustrate the trade-offs between delay and clarity that occur with different jitter buffer sizes. A smaller jitter buffer will result in lower delay, but could increase the number of dropped packets which would result in poorer clarity. A larger jitter buffer will decrease the number of dropped packets, but at a cost of added delay. By performing delay and clarity measurements with various jitter buffer sizes on the VQT, one can assess these trade-offs and determine the optimal jitter buffer size. Then one can determine if delay is still a problem that needs addressed elsewhere.

If VQT delay measurement results are high, use the IPTA as shown in **Figure 1** to measure VoIP packet jitter and loss at various links. High voice delay may be the result of IP network congestion, which could result in increased packet jitter or loss. While it is possible for IP packet transmissions to experience high latency while maintaining minimal jitter, it is more likely that as packet latency increases, so also does jitter. If high packet latency is a result of network congestion, then high packet loss may also exist.

Measuring delay with the VQT as shown in **Figure 1** will provide an assessment of VoIP end user delay, but it may also be helpful to know where in the network delay may be a problem. The VQT can be used to segment an end-to-end call path to isolate delay measurements to individual segments. This technique is illustrated in **Figure 2**.

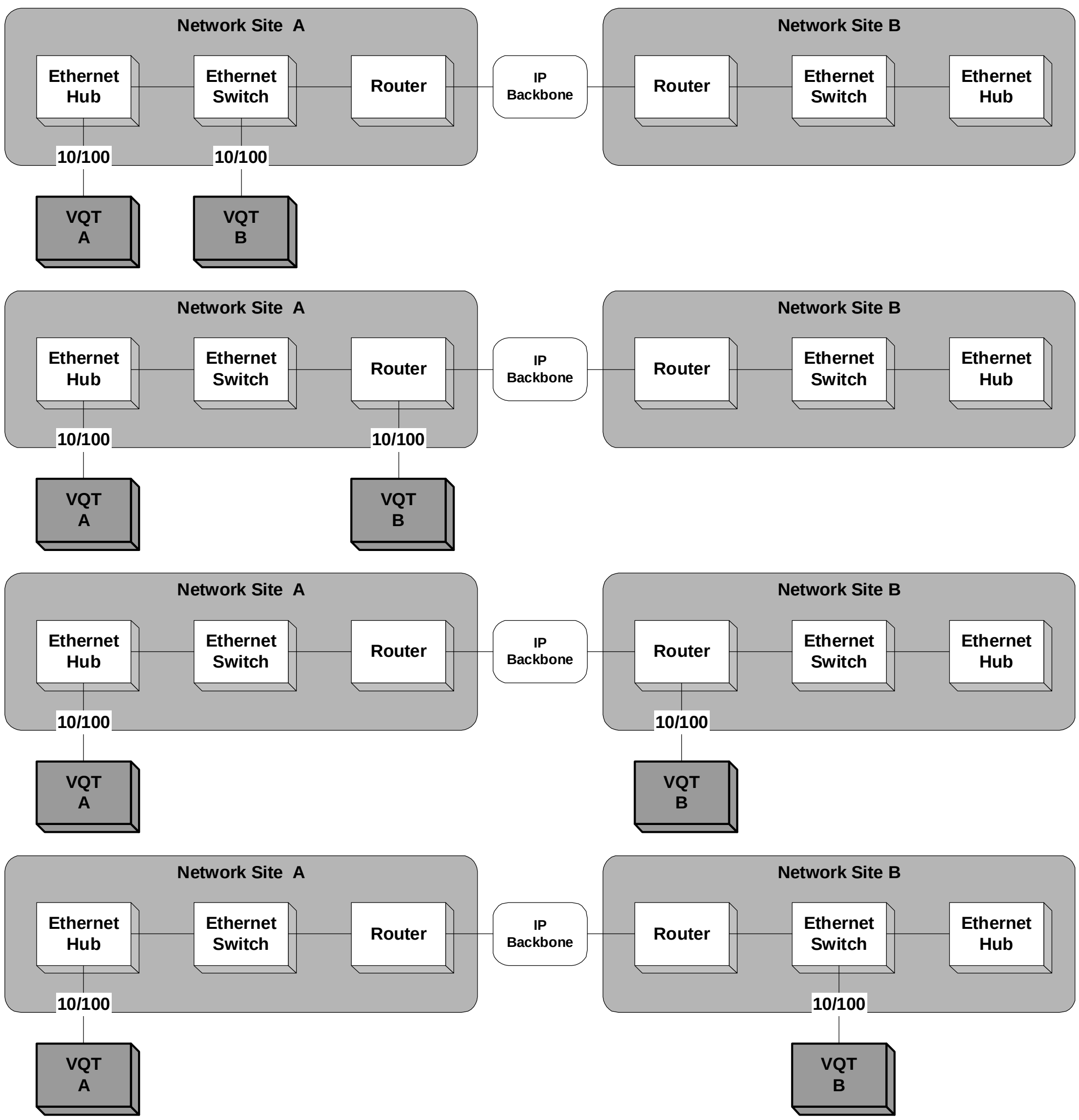

Figure 2

By measuring delay between two VQTs at different access points within the network, as shown in **Figure 2**, one can determine if there is a problem spot in the network. The VQT delay measurement result will not represent only the IP network delay, but will include VoIP process delays (e.g., codecs and jitter buffers) that are embedded in each VQT. The benefit of this measurement technique is that a consistent and known delay value within the VQT will be applied to each measurement shown in **Figure 2**, and therefore the *differences* in measurement results can be attributed to the differences in network segments.

Baseline Performance with Trending

To baseline the network's performance over time, and to determine any variance in performance due to network usage, perform clarity and delay trending measurements with the VQTs. While testing with the VQTs, continuously monitor the IP network performance in terms of VoIP packet loss and jitter with the IPTAs. Then correlate the VQT's clarity and delay measurements with the IPTA's packet loss and jitter measurements over time.

One useful tool for this measurement correlation is the Agilent IP Telephony Reporter. Using measurement results from the VQT and the IPTA, the IP Telephony Reporter time-correlates measurements results and provides graphs that show how IP network performance impacts voice quality over time. Such data provides valuable insight in determining what parameters in the IP network will impact voice service quality, where in the network the problems occur, and at what time of day the problems occur.

Alternative Techniques

Many other techniques may be applied for pe-VoIP assessment under different circumstances or for additional objectives. These alternative techniques are described as follows:

Assessments with VoIP Equipment

The preceding core techniques describe using the VQT 10/100 interface for voice quality testing, in which the VQT itself provides the SIP or H.323 call signaling and VoIP gateway processing. This is beneficial for pre-VoIP assessments because no VoIP equipment on the network is needed.

However, one may need to assess a network using a particular VoIP gateway. The VQT can also support this requirement. If a VoIP media gateway (along with means for completing VoIP calls) is already available for the assessment, the VQT can test voice services over the media gateways. The VQT can generate calls on analog FXO, analog E&M, T1, E1, and ISDN PRI telephony interfaces. The same techniques described previously for testing clarity and delay can be used.

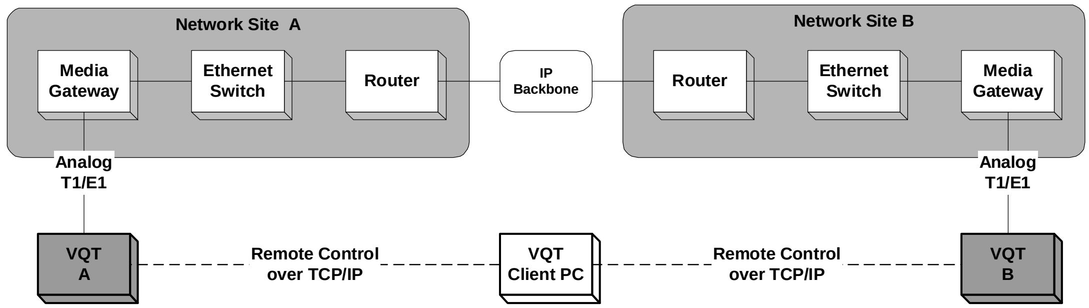

Figure 3

Figure 3 illustrates the use of VQTs to assess a network with media gateways. Connect the VQT's PSTN interfaces (analog, T1, E1) directly to the media gateways, or via intermediary equipment such as circuit switches or cross-connects. Then establish calls between the VQTs and perform the same measurement techniques described previously.

Delay measurements using the technique in **Figure 3** will have a key difference from those using the technique in **Figure 1**. In **Figure 3**, there will be no significant internal delay added by the VQT. The VoIP codec and jitter buffer processing takes place in the media gateways. However, the delay measurement results should be similar, because the processes that are included in the delay measurement are the same; the difference is whether these processes are located in the VQT or in the media gateways.

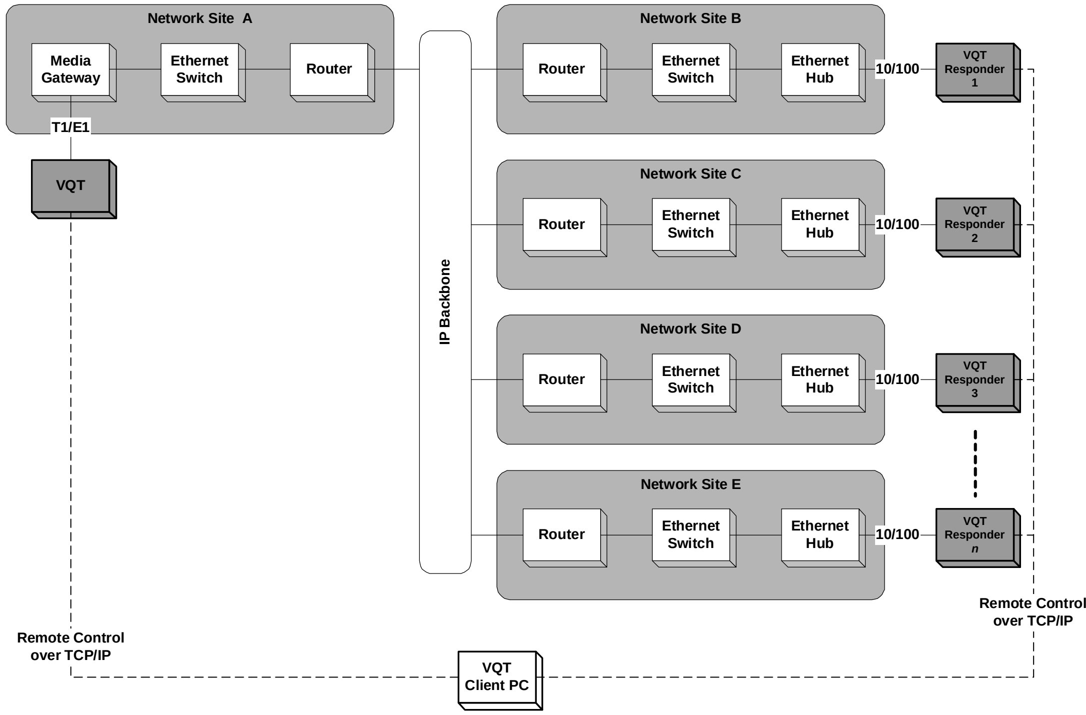

Figure 4

Assess Performance Against Background Traffic

It is valuable to assess the performance of a VoIP network against a background of actual traffic. **Figure 4** illustrates using the VQT T1/E1 interface and VQT Responders to accomplish this. VQT Portable Analyzers or Advisors can be used in place of VQT Responders.

Connect a VQT T1/E1 interface directly to a media gateway, or via intermediary equipment such as a circuit switch or cross-connect. Connect a multitude of VQT Responders to equipment at other network access points. Either the VQT Responder 10/100 interface or analog FXO interface can be used at each access point, depending on the network equipment available for connection.

Next, from the VQT T1/E1 interface, generate simultaneous calls to each VQT Responder or other VQT. Up to 48 calls from a VQT T1 interface, or 60 calls from a VQT E1 interface, may be generated.

Alternatively, standard telephones or any other end user telephony terminal may be used in place of VQT Responders or VQTs, to answer calls from the VQT T1/E1 interface. Only one other VQT or VQT Responder is needed to actually measure voice quality, while the other terminations serve as background traffic. Using VQT Responders for terminations enables voice quality measurements to be made to each access point at which a VQT Responder is connected. A combination of VQT Responders, VQTs, and telephones may serve at terminations for the VQT T1/E1 calls.

When the VQT T1/E1 calls have been established, load each active call with actual voice traffic using the traffic load feature in the VQT T1/E1 Call Control task. Then proceed to perform clarity and delay measurements on selected calls. Use the same techniques described previously (measurement trending, graphical analysis, changing codecs, measuring packet loss and jitter, etc.) to assess network performance against a background of actual VoIP traffic.

Assessments with IP Phones

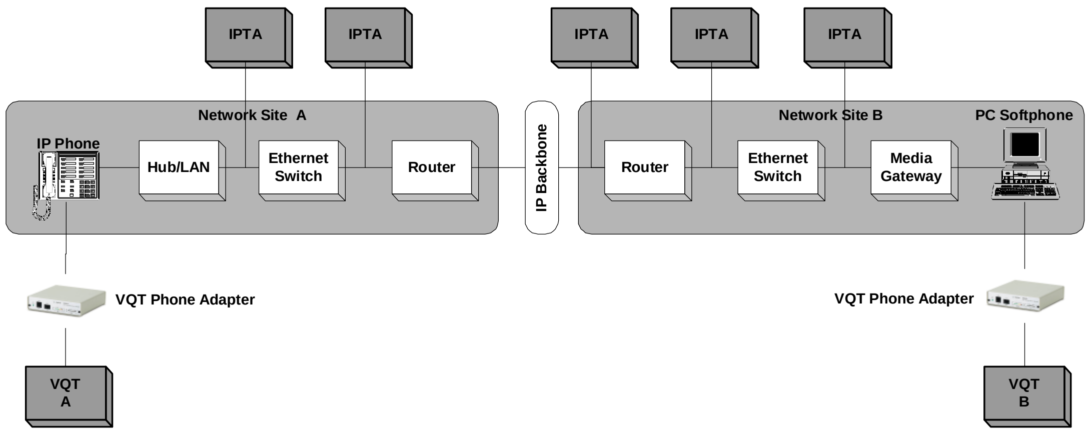

Figure 5

If IP phones are being deployed and are available (along with means for completing VoIP calls) for the assessment, the VQT can assess network performance using IP phones or PC softphones. As illustrated in **Figure 5**, the VQT Phone Adapter is used to connect a VQT analog port to the handset interface of an IP phone, or to the soundcard of a PC. Testing can be performed between two IP phones or PCs, or between an IP phone and a VQT 10/100 interface, or between an IP phone and a VQT analog/T1/E1 interface via a media gateway. The VQT Phone Adapter is simply another VQT telephony interface. Any combination of VQT telephony interface testing can be performed.

To test between an IP phone and a PC softphone, as illustrated in **Figure 5**, establish a phone call between the two terminals (i.e., the IP phone and the PC softphone). With a VQT connected through a VQT Phone Adapter to each terminal, perform clarity and delay measurements as described in the "*VQT Phone Adapter Users Guide*". Perform network assessments using the same techniques as previously described.

Testing for Echo

Speaker echo in a telephony network is generated on analog two-wire/four-wire hybrid junctions. Echo impacts *conversational quality*, and the degree of impact is proportional to the echo signal's level and delay. The greater the echo signal level (or lack of echo return loss), and the greater the echo signal delay, the greater the impact on conversational quality.

On the PSTN, echo cancelers are deployed to detect and minimize echo.

So why is echo an issue on a VoIP network?

A call originating on a VoIP network may terminate to a PSTN two-wire analog line, known as a "tail circuit". The two-wire/four-wire hybrid junction of this tail circuit will generate an echo signal.

But shouldn't echo cancelers on the PSTN take care of this echo?

Not always. Echo cancelers are expensive resources. PSTN carriers typically deploy them only where they are needed. Because echo is only perceivable if it is delayed 20 milliseconds or more, a PSTN carrier may only deploy echo cancelers on routes in which a transmission delay can exceed 20 milliseconds. Typically, this only occurs on long distance routes. Thus, many local PSTN networks do not include echo cancelers.

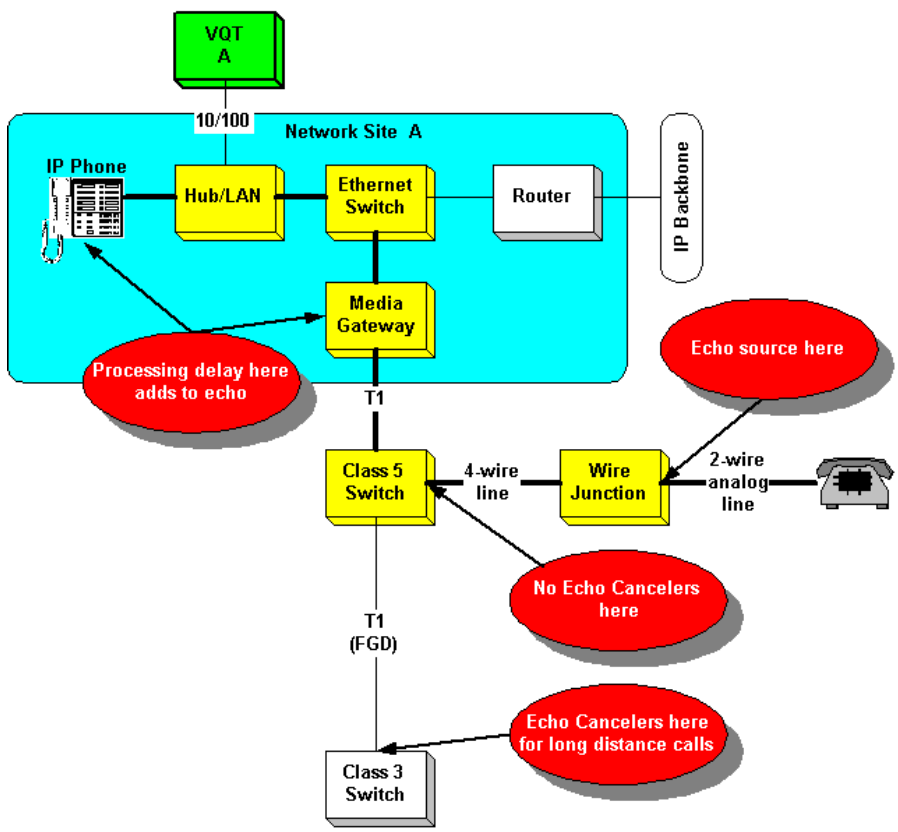

Figure 6

Figure 6 illustrates a scenario in which echo can be problematic. When a VoIP call generated by an IP phone terminates to an analog line on the PSTN, an echo will be generated by the two-wire/four-wire hybrid junction on the analog "tail circuit". This echo is returned to the speaker on the IP phone. The time from when the speaker speaks to when this echo is returned to them is known as the echo return delay. The attenuation of this echo signal (compared to the speaker's signal level) is known as the echo return loss.

On a local network, this echo may not be treated with an echo canceler. However, even for a very short transmission distance, the processing delay introduced by VoIP can cause the echo return delay to exceed 20 milliseconds and become disruptive.

Therefore, a complete pre-VoIP network assessment should include testing for echo to determine if echo cancelers will be needed in a VoIP deployment. Using only a single VQT, the echo on a VoIP network can be assessed, so that decisions regarding echo canceler deployments can be made.

If a media gateway and VoIP call capabilities are available for the assessment (as shown in **Figure 6**), then connect a VQT 10/100 interface to a hub, switch, or router in the IP network. Place a call to a local PSTN two-wire analog termination. Perform the VQT Echo PACE measurement and determine the echo signal level, delay, and impact on perceptual quality using PSQM+ or PESQ. In particular, determine if any received echo exceeds ITU recommendation G.131 echo tolerance curve.

Refer to the Agilent Application Note 5988-3036EN "Testing Voice Quality on Next Generation Networks Using the Agilent VQT" for more techniques on measuring echo and testing per ITU G.131 with the VQT.

Place calls to several different local PSTN terminations and perform echo measurements to determine if echo cancellation on the PSTN is adequate or not. If it is not, then deployment of echo canceler resources with the VoIP network should be considered.

Next, place a series of calls to distant PSTN two-wire analog terminations over long-distance networks and perform echo measurements. Again determine if echo cancellation on the PSTN is adequate or not.

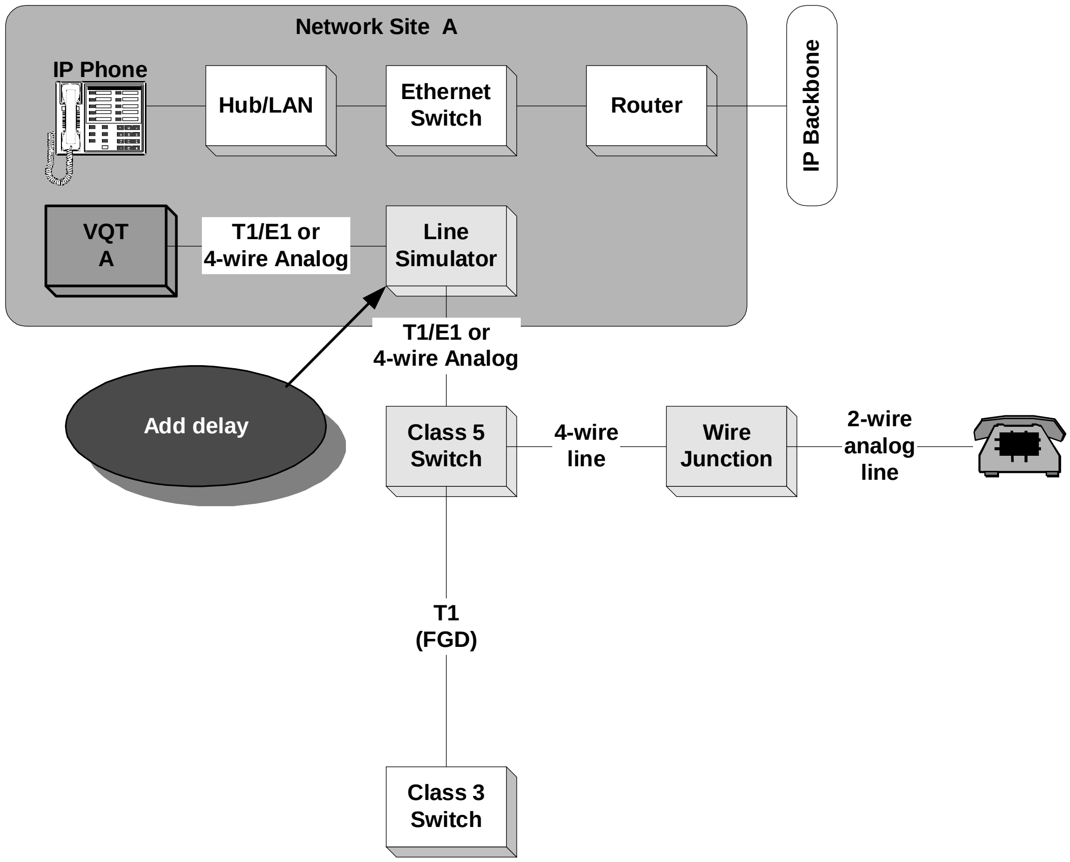

Figure 7

If a media gateway is not available for the assessment, then the setup shown in **Figure 7** can be used to test echo. Connect a VQT four-wire interface (analog E&M, T1, or E1) to one port on a telephone line simulator. Connect a second port on the telephone line simulator to the PSTN. The line simulator should have the capability of switching a call between the two ports and adding delay to the signal. This will emulate the media gateway shown in **Figure 6**. The delay applied by the line simulator in **Figure 7** should approximate the delay added by the VoIP processing in the media gateway and the VQT 10/100 interface in **Figure 6**:

VQT VoIP encoding	25 msec
Media gateway jitter buffer/decoding	30 msec
Media gateway VoIP encoding	25 msec
VQT jitter buffer/decoding	30 msec
TOTAL	110 msec

These values assumes G.729 two-frames/packet encoding.

The test signal transmitted by the VQT will be input to the PSTN with a delay, so as to emulate the delayed signal input to the PSTN in **Figure 6**. A signal echo will be generated at the tail circuit hybrid wire junction and returned to the VQT through the line simulator. The line simulator will add another delay to emulate media gateway VoIP signal processing and the VQT 10/100 interface VoIP signal processing.

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