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Application Notes

Title **Managing Wireless LANs & Wi-Fi Services**

Series VoIP Performance Management

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Overview

This application note describes the typical performance issues that enterprises and service providers encounter when deploying VoIP over wireless LANs (VoWLANs) and Wi-Fi services and introduces a management framework that enables them to detect, address and resolve these problems.

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Introduction

Voice over IP is steadily making its way onto both wide-area and local-area networks. VoIP is enabling enterprises to converge voice and data traffic not only for telecom and operational savings, but also for the productivity benefits of integrated computer-telephony applications.

As such, VoIP is already being implemented between wireless extensions to the corporate LAN, too. IEEE 802.11-based wireless LANs, also informally known as Wi-Fi networks, are quickly expanding from their traditional niche applications in warehouse, retail floor, and health care environments and into mainstream areas of all types of enterprises.

As the traditional workplace becomes mobile, wireless LANs are moving into the carpeted office and also into satellite offices and residences where telecommuters and remote workers are increasingly found conducting business. VoIP calls will soon ride those cable-free networks.

Synergy Research Group, for example, stated in its September 2004 quarterly wireless forecast: “Voice applications will continue to drive the adoption of wireless LANs in the enterprise.” The firm predicted a 36% compound annual growth rate for VoIP over wireless LAN (VoWLAN) phones for the next five years.

This Telchemy application note describes the typical performance issues that those in charge of maintaining voice quality on VoWLANs are likely to encounter when deploying and operating them. Those individuals might be internal enterprise IT staff or a service provider that has deployed a managed VoIP service, such as IP Centrex, and may or may not have included wireless LAN as part of the package.

In addition, this application note introduces a management framework that enables network operators to detect, address and resolve voice-quality problems.

Performance Management & Wireless LANs

VoIP has fundamental sensitivities to latency, jitter and packet loss that must be accommodated even in a cabled packet-switching environment. (See related Telchemy application note, "VoIP Fault & Performance Management: Managing Enterprise IP Telephony," for a testing and deployment primer.)

The situation becomes more challenging when using the less-predictable radio-frequency (RF) medium, which is susceptible to a condition called multipath. Multipath refers to the delays in the delivery of transmitted signals to a receiver due to obstacles and reflectors in the wireless propagation channel, which cause signals to arrive from various directions over several diverse paths.

Delays are further introduced by handoffs between *wireless access points* (APs) -- network infrastructure backbone radios that bridge clients to the wired network -- as mobile users roam. Similarly, more recently, *wireless mesh backbone nodes* have emerged for using 802.11 technology to provide wireless backhaul in enterprise environments where cabling is difficult or expensive. Wireless meshes replace cabled connections from APs to Ethernet switches, and hops between nodes introduce additional latency.

Furthermore, 802.11 nodes must deal with frame corruption because of interference and out-of-sync channel-frequency hops. To detect corruption, the receiver runs a checksum to validate the frame contents of every frame received.

When frames are not corrupt, an acknowledgement (ACK) is sent to the transmitter. When a frame is corrupt, no ACK is sent, and, after a brief backoff, the frame is re-transmitted.

Delay and retransmission can cause jitter, which is problematic for VoIP. Jitter is a variation, or degree of unpredictability, in delay. For these reasons, the wireless medium is often considered unreliable, compared to traditional cabling.

802.11 standards are emerging to help mitigate quality-of-service (QoS) issues with Wi-Fi, though these efforts are focused largely on how to prioritize real-time traffic in 802.11's shared-medium, fair-transmission environment. And newer standards-development efforts have recently gotten under way to speed inter-AP roaming times.

However, the RF environment will always remain dynamic and somewhat unpredictable. For example, wireless LANs can be prone to dead coverage spots, such as in out-of-the-way areas where APs have not been placed but users choose to roam. Also, coverage holes can be intermittent, as an AP becomes temporarily loaded and signal strength becomes too weak for a user to associate with it. Unlike more forgiving data applications, VoIP cannot tolerate any breaks in coverage.

A poor-quality VoIP phone call can be caused by degradation anywhere in the network from point A to point B -- in the wireless handsets, across the Wi-Fi network, the wired LAN, and the WAN. Any number of elements or conditions in the network path along the way can be the root of the cause.

Yet voice is perhaps the oldest mission-critical application there is, and VoIP performance must be successfully managed enterprise-wide, which increasingly will include VoWLAN. To this end, IT managers and managed service providers need tools that can isolate problems to the wireless LAN, if that is indeed where they are originating.

Quite a number of service providers, for example, are offering managed IP services, including WAN VoIP services that extend to LAN-based IP telephony. The next natural step is for one or more 802.11 APs to make their way onto those customers' premises, either by the service provider's design or by the customer's self-installation.

Hence, some service providers may sell VoWLAN services into the customer environment expressly or indirectly; in either case, they might receive wireless-related help desk calls concerning VoIP quality. Because the service provider must think ahead to the management and cost of technical support, that provider requires a way to quickly and remotely detect whether a call has been placed over a wireless LAN and then handle the technical issue in the appropriate manner.

This requires a VoIP performance management architecture that includes real-time monitoring of the service quality delivered by the wireless LAN in addition to the many other elements of the overall network infrastructure for fast, easy problem diagnosis.

To create such an architecture for use by network managers -- those in either enterprise or service provider environments -- it is imperative that wireless LAN system manufacturers embed VoIP monitoring software into their wireless LAN network elements, just as their wired VoIP manufacturing counterparts are doing.

Among the wireless LAN components involved in the VoIP transmission path:

- Wireless VoIP handsets and other client devices with softphone capabilities
- APs
- Wireless LAN switches
- 802.11 mesh backbone nodes/routers
- Wireless LAN telephony gateways

The New VoIP Performance Management Architecture

A new standards-based framework has emerged within the IP industry for VoIP performance management. Using concepts and technology initially conceived by Telchemy and now accepted by several standards bodies, including the Internet Engineering Task Force (IETF), International Telecommunications Union (ITU) and the European Telecommunications Standards Institute (ETSI), this framework uses a distributed software-probe architecture to provide cost-effective, real-time call-quality feedback with maximum network coverage.

The new framework's monitoring functions provide real-time visibility of network performance, detection of transient problems and comprehensive diagnostic data.

The VoIP Performance Management Architecture incorporates QoS reporting protocols into numerous network elements along the network signaling, transmission and management paths.

These protocols send key performance data to network management and call control systems with minimal network traffic overhead. Among these protocols, which are described in more detail in the section "Performance Management Reporting Protocols:"

- Real-time Transport Protocol Control Protocol Extended Reporting (RTCP XR), a media-path reporting protocol that exchanges call quality metrics between VoIP endpoints
- H.460.9 Annex B, a QoS reporting protocol for the H.323 multimedia signaling protocol
- H.248.30, a QoS reporting protocol for the Megaco signaling protocol
- SIP RTCP Summary Reports, a QoS reporting protocol for the SIP signaling protocol
- Simple Network Management Protocol (SNMP)

The concept behind this framework is that the management of real-time packet traffic requires more segmented and granular call detail information than what has been traditionally gathered in IP data networks so that any degradation in VoIP (or real-time video) quality can quickly be isolated and resolved.

Most data applications are far more forgiving in terms of delay, packet loss and jitter, so often, collecting data only in a few key elements has traditionally been sufficient -- as has often been after-the-fact troubleshooting.

By contrast, VoIP performance issues must be isolated and resolved right away to keep overall voice quality consistently high.

To help achieve this goal, in the VoIP Performance Management Architecture, lightweight monitors, or agents, are integrated directly into the wireless LAN elements mentioned in the previous section, as well as traditional wired network elements, such as routers, gateways, wired IP phones, IP PBXs and so forth. Having the ability to gather performance data about the packet voice stream remotely from each of these elements in real time enables fast isolation of both transient and chronic issues that can then be quickly resolved.

Embedded Monitoring Function

Telchemy VQmon® technology is a major building block of the new VoIP Performance Management Architecture. It has formed the cornerstone of several emerging industry-standard protocols for use in VoIP performance management (see appendix).

VQmon is a call quality monitoring and diagnostic agent that an equipment maker -- such as a manufacturer of a wireless LAN access points, switches, client devices, and mesh routers -- can embed in any equipment that sits in the path taken by a VoIP packet. VQmon agents are small and highly efficient and can be integrated into equipment without requiring additional CPU or memory resources. The software can scale from systems that monitor a single call to those that monitor hundreds of thousands of calls.

From there, VQmon enables a network operator, whether it is an internal IT network administrator or a remote service provider, to see call-quality problems in real time and identify the root cause on active or completed calls (Figure 1 below). Figure 2 on the following page shows the enterprise with wireless LAN plus the service provider network.

The technology measures the key characteristics of the packet voice stream and calculates real-time performance data, which network operators can use to detect, characterize and report problems that might be affecting service quality.

VQmon provides detailed service-quality metrics such as listening and conversational quality scores. It also provides diagnostic information such as the severity and distribution of packet loss and packet discards due to jitter. It is able to do this using less than 0.1 % of network bandwidth because it uses a lower reporting frequency than older technologies such as RTCP SR/RR.

VQmon is an advanced VoIP perceptual quality estimation algorithm that incorporates support for key international standards including ITU-T P.564, ITU-T G.107, ITU-T G.1020, ETSI TS 101 329-5 Annex E and IETF RFC 3611. VQmon incorporates support for time varying IP impairments (typically caused by network congestion) and has been independently shown to provide significantly more accurate and stable metrics than other algorithms such as G.107 (E Model).

There are two versions of Telchemy VQmon technology: VQmon/EP (End Point) and VQmon/SA (Stream Analysis). Both can be used for active testing of live calls and passive testing of scheduled or on-demand test calls.

Wireless VoIP Handsets, Wireless LAN Telephony Gateways: VQmon/EP

Leading device manufacturers are integrating VQmon/EP into wireless network elements where calls terminate, such as wireless VoIP handsets and wireless LAN telephony gateways, which convert a traditional PBX call to IP packets and forward them over the Wi-Fi network. VQmon/EP software monitors the received packet stream and extracts other vital information from the VoIP codec. It can use this data to calculate accurate call-quality estimates and supporting diagnostic data. It also sends QoS reports to management systems, such as the Telchemy SQmediator® application, via RTCP XR or the VoIP signaling QoS protocols mentioned earlier.

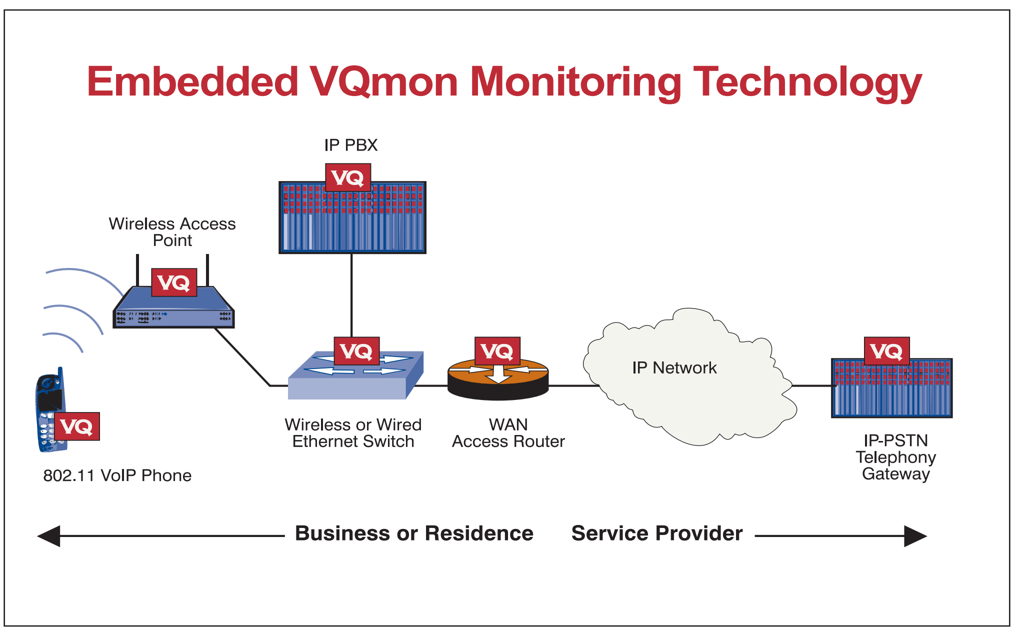


Figure 1: The New Performance Management Architecture For Enterprises With Wireless LANs

Wireless APs, Switches/Controllers, Probes, Analyzers and Routers: VQmon/SA

VQmon/SA is the core VoIP analysis software that is appropriate for use in wireless and wired devices that forward IP packet streams, as well as in diagnostic equipment where packet capture/decoding/analysis takes place.

Among these network elements are Wi-Fi APs, switches/controllers, mesh routers, handheld wireless LAN analyzers and distributed wireless LAN probes that communicate to centralized management stations.

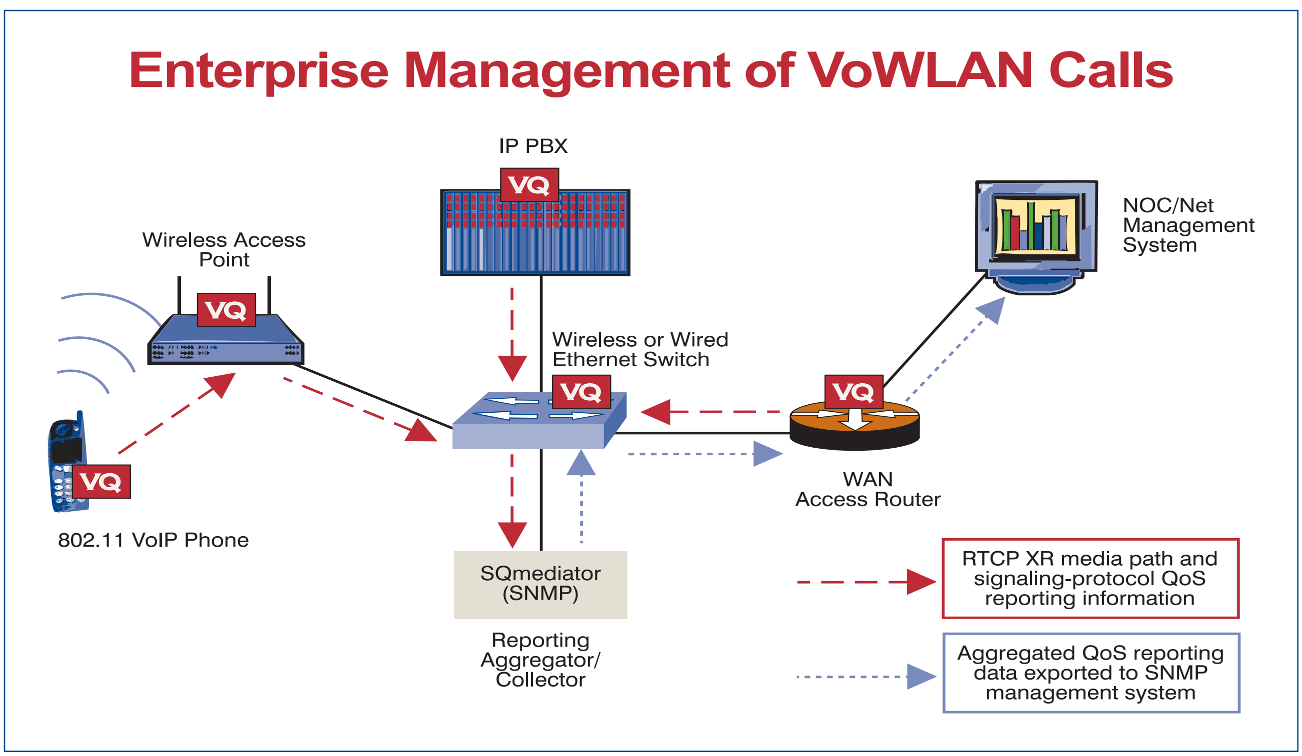
VQmon/SA monitors the packet stream, automatically recognizing individual call streams and the types of codecs in use. It determines which packets were lost or discarded due to jitter and uses this data to determine a call-quality estimate using the same algorithms as VQmon/EP.

Common VoIP Performance Metrics

The new VoIP Performance Management Architecture incorporates a common set of VoIP performance metrics supported by multiple QoS reporting protocols (Figure 2 below). In other words, the same QoS data is available independent of which protocol is used for reporting. The metrics measured and reported by the architecture include the following:

- Percentage of packets lost by the network and percentage of packets discarded by the jitter buffer due to late arrival. Both metrics help identify the degree to which a call is being affected by network packet loss or jitter. They also eliminate the need to guess at how much effect jitter is having on the packet discard rate.

Figure 2: The New VoIP Performance Management Architecture For Enterprises With Wireless LANs, Possibly Across a Service Provider Network.



- Mean length and density of bursts. A burst is defined as the interval of time during which the packet loss/discard rate is high enough to cause audio quality degradation. Transient network congestion is a common problem resulting in a period of high loss/discard lasting for several seconds during which call quality is degraded. These transient call quality-related problems are reported as "bursts."

- Mean length of gaps between bursts and the density of packet loss/discard within these gaps. These metrics help identify the extent to which the call is degraded by packet loss/discard and provide insight into the user experience.

- Round-trip delay between VoIP end points and end-system delay within a VoIP endpoint. These metrics help identify the sources of excessive delay that can lead to conversational difficulty and intensify the effects of echo.

- Signal, noise, and echo levels. These metrics enable detection of problems due to excessive variations in signal, noise or echo level. They also allow midstream probes to detect signal-, echo- and noise-level problems without needing to decode voice packets.

- Call-quality metrics using R factor and Mean Opinion Score (MOS). This data provides an immediate view of call quality. The R factor is a call-quality rating score determined by taking objective measurements of latency, jitter and packet loss and determining their impact on call quality while also accounting for codec type.

  MOS is normally obtained by conducting a controlled test in which a large number of human listeners make calls to each other and rate the voice quality on a scale of 1 to 5, with 5 the highest and 4 considered to be "toll" quality; however, technologies such as VQmon can estimate a MOS score. Both R and MOS can be calculated by VQmon embedded software-agent technology.

- Jitter-buffer configuration and packet-loss concealment algorithm. This information is used to determine if poor call quality is attributable to an incorrectly configured end system and to allow mid-stream probes to automatically detect endpoint configurations.

Performance Management Reporting Protocols

As mentioned, reporting protocols have been developed for the media path, signaling system and network management. These protocols are complementary and have been designed to work together. Among those supported in VQmon:

- **RTCP XR for Media Path QoS Reporting** -- Specified in IETF RFC 3611, this media path reporting protocol exchanges call-quality metrics between VoIP endpoints, such as handsets, gateways IP PBXs. Among its benefits:

  -- Enables remote endpoints to collect and generate call-quality reports

  -- Allows performance reports to be passed transparently through firewall routers

-- Supports diagnosis of complex problems, such as echo

-- Enables network probes to obtain analog signal information without the need to decode voice packets

-- Is compatible with the emerging IETF-Standard Secure Real-time Transport Protocol (SRTP) framework for encrypting voice conversations using the Advanced Encryption Standard (AES).

- **H.460.9 Annex B, H.248.30 and SIP for Signaling QoS Reporting** -- Several new QoS reporting protocols have been developed within the ITU and IETF that support call-quality reporting to call-management systems such as softswitches. These protocols provide call-quality information directly to the systems that maintain call-detail record (CDR) databases and link service quality information directly to specific customers and their calls.

  QoS reporting protocols for the H.323 multimedia signaling protocol (H.460.9 Annex B) and Megaco (H248.30) were approved in early 2004, and new protocols for the Session Initiation Protocol (SIP) and Media Gateway Control Protocol (MGCP) are expected in early 2005.

- **SNMP and the RTCP XR Management Information Base (MIB)** -- A RTCP XR MIB is under development within the IETF for use in gateways and probes to support the retrieval of metrics via SNMP.

  For example, RTCP XR could be used to relay call-quality information from an IP phone to the gateway that forms the network end of a VoIP connection, and SNMP could be used to retrieve call-quality information from the gateway for both inbound and outbound packet streams.

Applying the VoIP Performance Management Architecture to VoWLANs

The new VoIP Performance Management Architecture is essential to providing real-time visibility of user-perceived quality in VoWLAN networks, whether they are installed and managed by the enterprise or come about as an intentional or unintentional byproduct of a managed IP service from a carrier.

An Enterprise Scenario

Enterprise IT departments that plan to use VoIP on their wireless LANs should request embedded VoIP monitoring agents in their 802.11 handsets and wireless LAN system equipment from their Wi-Fi suppliers. IT departments should also determine a VoIP QoS reporting strategy, which entails whether they wish to use media path, signaling or SNMP reporting -- or all three.

During an active call, service quality can be monitored by the enterprise using a network management application, such as Telchemy SQmediator, which collects RTCP XR and/or SIP RTCP Summary reports from the VoIP handsets and wireless telephony gateway, if present. In addition, for enterprises using VoIP test equipment (probes and analyzers), when problems are reported and detailed analysis is required, staff can use VQmon-enhanced equipment to capture and analyze call streams. Finally, call-quality reports may be forwarded through the H.323, SIP or other signaling protocol.

A Service Provider Scenario

A service provider offering managed services with some visibility onto the customer premises will be able to detect if a phone call has been placed over a VoWLAN handset. VQmon is integrated into VoWLAN handsets, APs, and into enterprise "edge" devices, or CPE—the traditional service demarcation point. The service provider can retrieve call-quality RTCP XR reports from the CPE using SNMP, providing visibility into both directions of the call.

Summary

The new VoIP Performance Management Architecture provides a much-needed solution for call-quality monitoring and problem diagnosis. Important call-quality and diagnostic data are obtained in real time from key points in the packet stream, including wireless LANs, and are routed to network management systems and CDR databases.

Network managers can obtain a reasonable level of problem diagnosis simply by using data from VoIP call-quality reports; however, in-depth problem analysis requires the use of VQmon based probes and analyzers that support the VoIP management framework.

Many of the networking industry's leading equipment manufacturers are already implementing elements of the VoIP management framework in both wireless and wired IP phones, gateways, routers, probes and other devices.

Notes

Acronyms

AES	Advanced Encryption Standard
AP	Access Point
CDR	Call Detail Record
IETF	Internet Engineering Task Force
IP	Internet Protocol
ITU	International Telecommunications Union
RF	Radio-Frequency
SNMP	Simple Network Management Protocol
SRTP	Secure Real-Time Transport Protocol
VoIP	Voice over Internet Protocol
VoWLAN	Voice over Wireless LAN
VPN	Virtual Private Network
VQmon/EP	VQmon End Point
VQmon/SA	VQmon Stream Analysis
Wi-Fi	Wireless Fidelity

References

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[2] ITU-T G.799.1 "Functionality and Interface Specifications for GSTN Transport Network Equipment for Interconnecting GSTN and IP Networks"

[3] ITU-T H.460.9 Annex B Support for online QoS-monitoring reporting within H.323 systems: Extended Performance Metrics

[4] ITU-T H.248.30 RTCP Extended Performance Metrics Packages

[5] IETF SIPPING draft-johnston-rtcp-summary.02.txt

[6] ETSI, "Quality of Service (QoS) measurement methodologies," ETSI TS 101 329-5 V1.1.1 (2000-11), November 2000.

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About Telchemy, Incorporated

Telchemy, Incorporated is the global leader in VoIP and IP Video fault and performance management with its **VQmon**® family of multimedia quality monitoring and analysis software. Telchemy is the world's first company to provide voice quality management technology that considers the effects of time-varying network impairments and the perceptual effects of time-varying call quality. Founded in 1999, the company has products deployed worldwide and markets its technology through leading networking, test and management product companies. Visit **www.telchemy.com**.

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