Application Notes

Title **Managing Cable Telephony Services**

- **Series** VoIP Performance Management
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Overview

This application note describes the typical performance issues that cable operators encounter when deploying cable telephony networks and introduces a management framework that enables them to detect, address and resolve these problems.

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Introduction

Packet Telephony is an exciting new source of revenue for cable operators, so it is essential to build fault and performance management systems that support quick problem detection and resolution and avoid costly truck rolls. Cable operators are aware that HFC networks can suffer from performance-related problems, but they are less familiar with the broader range of call quality issues associated with Voice over IP networks and the interaction between traditional cable system and packet telephony components.

This application note describes the typical performance issues that cable operators encounter when deploying cable telephony networks and introduces a management framework that enables them to detect, address and resolve these problems.

Performance Management & Cable Telephony

Cable operators are familiar with many of the problems they will face when deploying cable telephony networks due to their past experiences introducing cable modem service:

- Ingress noise on the cable upstream that leads to high packet loss rates.
- Weather problems i.e., the affect of temperature extremes and thunderstorms on cable service.

In addition, Cable Telephony brings both an increased sensitivity of the service to packet related problems and addition problems that are particular to VoIP.

- Wiring problems in subscribers' homes that require service calls. Existing cable modem management statistics can help determine remote problems, but they may not be sufficient for diagnosing IP telephony-related problems.

- Network problems are transient in nature and can occur at many places along the packet path. By the time a user reports a problem, the network condition causing that problem may have disappeared.

- Some problems e.g., echo, result from the interaction of the two endpoints (trunking gateway and IP phone) and the network.

MSOs and operators need a performance management architecture that supports real-time monitoring of the service quality delivered to subscribers' home locations and fast, easy problem diagnosis.

The New VoIP Performance Management Architecture

A new standards-based framework has emerged within the IP industry for VoIP Performance Management. It 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 framework incorporates QoS reporting protocols that send key performance data back to network management and call control systems with minimal network traffic overhead. High performance network probes are located within the core network and at head-ends to provide in-depth diagnosis they detect problems.

A key benefit of the new VoIP Performance Management Architecture is that small lightweight monitoring functions can be integrated directly into MTAs, ATAs, IP phones, routers, gateways and similar network elements. This "direct integration" provides probe functionality in equipment where it would normally be cost prohibitive e.g., every customer home.

Embedded Monitoring Function

VQmon® technology is a major building block of the new performance framework (Figure 1); it was the first and is the most widely deployed monitoring function for VoIP performance management today. VQmon's embedded monitoring technology enables network managers to see call quality problems in real-time and identify the root cause of the problem for both active and completed calls.

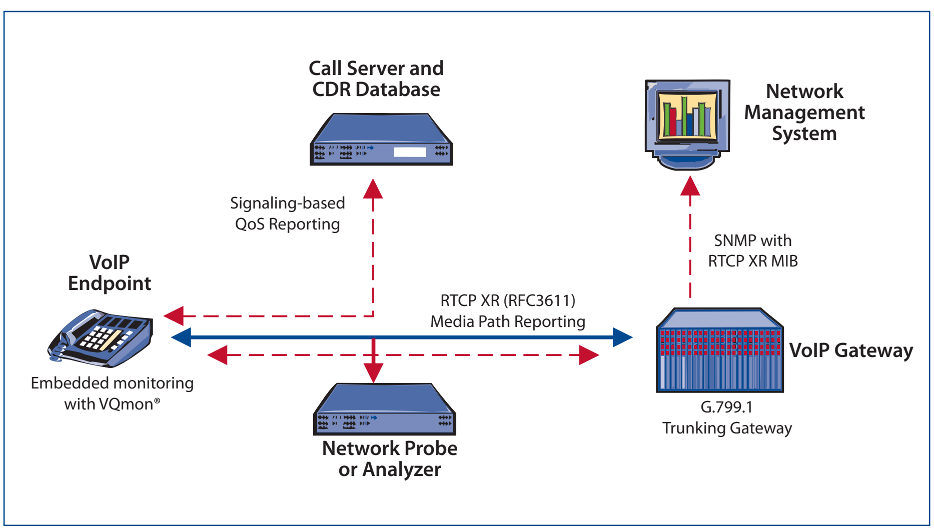

Figure 1: *The new VoIP Performance Management Reference Model*

The technology measures key characteristics of the packet voice stream and calculates real-time performance data that network managers can use to detect, characterize and report service quality affecting problems.
VQmon provides detailed service quality metrics and diagnostic information on transient problems; however, it uses a lower reporting frequency than older technologies thus requiring less bandwidth.

VQmon is a high performance standards-based non-intrusive call quality monitoring and diagnostic agent that can be integrated directly into VoIP CPE, SLA monitoring systems, probes, routers and generally any system that is on the path taken by voice packets. VQmon agents are small and highly efficient and can be integrated into existing equipment without requiring additional CPU or memory, scaling from systems that monitor a single call to those that support “hundreds of thousands” of calls.

VQmon agents produce call quality metrics, including listening and conversational quality scores. They also provide detailed information on the severity and distribution of packet loss and discards due to jitter and detect transient IP problems that affect service quality.

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 VQmon: VQmon/EP (End Point) and VQmon/SA (Stream Analysis). Both produce a rich set of diagnostic data, and support the common VoIP performance management metrics.

MTA, ATA & Trunking Gateways: VQmon/EP

Leading equipment manufacturers are integrating VQmon/EP into their IP phones and gateways. VQmon/EP monitors the received packet stream and extracts other vital information from the VoIP CODEC. VQmon can use this information to calculate accurate call quality estimates and supporting diagnostic data, and sends QoS reports back to management systems via RTCP XR or signaling based reports.

CMTS, Analyzers and Probes: VQmon/SA

VQmon/SA is the core VoIP analysis software used in many probes, routers, SLA monitoring systems and analyzers. VQmon/SA monitors the packet stream, automatically recognizing individual call streams and the types of CODEC in use. VQmon determines which packets were lost or would be 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 i.e., the same information is available regardless of the protocol used for reporting (Figure 2). These metrics are:

- Percentage Of Packets Lost By the Network

- Percentage Of Packets Discarded By the Jitter Buffer Due To Late Arrival

  Both these metrics help to identify the degree to which a call is being affected by network packet loss or jitter. They also eliminate the need to "guess" how much effect jitter is having on packet discard rate.

- Mean Length and Density Of Bursts (where a burst is defined as the interval of time during which the packet loss/discard rate is high enough to cause audio quality degradation)

- Mean Length and Density Of Gaps Between Bursts

  Both these metrics help to identify the extent to which the call is degraded by loss/discard and provide some insight into the user experience. 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 problems are reported as "bursts."

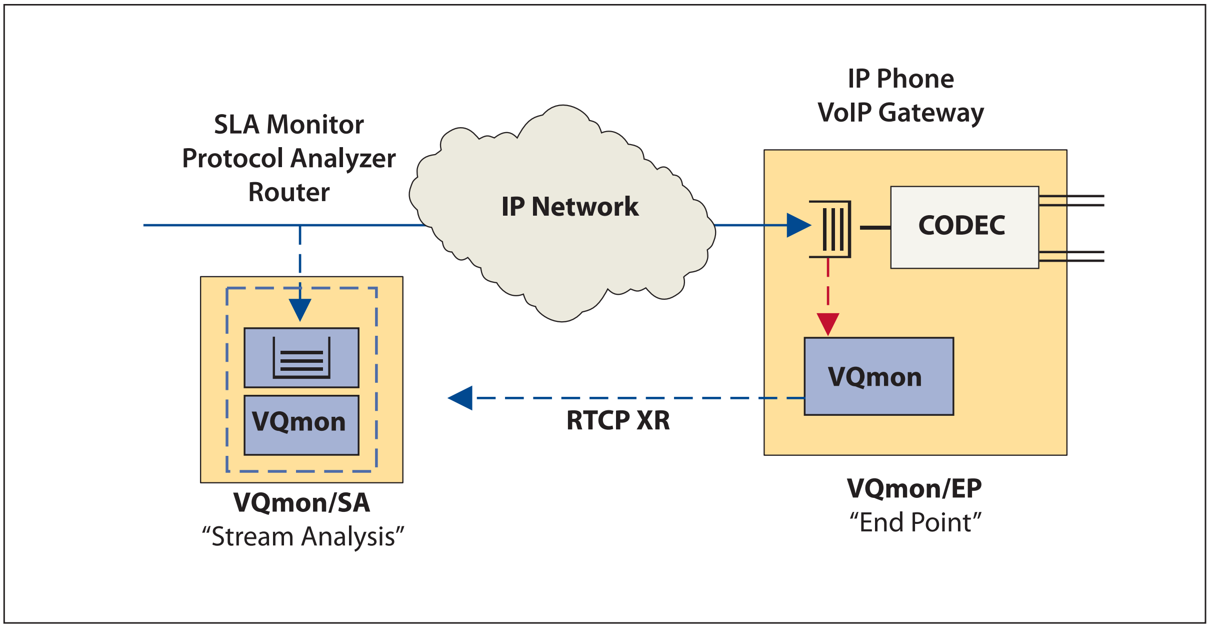

Figure 2: VQmon's embedded monitoring technology

- Round Trip Delay Between VoIP Endpoints
- End System Delay Within a VoIP Endpoint

  Both these metrics help to identify the sources of excessive delay which can lead to conversational difficulty and greatly intensify the effects of echo.

- Signal Level
- Noise Level
- Echo Level

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

- Call Quality Metrics In Both R and MOS Scaling

  These metrics provide an immediate view of call quality. If it is apparent that there is a problem, other metrics can be used for diagnosis.

- Jitter Buffer Configuration and Packet Loss Concealment Algorithm

  This information is used to determine if poor call quality is due to an incorrectly configured end system and to allow mid-stream probes to automatically detect endpoint configuration.

Performance Management Reporting Protocols

Reporting protocols have been developed for the media path, signaling system and network management. It is important to realize that these are complementary i.e., they are designed to be used together.

RTCP Reporting Extensions (RTCP XR)

The RTCP XR protocol (RFC3611) is a media path reporting protocol that exchanges call quality metrics between VoIP endpoints. RTCP XR provides several useful functions:

- Enables collection of call quality reports by the remote endpoint e.g., a trunking gateway or by intermediate probes
- Provides ability to pass performance reports transparently through firewall routers
- Supports the diagnosis of complex problems such as echo
- Enables network probes to obtain analog signal information without the need to decode voice packets
- Compatible with the emerging Secure RTP security framework.

Signaling Protocol QoS Reporting

Several new QoS Reporting protocols have been developed within ITU and IETF that support call quality reporting to call management systems e.g., softswitches. These protocols provide call quality information directly to the systems that maintain CDR databases and link service quality information directly to specific customers and their calls. QoS reporting protocols for H.323 (H.460.9 Annex B) and Megaco (H.248.30) were approved in early 2004, and new protocols for SIP and MGCP are expected in late 2004.

SNMP and the RTCP XR MIB

An RTCP XR Management Information Base (MIB) is under development within the IETF for use in gateways or 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 would be used to retrieve call quality information from the gateway for both in-bound and out-bound packet streams.

Applying the VoIP Performance Management Architecture To Cable Telephony

The new VoIP performance management architecture is essential to provide real-time visibility of user perceived quality in cable networks. (Figure 3)

- VQmon is integrated into the MTA or ATA in the customer's home to monitor the quality of each live call. RTCP XR reports are periodically inserted into the packet stream to provide real-time feedback to the service provider. In addition, call quality reports may be forwarded through MGCP/NCS messages back to the softswitch.

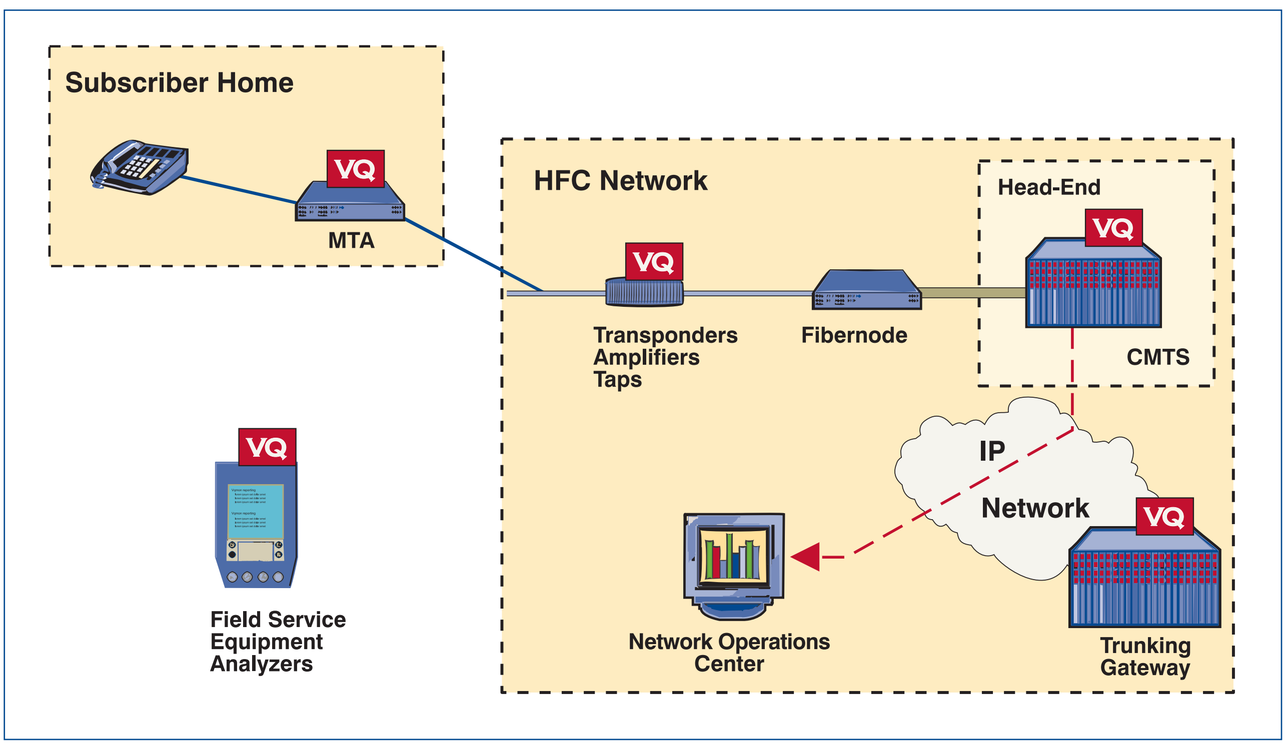

Figure 3: *The New VoIP Performance Management Architecture for Cable Telephony*

- VQmon may be integrated into a probe located at the head-end or directly into the CMTS. This is an ideal location to measure the quality of calls as they pass from the HFC network to the backbone and to observe RTCP XR data sent by MTAs.

- VQmon is integrated into the media/trunking gateway at the interconnection to the telephone network (meeting ITU G.799.1 requirements). During an active call, service quality for the customer-to-network direction of the call is monitored at the network's end-of-the-path.

- The Softswitch collects QoS reports sent via the signaling protocol. At the end of a call, a call quality report may be sent from the IP endpoint through the signaling protocol to the softswitch. This data can be incorporated into the Call Detail Record database.

- VQmon is integrated into VoIP test equipment (probes and analyzers). When problems are reported and detailed analysis is needed, Network Operations and Field Service personnel use this test equipment to capture and analyze call streams. VQmon can be used for both active i.e., on-demand or scheduled testing and passive i.e., live call testing.

- VQmon may be integrated into probes and transponders located within the HFC network to support fast detection and identification of HFC network problems that affect call quality.

Problem Resolution, Detection & Diagnosis

The new VoIP Performance Management Architecture provides the basis for detecting and diagnosing different types of call quality-related problems:

- **Upstream Ingress Noise**
—If there is an upstream ingress noise problem, then VQmon located in either the head-end or trunking gateway can immediately detect the problem and assess its impact on call quality. Ingress noise can appear as either a periodic or transient packet loss problem. VQmon will indicate a packet loss (versus packet discard) problem and provide statistics on the transient bursts of packet loss and their frequency.

- **Echo In PSTN**
—If there is a problem on a remote PSTN analog loop, the cable subscriber may experience audible echo. NOC staff can diagnose echo problems using VQmon-generated call quality reports sent from the trunking gateway that show echo level and the level's effect on conversational quality metrics as reported from the MTA.

- **Subscriber Reports "Noisy Call"**
—VQmon located in the MTA at the customer home can monitor incoming calls and provide reports back to the service provider through RTCP XR, the signaling system or SNMP. The data provided by VQmon clearly identifies if the network problem was caused by jitter, packet loss or noise in the signal.

Summary

The new VoIP Performance Management Architecture provides an ideal solution for service quality monitoring and problem diagnosis. Important call quality and diagnostic data is obtained real-time from key points on the packet stream and 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 probes and analyzers that support the new performance management framework as well as the use of VQmon for call quality analysis.

Many of the industry's leading equipment manufacturers are already implementing elements of the new management framework in IP phones, gateways, routers, probes and analyzers.

Notes

Acronyms

ATA	Analog Telephone Adaptor
CDR	Call Detail Record
CMTS	Cable Modem Termination System
HFC	Hybrid Fiber Coax
IETF	Internet Engineering Task Force
IP	Internet Protocol
ITU	International Telecommunications Union
MGCP/NCS	Media Gateway Control Protocol/Network-Based Call Signaling
MIB	Management Information Base
MSO	Multiple System Operator
MTA	Media Terminal Adaptor
NOC	Network Operations Center
PBX	Private Branch Exchange
QoS	Quality of Service
RTP	Real Time Protocol
SLA	Service Level Agreement
SNMP	Simple Network Management Protocol
VoIP	Voice Over Internet Protocol
VPN	Virtual Private Network
VQmon/EP	VQmon End Point
VQmon/SA	VQmon Stream Analysis

References

[1] IETF RFC3611 RTP Control Protocol Reporting Extensions (RTCP XR), November 2003

[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.

[7] ITU-T, "The E-Model, a computational model for use in transmission planning," Recommendation G.107

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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