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Application Notes

Title   **Managing IP Centrex & Hosted PBX Services**

■ **Series**   VoIP Performance Management

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Overview

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

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Introduction

As Voice over IP is deployed in Enterprise networks, IP Centrex and VPN services, it is vital to implement a performance management framework that is sufficient to deal with many operational problems that arise. IP Centrex or Hosted PBX services provide a cost effective solution for enterprise customers, minimizing capital outlay and outsourcing the headache of deploying and managing next generation telecom services. Telephony is mission critical for most businesses, and customers expect that new generation services deliver the quality and reliability that they had with traditional phone service.

IP Centrex services bring particular call quality and performance management challenges. The service provider is responsible for the call management (softswitch) function; however voice packets travel over both the customer LAN and service provider network. Problems are compounded by the transient and unpredictable nature of IP impairments that can make problem location and diagnosis difficult as well as the presence of firewall routers on the voice path.

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

Quality Management & IP Centrex Service

Customers expect that their IP Centrex or Hosted PBX service provider will manage the quality of the service they provide, thus requiring that the service provider monitor and manage call quality and troubleshoot any related problems. In today's market, customers often request a SLA to hold the service provider accountable for overall service and call quality.

Service level is specified at an interface or demarcation point on the network. For IP Centrex and managed IP services, this is usually the edge router located in the customer premises. Hence, the service provider is responsible for problems in the core network and on the access link, and the customer is technically responsible for problems on their LAN.

In an IP Centrex service, voice packets travel between an IP phone on a user desktop through a LAN, access link and core IP network to a trunking gateway which results in several problems:

- Since different network segments are "owned" by different entities i.e., the service provider and the customer, they may not know what effect their specific network segment is having on the IP phone call. In addition, the service provider may not have management access into their customer's LAN in order to retrieve diagnostic information from IP phones or gateways.

- 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.

- Service providers can't use existing management protocols such as SNMP due to both bandwidth and security issues.

For example, SNMP requests from the service provider to devices on the customer LAN can be blocked by the customer's firewall router.

- Some problems e.g., echo, result from the interaction of the two endpoints (trunking gateway and IP phone) and the network. Measuring echo and loss plan can be costly and impractical when Secure RTP is deployed and voice packet payloads are encrypted.

Due to these problems, IP Centrex service providers need a performance management architecture that supports contractual SLA monitoring at the service demarcation point, diagnostic service level monitoring at the IP phone and trunking gateway 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.

It also provides QoS reporting protocols with the ability to send data back to network management and call control systems with minimal network traffic overhead. The architecture features high performance network probes located within the core network and at major customer locations that provide in-depth diagnosis when they detect problems.

A key benefit of the new VoIP Performance Management Architecture is that small lightweight monitoring functions can be integrated directly into 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 desktop.

Embedded Monitoring Function

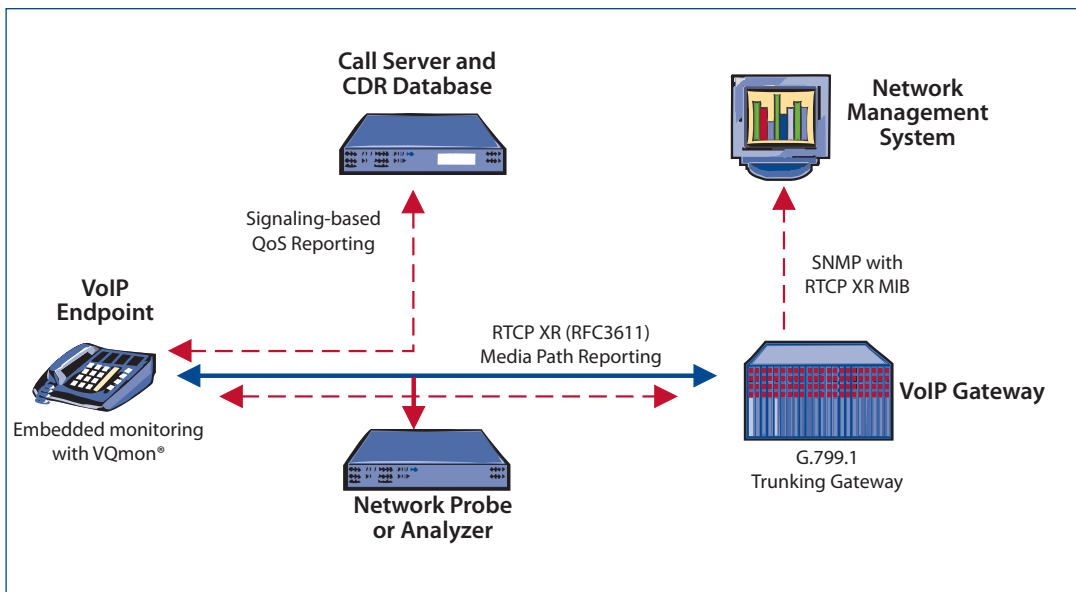**VQmon**® technology is a major building block of the new performance framework; it was the first and is the most widely deployed monitoring function for VoIP performance management today. VQmon's embedded monitoring technology enables network managers to see call quality problems in real-time and identify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

VQmon measures key characteristics of the packet stream and calculates real-time performance data that network managers can use to detect, characterize and report transient problems. The technology provides detailed information on service quality with less reporting frequency, 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 (Figure 1). 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 other essential diagnostic data. Most importantly, VQmon is able to detect transient IP problems and assess their effects on call quality.

VQmon supports 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).

Call Server and CDR Database

Network Management System

Signaling-based QoS Reporting

VoIP Endpoint

SNMP with RTCP XR MIB

RTCP XR (RFC3611) Media Path Reporting

VoIP Gateway

Embedded monitoring with VQmon®

G.799.1 Trunking Gateway

Network Probe or Analyzer

Figure 1: *The new VoIP Performance Management Reference 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.

IP Phones and Gateway Monitoring: 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 provide accurate call quality estimates and supporting diagnostic data.

SLA Monitors, 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 an interval of time during which the packet loss/discard rate is high enough to cause audio quality degradation)
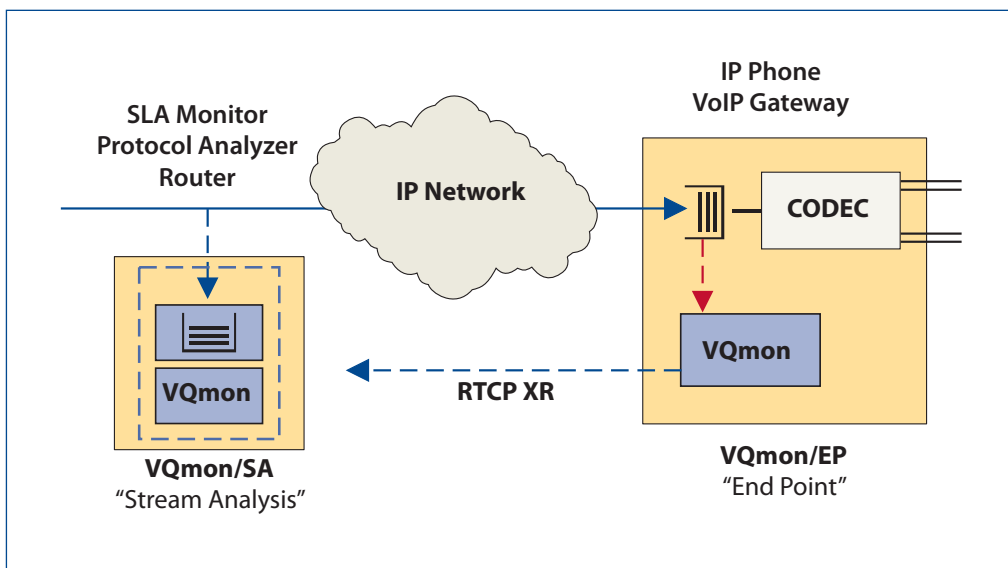
Figure 2: *VQmon's embedded monitoring technology*

■ Mean Length and Density Of Gaps Between Bursts

Both these metrics help to identify the extent to which a 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."

■ 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. These metrics will be essential when networks use the secure RTP framework.

■ Call Quality Metrics In Either R Or MOS Scaling

These metrics provide an immediate view of call quality. If it is apparent that there is a problem, the 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 transparently through firewall routers

■ Supports the diagnosis of echo-related problems

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

■ Compatibility 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 mid- to 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 New Performance Management Architecture to IP Centrex

The new VoIP performance management architecture is very useful in providing real-time visibility of user perceived quality in an IP Centrex application (Figure 3).

■ VQmon is integrated into the IP phone on a customer desktop or an integrated access device, edge router or multi-service gateway. During an active call, service quality for the network-to-customer direction of the call is monitored at the service demarcation point and 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 the signaling protocol e.g., H.323 or SIP.
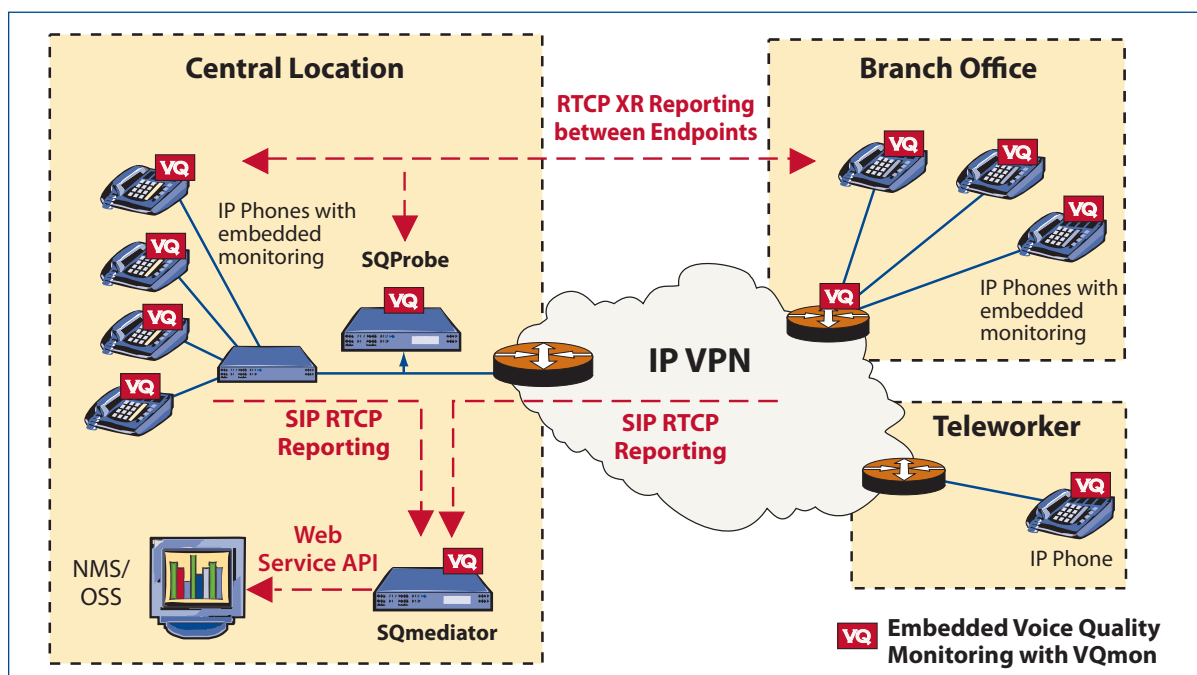
Figure 3: The New VoIP Performance Management Architecture for IP Centrex Networks

- VQmon is integrated into the media/ trunking gateway or a probe in the service provider network. During an active call, service quality for the customer-to-network direction of the call is monitored at the network end of the path, and RTCP XR reports from the customer premise equipment are collected and retrieved using SNMP. This provides visibility into both directions of the call.

- 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.
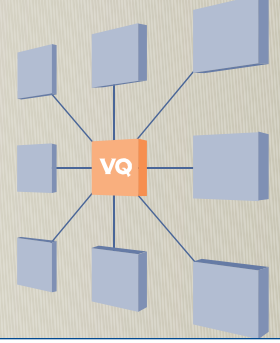
Problem Resolution, Detection and Diagnosis

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

- **Inbound Access Link Congestion**
  —If the access link (T1, DSL) from the network to the customer site has insufficient bandwidth, the jitter level will increase. This will typically cause the IP Phone to discard a high proportion of packets and can increase the size of the jitter buffer size adding more delay. Users can diagnose congestion problems using VQmon generated call quality reports sent from either an IP Phone or a router/IAD on the customer site which will show a high rate of packet discards.

- **Outbound Access Link Congestion**
  —If the access link (T1, DSL) from the customer site to the network has insufficient bandwidth, the jitter level will increase. This will typically cause the trunking gateway in the service provider network to discard a high proportion of packets and can increase the size of the jitter buffer, adding to delay. Users can diagnose congestion problems using VQmon-generated call quality reports retrieved from the trunking gateway which will show a high rate of packet discards.

- **Duplex Mismatch or Bad Ethernet Segment on Customer LAN**
  —If the customer's LAN is not configured properly, there may be a high rate of packet loss that severely impacts call quality. Users can diagnose duplex mismatches or bad Ethernet segments using VQmon-generated call quality reports sent from an IP phone on the customer site which will show a high rate of lost packets.

■ **Echo in PSTN**

—If there is a problem on a remote analog loop, the IP phone on a customer site may experience audible echo. Users can diagnose echo problems using VQmon-generated call quality reports sent from the IP phone and the trunking gateway which will show a high echo level.

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. ▲

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Acronyms

CDR	Call Detail Record
IETF	Internet Engineering Task Force
IP	Internet Protocol
ITU	International Telecommunications Union
PBX	Private Branch Exchange
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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▷ Information

- **Email**
  sales@telchemy.com
  info@telchemy.com
- **Phone**
  +1-678-387-3000

- **Telchemy, Incorporated**
  105 Nobel Ct
  Suite 100
  Alpharetta, GA 30005
  USA