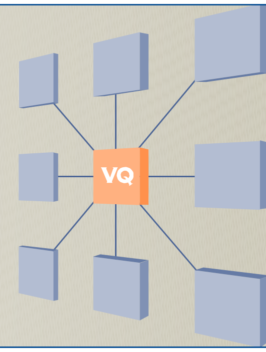

Application Notes

Title **Managing Enterprise IP Telephony**

Series Voice over IP Performance Management

Date Apr 2017

Overview

This Application Note describes typical issues that network managers encounter when deploying Enterprise IP Telephony and introduces the standards-based management framework that allows them to detect, address, and resolve these problems.

Contents

Introduction

Voice over IP brings many benefits to the Enterprise, including lower system maintenance costs, improved network integration between branch offices and teleworkers, and the ability to build distributed call centers. Telephony is mission critical for most businesses, and network managers expect their Voice over IP services to deliver same quality, reliability, and availability as traditional phone service. As VoIP is deployed in the Enterprise, it is critical to implement a performance management framework that is sufficient to deal with the many operational problems that arise.

Performance Management & Enterprise IP Telephony

Calls originating from IP phones are typically carried as internal VoIP traffic to branch offices or teleworkers in their home offices. VoIP traffic is often carried into telecom services as IP traffic, but may be translated back to "plain old telephone service" (POTS) for connection to a legacy analog or digital circuit.

IP Telephony is very different from conventional data applications in that call quality is particularly sensitive to IP network impairments. Existing network problems become much more obvious with the deployment of VoIP. For network managers, this means that LANs, access links, and network equipment will probably need to be upgraded and that more sophisticated management and diagnostic tools are needed when deploying and maintaining VoIP.

There are three basic categories of performance-related problems that can occur in Enterprise IP Telephony:

1. IP Network Problems

- **Jitter** — Variation in packet transmission time that leads to packets being discarded in VoIP end systems or to increases in delay. Jitter is usually due to network congestion, but it can also be caused by load sharing across transmission routes with differing delays.
- **Packet Loss** — Packets lost during transmission due to network errors, route changes, link failures, or buffer overflows in switches.
- **Delay** — Overall packet transmission "lag time" that leads to two-way conversational difficulty.

2. Equipment Configuration & Signaling Problems

- **VoIP Endpoint Configuration** — Performance impact based on codec type and packet loss concealment (PLC) algorithm, or jitter buffer configuration.
- **Router and Firewall Configuration** — Firewalls or incorrectly configured routers block VoIP traffic; routers need to be configured to allow RTP packets through in both directions.
- **SIP Awareness of Firewalls** — SIP-aware firewalls are able to understand SIP signaling and can intelligently handle and modify signaling messages to match media streams, which is important if IP addresses are being NAT'd.
- **Bandwidth Allocation** — Network may lack sufficient bandwidth to support peak traffic volumes.

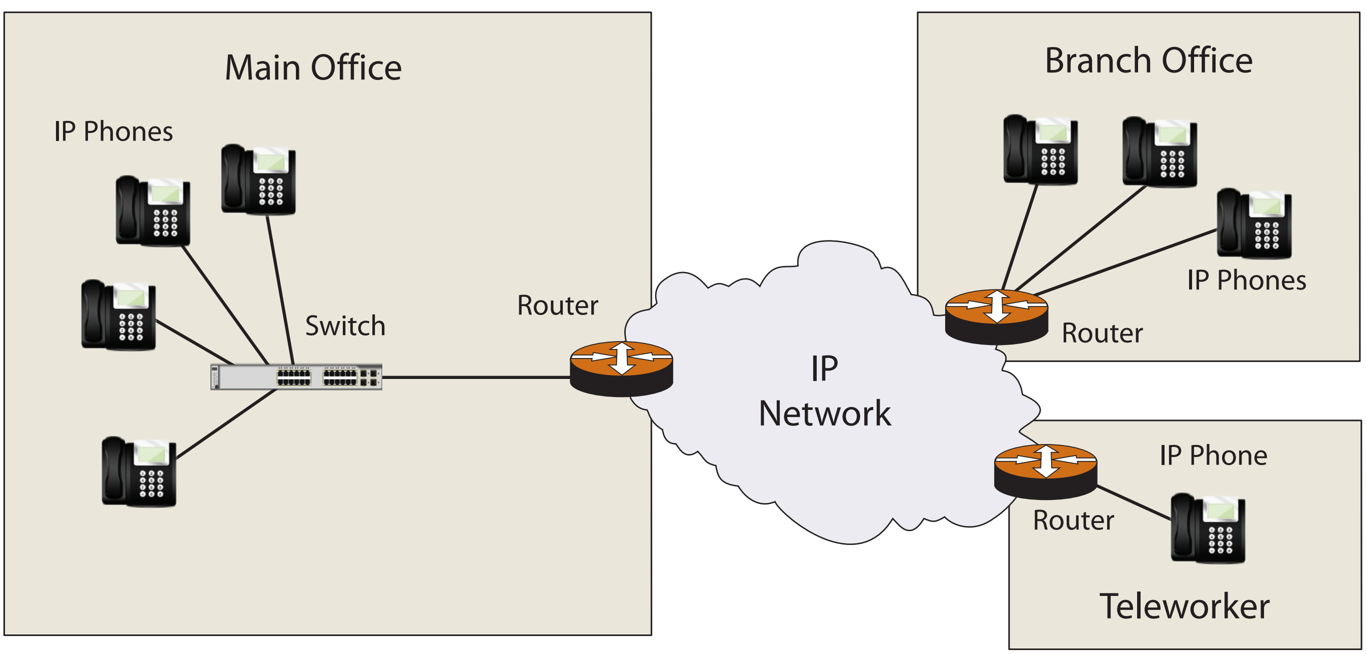

***Figure 1**: A typical Enterprise IP telephony network*

3. Analog/TDM Interface Problems

- **Echo** — Echo commonly occurs at the boundary between the digital network (VoIP or TDM) and analog local loops. This becomes very obvious and annoying with the additional delay introduced by the IP network problems described on the previous page.
- **Signal Level** — Abnormally high or low voice signal levels, "clipping," excessive noise, and echo may occur due to incorrectly configured gateway signal levels.

Network architects and managers should address call quality and performance management problems when they plan and deploy their IP networks, but they should be aware that these problems may also occur during normal day-to-day network operation after deployment.

Many VoIP-related problems are transient in nature and can occur anywhere along the network path. For example, a user accessing a file from a server may cause a period of congestion lasting a few seconds. This in turn can cause short-term degradation in call quality for other users on the network. Thus it is essential that network managers use performance management tools that are able to detect and measure these types of network impairments.

The transient nature of IP problems also means that they are not easily detected or reproduced. In contrast to traditional POTS, problems are not necessarily associated with specific cables or line cards – they can occur randomly due to the "collision" of several different factors. Network managers could attempt to use packet loss and jitter metrics to estimate call quality; however, these metrics alone do not provide an accurate measurement of user-perceived quality, nor enough diagnostic information to determine the cause of the problem.

Network managers use probes and analyzers located at specific network points to help detect and diagnose VoIP performance problems; however, it is not cost effective to place probes on user desktops, in small branch offices or teleworkers' homes.

VoIP Network Requirements & Pre-Deployment Testing

Data applications are not sensitive to real-time transmission problems like jitter, delay or congestion. VoIP performance, however, is very vulnerable to these problems. Before deploying VoIP, network managers need to carefully assess the IP infrastructure that will support the new service. In addition, they should be prepared for increased network traffic and potential congestion once their VoIP network is active.

When moving to VoIP, Enterprise network managers and architects should take the following steps to ensure success.

Step 1: Assess Inter-Site Connectivity

Many problems that affect call quality occur in access links or on limited bandwidth WAN or VPN links. Insufficient bandwidth can result in jitter or delay on inter-site connections, which may cause significant problems during and after VoIP deployment. Budget bandwidth usage between sites and verify that routers can prioritize RTP traffic.

Step 2: Assess Desktop Connectivity

Use a Gigabit (or at least a 100BaseT) Ethernet architecture. Even with the use of Ethernet, problems can still occur. Examine Ethernet switch statistics for evidence of packet errors or excessive collisions and upgrade equipment accordingly.

Step 3: Pre-Deployment Performance Testing

Before deploying the network, verify network performance using a pre-deployment tool such as Telchemy's DVQattest® to generate a realistic level of simulated traffic. In addition to analyzing performance, these tools can also highlight problems such as firewall configuration and calls with one-way media.

Step 4: Pilot Trial

After completing the previous three steps, conduct a pilot trial for the network. At this stage, the trial should be successful. Any problems during the pilot trial are normally the result of an incorrectly configured router or other network equipment rather than a network performance problem.

Step 5: Deploy "Live" Network

As deployment ramps up, carefully monitor any trends in key VoIP performance parameters, note any call quality degradation and investigate accordingly.

Step 6: Maintain VoIP Network

Because VoIP networks are constantly changing and evolving (e.g., network configuration, equipment configuration and network traffic), continued monitoring and maintenance of the network is essential to maintain quality voice service after its initial deployment.

To deploy and maintain a VoIP network successfully, network managers need a performance management infrastructure in which IP telephony issues and call quality problems can be quickly identified and resolved.

VoIP Performance Management Architecture

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

The framework relies on SIP PUBLISH (RFC 6035)[1] and RTCP XR (RFC 3611)[2] QoE reporting protocols that are able to send data back to network management and call control systems with minimal network traffic overhead. The architecture features lightweight embedded agents in IP phones and gateways, with optional high performance network probes deployed within the core network and at major customer locations to provide in-depth diagnosis when

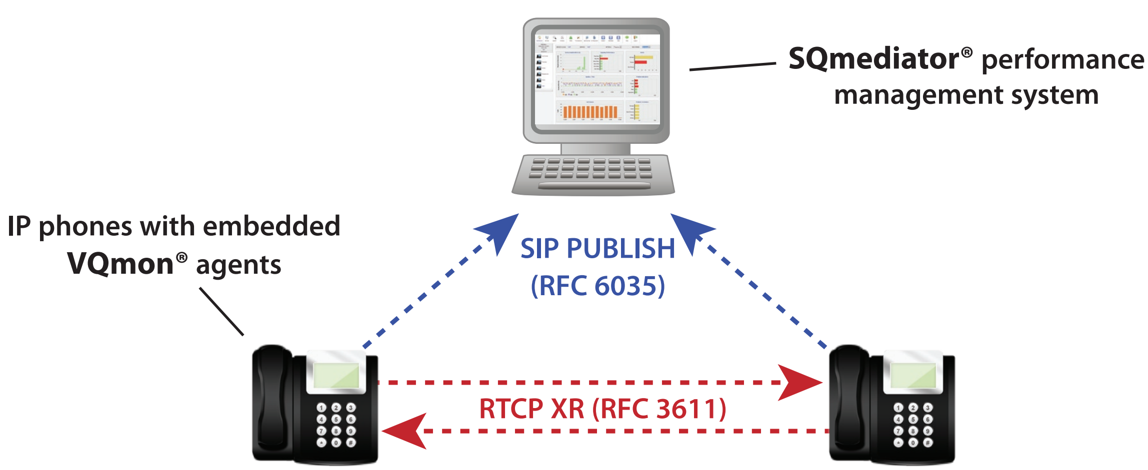

Figure 2: *QoE reporting using embedded software agents*

problems are detected. Embedded agents such as Telchemy's VQmon® provide quality and performance metrics directly from each user's phone or desktop and are extremely cost effective—most IP phone vendors include such agents as a standard feature, at no additional cost.

Figure 2 shows a typical scenario in which VQmon performance monitoring agents are embedded into SIP IP phones at each end of a VoIP call. The agents exchange RTCP XR reports with each other, and send SIP PUBLISH reports to a central performance management system, Telchemy's SQmediator®. (Following sections provide more detail on these protocols and their supported performance metrics.)

Embedded Monitoring Function

VQmon technology is a major building block of the VoIP Performance Management Framework. It was the first and is the most widely deployed monitoring function for VoIP performance management today, and formed the basis for the metrics used in the RFC 3611 and RFC 6035 protocols. VQmon's embedded monitoring technology enables network managers to see call quality problems in real time and identify the root cause of problems for both active and completed calls.

Key features of VQmon include:

- Accurate Mean Opinion Scores (MOS)
- Supports wide range of narrowband, wideband, and ultra-wideband codecs
- Unique perceptual model that incorporates the effect of transient impairments
- Small code size and tiny computational load

VQmon measures key characteristics of the packetized voice 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 a low reporting volume (typically one report per endpoint at the end of a call), thus using minimal 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,

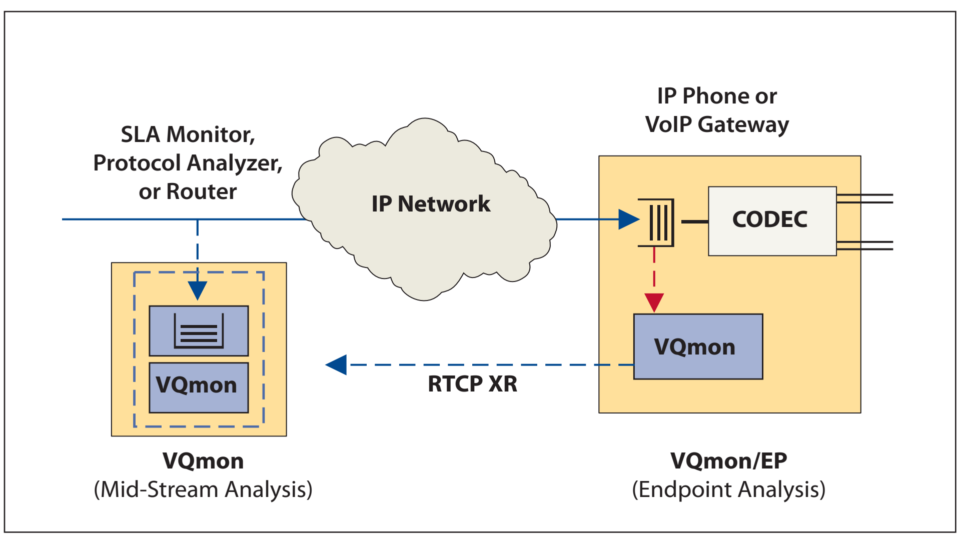

Figure 3: *VQmon's embedded monitoring technology*

routers and generally any system that is on the path taken by the voice packets (Figure 3). VQmon agents are small, 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, for a very wide range of codecs. 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 is an advanced VoIP perceptual quality estimation algorithm that incorporates support for key international standards including ITU-T P.564[3], ITU-T G.107[4], ITU-T G.1020[5], ETSI TS 101 329-5 Annex E[6], IETF RFC 6035 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).

Two versions of VQmon are available for VoIP performance management: **VQmon** for mid-stream performance analysis and **VQmon/EP** for endpoint analysis. Both produce a rich set of diagnostic data and support the common VoIP performance management metrics. VQmon is modular software that can also provide real-time network data analytics and video performance analysis for adaptive video streaming over HTTP, IPTV and IP Videoconferencing services.

SLA Monitors, Analyzers, and Probes: VQmon

VQmon is the core VoIP analysis software used in many probes, routers, SLA monitoring systems, and analyzers. VQmon monitors the packet stream, automatically recognizing individual call streams and the types of codec in use. VQmon identifies packets that were lost or will be discarded due to excessive jitter and uses this data to estimate call quality.

IP Phone and Gateway Monitoring: VQmon/EP

VQmon/EP is widely integrated into IP phones and gateways produced by leading equipment manufacturers, providing accurate call quality scores and supporting diagnostic data. Both VQmon and VQmon/EP use the same quality measurement algorithm, ensuring that call quality metrics are fully consistent.

Common VoIP Performance Metrics

The 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. These metrics are:

- **Packet Loss Rate** — percentage of packets lost by the network
- **Packet Discard Rate** — percentage of packets discarded by the jitter buffer due to late arrival

Packet loss and discard 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.

- **Burst Length/Burst Density** — mean length and density of "burst" periods during which the packet loss/discard rate is high enough to cause audio quality degradation
- **Gap Length/Gap Density** — mean length and density of "gaps" between bursts, when there is little or no packet loss/discard occurring

Burst and gap metrics help to identify the extent to which a call is degraded by packet loss and discard, and help to measure the distribution of loss/discard impairments over the course of the call. 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** — delay occurring between VoIP endpoints
- **End System Delay** — delay occurring within a VoIP endpoint

Round-trip and end system delay 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 occurring at call endpoints. These metrics are essential when managing networks that use secure RTP.

- **Call Quality Metrics** — QoE scores in either R or MOS scaling

R factors and MOS values provide an immediate indication of call quality. If it is apparent (due to low scores) that there is a problem, then other metrics can be used for diagnosis.

- **Jitter Buffer Configuration —** jitter buffer size and type (fixed vs. adaptive)
- **PLC Algorithm —** the technique used (if any) to conceal the effects of packet loss on speech quality

Jitter buffer and PLC information is used to determine whether poor call quality is due to incorrect configuration of the end system, and to allow mid-stream probes to determine endpoint configuration.

Performance Management Reporting Protocols

Reporting protocols have been developed for the media path, signaling system, and network management. It is important to note that these approaches are complementary.

RTCP Reporting Extensions (RTCP XR)

The RTCP XR protocol (RFC 3611) 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 the ability to pass transparently through firewall routers
- Supports the diagnosis of echo-related problems
- Enables mid-stream network probes to obtain analog signal, noise and echo measurements from call endpoints
- Compatible with the Secure RTP security framework (RFC 3711)

QoE reporting using SIP (RFC 6035)

RFC 6035 is a widely supported protocol that allows a SIP phone to send a call quality report at the end of a call to a "collector" such as Telchemy's SQmediator. This report contains metrics that include MOS, R factors, IP packet loss, jitter, burst metrics and other data.

RFC 6035 supports end-of-call reporting, interval reporting and alert-based reporting. It is strongly recommended to use end-of-call reporting for day-to-day quality monitoring. Interval reporting can drastically increase the volume of reports being sent and is generally used for testing or diagnostic purposes.

RFC 6035 can use either the SIP PUBLISH method or the SUBSCRIBE-NOTIFY method. The PUBLISH method (see Figure 4) is the most widely used;

Figure 4:
Example end-of-call report using SIP PUBLISH (RFC 6035)

```
PUBLISH sip:collector@example.org SIP/2.0
 Via: SIP/2.0/UDP pc22.example.org;branch=z9hG4bK3343d7
 Max-Forwards: 70
 To: <sip:proxy@example.org>
 From: Alice <sip:alice@example.org>;tag=a3343df32
 Call-ID: 1890463548
 CSeq: 4331 PUBLISH
 Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER,
  SUBSCRIBE, NOTIFY
 Event: vq-rtcpxr
 Accept: application/sdp, message/sipfrag
 Content-Type: application/vq-rtcpxr
 Content-Length: ...

 VQSessionReport: CallTerm
 CallID: 6dg37f1890463
 LocalID: Alice <sip:alice@example.org>
 RemoteID: Bill <sip:bill@example.net>
 OrigID: Alice <sip:alice@example.org>
 LocalGroup: example-phone-55671
 RemoteGroup: example-gateway-09871
 LocalAddr: IP=10.10.1.100 PORT=5000 SSRC=1a3b5c7d
 LocalMAC: 00:1f:5b:cc:21:0f
 RemoteAddr:IP=11.1.1.150 PORT=5002 SSRC=0x2468abcd
 RemoteMAC: 00:26:08:8e:95:02
 LocalMetrics:
 Timestamps:START=2004-10-10T18:23:43Z STOP=2004-10-01T18:26:02Z
 SessionDesc:PT=18 PD=G729 SR=8000 FD=20 FO=20 FPP=2 PPS=50
                  FMTP="annexb=no" PLC=3 SSUP=on
 JitterBuffer:JBA=3 JBR=2 JBN=40 JBM=80 JBX=120
 PacketLoss:NLR=5.0 JDR=2.0
 BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500 GMIN=16
 Delay:RTD=200 ESD=140 SOWD=200 IAJ=2 MAJ=10
 Signal:SL=-21 NL=-50 RERL=55
 QualityEst:RLQ=90 RCQ=85 EXTRI=90 MOSLQ=4.2 MOSCQ=4.3
   QoEEstAlg=P.564
 RemoteMetrics:
 Timestamps:START=2004-10-10T18:23:43Z STOP=2004-10-01T18:26:02Z
 SessionDesc:PT=18 PD=G729 SR=8000 FD=20 FO=20 FPP=2 PPS=50
                  FMTP="annexb=no" PLC=3 SSUP=on
 JitterBuffer:JBA=3 JBR=2 JBN=40 JBM=80 JBX=120
 PacketLoss:NLR=5.0 JDR=2.0
 BurstGapLoss:BLD=0 BD=0 GLD=2.0 GD=500 GMIN=16
 Delay:RTD=200 ESD=140 SOWD=200 IAJ=2 MAJ=10
 Signal:SL=-21 NL=-45 RERL=55
 QualityEst:RLQ=90 RCQ=85 MOSLQ=4.3 MOSCQ=4.2 QoEEstAlg=P.564
 DialogID:1890463548@alice.example.org;to-tag=8472761;
    from-tag=9123dh311
```


it allows the IP phone to "publish" the message containing the QoE metrics to the collector. The SUBSCRIBE-NOTIFY method requires the collector to subscribe to each device in order for the device to send QoE metrics in a NOTIFY message.

Applying the Performance Management Architecture To Enterprise Networks

The VoIP Performance Management Architecture is essential to provide real-time visibility of user perceived quality in an Enterprise network (Figure 5).

- VQmon is integrated into the IP phones on each user desktop at the main and branch office locations and at teleworkers' homes. During an active call, quality is measured at the IP phone at each call endpoint.
- During the call, RTCP XR (RFC 3611) messages are exchanged between the IP phones, ensuring that each phone has information about call quality in both directions.
- At the end of the call, the performance metrics locally measured by VQmon, and those received from the remote endpoint, are formatted into an RFC 6035 message and sent to the VoIP performance management system (e.g. SQmediator).

Problem Resolution, Detection & Diagnosis

The VoIP Performance Management Framework provides the basis for detecting and diagnosing different types of call quality problems.

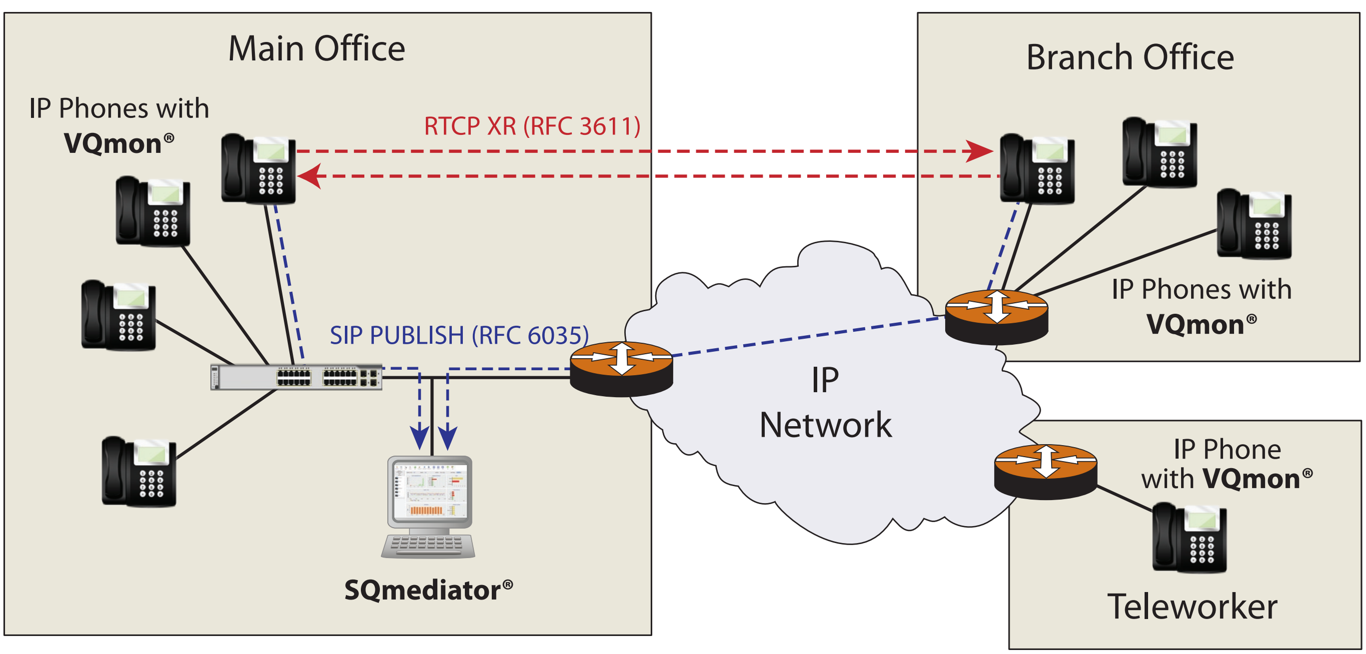

Figure 5: *Performance Management Architecture for Enterprise Networks*

- **Access Link Congestion** — If the access links to the IP network site have insufficient bandwidth, the jitter level will increase. This will typically cause IP phones to discard a high proportion of packets and can increase the size of the jitter buffer, adding more delay.
- **Bad Ethernet Segment on LAN** — If Ethernet switches are not configured properly or Ethernet segments are too long, there may be a high rate of packet loss that severely impacts call quality.

Acronyms

CPE	Customer Premises Equipment
ETSI	European Telecommunications Standards Institute
IETF	Internet Engineering Task Force
IP	Internet Protocol
ITU	International Telecommunications Union
LAN	Local Area Network
MOS	Mean Opinion Score
PLC	Packet Loss Concealment
POTS	Plain Old Telephone Service
PSTN	Public Switched Telephone Network
QoE	Quality of Experience
QoS	Quality of Service
RFC	Request for Comments
RTP	Real Time Protocol
RTCP	RTP Control Protocol
RTCP XR	RTP Control Protocol Extended Reports
SIP	Session Initiation Protocolt
SLA	Service Level Agreement
TDM	Time Division Multiplexer, -ing
VoIP	Voice Over Internet Protocol
VPN	Virtual Private Network

- **Echo in PSTN** — If there is a problem on a remote analog loop, the IP phone on a customer site may experience audible echo.

Measurement and monitoring tools are especially useful for isolating the specific location where poor quality calls are occuring—for example, whether they are affecting all calls in a particular area, or limited to one defective device.

Using the example in Figure 5 on the previous page, a network manager could use the VoIP Performance Management Framework—and in particular, perceptual call quality scores—to determine whether all calls in the branch office are experiencing quality degradation, or if the bad calls are occurring on one particular IP phone. Once the scope of the problem is determined, additional metrics (such as those described in the **Common VoIP Performance Metrics** section on page 6) can be used to diagnose the root cause.

Summary

The VoIP Performance Management Architecture provides an ideal solution for quality monitoring and problem diagnosis for Enterprise IP telephony service. Important call quality and diagnostic data is obtained in real time from key points on the packet stream and routed to network management systems.

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 Performance Management Framework as well as the use of VQmon for call quality analysis.

Many of the industry's equipment manufacturers have already implemented elements of the management framework in IP phones, gateways, routers, probes and analyzers.

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About Telchemy, Incorporated

Telchemy® is the global leader in Voice and Video over IP performance management technology with its VQmon®, DVQattest®, SQprobe® and SQmediator® families of service quality monitoring and analysis products. Telchemy has led the use of embedded software probe technology and the application of big data and analytics for VoIP performance management, and is positioned to be a leading provider of voice and video performance monitoring technology for the emerging SDN. Founded in 1999, the company has products deployed worldwide and markets its technology through over 140 leading networking, test and management product companies.

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