NetQoS
Performance Experts

WHITEPAPER

VoIP Quality of Experience

Ensuring Your Network Supports Optimal VoIP Performance

More than any other type of application, Voice over IP (VoIP) requires optimal network performance to deliver an acceptable quality of experience for users. There are many reasons why VoIP is so demanding. Unlike most data applications, VoIP is sensitive to latency, jitter, and packet loss. The maximum allowable latency of 150 milliseconds is quickly consumed by codecs, serialization delays, router queuing, and by transmission delay between locations. Firewalls and network address translation add latency to the processing of each packet. Network elements induce jitter by temporarily storing packets in buffers and queues. Congestion or the misconfiguration of Quality of Service (QoS) policies may result in bursts of packet loss. In short, VoIP is the most demanding application in your portfolio requiring optimal network performance, and the highest possible quality of service levels to deliver an acceptable quality of experience. As such, VoIP and its related applications can present a considerable challenge for those charged with managing the performance of business-critical data applications running on your converged enterprise network.

Converged Management for Your Converged Voice and Data Network

There are many performance monitoring tools on the market—some help you monitor and manage network performance, others focus only on VoIP applications. However, few provide the integrated metrics and visibility required to deliver a superior VoIP quality of experience while ensuring network performance. By taking a performance-first approach to managing both VoIP and your network using NetQoS products, you will be more successful in deploying and supporting VoIP, while protecting overall network and application performance.

Network Performance Management Tools

Achieving your goals for network and application performance starts with the tools you select. Traffic analysis tools, such as NetQoS ReporterAnalyzer™, leverage Cisco IOS® NetFlow data to give your network operations group insight into the composition of traffic across major links, broken out by protocol and by class of service. These products allow you to monitor for the proper assignment of Type of Service (ToS) settings to the traffic that is highest priority (which should always include voice). They also enable you to monitor the unmarked, low priority traffic to establish whether any high-priority VoIP traffic is mistakenly included in that class of service. And, they allow you to profile and monitor the overall amount of VoIP traffic compared to other protocols on network links.

Some infrastructure monitoring products, such as NetQoS NetVoyant®, use SNMP to shed light on network performance issues. In addition to providing data on network device health, SNMP is the access method for two critical pieces of network performance information: Cisco IP SLA and Cisco Class-Based QoS (CBQOS).

As a built-in capability of Cisco routers and other devices, IP SLA sends synthetic network transactions from one device to another, and returns data about the observed performance. IP SLA supports a variety of tests, including the emulation of voice traffic for several different codecs and circumstances. While not a measure of actual production traffic, it provides a consistent and repeatable characterization of the performance of any given path in the network.

CBQOS is a complex MIB which contains information about the performance of the class-based queuing mechanisms in a Cisco router. With this information, you can verify whether or not packets are being dropped under load, and that any packets dropped are from the lowest priority traffic queues, for example.

VoIP Quality of Experience Monitoring Tools

There are far fewer options when it comes to managing VoIP performance. Often, network managers must infer VoIP performance from infrastructure performance metrics. However, monitoring VoIP call setup and call quality is critical; monitoring VoIP performance without measuring actual call quality is like preparing a recipe without tasting the result.

And, with the increasing adoption and strategic value of VoIP applications, the need for VoIP performance management is not likely to change. It has long been the quest of networking professionals to tie the network infrastructure clearly to business issues to position themselves as an area of strategic importance. Until recently, network investments were often seen as exotic forms of overhead, unrelated in any direct way to key business functions, like Order Entry or

Sales. With the shift to VoIP, this changes. Use of the telephone is essential to business today; consequently, the performance of VoIP instantly puts the network in the limelight. This brings a new layer of possible user complaints. In many cases, it is also the first time the networking organization has primary responsibility for the performance of an application.

Understanding VoIP Quality of Experience

"Quality of experience" is a term that can be applied to what your users are experiencing with a VoIP telephone system. While many aspects of that experience rely on your network's performance, the subjective nature of human use of the telephone warrants understanding these issues as a "layer" in their own right.

Call setup is a good example. Long ago, in the old Bell System, the time it took to get a dial tone was quite long. This has improved greatly over time and today, most users expect to hear the dial tone immediately after picking up the handset. The typical standard for this delay is actually two seconds. If load, congestion, or even latency induced by distance causes an IP PBX to respond more slowly, and the user does not hear the dial tone until four or five seconds have elapsed, they will note this as a problem. Typically, users will hang up the phone after three seconds and try again—repeatedly. And, they'll report the quality of system performance as poor—even if it's actually working pretty well after the brief delay.

The basic audio quality of a VoIP phone call is the other major dimension of VoIP quality of experience. As the joke goes, the poor audio quality of cellular telephones prepared users for Voice over IP. In practice, people will quickly notice and respond to audio problems on a telephone call. Long delays (like those experienced with old trans-Atlantic calls) will cause people to revert to protocols familiar to walkie-talkie users ("Over"), while noticeable echo levels distract people from what they are trying to say.

Mean Opinion Score

To measure both subjective and objective aspects of VoIP audio quality, the Mean Opinion Score or MOS value is the industry-standard measure of audio quality. Virtually all IP telephony equipment uses some form of quality measurement which is mapped onto the MOS scale. The quality measurements take into account the underlying QoS issues like packet loss and jitter, as well as the ability of the phone to compensate (with jitter buffering, for example).

These very specific, non-network-centric measurements for VoIP performance are the key drivers behind the need for specialized VoIP performance monitoring products like the NetQoS® VoIP Monitor. NetQoS VoIP Monitor assesses your users' quality of experience and relates those measurements to network conditions to help you monitor VoIP performance and troubleshoot the source of any problems. It provides comprehensive metrics reporting for MOS, call setup performance (including time-to-dial-tone and time to connect), jitter, latency, and packet loss for every call, to help you to better understand how VoIP is performing on your network. By using NetQoS VoIP Monitor with the NetQoS® Performance Center product suite, you gain global visibility, via a single web-based management console, into all of your applications—data, video, and voice—across your entire network infrastructure.

Addressing the Social Challenges Associated with VoIP

In addition to many technical challenges, the deployment of VoIP is usually accompanied by organizational changes, under the general heading of "convergence." Running voice traffic over an IP network brings together the concerns of these traditionally separate groups within IT. Telecom staff, used to dealing with the traditional PBX and phone company issues, are suddenly working side-by-side with datacom staff, who speak the language of IP and routing.

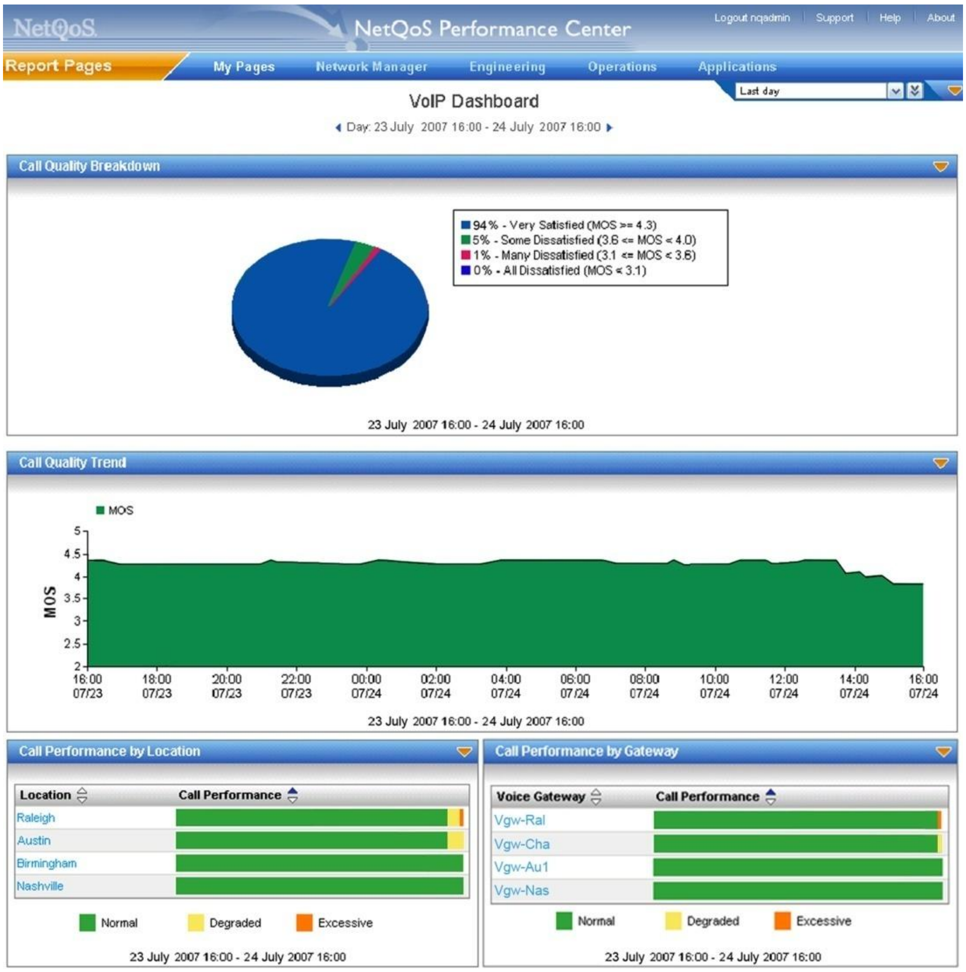
This organizational change plays out differently in different organizations. In some organizations, convergence with IT happens with a high degree of cooperation and success. In other organizations it happens with more finger-pointing. One aspect of the problem is a lack of tools that relate the issues, traditions, and metrics of one group, to the issues, knowledge, and metrics of the other. It's easy to see why VoIP management presents multiple challenges.

Making the Connection between VoIP Quality of Experience and Network Performance in the Real World

With a performance-first management product that looks at the traditional telecom quality of experience metrics, like MOS, and relates them to network-layer problems like packet loss and network paths, it is possible to manage VoIP across technical and organizational boundaries.

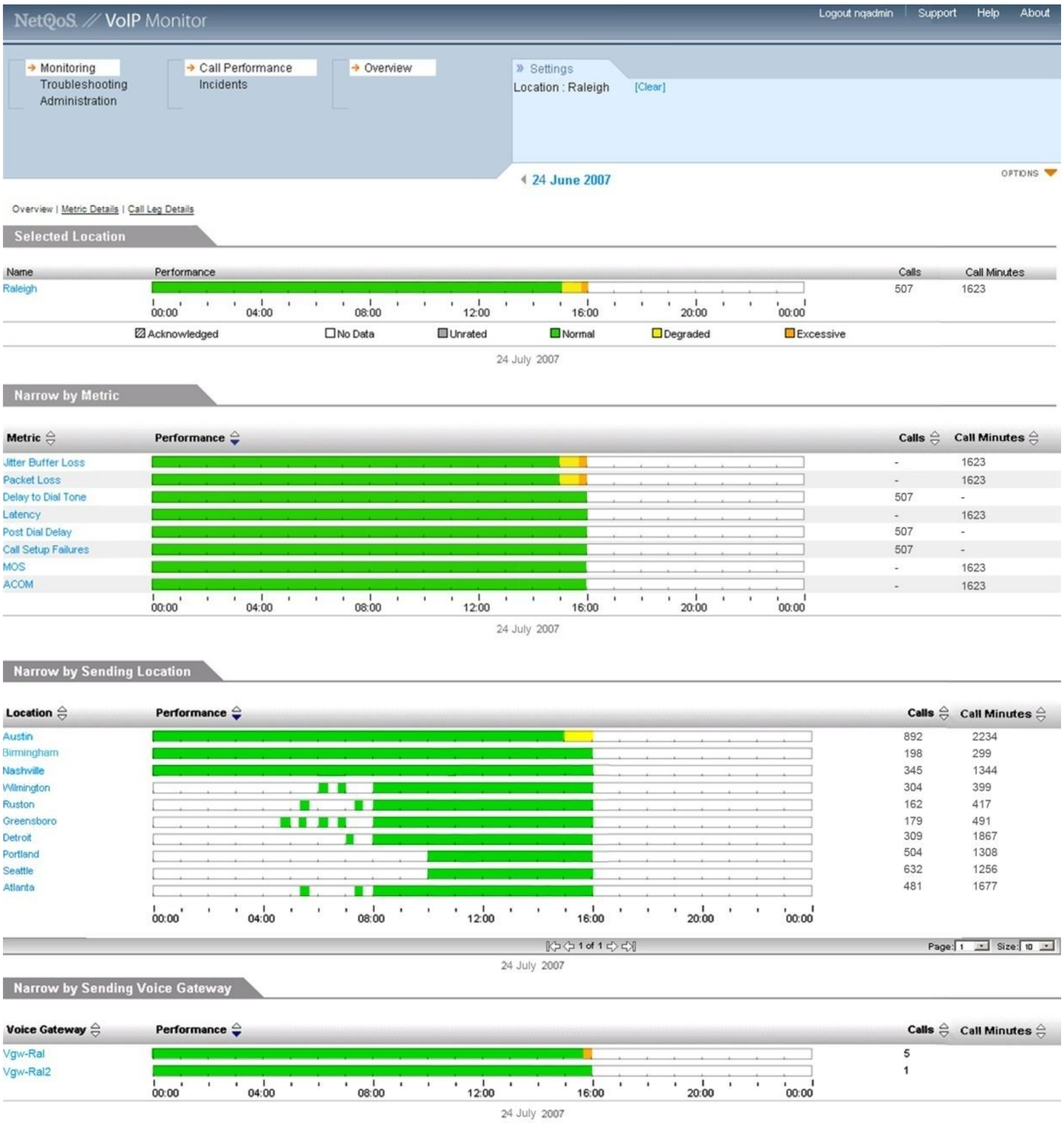
Here's one use case scenario that illustrates how VoIP quality of experience and network performance can be connected. An engineer on the converged voice team of a large services organization is responsible for managing VoIP performance. To quickly identify VoIP problems and pinpoint underlying network issues, he needs access to VoIP application performance metrics such as call quality and call setup.

1) Using the NetQoS Performance Center, and the call quality data available from NetQoS VoIP Monitor, the network engineer notes that call quality overall is slightly impaired and has actually been declining over the past hour.



By glancing at the lower left view which lists the worst performing locations from the complete set of locations in the enterprise, Raleigh and Austin are the suspects for the locus of this problem.

2) A click on the Raleigh link in the NetQoS Performance Center takes the engineer directly to the NetQoS VoIP Monitor product, with Raleigh selected as the location of interest.



The NetQoS VoIP Monitor Performance Overview for Raleigh clearly shows some kind of degradation in the last hour. The metrics summary, which is based on all calls with Raleigh as the location, points to jitter, buffer loss, and packet loss. Austin is the only location with a corresponding quality issue, so, to narrow the search, the engineer clicks on the Austin data.

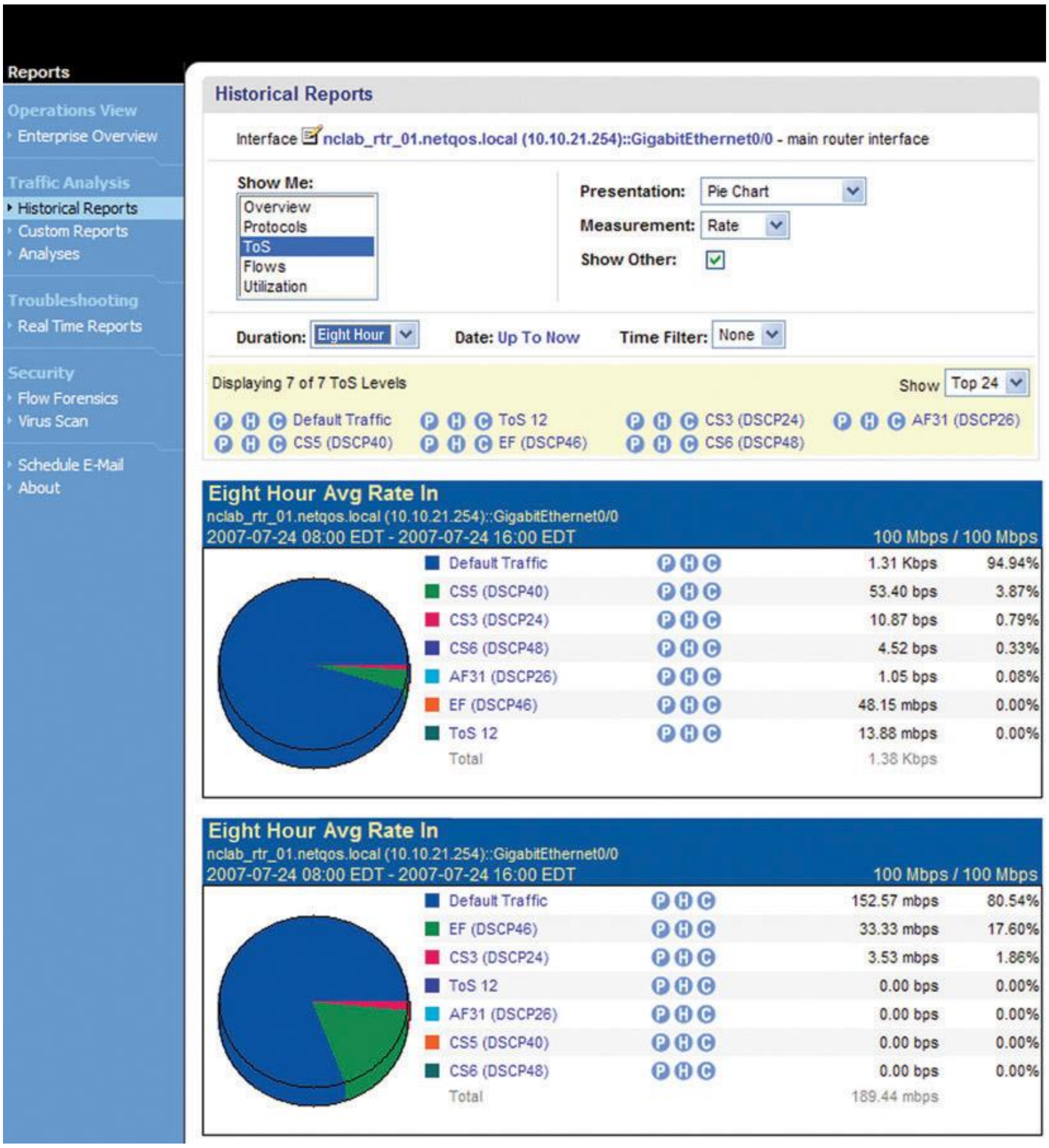
3) By selecting the Metric Detail view, the engineer sees that for Raleigh, it is apparent that packet loss rose in the last hour, while MOS degraded, during conversations with people in Austin. (The Metric Detail page also includes many additional statistics not shown here.)

Packet loss can arise from various sources. Queue discards are a possible source of packet loss, but VoIP traffic runs in the "Platinum" class of service, so there should almost never be any packet loss from the source. Nevertheless, the data warrants further investigation.

4) The network engineer returns to the NetQoS Performance Center to drill into traffic analysis for the interface connecting Raleigh with Austin.

Looking at traffic grouped by class of service (ToS marking), the engineer sees that voice traffic is marked and reported in the highest class of service (EF DSCP46 in the screen below).

However, packet loss could result from a misconfiguration of some kind, so he looks into the protocol distribution on this link for the default class of service—for traffic that receives no special quality of service handling.



5) Looking into the default class of service on this Raleigh-Austin link, the Protocol Distribution for this class of service shows that VoIP traffic recently appeared in this class.

This should never happen, since the latency requirement of VoIP traffic demands the best QoS the network can produce. So the appearance of VoIP traffic in this class is the problem which requires corrective action.

Resolution

Unrelated maintenance or re-configuration of the Raleigh-Austin link one hour ago appears to have inadvertently dropped VoIP traffic from the "Platinum" class of service it requires, to the "default" class of service, resulting in packet drops.

The engineer restores the QoS policy on this interface to the correct values, and VoIP traffic is restored to the needed priority, ending the call quality problem quickly.

Conclusion

The move to VoIP is accelerating with more and more organizations adopting VoIP each day. Telecom and datacom staff are being converged just as networks are being converged. If it hasn't already done so, VoIP may soon impact the performance of your network and the requests made of your network operations group. Finding a way to connect all the moving parts so that the network performance continues to perform well and support new business applications—telephony first among them—is a challenge that requires visibility into new metrics and integration of your existing network performance metrics. Organizations that can successfully connect VoIP quality of experience with network quality of service will be successful in using and supporting VoIP.

About NetQoS

NetQoS is the fastest growing network performance management products and services provider. NetQoS has enabled hundreds of the world's largest organizations to take a Performance First approach to network management—the new vanguard in ensuring optimal application delivery across the WAN. By focusing on the performance of key applications running over the network and identifying where there is opportunity for improvement, IT organizations can make more informed infrastructure investments and resolve problems that impact the business. Today, NetQoS is the only vendor that can provide global visibility for the world's largest enterprises into all key metrics necessary to take a Performance First management approach. More information is available at **www.netqos.com**.

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