Effective VoIP Management

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Introduction

IT professionals are increasingly concerned with application management as organizations come to realize that application performance, which directly affects the ability of users to do their jobs, is critical to business success. But the old strategy of deploying additional bandwidth on the assumption that scarce bandwidth is the major determinant of application performance no longer suffices. The increasing decentralization of employees, the consolidation of IT resources, the emergence of more complex traffic patterns, and the adoption of more demanding applications have all combined to force IT organizations to focus not just on the factors that impact the availability of networks, but also on the factors that impact the performance of both networks and applications. As a result, IT organizations need to determine how they can best measure and manage application performance, with the goal of detecting and addressing problems before end-users complain.

One of the applications that has driven IT organizations to focus on performance is Voice over IP (VoIP). Voice is, by its very nature, a mission-critical application. Based on their experience over the last century of experience with circuit-switched public telephony, users have come to expect nearly 100% voice availability, fast call set-up and excellent quality. However, when run over a packet network, voice does not always perform as well as it does when run on a circuit-switched network. The resulting user frustration with VoIP issues can have a major impact on how the rest of the company views the IT organization. As will be described in this brief, because of the sensitivity of voice to a range of network parameters, adding bandwidth is even less likely to guarantee acceptable VoIP quality than it is to guarantee the successful performance of other key applications. As a result, IT organizations need an application performance management solution that will specifically monitor voice performance.

But even beyond the technical demands of VoIP, an added sense of urgency arises from the increasingly strategic nature of VoIP implementations. The original impetus for VoIP adoption was basically tactical: the promise of low cost calls across a packet-switched WAN. However, recent market research [1] indicates that the tactical motive is receding into the background as companies discover the strategic benefits of VoIP implementation. According to this research, the top benefits of VoIP deployment, in order of importance, are:

- Mobility and flexibility can be provided to employees (strategic)
- Cost of moves/adds/changes will drop significantly (tactical)
- Enhanced/converged business processes (strategic)
- Easier to deploy new integrated/multimedia applications (strategic)
- Deploying enhanced voice functions (strategic)
- Ongoing cost of upgrading and maintaining our traditional PBXs will drop significantly (tactical)
- Employee productivity can be increased (strategic)

Five out of seven of the top benefits are strategic. In addition, a number of tactical benefits that in earlier surveys were listed as a key factor causing companies to deploy VoIP did not make the list of key factors in this latest survey. These factors include lower-cost domestic calls, lower-cost communications operations, lower-cost wiring, etc. This research is just one more indication that VoIP deployment is becoming a more strategic imperative for many companies.

[1] 2005-2006 VoIP State of the Market Report, http://www.webtorials.com/abstracts/2005-2006VoIPSOTMReport.htm

VoIP Characteristics

As noted, VoIP poses unique management challenges for two primary reasons: the new protocols it requires to perform tasks such as call set-up, and its stringent availability and performance requirements. For instance, there are many different coding algorithms available to handle the task of converting a conversation from analog to digital and back to analog again, and both sides of the call must use the same algorithm. The negotiation to ensure that both sides use the same algorithm is handled by a set of protocols that are involved with call setup. This includes H.323, Cisco's Skinny Client Control Protocol (SCCP), MGCP, and increasingly, the Session Initiation Protocol (SIP).

An important consideration is that because of its real-time nature, VoIP almost universally relies on UDP (User Datagram Protocol) rather than on TCP (Transmission Control Protocol). This poses particular problems for voice management, because unlike TCP, UDP does not offer any feedback information about whether or not packets sent have been received nor does it not have the capability to limit transmission in the presence of congestion. Because it lacks flow-control algorithms, UDP based applications can overwhelm a WAN link.

In addition to these protocol challenges, VoIP is extremely sensitive to a number of network parameters and, as noted, users expect virtually 100% VoIP availability, immediate dial tone, and fast call set-up. However, relatively low levels of packet loss can severely impact voice quality. End-to-end delay is also critical. At about 150 ms, voice quality will likely begin to degrade, and beyond 250 ms the link will almost certainly be unusable—levels of latency that have a barely noticeable affect on most transactional applications. Another parameter that must be managed in order to ensure acceptable VoIP quality is jitter. Jitter, which is the variation in arrival time from packet to packet, has little if any impact on traditional transactional applications. Too much jitter, however, can cause unacceptable VoIP quality.

Network Design for VoIP

The end result of these challenges is the requirement for a much more sophisticated approach to both network design and network management. The detailed design requirements of VoIP are beyond the scope of this brief, but it is worth reviewing the high-level considerations that can not only guide a pre-implementation baseline and assessment process, but can also play a part in selecting an effective voice management solution.

The demand for 100% uptime puts particular stress on how IT organizations must design for availability. A two-tier approach, focusing both on avoiding single points of failure at a device level (e.g. N+1 redundancy for power and switching elements) and multiple-path redundancy with fast fail-over capabilities through the network is a best practice for VoIP design. Security best practices, as outlined by both the Voice over IP Security Alliance (VOIPSA) and the National Institute of Standards and Technology (NIST), may mandate separate VLANs for voice or even separate access paths to the WAN.

The primary consideration, however, in designing a network to support VoIP is effective Quality of Service (QoS). With voice this not only means prioritizing voice traffic, but also limiting the bandwidth consumed by other, less time-critical applications such as email or file transfers. Proper QoS management requires three key pieces of functionality. The first piece is effective traffic classification, to discriminate between critical and non-critical applications. The second piece is traffic marking, to inform the elements responsible for QoS enforcement of the status of a given traffic flow. The third piece is management functionality that can accurately monitor the QoS status and assess the results of QoS policies.

Managing VoIP Effectively

The user's expectation of 100% availability and effective call set-up combined with VoIP's sensitivity to network conditions demands a new approach to network management. This new approach has technical and organizational components that are tightly intertwined.

The technical challenge of effective VoIP management arises not only from the need to adopt a performance-based approach to voice, but also from the organizational demands of convergence, the impact of which can be overwhelming if not planned for. Because of VoIP deployment, traditionally separate groups within IT find themselves working side-by-side. Telecom staff, used to dealing with the traditional circuit-switched PBX and phone company issues, must communicate with network engineers whose expertise includes IP and packet-switched routing. How well this organizational convergence plays out depends not only on managerial commitment to cooperation, but also on adopting network management tools that integrate the metrics of interest to each team into a holistic overview. By enabling the management of VoIP across technical and organizational boundaries, the traditional finger pointing that tends to exist between disparate groups can be reduced and potentially eliminated [2].

Enabling the management of VoIP across technical and organizational boundaries requires that IT organizations avoid the all-too-common fragmented approach to network management. This approach generally results from the incremental adoption of point solutions to address new problems on an ad hoc basis. In order to be successful with VoIP, IT organizations need an integrated management solution. This solution must relate voice-specific metrics such MOS (Mean Opinion Score) values to the underlying network behavior that influences these metrics.

To deliver this integration, a voice management solution must, at a minimum, deliver information from three sources: call signaling, NetFlow, and SNMP (Simple Network Management Protocol), and, even more important, relate them one to another. This information is required in order to ensure that the IT organization can manage call setup, ongoing call quality, and the underlying network conditions that affect them both.

Call signaling is typically implemented using one of the previously mentioned protocols: H.323, SCCP, MGCP or SIP. IT organizations need the ability to monitor whatever call signaling protocols are part of their VoIP implementation. For example, during call setup the two ends of the conversation negotiate a common coding algorithm, establish the channels that will be used for transmitting and receiving, and generate a number of status codes. This information can be used to derive important measurements like delay to dial tone, or to detect call failure. Without this data, IT organizations won't have the ability to determine what went wrong if users, for example, start complaining that they can't get a dial tone.

Being able to monitor call signaling also implies being able to receive and integrate data from the IP PBX. This is particularly important for monitoring voice-specific metrics such as MOS. The Mean Opinion Score is a well-understood metric that, although originally derived from the subjective reports of human testers, can now be derived by objective, quantitative measurements. Knowing that the MOS values are dropping is one technique that IT organizations can use to detect poor voice quality before it impacts users.

To relate these VoIP-specific metrics to network conditions, IT organizations need network data. NetFlow or RMON-2 data can give IT organizations insight into the protocol and class-of-service composition of the traffic, a necessity for effective QoS management. And, given the meshed nature of VoIP traffic, the ability of NetFlow or RMON-2 to deliver management data from many points in the network, as opposed to just from a central probe, is critical to successful VoIP management.

SNMP is also a requirement. Not only does it deliver data on the health of the network devices, but it can be used to access data from other sources of information for both data and voice management. For example, in a Cisco-based network, SNMP can give information from both the Cisco IP SLA and the Cisco Class-Based QoS (CBQOS) MIB. Cisco IP SLA generates synthetic transactions that can be used to emulate voice traffic across key links and derive metrics critical to understanding voice quality. The CBQOS MIB gives IT organizations information about the class-based queuing mechanism in a Cisco router, enabling them to ensure that their critical traffic is being treated appropriately when bandwidth is in short supply.

Having all of this data is necessary for success, but it is not sufficient. IT organizations must also have a way of integrating all of the data into a useful overview of voice and network performance. IT organizations should be able to, for instance, detect that MOS values are dropping on the link between headquarters and a major branch office, and bring up NetFlow data from the appropriate network devices to check the traffic composition on those links. This involves answering questions such as are the links being flooded by packets from a scheduled

[2] Eliminating Roadblocks To Effectively Managing Application Performance, Jim Metzler, BCR, January 2007

backup, or rogue traffic from an illicit application? It also involves using SNMP to check, for instance, CBQOS data to make sure the QoS settings are both appropriate and are being applied correctly.

Where To Go From Here

The addition of voice traffic to an IP network poses a number of challenges, both organizational and technical. Key to surmounting those challenges is the adoption of two related strategies that address the demands of a converged network.

As part of the first strategy, senior IT management should carefully analyze the organizational alignment and consider how it may have to change to ensure that the requisite management data flows freely across both technical and organizational boundaries. As part of the second strategy, IT organizations should implement a management solution that integrates key information from both the company's VoIP and network equipment in such a way that the solution avoids the fragmentation that results from the adoption of individual point solutions. This data includes information about the call signaling process as well as a combination of NetFlow and SNMP data.

When evaluating solutions, an important consideration is the ability of the solution to provide a holistic overview that can relate voice quality to network performance, and vice-versa. This holistic overview should be a part of every component of the solution - from the management console to the reports the solution generates. In addition, the solution should be able to generate reports that provide detailed information that further helps IT organizations manage VoIP quality.

Representative Product

NetQoS is one of the leading vendors of VoIP management products. One of the strengths of their VoIP management product, NetQoS VoIP Monitor, is that it monitors call signaling based on actual calls. As a result, if users complain about either a lengthy delay to dial tone or excessive post-dial delay, VoIP Monitor can validate if indeed there is a problem and can help the IT organization identify the root cause of the problem.

NetQoS VoIP Monitor also provides IT organizations with the ability to monitor MOS as a means of ensuring acceptable voice quality. As previously noted, just knowing that the mean opinion score is trending lower is helpful, but that alone does not enable the IT organization to resolve the cause of the degradation. To overcome this limitation, VoIP Monitor provides IT organizations with the ability to link MOS with key network parameters including delay, jitter and packet loss. In addition to merely providing this ability, VoIP Monitor also provides contextual information to help network managers quickly identify and resolve the cause of the voice degradation.

A Word from the Sponsor – NetQoS

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Kubernan™, a joint venture of industry veterans Steven Taylor and Jim Metzler, is devoted to performing in-depth analysis and research in focused areas such as Metro Ethernet and MPLS, as well as in areas that cross the traditional functional boundaries of IT, such as Unified Communications and Application Delivery. Kubernan's focus is on providing actionable insight through custom research with a forward looking viewpoint. Through reports that examine industry dynamics from both a demand and a supply perspective, the firm educates the marketplace both on emerging trends and the role that IT products, services and processes play in responding to those trends.

Kubernan is the Greek root word for *helmsman* as well as the phrases to guide and to steer. As such, the name Kubernan reflects our mission of guiding the innovative development and usage of IT products and services.

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Kubernan Briefs
Vol.1, Number 5

Published by Kubernan
www.Kubernan.com

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