Bridging
the Network Application
Integrity Gap
for VoIP

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Ensuring the availability of mission-critical applications lies at the heart of business productivity. Traditional network management focuses on treating the IT infrastructure as individual silos comprised of network, systems and applications groups. This traditional approach of managing infrastructural resources presents some challenges.

While this approach has helped ensure the availability of infrastructural components (routers, switches, hubs and servers), it has done little to provide a cohesive framework for correlating infrastructural performance with the availability of mission-critical applications. IT staff is held accountable for ensuring availability of applications, and the operations team is charged with ensuring the availability of the underlying infrastructure required to support them. Communication between these silos is often minimal until a problem occurs.

Many times these groups spend endless hours trying to triage the problem and determine ownership of the solution. Therein lies the ultimate challenge, where IT staff struggle with trying to understand how the underlying infrastructure impacts application performance.

"Network Integrity systems are essential to all mission-critical application environments that depend on network uptime," states The Yankee Group.[1] That is why the ideal application integrity management solution will help to bridge the network application integrity gap by correlating the performance of the underlying infrastructure with the performance of the mission critical applications that run across it.

The proliferation of Voice over IP telephony (VoIP) as a mission-critical application has become an attractive alternative to managing data and voice across separate networks. Running a converged network presents many challenges, like balancing high-speed, bursty data requirements with the need to assure high-quality connections for voice conversations. Customers are now demanding robust solutions that can effectively monitor and manage both voice and data performance in a cost-effective manner.

While these challenges are somewhat unique, it is widely understood that although VoIP has its own unique requirements, it is simply "just another IP application."[2] In a recent article published by *Business Communications Review*, an industry journal, the following metrics were deemed instrumental to the successful measurement of voice quality: jitter, packet loss, echo and delay, and voice signal level and noise.[3]

Customers are also realizing that the performance of the underlying infrastructure is paramount to VoIP implementation success. The ideal application performance management solution must be able to bridge the Network Application Integrity Gap for VoIP by providing an integrated solution that can cost-effectively monitor and manage both voice and data traffic. Before examining some of the issues of rolling out a converged network, it is worth highlighting the exponential market growth in VoIP and examining some of the market drivers that are contributing to it.

Network Application Integrity Gap

The Network Application Integrity Gap is based on the concept that business productivity and the availability and performance of mission-critical applications is directly correlated to the performance of the underlying infrastructure used to support them.

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VoIP Growth & Market Drivers

The worldwide VoIP market is exceeding 20% annual growth rate, driven by reduced configuration management costs, richer applications and infrastructure cost savings.

Reduced Configuration Management
Some of the main advantages to rolling out VoIP are reduced complexity, greater standardization and consolidation of valuable resources. VoIP is extremely beneficial in terms of reducing the burden on the IT staff. Traditionally, IT staff was divided along voice and data skill sets. The IT staff of the future will be proficient in managing both voice and data services.

Richer Applications
VoIP paves the way for deployment of richer applications. Mobility combined with simplicity merge together in a VoIP world where the telephone is migrating to an IP-enabled device. This facilitates the roll out of "find-me, follow-me" type services, enabling employees, for example, to access their contact information, check voice mail and join conference calls by way of an IP telephone. The availability of these integrated applications signifies a dramatic shift from the traditional world of Private Branch Exchanges (PBXs).

Cost Savings
Many customers who wish to leverage their existing underlying infrastructure (MPLS or Internet) can realize significant cost savings by combining voice and data over the same global network. Customers can conserve bandwidth by adding additional capacity only when it is needed. With cost savings like this, it is with little surprise that a September 2004 study conducted by Nemertes Research showed that when respondents were asked "are you using or planning to use VoIP?" over 80% responded "now" as their timeframe for doing so.[5] Conclusively, the future of VoIP is now.

VoIP services are taking off!

10% of all voice traffic is now transmitted with VoIP technology (IDC)[4]

80% of enterprises will deploy VoIP over the next 5 years (Gartner Group)

VoIP will account for about 75% of world voice services by 2007 (Frost & Sullivan)

21.9% worldwide growth of the VoIP market by 2012 (Dittberner Associates)

Unique VoIP Requirements

Successful VoIP deployments present a host of unique challenges that must be managed at the applications and infrastructure levels. These challenges need to be managed prior to deployment and on an ongoing basis. The process starts with a VoIP readiness assessment. End-users must also be able to track, as well as troubleshoot, critical performance metrics related to the time-sensitive nature of voice, and manage any quality of service techniques.

Network VoIP Readiness Assessment
Is your network ready for VoIP? One of the first issues to consider when planning a VoIP rollout is to assess the readiness of the network. One of the first basics is to understand capacity requirements, and whether or not the network supports Layer 2 or Layer 3 Quality of Service (QoS). With respect to capacity requirements, it is critical to understand router configurations, port utilization and how bandwidth is currently being consumed. Having a dynamic view of how all applications interact on the network is a key consideration for capacity planning. Interoperability and interdependencies among existing applications - be they mission critical or unauthorized - impact bandwidth requirements.

Realistically, capacity planning should be a continuous process in order to extrapolate trends and plan sufficient bandwidth for new application rollout.

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Measuring VoIP performance has become more standardized based on the performance reporting protocols developed by the International Telecommunication Union (ITU) to include Session Initiation Protocol (SIP), H.323 and Megaco. For these reasons, the ideal application performance solution will provide extensive detail and analysis in the area of traffic utilization and capacity planning, by analyzing bandwidth consumption across the physical, port and circuit layers (Layer 1-3). Key factors include the ability to:

- Monitor and measure VoIP protocol activity, including RTCP, RTP, H.225, H.245, H.323, and SIP;
- Determine the amount of packet voice traffic, categorized by channel and circuit;
- Plan capacity to determine whether a link can accommodate packet voice either currently or with a speed upgrade.

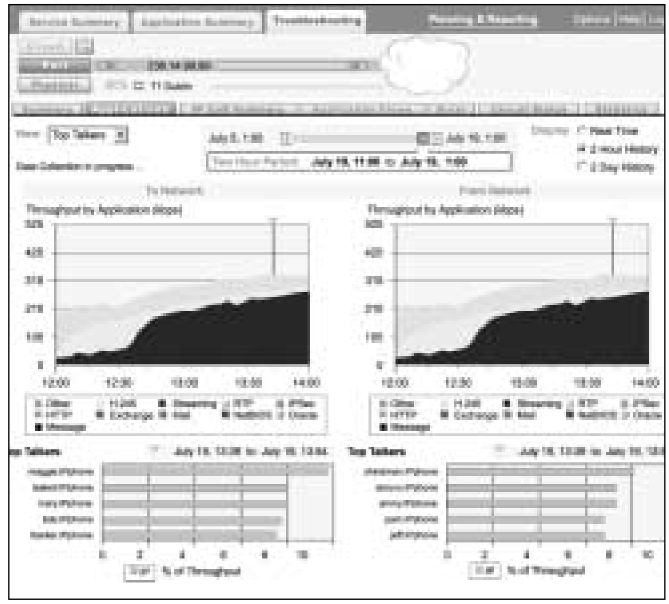

Figure 1

Measure bandwidth utilization and top talkers who are "taking away" valuable bandwidth needed for the Voice (RTP) traffic.

Providing details into how much bandwidth is being consumed for each application is also an integral part of bridging the network application integrity gap **(see figure 1)**. In this example, users watching a basketball tournament are causing a spike in streaming traffic, thereby minimizing the bandwidth available for voice traffic. This is important because the performance of a mission-critical application, in this case, VoIP, is directly related to the availability of resources in the underlying infrastructure. Closing the network application integrity gap depends on the ability to provide application and network visibility on an ongoing basis and then correlate this information into a meaningful framework that drives business productivity.

In a recent survey conducted by *Network World*, over 75% of respondents said their knowledge of the applications running on their network ranged from only "knowing some" to "do not know."[6]
In the same survey, customers were asked to rate their satisfaction with current network monitoring tools. Over 60% replied with answers ranging from "not at all satisfied" to "somewhat satisfied." In the case of VoIP, it is imperative that customers have visibility into each IP phone conversation and each IP flow **(see figure 2)**.

Figure 2

Identify which VoIP users are impacted by others watching the basketball game online.

This level of visibility is required to pinpoint bandwidth hogs and eliminates the frequent assumption that more bandwidth is needed. Providing application flow visibility helps to combat the concerns associated with security breaches by pin-pointing which end-users and locations are infected. IT staff needs to know the answer to three dimensions of the following question: which users are using which applications running on which servers.

VoIP is a peer-to-peer application so this makes it even more challenging.
While other applications tend to be centralized, VoIP requires application visibility at each site. Centralized solutions fall short of this by giving head-end only visibility. Providing visibility on a real-time and historical basis is critical since traffic patterns and changes tend to be more dynamic for peer-to-peer applications like VoIP.

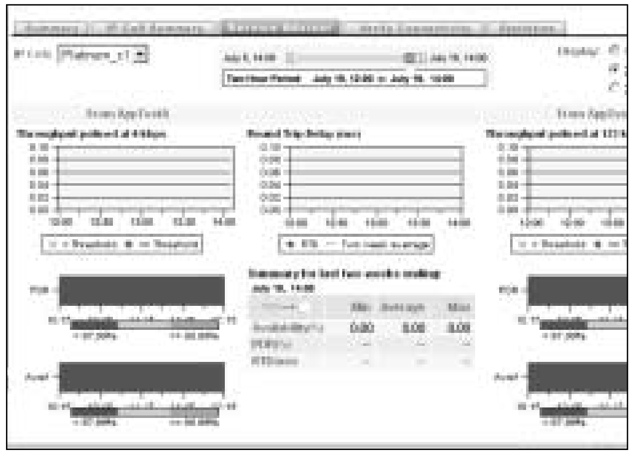

Figure 3 Up-to-the-minute views of SLA parameters including delay, throughput and availability

Several articles have been published on the importance of performance metrics for detecting and monitoring voice/call quality. The quality of voice traffic and effective measurement thereof is a major concern in planning a VoIP rollout. To reap the greatest benefits from VoIP, an enterprise needs the right network foundation so voice quality is not compromised by delay, jitter, packet loss or overall unreliability.

Other metrics include echo, voice signal level and noise. Jitter is extremely important since it is a measurement of the variance in delay. “While data is tolerant of delay, voice is not,” said analyst Steven Taylor and Larry Hettick.[7] The advantage of jitter-passive monitoring is the fact that per call estimates of transmission quality can be made with negligible impact on the service being monitored. Delay is also a key voice performance metric. When measuring and validating SLAs, it is fundamental to measure packet delivery ratio (PDR), availability and round trip delay (RTD) because these metrics are also key to assuring VoIP application integrity **(see figure 3)**.

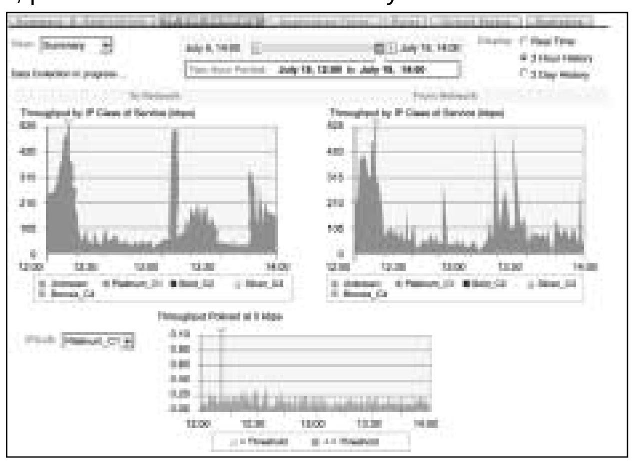

Figure 4 Configure appropriate QoS levels for VoIP traffic.

The ITU recognizes the importance of delay for VoIP. They recommend a maximum one-way delay of 150 ms, or 300 ms round-trip. Round-trip delay is best at no more than 250 ms because conversation quality declines at that point. A VoIP phone typically adds 60 ms of delay, but this figure varies by vendor and the size of the jitter buffer (see "Codecs: The Bandwidth + Delay Equation," Appendix A).[8]

Quality of Service (QoS) Topology and VoIP
The ability to support QoS is a key factor for successful VoIP management. In all cases IT managers need to know voice traffic is getting precedence across the network. Packet voice is typically provisioned on a real-time, Class-of-Service (CoS) network that uses IP differentiated services (DIFFSERV). **Figure 4** shows an example of a VoIP solution, which provides a QoS topology. By supporting the validation of traffic by class of service,

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IT managers can confirm that their routers are properly configured to handle the appropriate QoS levels. This is a key requirement for successful VoIP management.

Integrated Management Solution for Voice and Data
The exponential market growth in VoIP services has created the need for robust performance management solutions which have the ability to monitor both voice and data traffic. Visual UpTime® *Select*™ provides an integrated solution for monitoring both voice and data traffic.

Application performance solutions in general typically use appliances to collect mission-critical application and network data. These solutions must also provide a way of viewing that data through a user-friendly interface that is both intuitive and easy to navigate.

Businesses also need management solutions that perform real-time and historical assessments of network requirements. This is because understanding capacity requirements and performing an assessment of the network are among the first challenges of rolling out VoIP.

Visual UpTime *Select*
Bridge the Network Applicaiton Integrity Gap for VoIP

Visual UpTime *Select* delivers innovative pre-launch VoIP assessment tools and real-time VoIP application integrity monitoring. A flexible licensing structure enables enterprises to add system functionality where and when it is needed.

Why is Visual UpTime *Select* a leading performance management solution for bridging the network applications integrity gap for VoIP?

Integrated Solution for Voice and Data
Visual UpTime *Select* provides visibility into Layers 1-7 via an integrated management solution, monitoring application performance for both voice and data traffic. Visual UpTime *Select*'s Analysis Service Elements (ASEs) are intelligent devices which are deployed as either DSUs or probes, depending on the requirements of the network being managed. These devices lie at the core of the Visual UpTime *Select* solution and report to a single back-end database giving a range of performance metrics and information. This information can be easily viewed through a Web console facilitating easy access from anywhere across the infrastructure.

Assessing Network Readiness for VoIP
Rolling out VoIP can be a daunting task requiring an in-depth understanding of traffic patterns and bandwidth consumption across the entire infrastructure.

Visual UpTime *Select* provides:

- Real-time/zero-wait-time reporting and historical reporting so IT managers can monitor traffic patterns and bandwidth utilization in real time as well as scroll back-in-time;
- Site-specific application and network performance information through instrumentation of each location across the infrastructure.

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As a peer-to-peer application, VoIP requires instrumentation across each site so that IT staff can have visibility into each IP phone conversation and each IP application flow. The Visual UpTime *Select* Apps Flow module offers this visibility **(see figure 2)**.

Performance metrics and SLAs – Benchmarks for Success

The ITU has put forth a standard for VoIP monitoring and performance. Visual UpTime *Select* monitors and measures VoIP specific protocol activity, including RTCP, RTP, H.225, H.245, H.323, and SIP. The ability to do this at both the port and circuit layers equips IT staff with the knowledge needed to analyze the performance of packet voice across the entire infrastructure. To assure successful VoIP monitoring, Visual UpTime *Select* supports an array of passive jitter measurement and SLA metrics. These include end-to-end IP circuit jitter, which reports the daily average of passive jitter measurements on all types of user IP traffic for a selected CoS, and end-to-end IP circuit CoS, service level verification data by reporting on RTD, PDR, jitter and circuit availability for a specified number of days in a month for each class of service.

Providing a QoS topology

One of the key concerns for successful VoIP application rollout is the ability to prioritize traffic across the network. Voice must get precedence in order to ensure an effective quality of voice service. IT staff can benefit from *Select*'s ability to engineer a real-time network (be it Internet or MPLS), by providing such a framework through its CoS module **(see figure 4)**. Put another way if you choose to use MPLS CoS to engineer a real-time network, Visual UpTime *Select* is the solution to meet your needs.

Conclusion

VoIP services have taken off. Successful VoIP implementation and management depends on a network application integrity management solution, which can correlate network and application performance in order to drive business productivity.

Visual Uptime *Select* is the application delivery management solution that can help bridge the network application integrity gap by enabling IT managers to:

- Determine if the network is ready for VoIP;
- Measure performance metrics on an ongoing basis to help the network ensure continued application visibility and business productivity;
- Provide a QoS topology for assuring that VoIP and all other mission-critical applications get the prioritization required;
- Use a single, integrated solution for managing both voice and data traffic.

Information

www.visualnetworks.com or 1-800-240-4010.

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Appendix A

Bandwidth + delay equation

Codec	Total Ethernet Bandwidth	Total Predictable End-to-End Delay
G. 711	85 Kbps	61.5 ms
G. 729	32 Kbps	75.0 ms

Source: "Prepare Your Network for VoIP - Early assessment is the key," *Digital Convergence,* July 8, 2004, by Peter Morrissey.

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