SevOne

Whitepaper

A Guide to Ensuring Perfect VoIP Calls

VoIP service must equal that of landlines in order to be acceptable to both hosts and consumers. The variables that affect VoIP service are numerous and include: touching the network infrastructure, server performance, security, bandwidth, and the threshold KPIs of latency, jitter, packet loss, MOS and R-value scores. Setting up a series of best practices is the key to optimizing VoIP service. Performance management is essential to ensuring that these best practices remain on course continually, and that when any are breached, proactive action is instituted before any degradation of service is experienced.

People have become tolerant of and, some might say, even resigned to sub-optimal service on cell phones, thanks to dead zones and oversubscribed traffic volumes. But just as with standard landline phone service, users expect VoIP calls to be flawless in both transmission and reception. Any communication glitch, no matter how small — from static to breakups, and especially the deadly dropped call — is cause for complaint. Persistent problems with service will lead to a flood of complaints and in the case of service providers, a possibility of account cancellation. Whether you are a VoIP service host or consumer, you want VoIP to work perfectly because you know it can and you insist it must. The secret to getting VoIP service to perform consistently without error or disruption is effective monitoring of all the components that affect its operation.

VoIP is Highly Dependent on Everything Else

Monitoring VoIP is a lot like juggling: you have to keep your eye on the balls, *all* the balls, *all* the time. Flawless VoIP operation involves many interrelated components:

- the functioning network,
- the network infrastructure,
- the servers,
- the security protocols,
- bandwidth consumption,
- VoIP traffic, and
- all other traffic.

To that list must be added all the individual key performance indicators (KPIs) — such as jitter and latency — that directly affect VoIP performance.

If one already has VoIP and it's functioning well, one might ask why he or she is so lucky? One might also ask when will the luck run out? When considering adding VoIP to the network the person in charge of the project might second-guess his or her decision to join what may look like a high-tech circus. The ringmaster who can keep all the acts in all three rings in synch and delivering their specialties well goes by the name of Performance Management.

Under the plentiful and targeted monitoring capabilities of a superior performance management solution, all the components that contribute to flawless VoIP can be made to work together even during peak periods and under critical traffic loads. The key is to put in place a comprehensive set of best practices for each of the components essential to successful VoIP.

Optimizing Network Metrics Is Key to VoIP Performance

At its most basic level, VoIP depends on optimal network performance. Monitoring the metrics by which all VoIP service is measured is necessary to prevent the degree of deviation from the norm that will cause service deterioration. These measurements include latency, jitter, lost packets, MOS, and R-value, all of which must be kept within optimal operating ranges established by a set of best practices guidelines to ensure a quality VoIP experience:

- **Latency** — the measure of time delay in moving packets from the transmitting agent to the receiving agent; the maximum duration of latency that a VoIP system can sustain without deterioration of service is 150 ms in any one direction.

- **Jitter** — a variation in packet transit delay caused by queuing, congestion, timing drifts, route changes and serialization effects on the path through the network; the maximum allowable duration of jitter is 40 ms before deterioration occurs.

- **Lost packets** — the failure of one or more packets to reach their destination across the network; the maximum allowable packet loss is less than 1% for WANs and less than 0.05% for LANs.
- **MOS** — mean opinion score is a subjective measure of voice quality that gives a numerical indication of the perceived quality of the media received; MOS is expressed as a number from 1 to 5, with 1 (bad) being the worst and 5 (excellent) being the best. (Mathematically, 4.41 is the maximum MOS for G711, while the maximum is 4.07 for G729.)
- **R-value** — a quantitative expression of the subjective quality of speech in communication systems for digital networks that carry VoIP or for which VoIP is under consideration; R-values range from 1 (worst) to 100 (best), and the metric is often used in conjunction with MOS, although the R-value is considered a more accurate portrayal of the effects of packet loss and latency.

If any metric falls below the allowable range for VoIP, deterioration is likely to occur. A targeted performance management solution will monitor the levels of these metrics continuously. Successful performance management solutions include alarm features that alert network staff of the developing problem, troubleshooting features that assist in rapid determination of the source and cause of the problem, and corrective features that interact with remedial protocols to initiate actions to rectify the condition.

Inadequate Network Infrastructure Can Kill VoIP Calls

The KPIs important to VoIP are dependent on the makeup of the network infrastructure. Bandwidth capacity, CPU and memory size, routers, switches, servers — all play a part in how VoIP functions on your network. For example, if VoIP traffic or traffic from other applications utilizes a high percentage of your CPU and/or memory, VoIP service is likely to become disrupted or degraded. Similarly, problems in backplane utilization can indicate an overused switch, which can adversely affect VoIP performance.

Congestion on the network causes packet drops or discards and, if high enough, can kill VoIP calls. Buffer errors relate to discards or delayed packets, and a high buffer error rate indicates inadequate memory. Likewise, interface errors are indicators of bad hardware, cable problems, connector issues, etc., and can result in discards. The solution — hardware replacement or upgrades.

Servers, too, can disrupt VoIP service. High CPU utilization on a server is a good indicator of the workload of the CPU overall, and high memory utilization on servers can increase jitter and latency. Use of a server's network card can also spell VoIP trouble: 75% utilization over 15 minutes or discards/errors greater than .05% are detrimental to VoIP. Even the temperature of the hardware can have a negative influence, including anything over 35 degrees C or cooling fans operating higher than 70%. Disk usage, other applications running, database issues, as well as the number of calls — simultaneous, attempted, completed, currently active, in progress, and attempts — all these can have a direct impact on VoIP quality.

Probably the most common component of network infrastructure that can be detrimental to VoIP service is inadequate bandwidth. If the network is using most of the available bandwidth, there are likely capacity issues and a circuit upgrade is probably in the near future.

Unless… a performance management solution can provide insight into how resource allocation can be improved. Bandwidth allocation, component reassignment, traffic control, monitoring optimal server and network infrastructure metrics — these are just some of the ways that performance management can help one do more with the existing network infrastructure and, at the same time, ensure successful deployment of a VoIP application, including identification of needed hardware replacement or upgrades.

How to Identify Network-Related VoIP Problems

There are some straightforward questions about how VoIP is working on the network that will tell users whether problems are likely to be in the future. The little time needed to ascertain the answers to these red flags will help one see how well the network is accommodating the VoIP traffic alongside all the other traffic traversing it:

- *What is the average bandwidth consumption for one VoIP user across the interface?*
- *How many concurrent VoIP sessions does one have per application across the interface?*
- *What is the recommended number of concurrent sessions to configure by CPU? By RAM?*
- *How many locations or address pairs (origin and destination) are in a concurrent VoIP session?*
- *What is the latency associated with one user per VoIP application?*
- *When are VoIP applications mostly used, during normal business hours or off peak?*

Call capacity can easily be determined by first subtracting the average interface bandwidth of non-VoIP related traffic from the total bandwidth; then determine how many G729 or G711 calls can be made from the remainder, using these baseline figures: G729 = 32.2k per call, and G711 = 87.2k per call.

Assessing the responses to these questions against the existing infrastructure of the network will tell how adequate the network is to accommodate the VoIP traffic traversing it during peak business hours and at other times. If existing infrastructure is not up to necessary VoIP needs, one should consider hardware or bandwidth upgrades.

Alternatively, it might only be necessary to manage the performance of the existing system.

Specific Tests to Determine VoIP Quality

Specific tests can assess the quality of VoIP service. These tests can run independently, or automatically with a performance management solution. Either way, there are some basic assessments to be made:

- **ICMP or ping test for each phone** — to establish KPIs, including latency, jitter, packet loss, and MOS score
- **IP SLA tests** — to get an understanding of the health of the paths, capturing voice specific data and what should be expected from a latency, jitter, packet loss and MOS score
- **Qualities of Service (QoS) tests** — to look for errors, drops, and interface utilization, and understand how many QoS queues are configured
- **Network Traffic Assessment** — to determine how calls are routed through the system

Deploying a full mesh of IP SLA tests can be an overwhelming task. If there are 800 sites in the system, for example, the number of IP SLA tests needed to cover a fully meshed environment would be 319,600 (799+798+797+...1). So, subsets are more practical, especially if there is a hub and spoke architecture in which tests from the hub to all attached sites can be performed, and then a full mesh of tests to each of the hubs.

These tests are not one-time events. Quality of VoIP service needs to be assessed on an ongoing basis because the networks that carry the calls are in constant flux. The only foolproof and cost-effective way to execute this constant monitoring of VoIP quality is with a performance management solution.

Optimizing VoIP with Performance Management

Leveraging performance management solutions to evaluate and determine if the network is ready to support a successful VoIP rollout or to assess the quality of existing VoIP service. Remember, the goal is for the quality of the VoIP service to equal that of a landline, so one wants to measure all the factors that affect VoIP described above and ascertain that they are within acceptable ranges. If problems are encountered with any of the metrics, the issues must be addressed either through hardware, software, or bandwidth upgrades. The tests must be run while the network is in typical operational mode to measure VoIP under normal network conditions.

All network performance management solutions are not capable of monitoring all the aspects of VoIP necessary to maintain and deliver superior service, but the performance management solution from SevOne is particularly suited for VoIP. In addition to monitoring the quality of calls, SevOne's solution can perform an ICMP for each phone to set KPIs at the call manager level where SevOne gathers metrics for latency, jitter, packet loss and MOS score.

Being proactive in monitoring VoIP service means relying on alerts from the solution rather than a phone call from a dissatisfied customer to signal something is wrong. SevOne uses the KPIs to establish static and baseline thresholds that alert when breached.

This approach of alerting when any deviation from baseline metrics occurs across any set time duration is a more sophisticated technique of identifying problems than merely setting alerts for any exceeding of peak values.

Unlike SevOne's solution, not all performance management monitoring tools allow for dynamic thresholds based on "normal" performance in the environment.

The SevOne performance management solution also looks at the metrics on routers and switches, and can create call reports to aggregate PBX by source IP, by destination, etc. The solution also monitors the entire network infrastructure the voice packets traverse, which allows for capturing the health of the systems encountered as well as interface KPIs and, importantly, QoS elements, including errors, drops, and interface utilization.

Successful VoIP Starts with Best Practices

SevOne's monitoring of VoIP metrics also allows for capacity planning in addition to identifying trends in data consumption as applied to VoIP in QoS. SevOne uses NetFlow data to create visibility into the traffic and identify whether calls are going through the appropriate QoS and, more importantly, if other flows are doing so that shouldn't. SevOne recommends that voice-only queues be used for both clarity and troubleshooting. Use of NetFlow data also makes troubleshooting easier using SevOne's Metric to Flow capability, which streamlines the troubleshooting process by pre-filtering NetFlow data to only that which is important and relevant.

The best practices that affect the quality of VoIP service described in this paper are key to ensuring that hosts and consumers of VoIP deliver and receive the quality voice communication that rivals and equals that of landlines. The most efficient and safe way to ensure that those best practices are followed and that any breach is identified and rectified before any disruption of service occurs is to utilize a performance management solution. With its highly scalable capabilities, SevOne is uniquely positioned to provide performance management for VoIP systems, from small installations to large-scale systems. Users can see how well the VoIP solution can sound as one imagines the CIO singing your praises for a solid improvement of the VoIP service.

The essence of successful VoIP service is identifying and dealing with a problem before it reaches customers. Doesn't VoIP service deserve the best practices approach that SevOne can provide?

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