



One Click Management™ for VoIP

A White Paper Commissioned by Codima

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The VoIP Management Challenge

As organizations assess and then start to deploy VoIP in earnest on their networks they face two major inter-related challenges. The first challenge is an economic one: how to test and manage the roll-out of a young and sometimes problematic technology in an affordable way? The second is technical: what is the most practical way to monitor and identify the causes of low voice quality for a large number of IP phones and calls.

The technical challenge is significant as even on a medium-size network there are multiple sources of delay, jitter and other QoS problems. It's difficult to identify these in the first place and mostly they are temporary, i.e. the problem is not always present. Even when identified, fixing one or more of these issues may not be enough to solve user quality problems.

Different vendors offer different functions and techniques for VoIP management. Some are quite costly to deploy, and most do not cover all the functionality required for effective VoIP management. What is needed is a practical way to manage a medium-size or larger VoIP installation. A way that is both easy to manage, and yet combines and correlates all the necessary functions required.

It is to tackle precisely these issues that Codima has introduced autoVoIP™ and One-Click Management™.

In general, VoIP management products focus on either an active or passive approach. Simply put, active monitoring systems generate calls and report on their quality. Passive monitoring systems just monitor existing traffic.

The Active Approach to VoIP Management

The active approach is an intrusive approach.

It consists of installing a probe or thin client at different points on the network so that targeted testing of specific network links and point-to-point paths can be undertaken. The local readings generated from these tests are sent back to a central server for evaluation and a perceptual evaluation of speech quality is provided. An example of an active approach is defined in P.862, where a known voice file is sent over a test connection and then the original and impaired files are compared using a scoring system similar to MOS (see box).

Methods for Measuring Voice Quality

Subjective Testing

The "human" method of measuring voice quality is a specialized and costly process which averages the results of human evaluations of call quality to produce a mean opinion score (MOS).

A MOS score ranges from 1 for an unacceptable call to 5 for an excellent call. A satisfactory range for VoIP is between 3.5 and 4.2.

Active Testing

P.861/PSQM (Perceptual Speech Quality Measure) and the newer P.862 analyze the distortion on test voice signals transmitted through a VoIP network to produce an estimated MOS score. It is an intrusive method that requires calls to be set up between specific agents on the network. Its focus is on measuring the quality between two defined end-points rather than individual IP phones or calls.

Passive Monitoring

Passive or non-intrusive monitoring examines a stream of voice traffic and produces a transmission quality metric that can be used to estimate a MOS score. This method monitors all IP phones and calls in a network without any additional network overhead. The method calculates an 'R' factor on a scale of 1-100 based on a number of variables such as equipment impairment, jitter, delay and other elements. Scores of 70 to 94 are considered good with anything less than 50 being unsatisfactory.

Both at a practical and a logical level, the active approach has substantial flaws:

- The probes or agents do not replicate actual call conditions. Only the user or phone can assess those. The probes are a proxy for those conditions.
- The best proxy is to locate a probe or agent behind every data switch that aggregates phones on the network, but this is economically unattractive
- To reduce the deployment impact of the active approach, probes are typically located at the ends of WAN links, where most jitter or delay is expected. In doing so, organizations are partially replicating information already available to them in the form of service-level reports from their service providers.
- Active testing takes up bandwidth

The Passive Approach to VoIP Management

Passive monitoring systems monitor existing traffic. By measuring existing traffic they can statistically extract voice quality metrics such as packet loss and jitter. Their key advantage is that they are non-intrusive. Frost & Sullivan and other industry analysts have reported rising demand for these systems as they are inherently less costly to deploy.

This reduced cost is because of the way the voice traffic they monitor is conducted on the network. The voice set-up protocols, such as SIP and H.323, set up calls via the call server or soft-switch.

Thus locating a passive monitoring system in front of this entity is the obvious place to monitor this process. But what about the phones – surely they speak to each other directly once the call has been set up?

Most VoIP phones, have on-board agents, which use a protocol called RTCP (Real-time Transport Control Protocol). RTCP agents automatically generate voice-quality reports that can be read by a passive monitoring system or other interested parties.

This also means that only one passive monitoring system may be needed to monitor call quality on a single voice network. A key advantage of using these embedded RTCP agents is that they report on the actual user experience, as measured by the phone's agent, for each and every phone and call made on the network. This is in sharp contrast to the active approach of probes and synthetic agents, which report on transactions between two (non phone-specific) end-points. RTCP is thus actually a better

Critical Quality Metrics

Jitter

Jitter is a variation in packet transit delay caused by queuing, contention and serialization effects on the path through the network. Jitter is most often caused by congestion on the network.

Delay

Excessive delay (generally over 200 ms round trip) causes conversational difficulties including making echo obvious. It may be caused by congestion, routing or configuration issues.

Packet Loss

Packet loss tends to be 'bursty' in nature and is most often associated with buffer overflows in network equipment.

measurement than the active approach of what the end-user is experiencing since it is directly generated by the phone.

The passive approach is not without its flaws however:

- An active approach will provide better localisation of problem areas as it measures the results from a specific path through the network as opposed to just the experience of the end-phone-point. This is the major problem of a passive only approach. As we shall see below however, Codima's integrated product does not suffer from this disadvantage.
- Some VoIP phones do not implement RTCP.
- RTCP agents generate do not support all types of quality metrics. Its functionality is extended by RTCP XR (Extended Reports) but very few phones implement this yet.

Clearly RTCP needs to be supported to use this approach and most VoIP phones do implement it. RTCP monitoring can be extended through the monitoring of RTP (Real-Time Transport Protocol) and Codima's multi-port system accommodates this. In general, RTCP monitoring represents a practical and very effective way of monitoring large numbers of IP phones and calls.

The Value of Correlation

Regardless of whether an active or passive approach to VoIP management is taken, neither method on its own is particularly effective at localising the root cause of quality issues. The active approach will indicate that a specific path through the network has quality issues and the passive (RTCP) approach will identify a phone or group of phones that are experiencing quality problems. For more precise problem identification, a number of different functions are required:

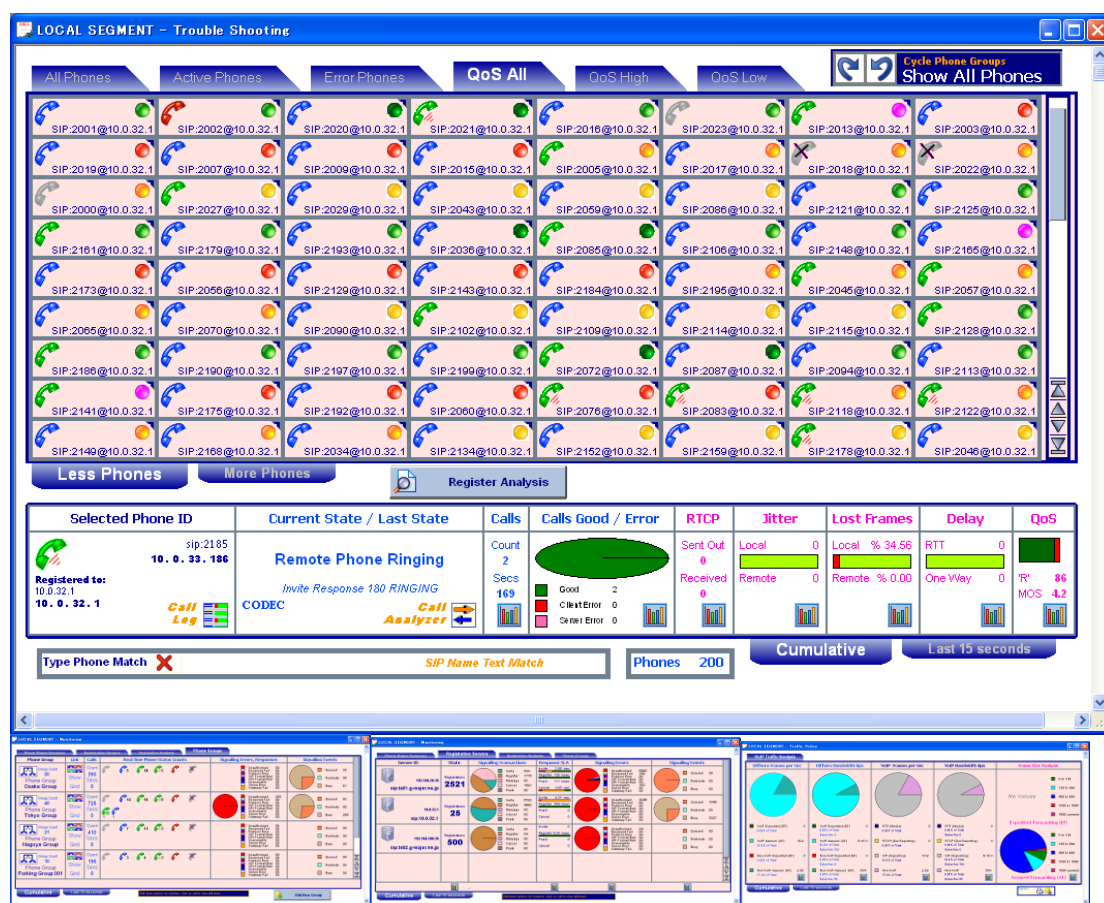
- Internal reports on network elements showing congestion levels, dropped packets and the actual QoS configuration, if implemented.
- A topology discovery system that can identify which network elements are in the actual voice transmission path (and thus the potential source of quality degradation issues).
- Application level response time and transactional analysis for call set up protocols such as SIP.
- General network statistics, particularly regarding access to the call server, so that bandwidth, congestion, response time or number of calls placed can be determined.

These additional functions provide the necessary additional information to localise quality degrading conditions on the network. Isolated sets of information, however, will still not pinpoint the causes of quality problems. For instance, a network switch may be dropping packets or there may be high utilisation of bandwidth on the link that provides access to the call server. Yet neither may actually be degrading call quality. This reveals a problem with management systems that use thresholding to localise issues. Breaching a specific threshold may not have any impact on the problem being targeted, and thus solving the cause of the breach may not lead to its resolution.

What's needed is a way of automatically correlating the different data sets with call quality levels over time. If a management system shows that a certain level of congestion is time-correlated with a specific degradation in quality then, and only then, is a deterministic problem-solving scenario revealed.

The problem for managers with responsibility for VoIP deployments is that most VoIP management systems do not do this. Many products are not in a position to do so as they are only monitoring a small subset of the data required for effective problem identification. In some cases, where parts of the relevant data are available, it is just too difficult and time-consuming to assemble and manually correlate the necessary data sets.

autoVoIP™ and One-Click Management™



Cost of deployment and effectiveness in problem resolution are the two most important criteria for most organisations when choosing a VoIP management system. Codima's autoVoIP™ products addresses these concerns directly. It also introduces the concept of One-Click Management™, which both integrates multiple functions, and makes them extremely easy to use.

The key architectural and design elements that have been incorporated into autoVoIP™ are:

- Automatic monitoring of RTCP, which is a non-intrusive and easy to deploy mechanism to monitor voice calls. It also has the advantage that it reports on end user experiences rather than between test probes placed on the network.
- Passive monitoring of RTP and performance statistics regarding network operations such as bandwidth utilisation, response times and protocol utilisation.
- Integration of SNMP, which is the best way to look at the internal performance records of network infrastructure devices such as switches and routers.
- Integration with a topology discovery function, which is a major time-saver when trying to identify the source of quality degradation issues. It determines which network devices in the transmission path are relevant for problem analysis purposes.
- An automated correlation function, which sifts through the multiple sets of data to achieve a deterministic result for root-cause analysis and problem resolution.
- Zero to minimal configuration from the user.
- Access to all functions by either clicking on a browser-like tab to sort or filter results, or by clicking on an icon to view more detailed reports.

In sum, autoVoIP™ supports both a cost-effective way of gathering the necessary data to monitor call-quality levels and integrates the means to identify the likely causes of call quality degradation. It automatically correlates its sources of data for effective problem determination and with One-Click Management™, truly redefines ease-of-use for a multifunction product.