White Paper

QoS: The Missing Link

Vital VoIP Knowledge for Managing VoIP and UC

By Sue Bradshaw, Technology Writer, Integrated Research and Tom Cross, CEO TECHtionary.com

This white paper takes you on a journey that explains the importance of achieving real time performance of voice and video to deliver quality Unified Communications.

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Introduction

If you're considering implementing Unified Communications (UC) or even if you're completely new to Voice over Internet Protocol (VoIP) this white paper is for you.

Co–authored by Sue Bradshaw of Integrated Research and Tom Cross, CEO of TECHtionary, it will help you translate your years of experience managing servers and applications to be meaningful in the new world of VoIP, IP telephony and UC.

And it *is* a new world if you're from the network, server or application sides of the house. With potentially confusing concepts and terminology, UC also redefines what real time really means.

This white paper takes you on a journey that explains the importance of achieving real time performance of voice and video to deliver quality UC. To assist you further it also includes Tom's Top 10 Recommendations for UC implementations and a Prognosis VoIP monitoring checklist to ensure visibility and control of your UC ecosystem.

About the Authors

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Sue's extensive technology background includes Datapoint Corporation, Memorex Telex and Microsoft where she worked as a technical specialist for almost 6 years.

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Thomas B. Cross

Thomas is the CEO of TECHtionary.com, LyncUserForum.com, Pictagrams.com and CrossPinPoints.com. He has three decades of experience in startups and as a consulting advisor with leading providers and venture capital companies.

This experience includes market planning and development, hardware/software design and development, project management, intellectual property in telecommunications, information technology, conferencing, teletraining, telecommuting, groupware, networks, call centers, internet, artificial intelligence and other fields.

Thomas has also managed the successful development of more than 10 software, hardware and internet products to market and received industry awards for this work. He has authored 13 books, written, produced and directed 15 commercial videos and is creator and producer of the World's Largest Animated Knowledge Source on Technology – http://www.techtionary.com.

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About Integrated Research and Prognosis

Integrated Research is the creator of Prognosis® multi-vendor UC ecosystem management for Avaya Aura™, CS1000, Cisco and Microsoft UC platforms. Prognosis helps you manage the infrastructure needed to implement and maintain successful UC and SIP solutions. For more information visit www.ir.com/uc

When it's Really, Really Real Time

Real-time is different for UC. The closest real-time experience you may have had to date is implementing virtual desktops for application delivery, email and instant messaging where user satisfaction and tolerance are measured in seconds. However, when it comes to measuring voice quality; user satisfaction and tolerance are measured not in seconds, but in milliseconds.

It's hard to imagine how short a millisecond is. A flash from a camera is about a millisecond and a housefly's wings flap once in about three milliseconds.

It takes five milliseconds to blink your eyes.

Because of these tiny increments, even 100-150 milliseconds delay between words or sentences during a conversation are noticeable to us. Traditional analog systems used by businesses for over 100 years have a normal, tolerable latency of 45 milliseconds (ms).

That short delay is not even perceived by the human ear. Considering that an analog call is the benchmark by which VoIP is judged, 75 or even 100 ms of latency is acceptable. So what do you do if latency is greater than this? How do you measure it, identify its causes and resolve it?

Understanding, Identifying and Resolving Latency

From all the research available, delays of more than 250 milliseconds are generally unacceptable to people as they think the connection is lost or start to experience the "can you hear me now?" syndrome. Since the internet is at best 'best efforts' – designing for and managing latency is required even before you begin your UC deployment. That is, it's a good idea to benchmark your network now to see if it is UC-ready. Certainly most core corporate IP networks may have sufficient bandwidth to handle the load but it's often remote offices that will need attention. They may have old cables, switches, endpoints and even power that isn't quite ready for a complete UC loading during peak times of the day.

Latency measurement can be achieved in a number of ways, and the most commonly-used in VoIP is MOS. MOS, meaning Mean Opinion Score, is used to quantify voice quality, and measures latency as one of its most common impairments. Interpretation of individual MOS values is based on a scale of 1-5, where 1 is low and 5 is high. A value of 4 is considered equivalent to toll quality. However, it should be noted that the word 'opinion' is really the view of a person, not of a measurable fact or quality. In other words, MOS is just one set of values, because if the CEO is experiencing stuttering, fluttering, echo and other common IP telephony problems, then it's his or her opinion that really counts!

Once you can measure the latency factor in MOS you can identify where it's coming from. You can achieve this using media-path diagnostics, which maps the calls' paths so you can correlate network performance with voice quality. With the limiting point or points identified, you can analyze the causes and apply forensic-style analysis to the problem.

Often it's an individual phone – known in VoIP terms as an end point, that's at fault. It is also likely to be related to VoIP configuration on the local area network (LAN) or wide area network (WAN). For example, if a router or firewall cannot pass the number of packets presented without delay, and packet loss occurs, it can lower the quality of voice or result in noticeable delays.

A common firewall configuration error is where one user can hear the caller but not the other way round. Resetting the firewall to open transmission control protocol (TCP) ports in both directions often solves this easily. As you troubleshoot these types of problems you also start to realize there are many players in this process. The installer, consultant, firewall manager, active directory (AD) specialist, Exchange specialist, cable/wiring installer, server manager to name just a few. Latency in this sense also includes any "finger-pointing" between the players.

Latency is just one of the factors measured in a MOS calculation. MOS tests for voice are specified by the International Telecommunications Union ITU-T recommendation P.800 and are a subjective measurement of voice quality. MOS is based upon ratings of call quality by a number of listeners who read test sentences over the communications circuit. Some of the suitable English-language phrases suggested by ITU-T are:

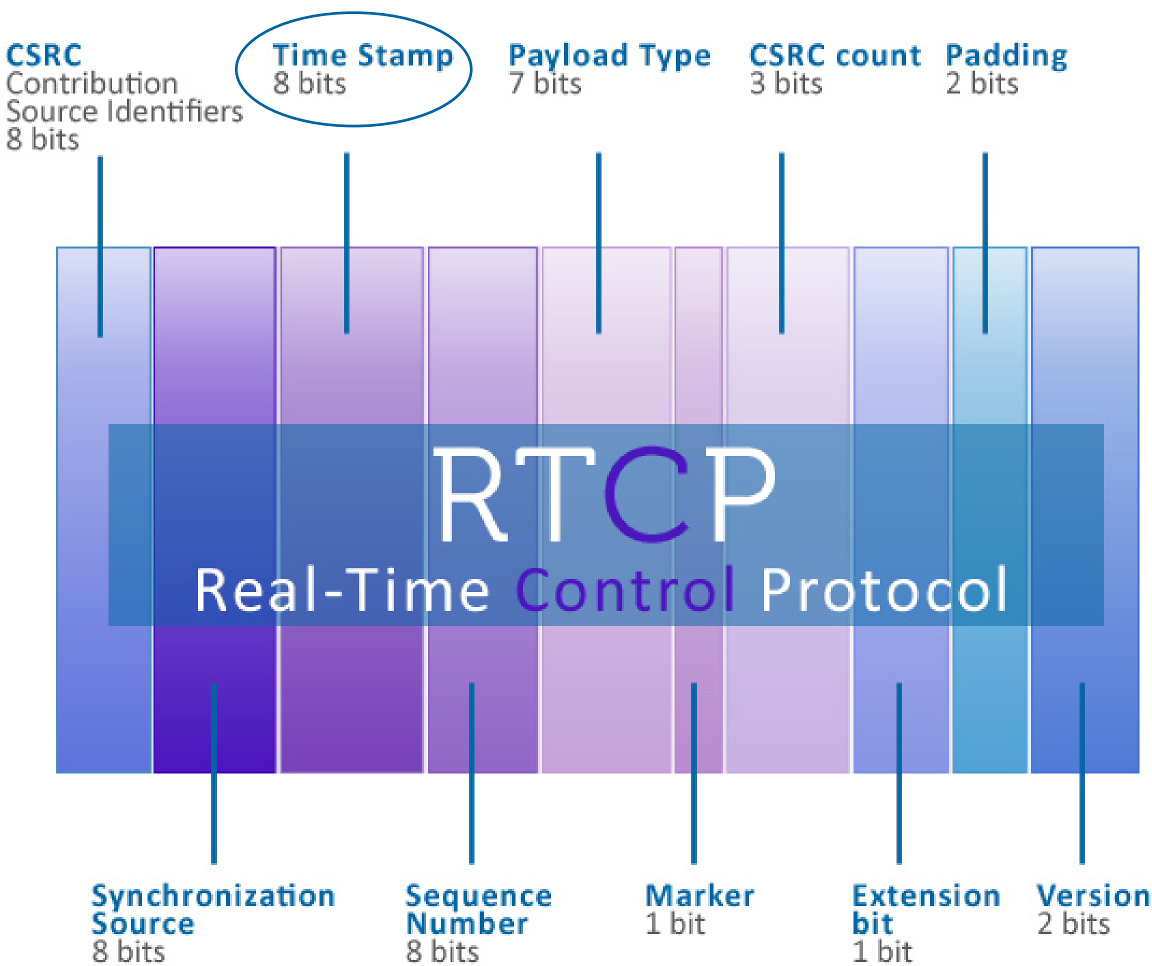
"You will have to be very quiet." "There was nothing to be seen." "They worshipped wooden idols." "I want a minute with the inspector." "Did he need any money?"

The arithmetic mean of the ratings is used to determine the MOS.

To translate this to be meaningful in a computer environment, an algorithm is applied. The algorithm incorporates packet size, jitter, lost packets and latency that are read from Structured Query Language (SQL) data, Call Detail Recording (CDR) database, realtime transport control (RTCP) packets or simple network management protocol (SNMP) Traps depending on the individual VoIP platform.

RTCP Explained

Below is a graphic image of RTCP. We find it helps to show what packets really look like to gain a better understanding of their features and functionality. Some of the packet functions are simple enough. However, for this discussion the time stamp is probably the most relevant.



As you may remember from earlier in this paper, delay or latency can cause voice calls to stutter or video to jitter. If we are to solve this we need to be able to measure the delay in milliseconds. The time stamp in RTCP provides us with this information. Just like a worker punching the time clock, packets are ‘punched in’ as they cross the IP network and if they are late – just like a worker being late, stuttering or jitter occurs.

RTCP also has additional capabilities called RTCP- XR, (eXtended Reports) that give much more detailed information. This information, external to the actual voice, includes room noise, talker echo, one-way transmission delays, jitter buffer size and codecs. The resulting MOS values presented to the VoIP administrator subsequently measure the perceived quality of audio after it has been compressed by a particular codec, transmitted, and decompressed..

Understanding, Identifying and Resolving Jitter

When you’re talking to someone it’s important that they hear what you say in the same order that you say it, otherwise they won’t understand what you’re telling them. Unfortunately, jitter causes packets to arrive at their destination with different timings and sometimes even in a different order than they were sent (spoken), with some arriving faster and some slower than they should.

Jitter is a variation in packet transit delay caused by queuing, contention and serialization effects on the path through the network. In general, higher levels of jitter are more likely to occur on either slow or heavily congested links.

If latency is less than 150 ms and it’s regular, it’s tolerable. Our brains can adjust to the rhythm and compensate for the pauses. But when latency is irregular – as in jitter, it’s harder to tolerate. And it’s not just the irregular delivery of voice that disrupts conversations. Because of the fundamental way packets are sent across an IP network, not all voice packets for a conversation are guaranteed to take the same route. Network congestion and jitter can cause voice packets to arrive unevenly or out of sequence, disrupting the conversation and causing garbled sentences, choppy voice and dropped audio.

To correct the effects of jitter, VoIP endpoints collect packets in a buffer and put them back together in the proper timing and order before the receiver hears them.

Jitter buffers on routers, gateways and end-user devices can help alleviate the problem of irregular packet delivery. The jitter buffer, which is located at the receiving end of the voice connection, intentionally delays the arriving packets so that the end user experiences a clear connection with very little sound distortion. However jitter buffers need to be configured with the correct delay setting. If this is too short – for example 10 ms or less – packets can be discarded needlessly.

Since delays of up to 50 ms aren’t detectable by the human ear, a higher delay setting can resolve this problem. This works, but it’s a balancing act. Processing the buffer adds delay to the call, so the bigger the buffer, the longer the delay. However jitter buffers are just that – a ‘buffer’ against some of the problems from packet delay. If voice packets arrive when the buffer is full then packets are dropped and the receiver will never hear them. These are called lost or discarded packets. In addition, jitter buffers don’t recover packets that are lost in transit between two locations or solve other problems.

Understanding, Identifying and Resolving Packet Loss

Just as it’s important to hear what someone says in the order they say it, it’s also important to hear all of what they’re saying. If you miss one out of every 10 words or 10 words all at once, chances are you may still understand much of the conversation but you may lose the context or be distracted. This is an example of packet loss. Typically network planners and designers assume that there is not enough bandwidth and add more bandwidth. This is expensive, time-consuming, often doesn’t really improve the situation and just adds to the cost.

Diagnostic tools drill down to look at the packet flows over a period of time and give network planners more information. Just like congested automobile networks during ‘rush hour’, network planners and highway planners need to look at demand, not just supply. Similar to providing transit lanes for buses, cabs and hire cars, voice and video need to be given higher priority over email, file transfers and downloads.

Knowing the average packet loss for a call gives you an overall sense of the quality of the call. Small amounts of packet loss do not affect the achieved MOS significantly. It will take 4 percent of packet loss or 230 ms of latency or jitter to reduce MOS to 4, which is still toll-quality speech. However a call with less than one percent average packet loss will always sound better than a call with 10 percent loss.

A Quick Explanation of VoIP Codecs

If you've ever delved into music, video, digital projectors or PC monitors during your IT career, you will have already encountered codecs.

Simply put, it's the compression and decompression of a digital stream or signal, commonly known as a coder-decoder or codec. For example MP3 comes from the Motion Picture Experts Group and is a compression standard for music. Either way, it's an integral part of received signal quality which is why it's so important to voice. In voice, there are greater codec choices, each of which is an inescapable factor in the maximum achievable voice quality.

With today's codecs, the highest achievable MOS is 4.4, using the G.711 codec, which only compresses at a 2:1 ratio and is the one that most closely matches traditional PSTN quality of 64 KBPS used in T1/E1 (transmission level 1 T for US and E for European) transmission. The G.xxx series comprises ITU standards; meaning it is designed for telecommunications services globally. G.711, is usually deployed in a high bandwidth environment, such as a LAN, for two reasons – its low compression ratio (2:1) and it is widely used in nearly all IP telephony devices.

On the other hand, G.729 is more typically used across a WAN or lower bandwidth environment. It has a higher compression ratio (16:1), and is a licensed codec that can still deliver acceptable quality speech transmission. However, in choosing a codec there are several key points. First, it is desirable that both endpoints (phones, headsets or speakerphones) use the same codec. However, they certainly can be and are often different, so most media gateways will perform a compression conversion such as from G.711 to G.729.

The digital signal processing (DSP) during the codec conversion can also add jitter and some latency, so MOS quality can be affected. Second, multiple codec conversions across multiple networks – known as transcoding, can add additional problems.

Transcoding can introduce latency that may affect voice quality and as the packet size decreases, more traffic is generated and the packet processing overhead on the host processor is greatly increased. This may result in broken or one-way voice when too much CPU time is consumed.

Once you know how busy your media gateway is, you can either increase its speed or the connection's bandwidth to accommodate a lower compression codec. You'll then be able to monitor the difference in voice quality. To do this you need a high level view of voice quality, a map of the call's path and drill-down information on the gateway, and if it's a virtualized server, information about other guests running on the host.

You need to ensure that enough processing power is available to handle transcoding without delay. It's useful to be able to see which transcoding devices are idle or in use, and when they are in use. Knowing the percentage of the gateway, transcoding devices and digital signal processes (DSPs) that are used as a percentage of capacity, will help you identify where potential issues exist in your network.

Deploying Quality of Service to Deliver Quality of Experience

Other terms that you will come across in the VoIP and UC worlds are Quality of Service (QoS) and Quality of Experience (QoE). Nothing new there for the network people, you've lived and sometimes faced QoS nightmares for years. This is because strictly speaking, QoS is a traffic engineering term that refers to resource reservation control mechanisms rather than the achieved service quality. However, QoS is sometimes also used as a quality measure, so you may be familiar with it being used to describe both service quality enablement as well as its measurement.

QoE on the other hand is how we measure the user's end-to-end experience, which is vital in VoIP and UC environments. For example if someone picks up a phone or presses a soft phone button, they'll expect to hear a dial tone in less than 3 seconds, as they are used to doing with traditional TDM telephony (Time Division Multiplexing). They will then expect to enter the called number, be connected quickly to the party or parties they need, experience high fidelity voice, then terminate the call as required and have the line available for the next time they need it. Users also expect to be able to confidently join a conference, create multi-party calls, video sessions and so on.

It's important that we don't test users' tolerance for the sake of technology! Unlike emerging technologies where users tolerate reduced quality to gain technological advantage, VoIP is now a mature technology. For example, early cell phone quality would score between 3 and 3.6 on the MOS scale but users tolerated the reduced MOS for the mobility and features it offers. Even today satellite phones often score less than a 3, but are tolerated for the very useful (and sometimes only) service they provide in remote regions.

VoIP is a fundamental piece of the UC application jigsaw of services and technologies. It's a given that it must work reliably and provide high quality in the four and five nines range – that is 99.99% and 99.999% availability.

Administrators should seek to insulate users from factors that can impair their experience. For example a user doesn't care about call admission controls that cause their call to be rejected because there are too many other calls in progress already. Although this is a measure taken to protect the quality of *all* active calls, it's only their call that matters to them. Similarly if their experience is tainted because of unacceptable delay to dial tone, users don't regard it as their concern. Whether it's because of busy or inadequate DSP resources, insufficient capacity on failover route or if gateways are choked, they just want to pick up the phone with the same confidence as in the old TDM world.

UC Planning and Design Helps Mitigate Risk

The basic structure of your LAN will go a long way to eliminating or even causing problems! Poorly thought out or legacy designs can affect both voice quality and signaling. It's no secret that the top issues affecting VoIP call quality are often not caused by the application itself, but are related to its configuration, conflict with other applications for network resources and misalignment between VoIP and network design.

If you're investigating ways to move to VoIP and Unified Communications, what quality do you expect to receive? In the early stages of VoIP deployment, network architects often aren't aware of previous calling patterns, which mean that network links can become congested, or that voice traffic does not receive the priority it needs.

Delayed voice traffic or non-prioritized traffic can result in significant quality loss. So it's common practice to take a best guess at capacity. And if you're an Exchange, Lotus Notes or Domino administrator, chances are you haven't had to worry too much about the underlying infrastructure that has supported your applications.

So to help you address some of the challenges you'll face, on the next couple of pages we've compiled the Top Ten recommendations for implementing UC and a VoIP monitoring checklist.

UC Implementations – Top 10 Recommendations

1 Tools – are the tools you need for your UC project. Before starting, or even if you have started, get all the tools you need for the job.

You should research and evaluate all the tools you need for your UC project, beginning with planning and diagnostic tools. Evaluate all the tools and find the ones that work for you. As previously mentioned – if QoS is not there before you begin your UC deployment, it will not occur organically during the process. Without taking the correct steps and using the right tools, there will be no QoS after implementation.

2 Tales – are about what others have encountered. It's human nature to want to know the good and bad, especially the bad, so it doesn't happen to us!

You want to avoid the possibility of an RGE (resume generating event!) which is the result of catastrophic network downtime, successful hacker attacks or worse.

This means that your job is on the line every day when you're working, and includes the multiple and capricious forces that make up UC. UC is not just about voice – it's a potential communications transformation! It includes evolving your existing IP network, merging your PSTN traffic and adding telepresence with your web traffic into a highly scalable, totally secure and high performance communications' solution for staff and customers.

3 Timing – is not your timing but getting in sync with corporate strategy. Social media integrated call centers are the latest trend in customer contact centers. Being on Facebook and clicking to contact via chat or voice with a vendor is a great solution. Responding to either good or negative Twitter 'tweets' into the contact center is not just a competitive edge but critical to corporate survival.

All this integration is just part of UC 'use' cases. Social media is teaching companies that if you ignore UC you will be left behind. Timing is about getting your UC strategy integrated with corporate business strategy and not lagging the trend.

That's quite a tall order! The point of this recommendation is to listen to the tales of other people and, at the very least, find out what they would recommend if they had to do it all over again.

4 Team – is not about the people on your team but the ones on all other teams. Certainly it is about having all the competencies on board your team before launching UC but there are still too many other 'silos' of different groups not on board. During a recent UC class taught by Tom, only the Exchange email group was there as they perceived UC to be just an extension of instant messaging.

Tom taught similar classes where just the PBX/PSTN and the IT groups were present. Much of each class's time was spent familiarizing the group with the terms, technologies and vendors of the other group. The IT team knew nothing about PBX trunks and telephony endpoints and the PSTN team knew surprisingly little about IP. The point is to get your teams ready, learn the language of each, cross-train each other and then begin UC.

5 Tips – are all the advice you will get and knowing what you need to know and what you don't. The best part of UC is there is plenty of information available. The best tip is to build your own lab where you can pilot everything you *need* to implement and what you would *like* to try. We have gone from a world where the 'Bell System' told you what to do to a totally DIY or 'self-serve'- pump your own gas environment.

And UC does have a lot of moving parts. The simpler and cleverer the solution is for the user, the more complex it is under the covers and the more work administrators have to do to keep all the components working smoothly.

6 Testing – is road, not lab testing or rather underground testing. Having a lab is recommended, however as they say 'airplanes are safe on the ground but that's not what airplanes are built for', you need to get going on some road trips. Scalable is often used to suggest that if it works one time then it will work a million times. It may well be that it does, but pilotless airplanes are not yet proven and pilots are still needed to be on the spot when something goes wrong. In a recent web seminar, a UC vendor said, "if UC works one time, there is no guarantee it will again." In other words, you need to do a lot of load testing.

7 Threats – refer to the coming onslaught of attacks on UC-enabled networks. Calljacking is a term Tom coined to explain that SIP calls can be hijacked and redirected to the wrong number, another site, used for toll fraud or worse. Calljacking is also the 'injection of hate, explicit, or other undesirable content'. The impact of just calljacking alone can likely lead to litigation if allowed to continue without adequate corporate security or due diligence. Be prepared or it will get ugly.

8 Tackle – is about taking the plunge. Even if everyone has all the answers or think they do, they don't know your network. It's you who lives and dies by your network. Tackling the toughest challenges is also stimulating, and brings something really exciting to your every-day.

9 Training – is simply – you can't do what you don't know. Although you may know everything you need to, does everyone on your team or on other teams? As IT organizations are merging their communications, network, and Microsoft server support teams, much of VoIP and UC terminology is unfamiliar to the teams concerned.

Training is all about helping everyone do their job better. It also helps avoid finger pointing and having problems bouncing between operational domains. Speaking a common language is a way of mitigating risk.

10 Trends – is all about the future. Since we don't know what the future is, then it's all about getting ready for it. We say you can't predict future but you can be ready when the future moves you in a direction that you are taken – get ready for a wild ride.

To help you control your destiny and make the ride a little less wild, you'll find a VoIP monitoring checklist on the facing page that will help ensure insight and management of your entire UC ecosystem.

Prognosis VoIP Monitoring Checklist

Real time monitoring

This is critical for VoIP so you can monitor voice quality and the factors that can impair it. This helps you to ensure availability and quality of VoIP as an integral part of UC services and applications delivery.

Forensic Replay

Prognosis delivers the forensic capability to go back to a point in time and diagnose issues that occurred prior, during or after that point. It also offers the flexibility to collect information at a defined interval, such as every five minutes, and then automatically summarize it to view the trends over an hour, a day a month or a year.

Proactive Insight

Prognosis empowers a team by providing proactive visibility so problems can be prevented. For example, a gateway may not have failed completely but not all components required for a route pattern are available. Prognosis will alert you to the failure of one or more components that are essential for the gateway to function as a complete entity.

Intelligent Automation

Prognosis can automate actions to maximize availability and reduce operating costs. It does this by monitoring your entire UC ecosystem for changes in performance, availability and quality over a defined period.

If any changes breach defined operating conditions, Prognosis will alert according to location, group, date, time and severity. This is important to avoid alert floods. Prognosis will then automatically execute rules to rectify a problem and alert administrators that it has attempted to do this, together with the outcome of the automated action.

Multi-platform support

Prognosis allows you to compare QoS and Service Level Agreement performance across all monitored platforms. It also ensures long term ROI regardless of your UC platform evolution. These are key differentiators when compared to vendor-supplied management tools that only monitor a single platform.

Proven Scalability

Prognosis' unique architecture enables it to scale to manage hundreds of thousands of endpoints. As well as its scalability it provides highly flexible management options. This means that you can manage your UC ecosystem from any location as well as roll up monitored information to one or more central locations.

Broad Extensibility

Prognosis leverages many APIs, including WMI, as well as its unique PACE toolkit to provide visibility to the entire UC ecosystem. It also comes with a rich set of tools that allows users to create and customize their own displays, dashboards and reports. In fact, users are provided with the same toolset that Prognosis' own design teams use.

In this way system administrators can design custom dashboards to suit the needs of individual stakeholders and ensure the information they need is rapidly delivered in real time. To support historical trend analysis, this same information, in equivalent granular detail, or in custom summaries provides operational and business-level reports.

Proven Experience

Prognosis has been a global leader in VoIP performance management for 10 years. Customers in more than 50 countries, including many of the world's largest organizations such as stock exchanges, banks, credit card companies, airlines and universities rely on Prognosis.

Complete UC Ecosystem Visibility

As complexity increases so does the time it takes to identify problems, convince the correct team to own the problem, take action and achieve resolution. Prognosis provides complete visibility into your UC ecosystem allowing you to minimize the overall mean time to repair (MTTR).

It does this in several ways –

- Provides a single pane of glass to monitor all vendors' UC platforms' performance
- Alerts you wherever you are to problems across your entire physical and virtual infrastructure
- Allows you to correlate underlying network performance with voice quality, trunk and server utilization and capacity, gateway performance and availability
- Enables custom designed dashboards and reports to provide vital and relevant information to stakeholders and groups

Conclusion

UC begins and ends with QoS. Without QoS, there is no quality VoIP and the accompanying threat of job loss. No QoS also equals disturbances during your free time, as you may well get calls in the middle of the night. Achieving long-time QoS is always going to be on the horizon as there is an infinite demand for bandwidth by users as they want video, then shared video, multi-screen video, immersion video and so on.

UC management facilitates high quality service delivery by empowering VoIP support personnel to identify issues more quickly and thereby reduce mean-time-to-repair. In this way, not only can you manage the application layer, and the real time events there, you can also monitor guest and host activity, load and capacity in virtualized servers within the global UC ecosystem.

Take a look at Prognosis for Unified Communications and Collaboration Management and you will find that our goal is to give you the ability to look deep inside your VoIP and UC ecosystems so as the network demands grow, you are ready for them.

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