

White Paper

Microsoft Lync and SIP trunking – Ensuring multi-vendor technology success with Prognosis

by Sue Bradshaw: Technology Writer, Integrated Research

Ensuring new technology like Lync delivers parity of service across your business, and is scalable, reliable and delivers cost-effective communications are all factors that contribute to its success. This short white paper explains how Prognosis gives you the invaluable insight needed to help you achieve these goals.



Starting out: Considering SIP trunking?

One of the ways in which you can reduce costs and simplify operations is by routing calls across SIP trunks. Connecting your business to the PSTN in this way delivers substantially cheaper long distance communications and simplifies on-premise telecommunications infrastructure.

While the task of providing consistent, reliable and robust off-net VoIP is transferred to the service provider, you're still responsible for a lot of groundwork. You need to collect vital details from vendors, staff and systems to ensure your VoIP network's requirements are met and that service level agreements will support operations.

As many choices and decisions await you on pricing, sizing, capacity and service level agreements, it's vital to gain an understanding of current call activity, peaks, troughs and spikes within your VoIP environment. This type of insight can be achieved through real-time monitoring and measurement, backed up by comprehensive call history.

Some of the many factors it's vital to understand in the existing environment include:

Server availability	Performance, usage and capacity, for individual servers and pools.
Inbound and outbound calls	Conference, audio, data and video channels.
Active calls	Statistics, activity, success or failure and session details.
Peak busy-hour demand	Voice streams and their quality, broken down into source and destination.
Detailed call information	Call duration and end time and numbers of attempted and completed calls.
Call impairments	Signal variation and noise levels, MOS, latency, packet loss and jitter.
Session Border Controllers	Status, inbound and outbound sessions and performance load.

On the SIP journey: Some useful signposts

VoIP traffic

The bandwidth required for a SIP trunk depends primarily on the number of concurrent calls that it has to carry, and there are various ways you can calculate this requirement. The good news is that you don't need to work it out yourself; Prognosis helps you remove the guesswork out of capacity planning and understand your environment before you change it.

Prognosis for UC collects call activity details with corresponding voice quality for summarization and analysis. You can use this collected data to review activity over a day, week, month or year and see peaks, troughs and spikes at a glance. This will give you the information required about the volume and type of calls you need to support and help you with capacity planning.

- ✓ Busy hour
- ✓ Calls attempted
- ✓ Calls completed
- ✓ Voice quality
- ✓ Trends
- ✓ Concurrent calls
- ✓ Number of calls
- ✓ Type of calls
- ✓ Trunk information
- ✓ Source and destination
- ✓ Comprehensive call history

End-to-end voice quality

Bad voice quality leads to frustration, transaction errors and increased call-handling time. It can also mean you may need to make additional infrastructure investments to resolve it, and to do that you need to know where the investment should be made.

Whether you're using traditional telephony, VoIP or UC, accessing the PSTN directly, or via a SIP trunk real-time insight enables you to investigate problems so you can resolve issues before they impact your business or customers. You need to know when things change so that you can manage your environment proactively; anticipating and preventing problems.

Insight to voice quality for all calls enables you to uncover patterns of unacceptable voice quality and narrow problems down to particular groups of users, servers, locations, gateways and time of day.

SNAPSHOT

Prognosis supports executive-level infrastructure decision making.

A global business was considering making a \$500,000 investment in a new Lync pool for Europe, based on voice quality complaints from end-users.

Prognosis showed voice quality was the same for all regions, and that it needed to be improved globally.

This insight supported executive-level infrastructure decision making, by being able to determine much smaller investments in existing infrastructure would improve voice quality for everyone. This averted a \$500,000 expense that was not only unnecessary, but would have made the UC environment significantly more complex with little or no added value.

Call information

If the number of completed calls equals the number of attempted calls, voice quality is good and your users are happy, things are going well. However if things go wrong and alerts start appearing on your screen or arriving by email you'll need to start troubleshooting without delay.

If calls are moving through your system but the quality is poor, you can filter on just those calls to reduce the amount of information in front of you and drill down to investigate the root cause. For example you can look at a range of calls over a specific time period and identify the reason for poor quality.

This may be caused by excessive round trip time taken by voice streams, or the session border controller's resources being overloaded. If the number of attempted sessions is higher than it could manage, latency will be a problem and some users will be unable to complete calls to the PSTN.

- ✓ Duration
- ✓ Signal and noise levels
- ✓ Route patterns
- ✓ Time of day
- ✓ Degraded calls
- ✓ Percentage
- ✓ Comprehensive call history
- ✓ SIP codes
- ✓ Conference ID

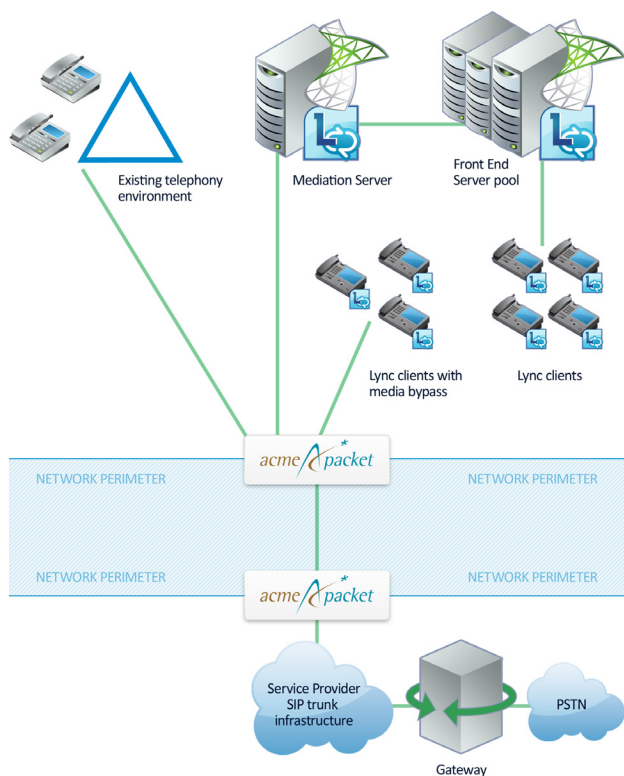
Server capacity and usage

The performance of your Lync Server and SIP trunk infrastructure is critical to achieving simpler and more cost effective communications. This is why it's important to monitor each server and component relationships – like pools. This means you can manage the load and follow the progress of calls within your network as they traverse your operational domains to the session border controllers at your network's edge.

This helps you identify what voice quality is like before it leaves your network which will help you address any issues more quickly with your service provider.

If on the other hand voice quality issues are occurring within your network you can analyze them from the Mediation Server to the Session Border Controller and identify at which point impairment factors like jitter, latency and packet loss impacted them.

The destination: Microsoft Lync Server ecosystem connected to a SIP trunk



- ✓ Real time information
- ✓ Voice quality
- ✓ Performance load
- ✓ Incoming user service requests
- ✓ Current SIP connections
- ✓ Calls recently attempted
- ✓ Number of calls processed
- ✓ Server availability
- ✓ Flow control
- ✓ Outstanding sends
- ✓ Incoming messages
- ✓ Incoming timeouts
- ✓ Average holding times
- ✓ Inbound and outbound calls
- ✓ Conferences
- ✓ Active audio, data and video channels

There are several ways to connect your Lync clients to a SIP trunk. The most common method employs key Lync Front End and Mediation servers and one or more Session Border Controllers (SBCs).

SBCs are placed between the Mediation Server and the carrier for session border control and management, additional security and to insulate the Lync Server topology from the internet.

SBCs also address many of the interoperability problems common in multi-vendor networks like transcoding, SIP protocol normalization and E.164/non E.164 number translation. It can also optimize utilization across multiple Lync Mediation Servers and trunks, and support media bypass and hardware-based media processing to reduce load on Lync servers.

In this environment you need to manage:

Front End Servers

Servers with this role are core to the Lync server environment and run critical tasks including user authentication and registration, as well as providing IM functionality and Presence information. When pooled, Front End Servers also enable scalability and failover.

With so many core tasks running on them some of the things you should know include real time information about their load, incoming user services requests and current SIP connections. Monitoring the Front End Server pool as an entity together with drilldowns to detailed information about each server within the pool summarizes the calls recently attempted, and the number processed by each server.

Intelligent alerts demystify potential issues before they become real problems. For example a Front End Server within the pool may be consistently using over 90 percent of its processing power and available memory. This can cause attempted calls to fail because of time-outs or insufficient resources. Such overloading is easily identified and administrators are alerted so they can investigate and rectify any load balancing or overall pool capacity issues.

Mediation Servers

A Mediation Server is a necessary component for implementing Enterprise Voice and dial-in conferencing. It translates signaling and, in some configurations, media between your internal Lync Server infrastructure and a PSTN gateway, IP-PBX, or a SIP trunk via an SBC.

This is why it's important to monitor Mediation Server health including availability status, CPU busy percentage, together with the top busiest processes, memory utilization and disk information. Monitoring inbound and outbound calls and calls in transit, with active channels broken down by type – audio, data and video assists in troubleshooting problems and capacity planning by providing the data to analyze peaks, troughs and trends over time.

Session Border Controllers

Session Border Controllers sit at the edge of your network and provide many functions including network address translation, firewall and topology-hiding, security and protocol translation.

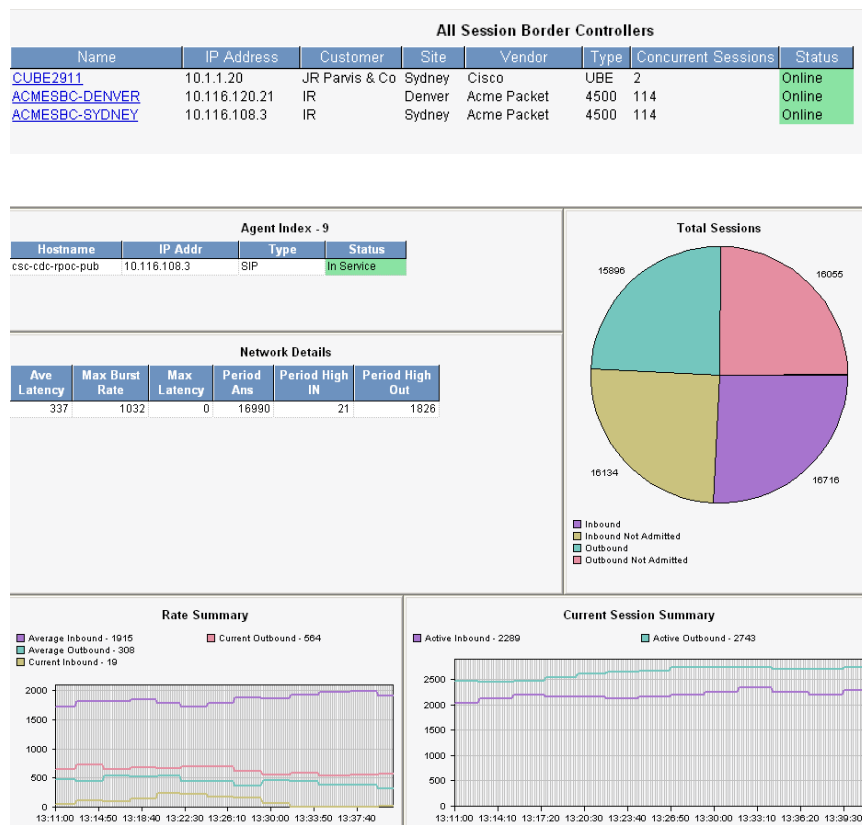
To manage their availability, performance and capacity, Prognosis provides information about the SBC itself as well as its active sessions. At-a-glance information includes the SBC's status and calls per second, CPU and memory utilization, gateway endpoints and redundancy status.

Understanding your SIP traffic is more important than ever. With the ability to see the number of concurrent sessions per SBC, broken down in each direction and matched to a customer name and/or site you are able to:

- Monitor the total number of calls leaving the Lync Mediation Server, Avaya and Cisco VoIP environments.
- Monitor the total number of sessions traversing the SBC
- Monitor the status, availability and details of all SBCs and their active sessions.

Multi-vendor PBX SIP trunking

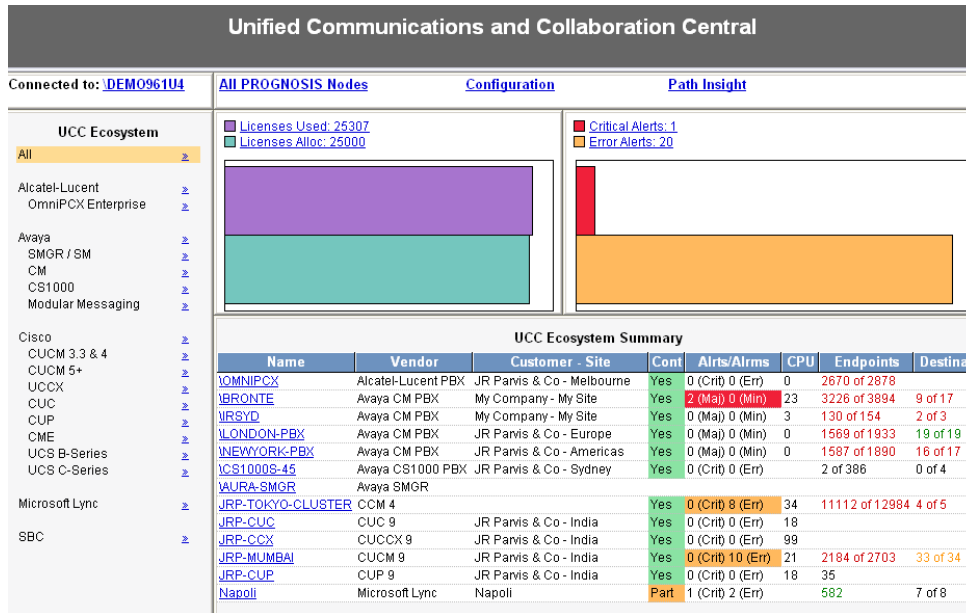
Your SIP trunking solution will be influenced by where you are in your VoIP/UC implementation, and what other PBXs you need to incorporate. Many enterprises have decades of investment in existing telephony infrastructure and use Lync for IM and Presence, while maintaining the SIP trunk connection to an existing PBX. Deploying Lync under these circumstances means having to manage a multi-vendor implementation by default.



- ✓ Real time status
- ✓ Concurrent sessions
- ✓ Incoming, outgoing calls
- ✓ Calls in transit
- ✓ SIP traffic stats
- ✓ Media resources
- ✓ DSP availability
- ✓ Transcoding activity
- ✓ Congestion occurrences

At-a-glance and drill down views of SBC status and activity

And as each technology comes with its own client interfaces, configuration and administration tools, terminology and user skill requirements, managing hybrid environments inevitably means more complexity. Multi-vendor VoIP, UC and legacy infrastructure complexity can place a heavy burden on IT staff. You can ease this burden by placing a single pane of glass across the entire multi-vendor UC ecosystem, to deliver insight about each component.



Ease multi-vendor performance management by placing a single pane of glass across the entire UC ecosystem.

Scalability and reliability

Depending on the location of sites and how much traffic you anticipate within your enterprise, you may not want to route all users through a central SIP trunk, you may choose to route some users through a SIP trunk at their local branch. So how do you know how many SIP trunks you need and how much capacity is required for each one?

Monitoring your traffic flows so you can size your trunk/s correctly is vital and a unified interface across multiple vendor SIP call sources delivers the following benefits:

- Tracks the total number of calls leaving the Lync Mediation Server pool, broken down by server, together with calls originating from Avaya and Cisco PBXs.
- Monitors and collects the total number of calls traversing all SBCs, providing vital information like the number of inbound and outbound sessions at any given time, and importantly for sizing purposes, the number of concurrent sessions.
- Identifies users by type – Enterprise Voice, remote call control, PC to PC and those with audio or video disabled as well as the calls that a user made, either as an individual or as part of a conference.
- Identifies the busy hour for each day, week, month or year, the number of calls completed compared to the number attempted and the voice quality of those calls.

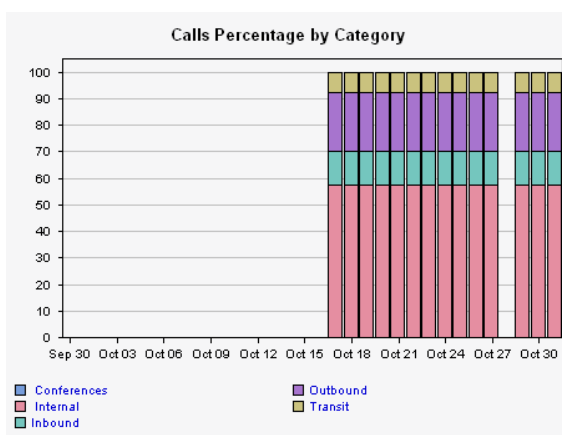
Troubleshooting

Customers report that as 80 percent of Lync issues have a root cause external to Lync it's vital to be able to combine insight into UC-specific problems with underlying network, gateway and trunk conditions. As many Lync administrators don't have extensive UC knowledge, because they have been Microsoft server and application focused, there is limited common language to communicate between the server, telephony, and network teams; resulting in finger-pointing.

Prognosis will help you translate your years of experience managing Microsoft servers and applications to be meaningful in the new world of Lync Server UC.

If you receive an alert about a particular condition you can step back in time using the same interval at which the data was collected, or summarize it into larger segments for a higher level view. While you replay this data you can verify all the Lync services are running, as well as view the network interface, disk and CPU activity for each monitored server.

If SIP trunks are accessed via an Acme Packet Session Border Controller, related voice streams are correlated into and out of it as well as to other vendors' PBXs within your site. In this way you can identify where voice quality issues are occurring and you can pinpoint where capacity, call control or configuration issues are affecting quality.



Calls Attempted by SIP Code		
Code	Response	Count
200	OK	165
404	Not Found	21
480	Temporarily Unavailable	11
487	Request Terminated	9

Conference Sessions Attempted by SIP Code		
Code	Response	Count
200	OK	13

What problems might occur?

- ! One way or missing audio
- ? Intermittent trunk failures
- ? Registration failures
- ! Poor voice quality

Problems can occur in the SIP environment including one-way or missing audio, trunks dropping intermittently, time-outs, registration failures, and poor voice quality caused by high levels of latency, jitter and packet loss.

All of these issues can be monitored, alerted and reported on by Prognosis. No longer do you need to try and find a needle in a haystack; a choice of at-a-glance information in real time, or a historical collection initiated on the fly or summarized from days, weeks or months of data can provide the answers.

How UC might affect future call volumes

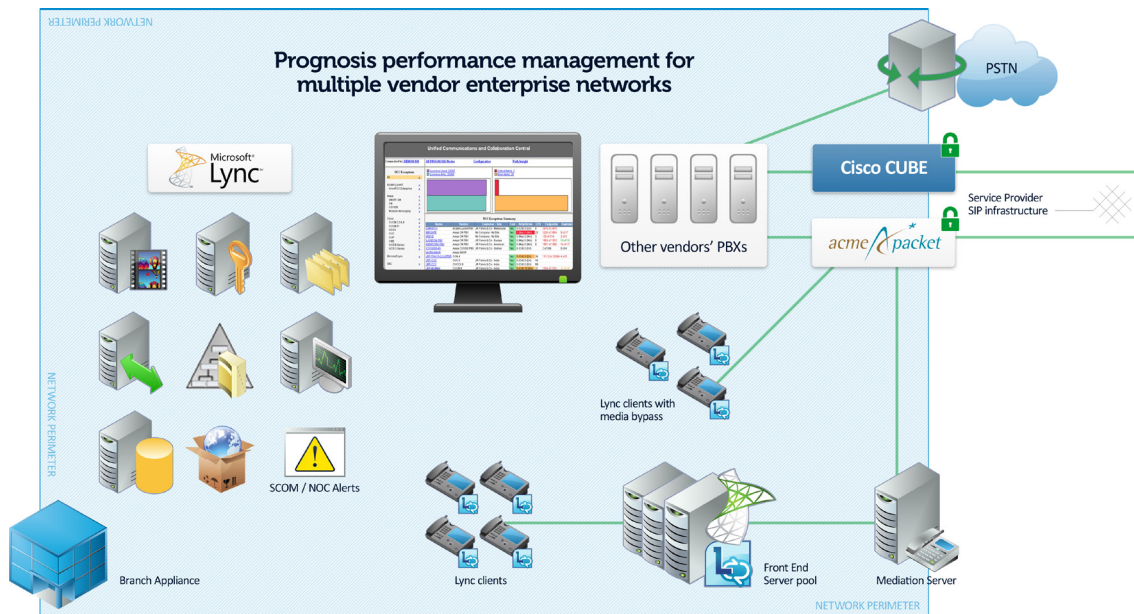
UC introduces new communications options and as a result, voice streams may well reduce in number and frequency. As most UC software platforms support peer-to-peer IP communications, voice-to-voice calls may be replaced by other communications types like IM and video, and the number of wasted calls significantly reduced through the use of Presence.

UC-based communications, including voice, can flow over the WAN or Ethernet backbone services and do not require SIP trunks. So at the same time as monitoring SIP trunk usage, other UC services like instant and unified messaging and video should be monitored. If voice traffic decreases, other types of communication will increase.

Key to understanding usage requirements is knowing what types of endpoints are in place, whether these are current analog devices, phones in common areas, IVRs, phones in elevators, fax machines and so on.

Summary

SIP provides enterprises with more options for multi-vendor integration, so management of a converged environment becomes even more critical. Successful interoperability will reduce complexity for users but administrators must manage multiple vendors and technologies, as well as versions, locations and user expectations. The flexibility of monitoring any multi-vendor PBX, Lync Server and SIP trunk interface enables you to manage device availability, interface status, capacity and usage and minimizes the impact of one device's failure on another.



If you'd like to read more white papers and case studies about optimizing UC operations and resources visit <http://www.prognosis.com/resources/uc-resources/white-papers>.



Contact us

Americas t: +1 (303) 390 8700 f: +1 (303) 390 8777 e: info.usa@ir.com
Asia Pacific/Middle East/Africa t: +61 (2) 9966 1066 f: +61 (2) 9966 1042 e: info.ap@ir.com
United Kingdom t: +44 (0) 1344 894 200 f: +44 (0) 1344 890 851 e: info.europe@ir.com
Germany t: +49 (69) 710 455 255 f: +49 (69) 710 455 450 e: info.germany@ir.com

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