Managing SIP Trunking Issues with the nGeniusONE Platform

SIP Trunking, a key component of many enterprises' Unified Communications and Collaboration (UC&C) strategy, provides many benefits. Often implemented to improve delivery of IP-based voice and video services, save on long haul telephony costs, as well as leverage new UC&C equipment feature sets, SIP Trunking can be broadly deployed in many locations across a global enterprise. However, SIP Trunking comes with its own intrinsic complexities for UC&C managers that require increased collaboration not only amongst the IT Operations teams but also with multiple third party providers who need to manage an increasing number of UC&C components, session border controllers (SBCs), UC&C servers, WANs and SIP Trunks.

Solving issues involving SIP Trunking requires the ability to quickly triage and identify the source of the problem. The innovative nGeniusONE® Service Assurance platform makes this possible. Powered by Adaptive Service Intelligence™ (ASI) technology for real-time, in-depth, smart data, nGeniusONE delivers a common user interface for smart analytics to seamlessly and contextually go from health status in dashboard views to service dependency maps to session analysis and service monitors to precisely pinpoint errors and service disruptions. This provides an in-depth view of service, network, signaling and application performance across complex multi-tier, multi-domain service delivery environments.

Challenges in Troubleshooting SIP Trunk Environment

Delivery of SIP Trunking communications services depends on multiple key devices, specifically, SBCs, network switches and routers, server components, and call servers. Managing UC&C relies on multifunctional teams who often times resort to point tools to assure the performance of their specific components in the UC&C service delivery chain.

Siloed management tools lack visibility into the complete picture of the problem, including the relationships and interrelated nature of the overall converged network infrastructure, SIP Trunks, application services, signaling and enabling protocols necessary to deliver UC&C. This makes it difficult to evaluate such things as:

- The SIP Trunk network edge interconnect points including Gateways and SBCs
- Service enablers like DNS, RADIUS and Active Directory/LDAP
- Network and Application Resource Contention in the converged network, e.g. use of QoS
- Server Dependencies with other UC&C services including instant messaging and multimedia conferencing services
- Application Dependencies
- The Distributed Service Delivery Environment

The multitude of vendor-specific management tools with their own unique interfaces and views of their service components leads to inefficient workflows with time consuming hand-offs where one team attempts to translate and verify the analysis of the other team. Examples of some user complaints and their potential root causes are provided in the chart below (see Table 1):

Problem Area or Complaint	Possible Cause (Security, Network, Server Performance, Interoperability)
"I can't dial out."	• Network – issues with endpoints communicating with the call server • Network – issue with call servers communicating with external peers • Session Border Controllers (SBCs) – interoperability issues between the call server and the SIP Trunk
"It's slow to dial out."	• Network – WAN issue at remote site causing slow response to call servers • Server – congested or badly balanced call servers • Server – SBC vendor interoperability / code issues • Network – incorrect network prioritization for signaling protocols
One-way calls - "There's no one at the other end of the call."	• Call Server or SBC – issue affecting call transmission (one-way call) • Network – diverse routing in the network with no return path
"That phone call was poor quality."	• Call Server or SBC – configuration issues negotiating wrong codec on call setup • SBC – incorrect QoS re-classification • Network – bandwidth or misconfigured QoS issues affecting call quality • Endpoint – echo cancellers not working

Table 1: UC&C Problems and possible root causes. Whose problem is it: Network, Server, SBC, ITSP, and/or the UC&C Team?

About SIP Trunking

SIP Trunking sets the foundation for delivery of a multitude of additional UC&C applications and services over a shared or converged infrastructure principally leveraging Session Border Controllers (SBCs). This consolidates the network access for both inbound and outbound calling with the incoming calls routed by the Internet Telephony Service Provider (ITSP) and outgoing calls routed by the SBC to be delivered over a private network or an MPLS network.

A key benefit of SIP Trunking is long distance cost savings for voice calls. However, once implemented, SIP Trunks are often used to deliver additional UC&C services including instant messages, multimedia conferences, Enhanced 9-1-1 (E9-1-1) emergency calls, as well as other SIP-based, real-time communications services (see Figure 1).

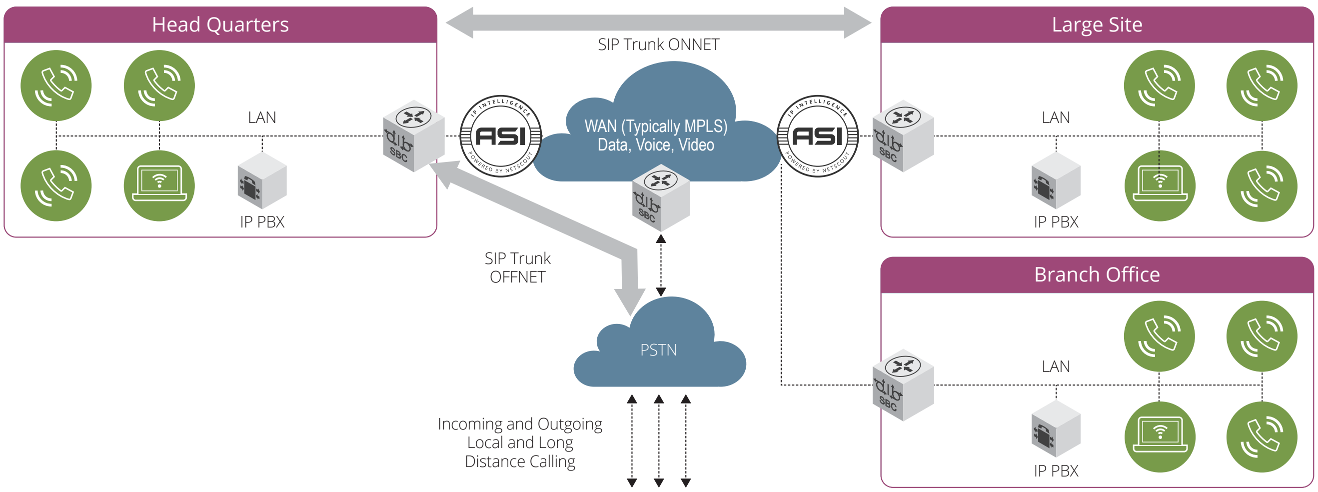

Figure 1: A sample SIP Trunk environment – User Communities, SIP Trunks and PSTN Gateways, and Server infrastructure for Call Management, Presence, Conferencing, Enablers etc. Areas of the network that will benefit from strategically deployed ASI performance management visibility are any locations with SBCs which are typically characterized by high traffic volume and critical conversations paths.

Rapid Diagnosis of Issues: nGeniusONE with ASI Technology for SIP Trunking

ASI technology provides unique metadata that includes Key Performance Indicators (KPIs), Key Traffic Indicators (KTIs), Key Server Indicators (KSIs) and Key Error Indicators (KEIs). ASI supports Session Initiation Protocol (SIP), an industry standard protocol used for IP-based signaling.

nGeniusONE with ASI technology provides contextual workflows and situational awareness that facilitates quick diagnosis of issues within the SIP Trunking environment. In this way, operators can identify issues that may impact successful completion of SIP Trunk calls which can include such sources as poorly performing Call Manager Servers, Session Border Controllers, SIP Trunks, congested WAN links, mis-configured QoS / CoS classes or VLAN assignments, problematic infrastructure network components or possible service level agreement (SLA) violations.

Once an Incident occurs, IT staff can make use of nGeniusONE's intuitive workflows between the real-time Service Dashboard, Service Monitors and Dependency Maps to troubleshoot, and perform SIP Sessions and IP Traffic-based deep dive packet analysis as necessary (see Figure 2). Handoffs can be made to other complementary component performance management tools (for servers or network devices) or to third-party vendors (IP Telephony providers, SBC manufacturers, or PSTN providers) to solve a given problem or provide evidence of an SLA concern.

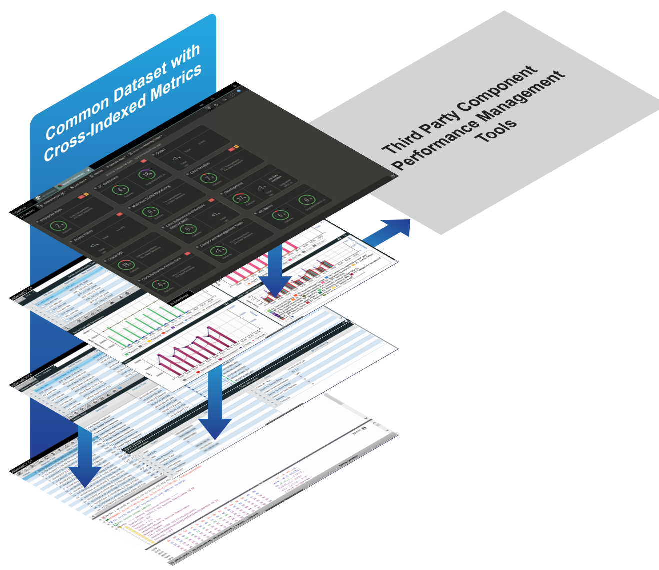

Figure 2: nGeniusONE provides a common workflow for incident management and root cause escalation for unified communications and collaboration issues, such as those that involve SIP Trunking, that are similar to workflows for other services. The workflow leverages the Service Dashboard, Service Monitors, Dependency Mapping, Session Analysis and, when necessary, packet decode.

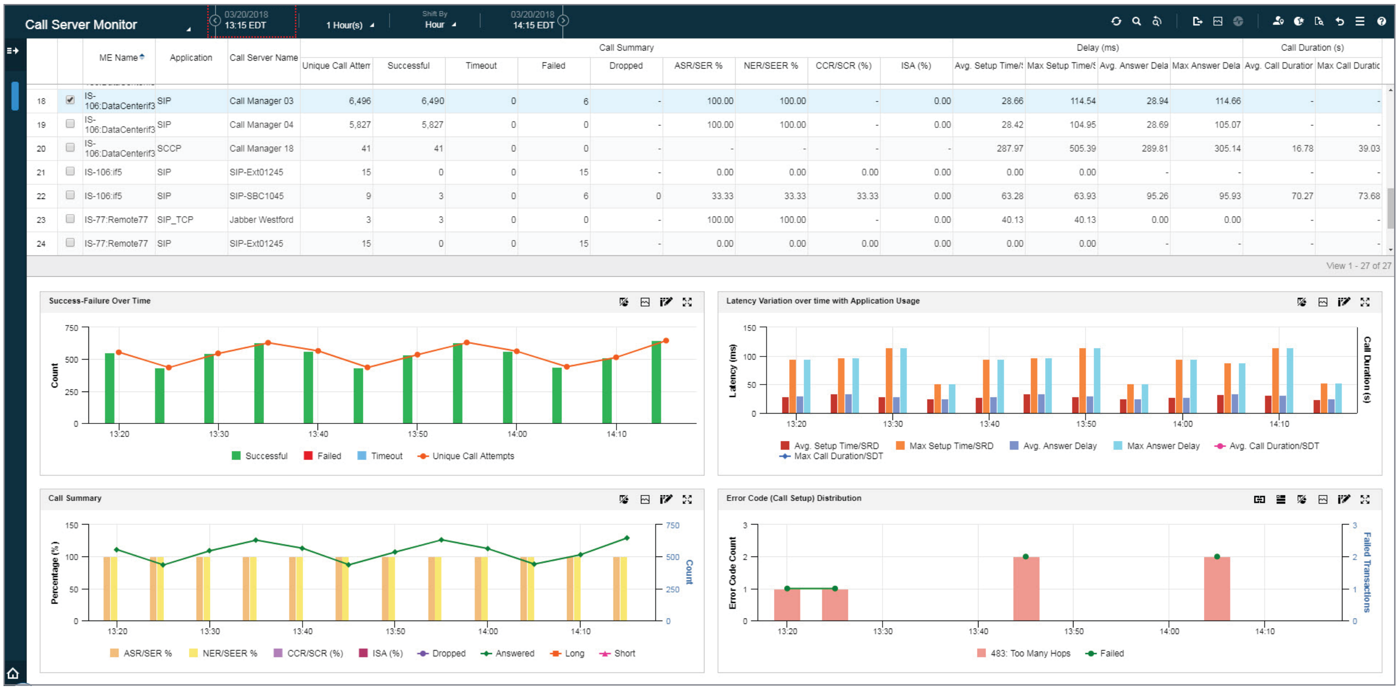

Figure 3: The Server and Community Summary in Voice Statistics Monitor show 6 failed sessions.

Real-World Use Case: How nGeniusONE Workflows Triages SIP Trunking Issues

There are a common set of problems that can impact users in enterprises using SIP Trunking, such as users having trouble making an outgoing call. When a phone is picked up and there is no dial tone, users perceive that the phone is dead and will likely hang up. Similarly, when making a call, if the call does not connect within few seconds, the callers get frustrated. User experiences, whether it is related to getting a dial tone, connecting calls, or phones ringing on the other side, are all dependent on the performance of call signaling protocols. The impact of such problems in a business environment can be as simple as lost employee productivity or as damaging as poor customer experience. The following use case demonstrates how logical workflows in the nGeniusONE platform can be used to quickly triage and resolve such real-world user experience problems.

Triaging Outgoing Call Connection Issues

A company that had a successful SIP Trunk solution deployed in their headquarters had just finished migrating some of their regional locations to SIP Trunking, and were finding that some users in those offices could not make external calls. Calls using company's internal networks (ONNET calls) were delivering good user experience. However, users were experiencing problems when making calls to customers, partners and vendors over a SIP Trunk (OFFNET calls). The IT staff leveraged the nGeniusONE platform to research the problems to discover which SIP Trunks the users were accessing to begin triaging the call connection issues.

By leveraging ASI technology, the nGeniusONE platform's Voice Statistics Monitor (see Figure 3) discovers and reveals essential metrics and details pertaining to the regional offices using SIP Trunks. The

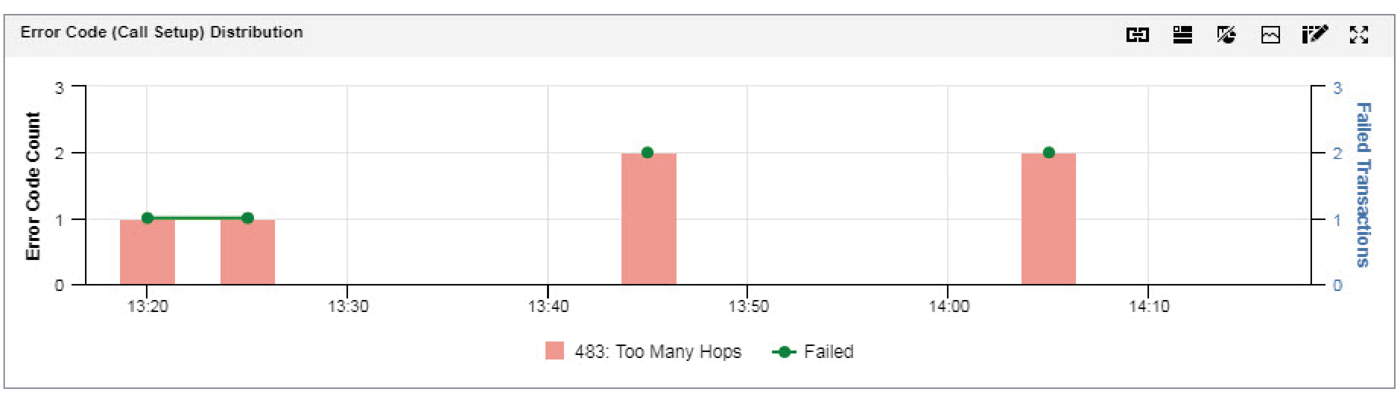

Figure 4: nGeniusONE Voice Statistics Monitor showing Errors for Call Manager 03.

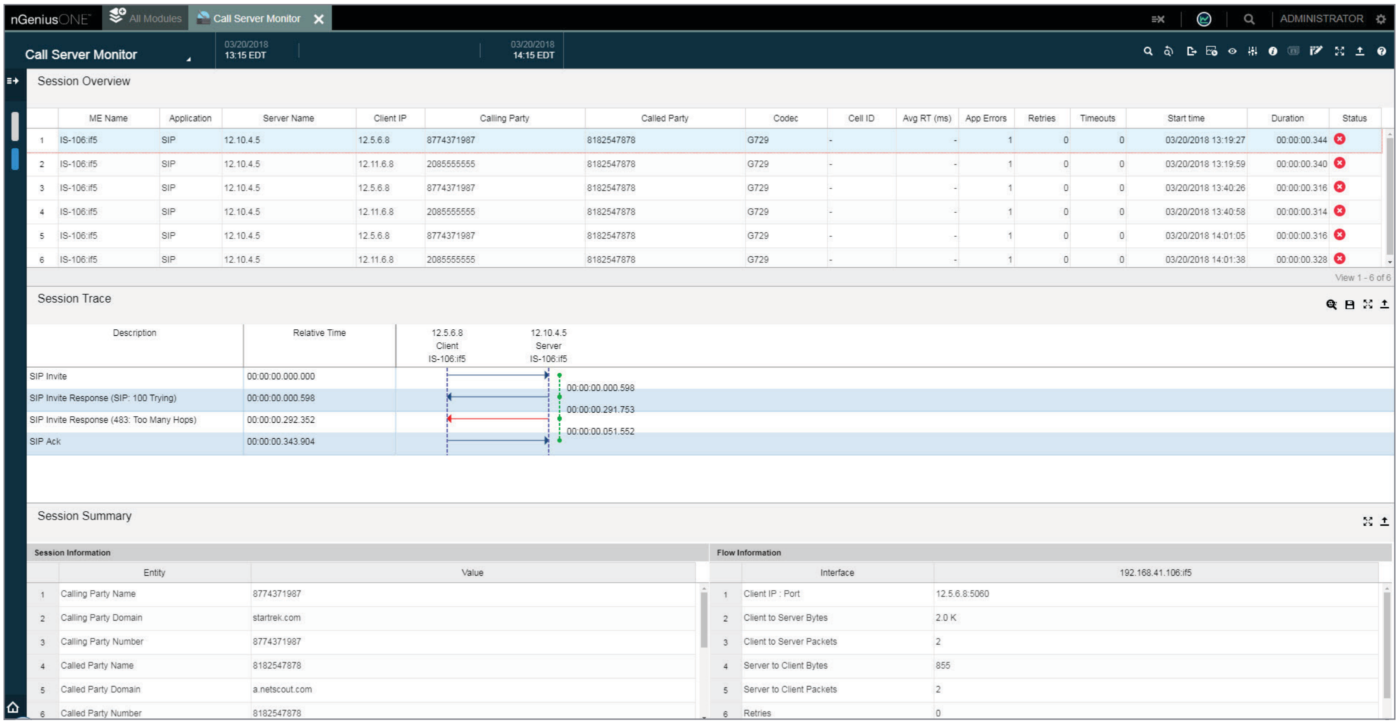

Figure 5: nGeniusONE Session Analysis provides multi-hop visibility providing additional evidence of the interaction between the Call Server, enterprise SBC, and external ITSP's SBC to effectively isolate the root cause.

table with the row containing "Call Manager 03" shows that there are 6 failures during this period. The Voice Statistics Monitor helps IT teams quickly assess the cause for SIP signaling failures as well as track key performance metrics related to the network and user behavior. For example, using the tabular data in the Voice Statistics Monitor, IT teams can determine how effective the network was at delivering SIP signaling traffic, what percentage of calls failed, and how many calls were not "setup" successfully, among many other key performance indicators. A key differentiator for the nGeniusONE platform over other point tools is that once a problem is identified in the Server / SBC Summary table, additional information is available in the graphs below for that particular server being investigated.

The Error Code Distribution graph (see Figure 4) provides an at-a-glance view of the health of the call setup and teardown messages flowing from the SBC toward the SIP Trunk service provider. This graph shows that there are a number of SIP 483 errors (Too Many Hops).

The final step in this particular workflow is to drill down into the Session Analysis for one of the calls experiencing the problem. The Session Trace (see Figure 5) shows multi-hop visibility of the performance and the interaction of the call server manager (IP address 12.10.4.5) with a Session Border Controller's (SBC) internal interface having the IP address of 12.5.6.8.

By examining the ladder diagram in the session trace screen (see Figure 6), IT teams can confirm that there are SIP 483 errors generated. The directionality of the SIP errors shows that the errors are being generated by the internal server. As a result of this, IT teams can quickly and effectively isolate the root cause of error number 483. The next step to correct this issue is to modify the current call routing. Had the error originated in the ITSP, IT staff could notify the SIP Trunk provider with evidence and have them change the SIP header to reflect the right number of proxies before reaching the destination.

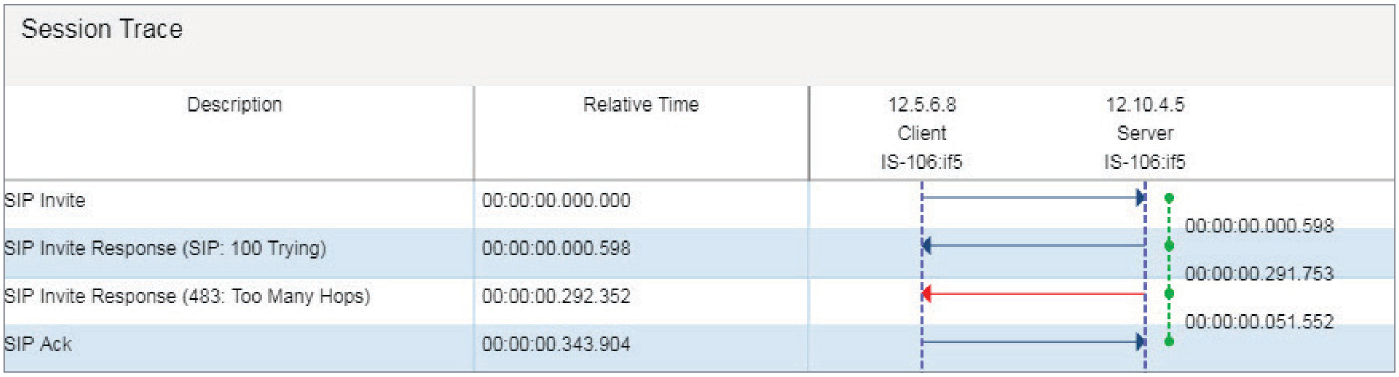

Figure 6: Expanded view of the ladder diagram from Session Analysis screen in Figure 5.

Gaining the Best Visibility for IT: ASI Instrumentation for Call Servers and Signaling Protocols

Enterprises with existing nGenius® implementations may already have the right instrumentation in the best locations for leveraging ASI with nGeniusONE platform to address UC&C service deployment and specific SIP Trunking issues as they arise. However, for those expanding an existing environment, the best visibility for IT organizations can be gained by adopting some combination of the following:

- Add InfiniStreamNG™ appliances, via TAPs or SPAN ports, for SIP signaling near the SBCs, Border Gateway, or SIP Trunks at headquarters, large sites and branch offices where SIP Trunking is implemented.
- The nGenius Packet Flow Switches deployed in the environment will provide the ability to connect multiple links to existing InfiniStreamNG and InfiniStream® appliances where appropriate to gain performance management visibility of the SIP Trunking environment to share with other tools as necessary.
- Leverage InfiniStreamNG and InfiniStream appliances elsewhere in the enterprise environment, e.g. near Call Servers, on either side of the SBC, to provide analysis of the performance of a Call Server to a client and to show whether problems originate in the enterprise network or in SIP Trunking service provider network.

Collaborate More Effectively to Pinpoint and Resolve Issues

The nGeniusONE platform provides a single, consistent set of workflows and views that IT, Server, Network and UC&C teams can use to troubleshoot service delivery over their environment. The SIP Signaling protocol for call setup and teardown is analyzed by ASI technology, DPI of rich packet-flow data, provided by the InfiniStream appliances.

With the nGeniusONE platform, IT staff is able to manage SIP Trunking alongside other UC&C and data services holistically. The common analysis, situational analysis, screens and workflows in nGeniusONE effectively break down the barriers between, not just the various IT operations teams, but also the multiple vendors involved in delivering UC services in a SIP Trunk environment. Where SLA agreements exist with external PSTN / SIP trunk service providers, information from the nGeniusONE platform can be used to provide evidence and insight so that problems can be resolved and rectified quickly.

IT staff and their vendors are empowered to collaborate more effectively, reducing the time consuming finger pointing that inevitably occurs in such complex deployments. nGeniusONE delivers actionable information often outside the reach of existing UC&C tools. This results in rapid identification of the root cause of issues affecting the end-to-end SIP Trunk services, quick escalation to the team or vendor responsible for service degradation, and ultimately optimal quality user experience returned as quickly as possible.

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