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Managing Voice, Video Call Quality Issues in Contact Centers with the nGeniusONE Service Assurance Platform

Contact Centers are an essential part of many businesses. Whether a person is calling to check on a payment received, place an order, file an insurance claim or transfer money at a bank, they are likely communicating with a business representative in a contact center. The importance of that exchange between a customer and the contact center cannot be under stated as it is the "face" of the business. Many contact centers are deploying next generation Unified Communications & Collaboration (UC&C) technology to improve customer experience and increase business productivity.

Although UC&C brings tangible benefits to contact center employees, the successful adoption of these services poses many challenges to IT operations teams. Since UC&C services rely on IP networks and share network resources with other business applications used by agents, IT should fully understand how changes in traffic patterns impact both UC&C and other business application performance. Additionally, voice and video services are extremely sensitive to network performance issues such as packet loss, jitter, and latency. There are certain other impairments such as background noise, echo, and speech levels that cannot be easily measured but have profound impact on user experience with severe consequences on the conversational quality of phone connections between the agent and the customer.

Any delay in making a connection to the contact center personnel, issues with the quality of the call itself, or unplanned disconnects will understandably frustrate a customer. These customer service impacting issues need to be avoided or at a minimum, corrected immediately.

Challenges Troubleshooting Call Quality Issues in Contact Center Environments

Call quality can be impacted by IP network congestion; errors in QoS provisioning; misconfiguration of UC&C equipment such as codec types, packet loss concealment configuration, jitter buffer configuration; or by the presence of excessive echo and speech levels, among many other issues. The chart below illustrates some of the common contact center complaints and their potential causes.

The root causes for call quality issues discussed in table 1 are transient in nature which makes detecting or reproducing the problem for troubleshooting extremely difficult. In response to this challenge, many IT teams typically resort to using vendor provided and/ or component specific tools to monitor call quality issues. In such environments, manually correlating data from disparate sources makes

Problem Area or UC&C complaint	Possible Cause (Security, Network, Server Performance, Interoperability)
"All trunks are busy" "My calls are getting abandoned or dropped"	 Network - bandwidth or misconfigured QoS affecting call sessions Call Server - Too many requests are being processed by UC&C server(s) There are delays in setting up calls Too many calls are failing Load balancer - Calls to contact centers are not well balanced or load balancer is struggling to keep up with the sudden spike in traffic
"That phone call was of poor quality"	 Network – Bandwidth or misconfigured QoS issues affecting call quality Call Server – Configuration issues negotiating wrong codec on call setup Gateway – Echo cancellers not working Endpoint – Bad microphone at remote end, soft client performance
"One-way Audio" "I can't hear anyone on the other end of the call"	 Network – Firewall and/or routers may be blocking voice traffic Network – Edge devices blocking traffic for external peering traffic Network – Routing issues causing difficulties in media streams reaching the destination SBC/Network – SBC blocking traffic due to failure or misconfiguration
"My video has been grainy all day"	 Network – Bandwidth or misconfigured QoS issues affecting call quality Equipment – Codec adaption to cope with network issues Bridge – Interoperability / codec selection issues

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

About Unified Communications and Collaboration in Contact Center Environments

UC&C environments, consisting of agent desktop/phones, UC&C Servers and IVRs, are typically very complex and difficult to manage. Additionally, many contact centers are replacing legacy PSTN networks with SIP trunking to reduce communication costs while enhancing the customer experience through HD voice and video. These environments typically use Session Border Controllers (SBC), and PSTN Gateways to enable agents to call customers over PSTN (see Figure 1).



Figure 1: A sample UC&C 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 locations characterized by high traffic volume and critical conversations paths.

troubleshooting difficult especially across distributed, multivendor platforms operating across multiple locations. Further, as there is a lack of common performance data shared across different functional domains, IT operations teams cannot efficiently triage service quality problems, which results in increased time to troubleshoot issues, ultimately delaying mean time to resolution (MTTR).

nGeniusONE Platform

The innovative nGeniusONE® Service Assurance platform makes it easy for IT teams to identify and quickly triage call signaling and media related problems. The nGeniusONE platform is powered by Adaptive Service Intelligence™ (ASI), NETSCOUT's next-generation Deep Packet Inspection (DPI) technology, which leverages the inherent richness of wire data to generate real-time metrics and metadata.

Using the ASI metadata, the nGeniusONE platform provides clear visibility into all sources contributing to call quality issues. Additionally, through correlated metrics along the call path, the platform helps IT organizations quickly identify major user experience issues such as one-way audio, no audio, short duration calls, and grainy video images.



Figure 2: nGeniusONE provides a common workflow for incident management and root cause escalation for unified communications & 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.

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Figure 3: nGeniusONE Call Server Monitor offers in-depth analysis into the communications of the new users in the regional office using SIP trunking.

Real-World Use Case: Using nGeniusONE to Triage Call Connection Issues Due to SIP Signaling Errors

Any delay in call setup is viewed as a failure of the company in meeting customer expectations. When such issues arise, customers may perceive that the company is offering inferior products and services.

nGeniusONE platform enables IT teams to quickly identify call signaling performance issues. With nGeniusONE, IT teams can monitor server load, SIP signaling error codes, network and application latencies, and data transmission errors. The following use case demonstrates how logical workflows in the nGeniusONE platform can be used to quickly triage and resolve call connection problems introduced during call setup and teardown.

Triaging Call Connection Issues

In order to reduce lengthy queue times, many contact centers enable customers to use the "call back" feature. nGeniusONE is used to solve call connection problems experienced by agents when making PSTN calls to customers that opted for a call-back.

When agents experience slow call connection times, help desk or IT teams usually hear that described as "calls are not connecting" or "phones are not working". Since many contact centers use multiple trunks for achieving redundancy and high availability, it becomes difficult to identify the servers/SBCs where call connection issues may be introduced. The nGeniusONE "Call Server Monitor" (see Figure 3) makes it easy to discover the specific UC&C component that is experiencing call signaling issues.

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6	ø	IS- 77:Remote77	SIP	SIP- SBC1045	16	8	0	8	(50.00	50.00	50.00	0.00	29.12	29.13	49.13	49.14	55.88	55.88

Figure 4: The Server and Community Summary in Call Server Monitor show failures for SIP-SBC1045.



Figure 5: nGeniusONE Call Server Monitor showing Errors for SBC SIP Trunk 1.

By leveraging ASI technology, the Call Server Monitor discovers and reveals essential metrics and details relevant to each UC Server including SBCs connecting different SIP Trunks. On inspecting the table on the top of the screen, the data for SBC "SIP–SBC1045" shows that there are 8 failures caused during a specific period. More importantly, Call Server Monitor enables IT teams to view all important information such as the server name, summary statistics, latencies, and average call setup and other signaling related information, in a single pane of glass (see Figure 4). As a result, IT teams can quickly make inferences, understand

the issue, and take action based on the information presented on the screen.

A key differentiator for the nGeniusONE platform over other point tools is that once a problem is identified in the Summary table at the top of the Call Server Monitor (Figures 3 & 4) screen, additional information is available for the server being investigated (Figures 3 & 5), which provides more contextual information correlated with other important metrics to help IT better understand the issue.

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Figure 6: nGeniusONE Session Analysis provides additional evidence of the interaction between the Call Server and the SIP Trunk SBC to isolate the issue.

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4		IS- 106:DataCent	Audio	San Jose UC	4.97	186	13.98	11.83	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	1
5		IS- 106:DataCent	Audio	Singapore Users	4.88	150	0.00	0.00	0.00	98.84	0.00	98.88	98.03	0.00	0.00	0.00	0.00	0.00	
6		IS- 106:DataCent	Audio	Dubai UC	2.39	74	0.00	18.92	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	
7		IS- 106:DataCent	Audio	Berlin UC	1.99	82	26.83	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	49.07	0.00	0.00	
8		IS- 106:DataCent	Audio	Toronto UC	1.74	52	0.00	57.69	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	
9		IS- 106:DataCent	Audio	SIP Gateway1	1.68	113	0.00	2.65	0.00	3.25	0.00	2.22	38.53	0.00	0.00	0.00	0.00	0.00	
10		IS- 106:DataCent	Audio	UC SIP Trunk A	1.16	36	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	100.00	0.00	0.00	
11		IS- 106:DataCent	Audio	Jabber Toronto	1.07	15	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	
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Figure 7: nGeniusONE Media Monitor provides deeper analysis into the call quality performance so IT teams can collaborate effectively. The data indicates that San Jose users are experiencing One-Way Audio Issues but the network is not the cause of the problem.

On the same page, a summary of the distribution of errors over time is displayed, providing an at-a-glance view of the health of the call set up and teardown activities for the calls traversing the SBC. The error code distribution graph shows several SIP 483 errors (see Figure 5).

The final step in this particular workflow is to obtain evidence. By drilling down to the Session Analysis for one of the calls experiencing the problem (see Figure 6), IT can view the session details confirming SIP error 483 between the call server and the SIP Trunk. The SIP error 483 results when there are too many hops over the SIP Trunk. This issue can be corrected by establishing a new call route that uses fewer hops over a proxy server or by changing the SIP header to reflect the correct number of proxies used in signaling before reaching the final destination.

Triaging Media Issues

The following use case demonstrates how logical workflows in the nGeniusONE platform can be used to quickly triage and resolve call quality issues. The Media Monitor Screen (see Figure 7), shows key performance metrics such as QoS Mismatch, Single Direction Calls, Short Calls, as well as MOS and voice quality degradation metrics that help IT identify potential root causes for call quality problems. Every column is sortable to allow the worst performing locations/users to bubble up to the surface. The summary view enables IT teams to quickly identify high priority issues that need their immediate attention.

In this case, the "Single Direction Call" column reveals that almost 12% of sessions at the "San Jose" location are experiencing one-way audio issues. Further examination of other columns in the summary view shows the one-way audio problem is not caused by impairments to the IP network or to the payload. This leaves the IT team with one more area to investigate - misconfiguration of one or more UC&C network elements that may be causing packets to be dropped.



Figure 8: View showing all conversations between different locations. This quickly shows the health of all incoming and outgoing calls from San Jose to other locations.

Drilling down to the Conversation View (see Figure 8) shows that voice streams coming into San Jose are unaffected while those leaving San Jose are experiencing "Single Direction" problems. Subsequently, the Streams View (see Figure 9) provides the evidence needed to prove that some calls originating from the San Jose location do not have corresponding media streams going back in the other direction.

Launching a Network View (see Figure 10) helps determine the source contributing to one-way audio issue experienced by San Jose users. In this case, the root cause is identified as the SBC with IP address 15.15.15.12. Now that the root cause is identified, the next step for the operations team is to use component tools specific to the SBC under investigation for final resolution.

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10	0	03/29/2018 14:05:21	00:00:08	IS-106:DataCenterif3	8.8.8.11	15.15.15.11	34	G.711_A-law	0.00	-	0.00	0.250	-	
11	0	03/29/2018 14:51:54	80:00:00	IS-106:DataCenterif3	8.8.8.11	15.15.15.11	34	G.711_A-law	0.00	-	0.00	0.218	-	
12	0	03/29/2018 14:48:34	00:00:08	IS-106:DataCenterif3	8.8.8.11	15.15.15.11	34	G.711_A-law	0.00	-	0.00	0.250		
13	0	03/29/2018 14:32:29	00:00:08	IS-106:DataCenterif3	8.8.8.11	15.15.15.11	34	G.711_A-law	0.00	-	0.00	0.250	-	
14	0	03/29/2018 14:14:45	00:00:08	IS-106:DataCenterif3	8.8.8.11	15.15.15.11	34	G.711_A-law	0.00	-	0.00	0.250	-	
15	0	03/29/2018 14:16:25	00:00:08	IS-106:DataCenterif3	8.8.8.11	15.15.15.11	34	G.711_A-law	0.00	-	0.00	0.312	-	
16	0	03/29/2018 14:50:34	00:00:08	IS-106:DataCenterif3	8.8.8.12	15.15.15.12	34	G.711_A-law	0.00	-	0.00	0.312		
17	0	03/29/2018 14:32:49	00:00:08	IS-106:DataCenterif3	8.8.8.12	15.15.15.12	34	G.711_A-law	0.00	-	0.00	0.250		
18	0	03/29/2018 14:02:21	00:00:08	IS-106:DataCenterif3	8.8.8.12	15.15.15.12	34	G.711_A-law	0.00	-	0.00	0.281	0	
19	0	03/29/2018 14:31:09	00:00:08	IS-106:DataCenterif3	8.8.8.12	15.15.15.12	34	G.711_A-law	0.00	-	0.00	0.218	-	
20	0	03/29/2018 14:00:41	00:00:08	IS-106:DataCenterif3	8.8.8.12	15.15.15.12	34	G.711_A-law	0.00	-	0.00	0.250	0	1
21	0	03/29/2018 14:45:34	00:00:08	IS-106:DataCenterif3	8.8.8.12	15.15.15.12	34	G.711_A-law	0.00	-	0.00	0.312	0	1
22	0	03/29/2018 14:29:29	00:00:08	IS-106:DataCenterif3	8.8.8.12	15.15.15.12	34	G.711_A-law	0.00	-	0.00	0.312		1
23		03/29/2018 14:36:10	00:00:08	IS-106:DataCenterif3	8.8.8.12	15.15.15.12	34	G.711 A-law	0.00		0.00	0.312		1

Figure 9: Evidence of San Jose calls terminating at the Session Border Controller having single direction calls.



Figure 10: The network view shows no audio streams exist between Session Border Controller and San Jose user.

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

Contact centers in enterprises with existing nGenius implementations may already have the right instrumentation in the best locations for leveraging ASI technology with the 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 Session Border Controller (SBC), Border Gateway, or SIP Trunks at contact centers, large sites and agent-based branch offices where SIP Trunking is implemented.
- nGenius Packet Flow Switches deployed in the environment will provide connectivity between multiple links and existing InfiniStreamNG appliances to gain performance management visibility from the SIP trunking environment and simultaneously share with other tools as necessary.
- Leverage InfiniStreamNG appliances elsewhere in the enterprise environment, e.g. near Call Servers, to provide analysis of the performance of a Call Server to a client or any other signaling device that may impact overall user quality of experience.

Collaborate More Effectively to Pinpoint and Resolve Issues

Through comprehensive correlation of performance metrics extracted by ASI, the nGeniusONE platform offers faster resolution to voice and video performance problems and helps IT deliver superior end-user experience. Unique to the nGeniusONE solution, this is achieved by providing real-time visibility into the performance of both call signaling and media quality for all legs of a conversation. As a result of this correlation, IT can quickly isolate the source of performance problems.

Additionally, the platform provides a single, consistent set of workflows that enables collaboration across the IT organization. Since key performance metrics are extracted in real-time by analyzing all IP packets traversing the network, IT teams can easily see how voice, video, and other line of business applications are interacting with each other to gain a better understanding of the service delivery environment. Consequently, IT can quickly pinpoint root cause for service degradation and proactively improve service quality, even when managing complex multi-tier, multi-vendor networks.

NETSCOUT.

Corporate Headquarters

NETSCOUT Systems, Inc. Westford, MA 01886-4105 Phone: +1 978-614-4000 www.netscout.com Sales Information

Toll Free US: 800-309-4804 (International numbers below) Product Support

Toll Free US: 888-357-7667 (International numbers below)

NETSCOUT offers sales, support, and services in over 32 countries. Global addresses, and international numbers are listed on the NETSCOUT website at: www.netscout.com/company/contact-us

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