8x8 Virtual Office

Analysis of Cloud Communications VoIP Quality Under Normal and Adverse Network Conditions

Tolly Report #217121
Commissioned by
8x8, Inc.

May 2017

8x8, Inc.

Virtual Office

VoIP Voice Quality Under Normal & Adverse Conditions

Tolly Certified

May 2017

THE BOTTOM LINE

8x8 Virtual Office delivers:

1. Best or nearly equivalent quality in normal conditions "On Net" and PSTN.
2. Best voice quality in 10 of 16 "On Net" packet loss tests across Apple iOS, Google Android, Windows and Apple Mac OS X platforms.
3. Best voice quality in 11 of 20 "On Net" jitter tests with quality always above 4 (good) across Apple iOS, Google Android and Apple Mac OS X platforms.
4. Dramatically better voice quality under adverse "On Net" bufferbloat tests across Apple iOS, and Apple Mac OS X platforms.
5. Results proving that only 8x8 and another vendor clients support switching networks during a call. This means that they can preserve call continuity when the LAN connection is unplugged and re-plugged to the same or to a different network, and also when calls switch from a LAN to a WiFi network.

Executive Summary

Customers expect a high level of voice quality when selecting a cloud-based business communications system. Often these customers are migrating from legacy PBX or other on-premise systems.

As networks have converged and a more distributed workforce has emerged, employees are using different network types to communicate and connect. This includes utilizing not only Ethernet from the office, but also home networks employing both consumer broadband and WiFi, mobile networks using 3G/4G LTE, as well as public WiFi networks such as the neighborhood Starbucks WiFi. This diversity in networks offers tremendous flexibility for modern users to work from anywhere and with a communications suite that allows them to communicate in ways that were never possible before.

As the modern worker is connecting to more networks in more places than ever before, it becomes critically important to utilize the services of a cloud communications provider that invests in technologies that optimizes quality of service for real-time communications. This is particularly important because network conditions may sometimes be less than ideal, or users can be switching/transitioning across different types of networks. Packet loss, delay and jitter, as well as other impairments can often be present and can potentially degrade voice quality. To deliver high-quality voice communication in the face of such conditions requires proprietary technologies and a deep understanding of the quality impairments that can affect users across different connections and clients.

Recognizing the increased requirements of today's cloud-based business communications systems, 8x8 has architected its solutions to meet these challenges.

8x8 commissioned Tolly to compare the voice quality of 8x8 Virtual Office in both "On net" (cloud-based VoIP-to-VoIP) and VoIP to public switched telephone networks (PSTN) environments. In the first scenario both users were cloud-connected. In the latter scenario, one user was cloud-connected while the other was connected to the traditional public telephone network.

Tests compared the 8x8 solution to Google Voice/Hangouts, RingCentral Office, and Microsoft Skype for Business Online and were run using four different client OS pairs: Apple iOS, Google Android, Apple Mac OS X and Microsoft Windows.

The 8x8 solution delivered the best voice quality in a majority of the test cases - across scenarios, client platforms and impairment conditions. In cases where 8x8 did not have the highest quality scores, its results were generally in line with the other solutions tested. In call continuity tests, 8x8 recovered calls in all test scenarios. Results are detailed in the following pages.

Test Results

For all tests detailed in this report, three different impairment scenarios were used and run as separate tests. Packet Loss - Tests were run with no packet loss and again with 2, 5 and 10% loss. Jitter - Tests were run with no jitter and again with average absolute jitter (Gaussian distribution) of 20, 60, 100 and 200ms. Bufferbloat (holding then bursting of packets) - Tests were run with no bufferbloat and again with four different flows with varying maximum bursts. Flow spikes were 0, 200, 400, 750 and 1500ms. (See the Test Methodology & Setup section for details.)

High-Definition Client - "On Net"

The first set of tests examined VoIP client-to-client communications with clients on opposite sides of the simulated network "cloud." The network simulator allowed test engineers to inject impairments into the network. In brief, all voice quality tests were run by playing a high-quality audio file from one client to another and recording the audio that was received for further analysis. State-of-the-art POLQA[1]-based voice quality assessment software was used to compare the received audio to the sent audio and calculate a voice quality score. POLQA predicts a Mean Opinion Score (MOS). In that scale a 4 or above is "good", 3 and above is "fair", 2 or below means "poor" or "bad".

Apple iOS Clients

Packet Loss

The 8x8 solution delivered the best quality in all scenarios. Even when 10% packet loss was introduced into the network, the 8x8 solution quality was nearly 3.5. This fair quality contrasts with the other solutions that all delivered sub-3.0, "poor" results with 10% packet loss. See Figure 1 for all Apple iOS "On Net" results.

Absolute Jitter

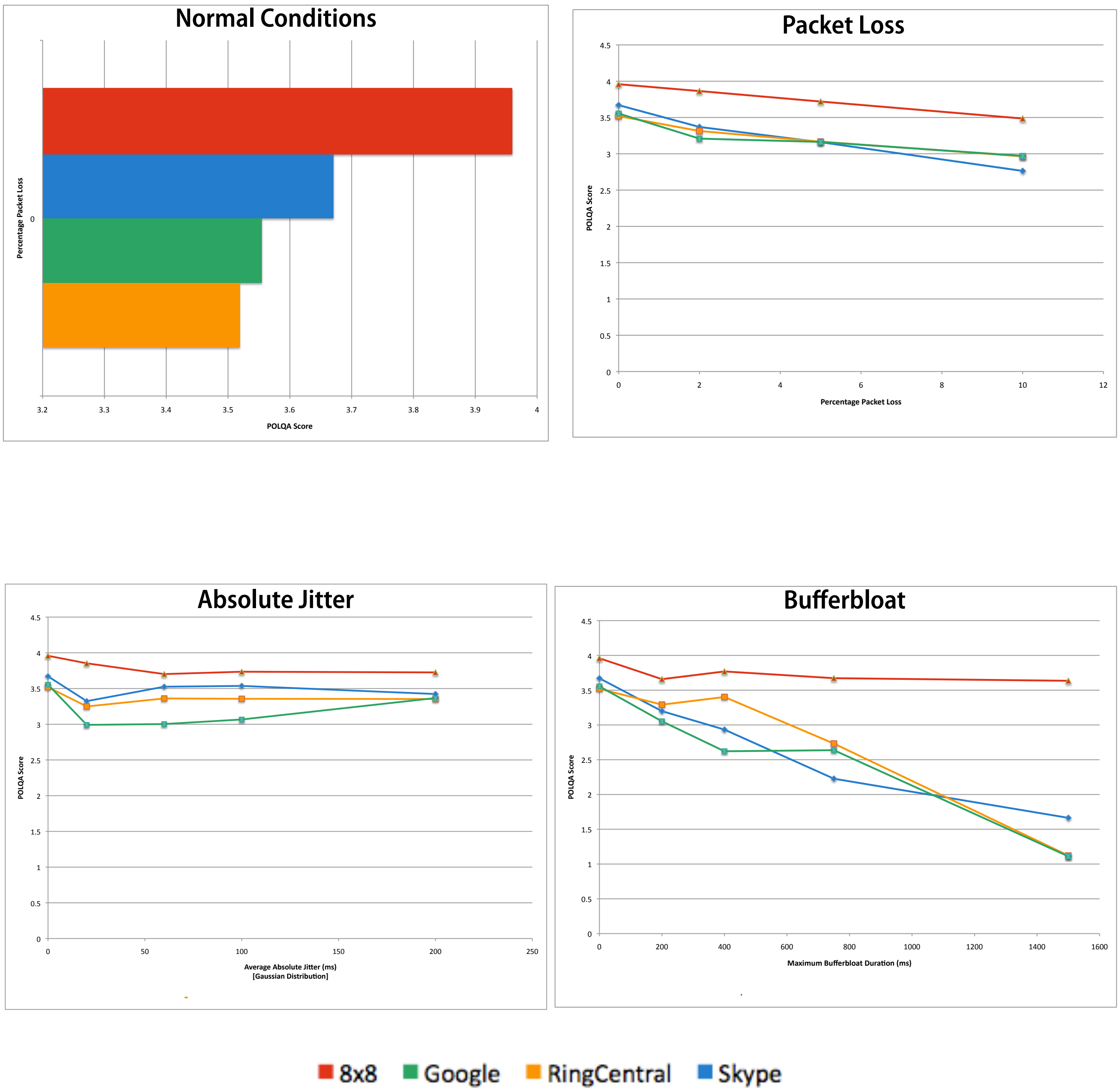
In the absolute jitter tests, the 8x8 solution delivered the best quality in all scenarios compared to the other solutions. As the jitter was increased, the 8x8 solution quality scores remained in the 3.7 to 3.8 range, even with 200ms of jitter. The other solutions ranged from about 2.9 to 3.5 across the range of tests.

Bufferbloat

In the bufferbloat tests, the 8x8 solution delivered the best quality in all scenarios compared to the other solutions but here the 8x8 quality scores were dramatically better as the spike duration increased. Across all scenarios, 8x8 quality remained above 3.5 where the others fell off dramatically with the quality quickly becoming "poor" or "bad" with increased spike duration.

[1] https://en.wikipedia.org/wiki/POLQA

Test Results - "On Net" - Apple iOS Clients



Source: Tolly, May 2017

Figure 1

Test Results

High-Definition Client - "On Net" - Continued

Google Android Clients

Packet Loss

The 8x8 solution delivered the best quality in all scenarios. Even when 10% packet loss was introduced into the network, 8x8 quality score was nearly 4. This quality contrasts with the other solutions that all delivered lower results with 10% packet loss. Interestingly both Skype and Google delivered poorer results at 2% and 4% loss than they did at 10% loss. See Figure 2 for all Android "On Net" results.

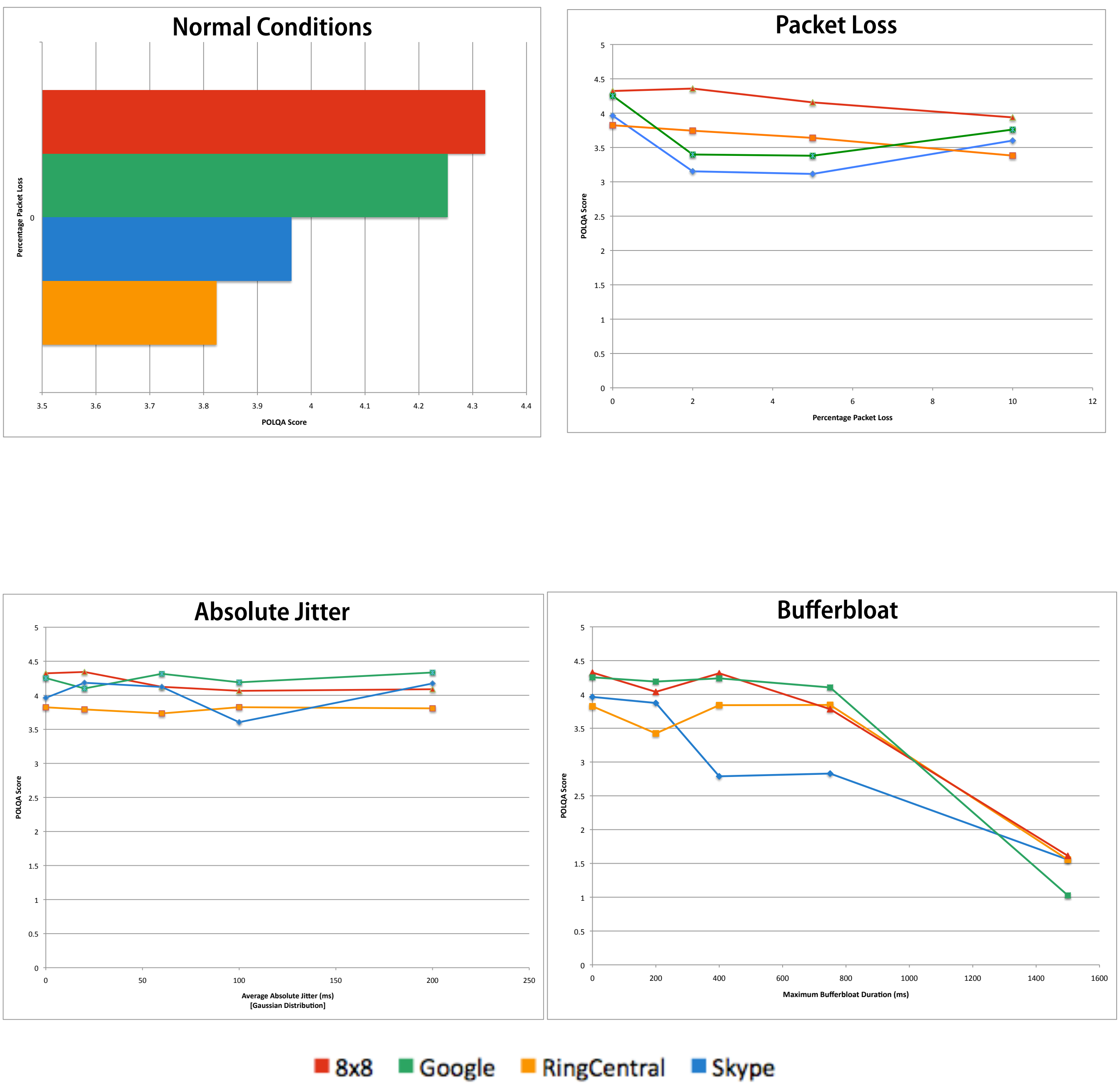
Absolute Jitter

In the absolute jitter tests, 8x8 delivered the best quality in the 20ms scenario and comparable quality in all other jitter scenarios with 8x8 quality scores always above 4 (good). This contrasted with RingCentral where scores were under 4 (~3.7 to 3.8) for all scenarios. Microsoft's quality scores dipped down for the 100ms test but improved with the 200ms test.

Bufferbloat

In the bufferbloat tests, the 8x8 solution quality scores above 4 through 400ms spike test. Except for Google, all the solutions dropped down below 4 with the 750ms spike test. And all the solutions had sub 2 scores with the 1500ms spike test.

Test Results - "On Net" - Google Android Clients



Source: Tolly, May 2017

Figure 2

Test Results

High-Definition Client - "On Net" - Continued

Microsoft Windows Clients

Packet Loss

The 8x8 solution delivered the best or equivalent quality through 5% packet loss and delivered second best quality in the 10% loss scenario. Even when 10% packet loss was introduced into the network, 8x8 quality score was nearly 3.8. See Figure 3 for all Windows "On Net" results.

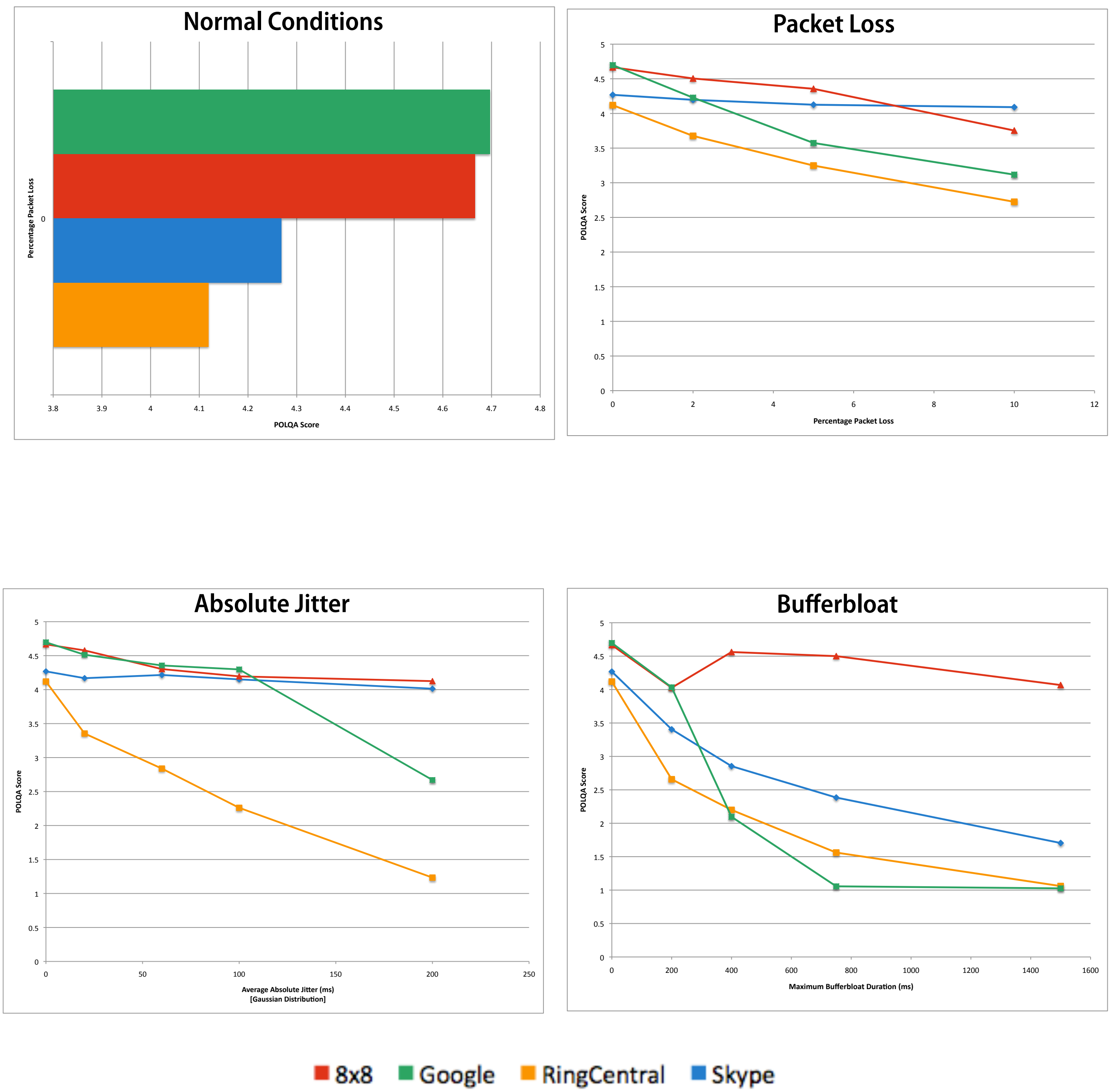
Absolute Jitter

In the absolute jitter tests, the 8x8 solution delivered the best or virtually equivalent quality in all scenarios. RingCentral's quality scores dipped below 3 even with 20ms jitter and degraded steadily as jitter increased. Google's quality remained consistent and above 4 until jitter was increased to 200ms at which time the score dropped precipitously to below 3.

Bufferbloat

In the bufferbloat tests, the 8x8 solution delivered the best quality in all adverse scenarios compared to the other solution but here 8x8 quality scores were dramatically better as the spike duration increased. Across all scenarios, 8x8 quality remained above 4 where the others fell off dramatically with the quality quickly becoming "poor" or "bad" with increased spike duration.

Test Results - "On Net" - Microsoft Windows Clients



Source: Tolly, May 2017

Figure 3

Test Results

High-Definition Client - “On Net” - Continued

Apple Mac OS X Clients

Packet Loss

The 8x8 solution delivered the best or equivalent quality in all scenarios with Skype’s score slightly higher at 4.09 compared to 8x8’s 4.01 at 10% loss. Even when 10% packet loss was introduced into the network, 8x8 solution quality was at 4.

It is notable that the results for RingCentral were in the “poor” range of 2.1 even under normal conditions. The RingCentral results remained consistently poor throughout. the packet loss tests. See Figure 4 for all Apple Mac OS X “On Net” results.

Absolute Jitter

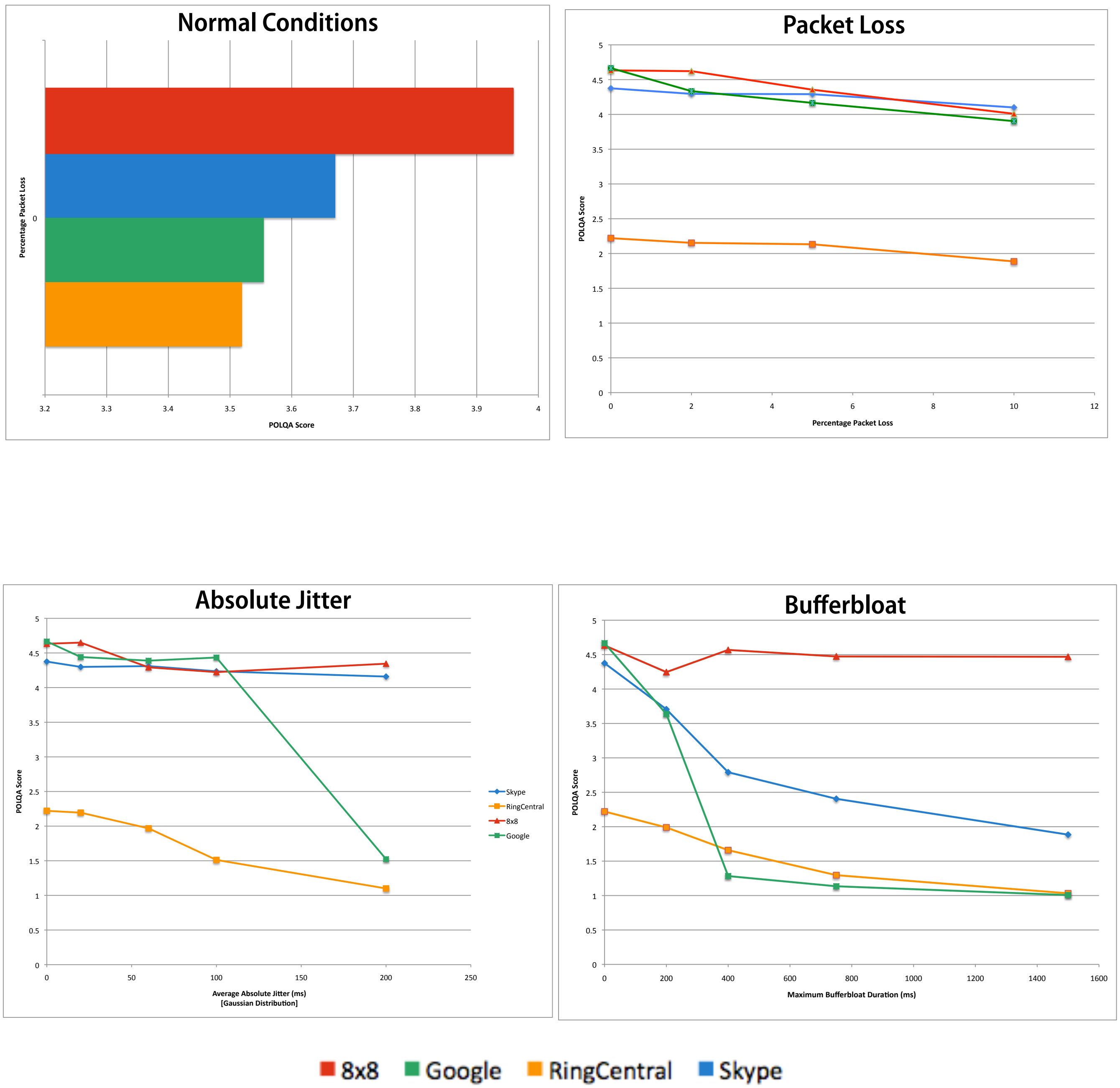
In the absolute jitter tests, the 8x8 solution delivered voice quality scores above 4 in every scenario even with 200ms of jitter. Google maintained results near 4.5 through the 100ms test and then dropped down to 1.5 with 200ms of jitter.

RingCentral’s best score was 2.2 with no jitter and its scores degraded even further as jitter increased. With 200ms of jitter RingCentral’s score was only 1.1.

Bufferbloat

In the bufferbloat tests, the 8x8 solution delivered the best quality in all adverse scenarios compared to the other solutions but here 8x8 quality scores were dramatically better as the spike duration increased. Across all scenarios, 8x8 quality remained around 4.5 where the others fell off dramatically with the quality quickly becoming “poor” or “bad” with increased spike duration.

Test Results - "On Net" - Apple Mac OS X Clients



Source: Tolly, May 2017

Figure 4

Test Results - "PSTN-to-VoIP"

For all tests detailed in this report, three different impairment scenarios were used and run as separate tests. Packet Loss - Tests were run with no packet loss and again with 5% loss. Jitter - Tests were run with no jitter and again with average absolute jitter (Gaussian distribution) of 100ms. Bufferbloat (holding and then bursting of packets) - Tests were run with no bufferbloat and again with a flow spike of 750ms. The chart labeled "Normal Conditions" represents the results of the test with no packet loss, no jitter and no bufferbloat spike. (See the Test Methodology & Setup section for details.)

The second set of tests examined PSTN-to VoIP-Client communications with clients on the far side of the simulated cloud linked to the PSTN network. The network simulator allowed test engineers to inject impairments into the network. As in the VoIP-to-VoIP tests, all voice quality tests were run by playing a high-quality audio file from one client to another and recording the audio that was received for further analysis. Calls were played from the PSTN network to the VoIP network. Google Voice replaced Google Hangouts for this set of tests.

Apple iOS Clients

Packet Loss (5%)

The 8x8 solution delivered a quality score of 3.6 in this test. Skype and RingCentral had scores of 3.9 and 3.7 respectively with Google Voice scoring 2.7

See Figure 5 for all Apple iOS "PSTN-to-VoIP" results.

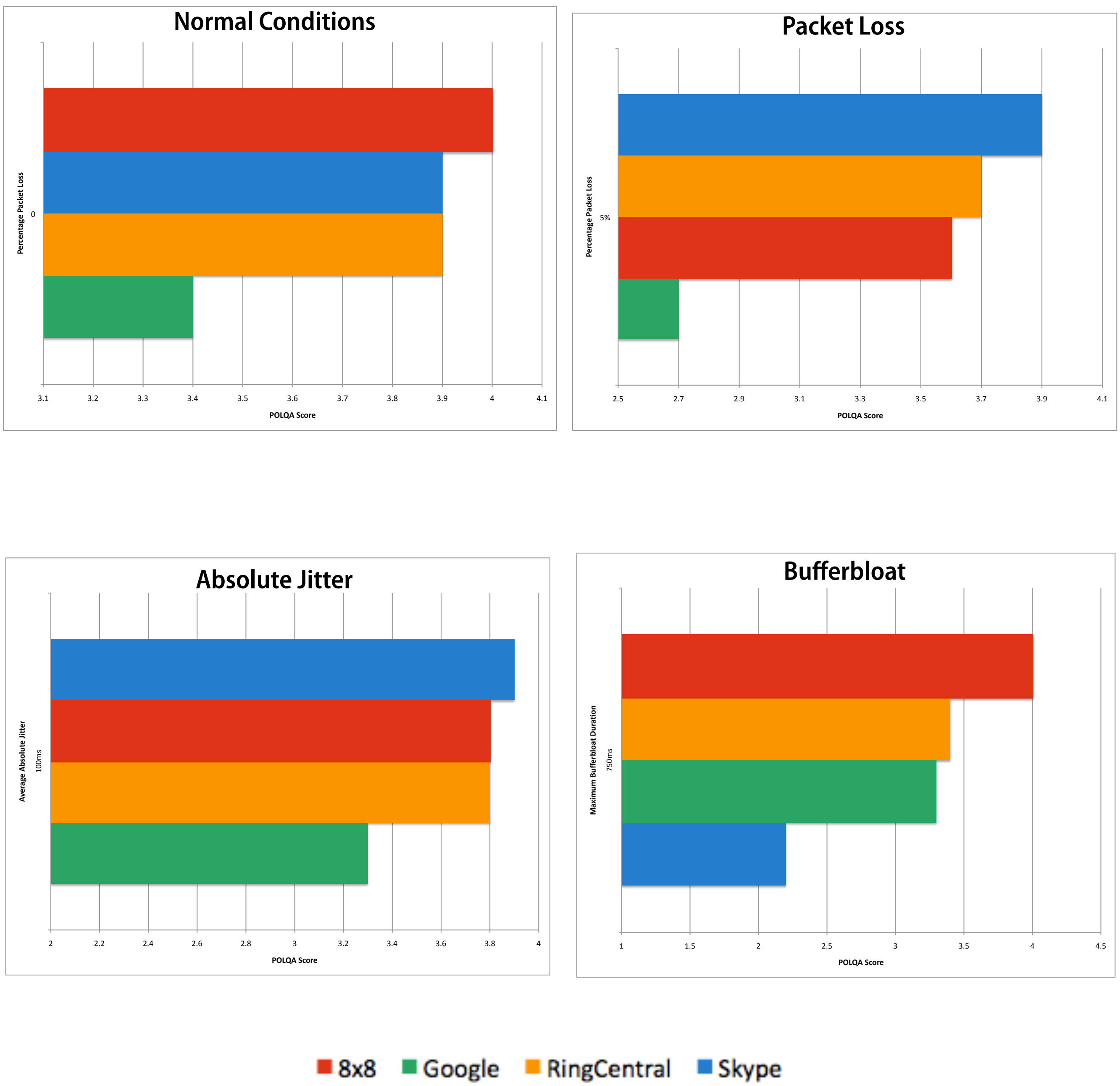
Absolute Jitter (100ms)

In the absolute jitter tests, Skype had the best score at 3.9 and 8x8 and RingCentral were slightly lower at 3.8. Google's score was 3.3.

Bufferbloat (750ms spike)

In the bufferbloat tests, the 8x8 solution delivered the best quality at 4.0. RingCentral and Google had scores of 3.4 and 3.3 respectively. Skype had the lowest score at 2.2.

Test Results - “PSTN-to-VoIP” - Apple iOS Clients



Source: Tolly, May 2017

Figure 5

Test Results - "PSTN-to-VoIP" Continued

Google Voice replaced Google Hangouts for this set of tests.

Google Android Clients

Packet Loss (5%)

The 8x8 solution delivered a quality score of 3.8 in this test. Skype and RingCentral also got the same score as 8x8 and Google Voice scored 2.7. See Figure 6 for all Google Android "PSTN-to-VoIP" results.

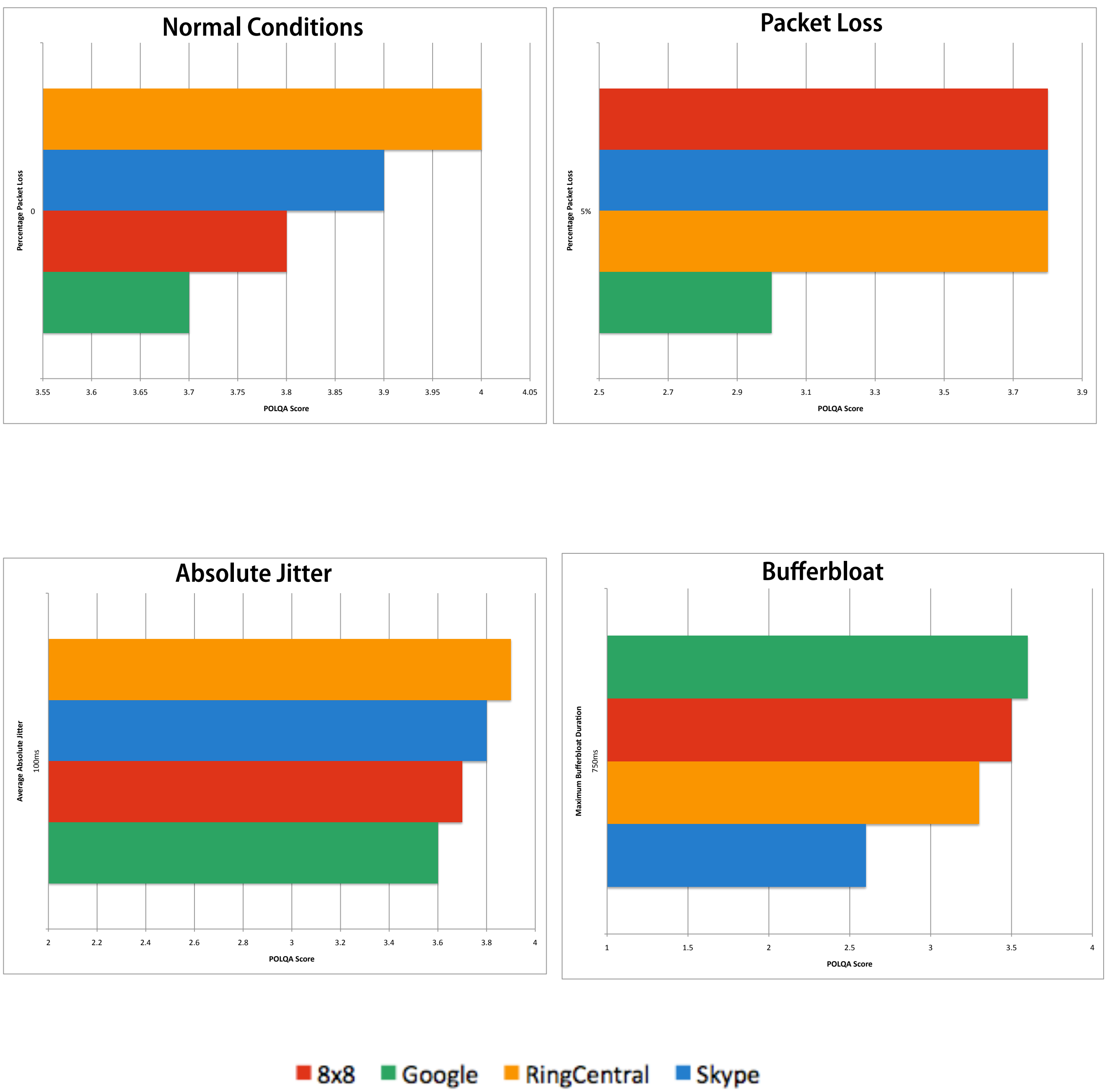
Absolute Jitter (100ms)

In the absolute jitter tests, Skype had the best score at 3.9 and 8x8 and RingCentral were slightly lower at 3.8. Google's score was 3.0.

Bufferbloat (750ms spike)

In the bufferbloat tests, the 8x8 solution had a quality score of 3.5 which was slightly lower than Google's score of 3.6. RingCentral had a score of 3.3 and Skype a score of 2.6.

Test Results - "PSTN-to-VoIP" - Google Android Clients



Source: Tolly, May 2017

Figure 6

Test Results - "PSTN-to-VoIP" Continued

Microsoft Windows Clients

Packet Loss (5%)

The 8x8 solution delivered the best quality score of 3.8 in this test along with RingCentral. Skype and Google had scores of 3.6 and 2.8 respectively.

See Figure 7 for all Apple iOS "PSTN-to-VoIP" results.

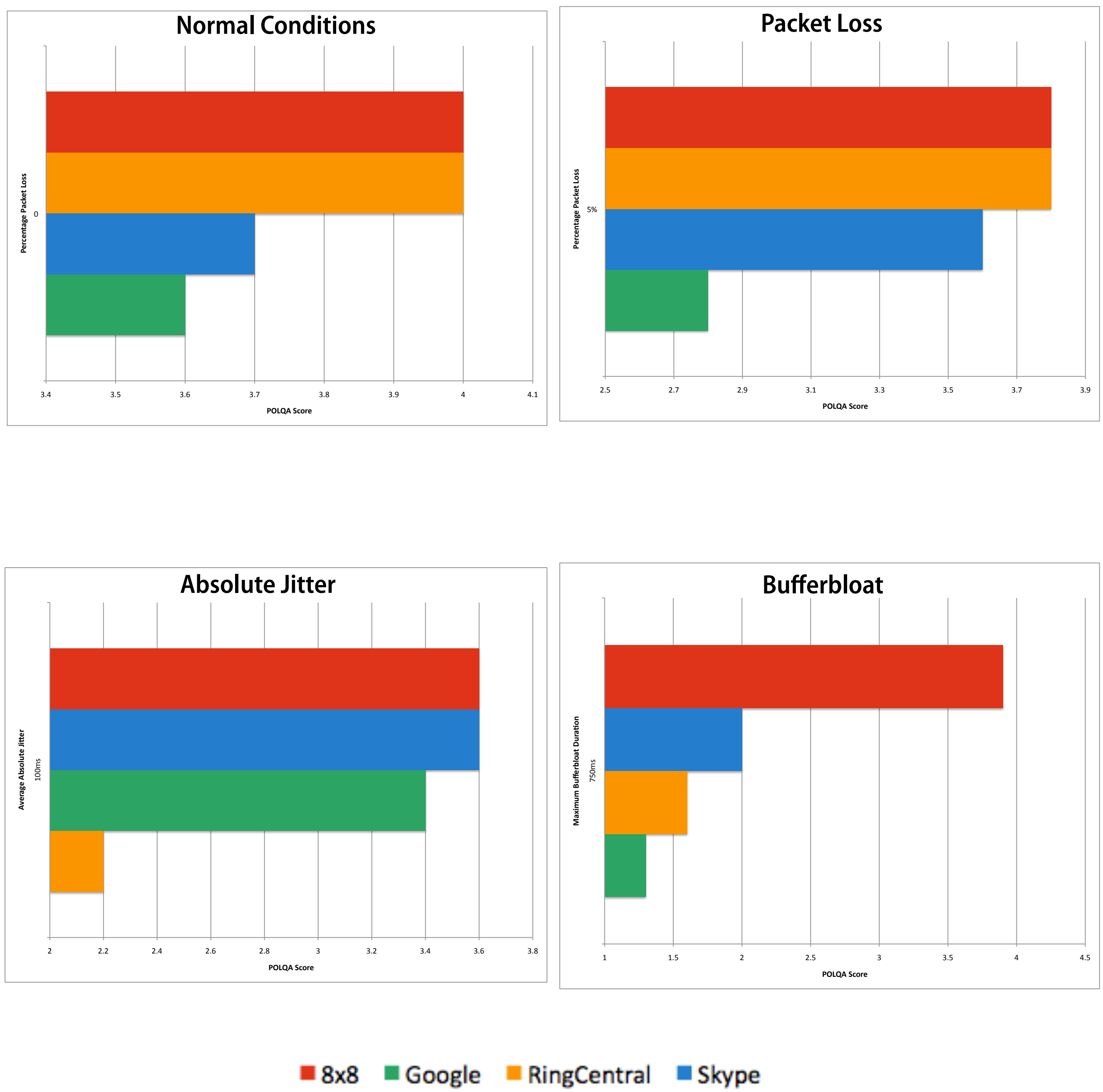
Absolute Jitter (100ms)

The 8x8 solution delivered the best quality score of 3.6 in this test along with Skype. Google and RingCentral had scores of 3.4 and 2.2 respectively.

Bufferbloat (750ms spike)

In the bufferbloat tests, the 8x8 solution delivered the best quality at 3.9. All of the other solutions had poor quality in this test. Skype and RingCentral had scores of 2.0 and 1.6 respectively. Google had the lowest score at 1.3.

Test Results - “PSTN-to-VoIP” - Microsoft Windows Clients



Source: Tolly, May 2017

Figure 7

Test Results - "PSTN-to-VoIP" Continued

Apple Mac OS X Clients

Packet Loss (5%)

The 8x8 solution delivered a quality score of 3.8 in this test with RingCentral having the best score of 4.0. Skype and Google had scores of 3.7 and 3.1 respectively

See Figure 8 for all Apple Mac OS X "PSTN-to-VoIP" results.

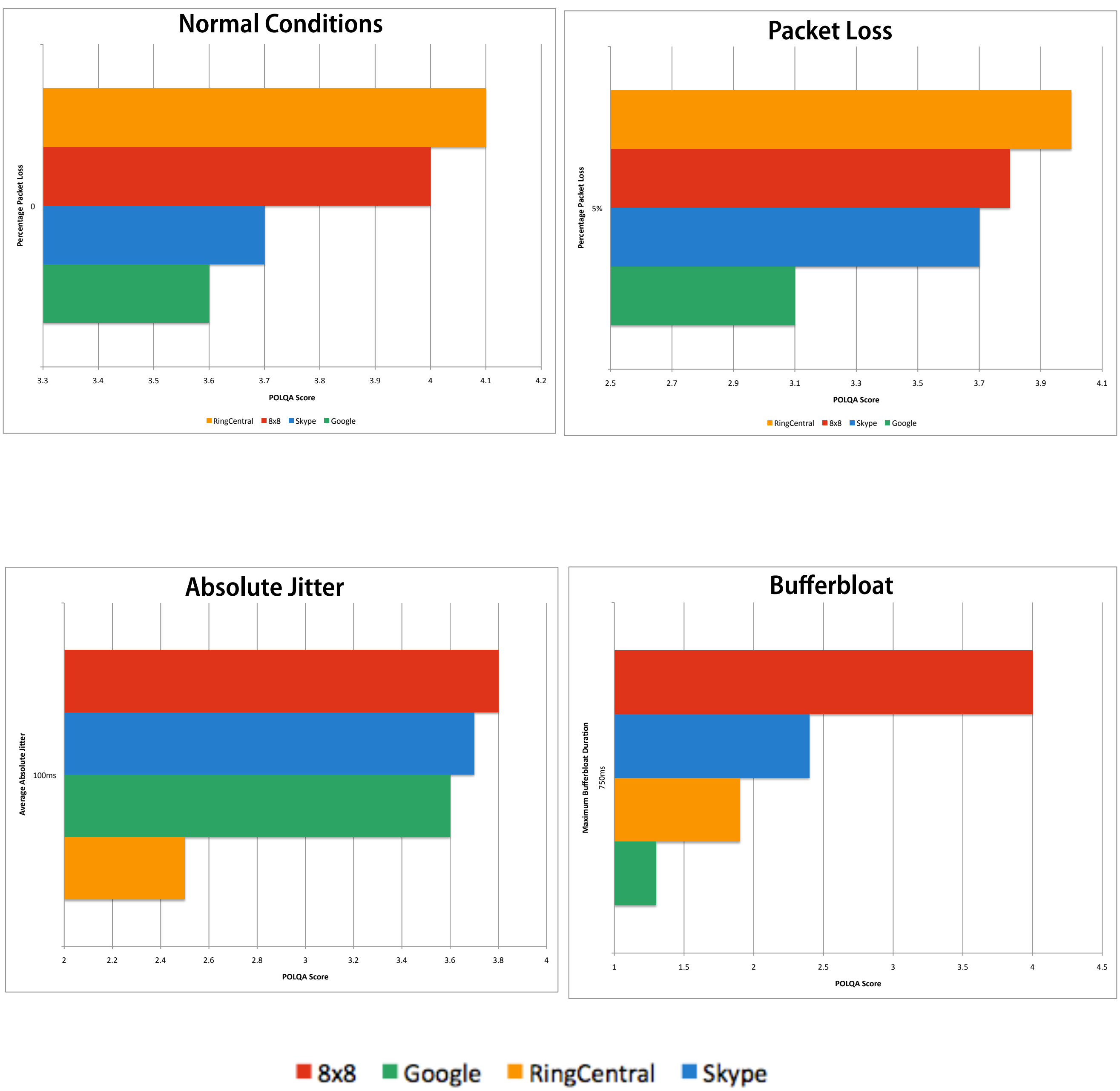
Absolute Jitter (100ms)

In the absolute jitter tests, 8x8 had the best score at 3.8. Skype and Google followed with scores of 3.7 and 3.6 respectively. RingCentral had the worst score at 2.5

Bufferbloat (750ms spike)

In the bufferbloat tests, the 8x8 solution had the high quality score of 4.0. The other solutions had poor quality scores with Skype delivering 2.4, RingCentral a 1.9, and Google a 1.3.

Test Results - "PSTN-to-VoIP" - Apple Mac OS X Clients



Source: Tolly, May 2017

Figure 8

Test Results - Call Continuity

High-Definition Client - Windows/Mac "On Net" -

These tests examined the impact on call continuity when the Ethernet LAN connection was disconnected and then reconnected during the call. Three variations were tested as outlined below. Approximately five seconds elapsed between the time the cable was disconnected and reconnected.

Tests were run using pairs and Windows clients and then using pairs of Mac OS X clients. Results were identical and, thus, are reported only once. Only 8x8 and Google Hangouts recovered the calls in all three scenarios.

Scenario 1: Unplug/replug Ethernet cable during call

All solutions were able to recover the call. See Table 1 for all call continuity results.

Scenario 2: Unplug/replug Ethernet cable during call changing IP subnets

8x8 and Google recovered the calls, RingCentral and Skype did not.

Skype simply dropped the call. RingCentral showed the call status as normal but there was no audio on the connection. Tolly engineers deemed this a dropped call.

Scenario 3: Unplug Ethernet cable during call and switch to WiFi

8x8 and Google recovered the calls, RingCentral and Skype did not.

For 8x8 and Google there was a very minor disruption as the call failed-over to WiFi. The 8x8 solution switched to WiFi faster than Google.

As with Scenario 2, Skype simply dropped the call. RingCentral showed the call status as normal but there was no audio on the connection. Tolly engineers deemed this a dropped call.

Call Continuity Test Results

"On-Net" Microsoft Windows & Apple Mac OS X Client pairs

Vendor	Solution	Test Scenarios		
		1: Unplug/replug Ethernet cable during call	2: Unplug/replug Ethernet cable during call changing IP subnets	3: Unplug Ethernet cable during call and switch to WiFi
8x8, Inc.	Virtual Office	Pass	Pass	Pass
Google	Hangouts	Pass	Pass	Pass
Microsoft	Skype For Business	Pass	Fail	Fail
RingCentral	Office	Pass	Fail	Fail

Note: Tests were run twice, once between pairs of Microsoft Windows clients and again between pairs of Apple Mac OS X clients. Results were identical and thus, are shown only once.

Source: Tolly, May 2017

Table 1

Test Methodology & Setup

Voice Communication Software Evaluated

Vendor	Solution	Software Versions by Platform			
		Apple iOS	Google Android	Microsoft Windows	Apple OS X
8x8, Inc.	Virtual Office	6.3.5	7.0.25000311	4.6.400	4.6.787
Google	Hangouts/ Voice	14.6.0	Dialer 19.0.154358895	Web-based, Chrome browser	Web-based, Chrome browser
Microsoft	Skype For Business	6.14.1.229	6.15.0.5	16.0.7766.7080	16.3.240
RingCentral	Office	9.0.2	9.0.2	8.4.5.25218	8.4.5.25218

Note: Apple iOS versions 10.3 and 10.2.1; Google Android versions 6.0 and 7.1.1; Microsoft Windows 7, Apple OS X 10.12.4.

Source: Tolly, May 2017

Table 2

Client Environment

All on net testing was run using the same pairs of clients for each vendor solution.

Apple iPhone: One iPhone 6s with iOS 10.3 and one iPhone 6s Plus with iOS 10.2.1.

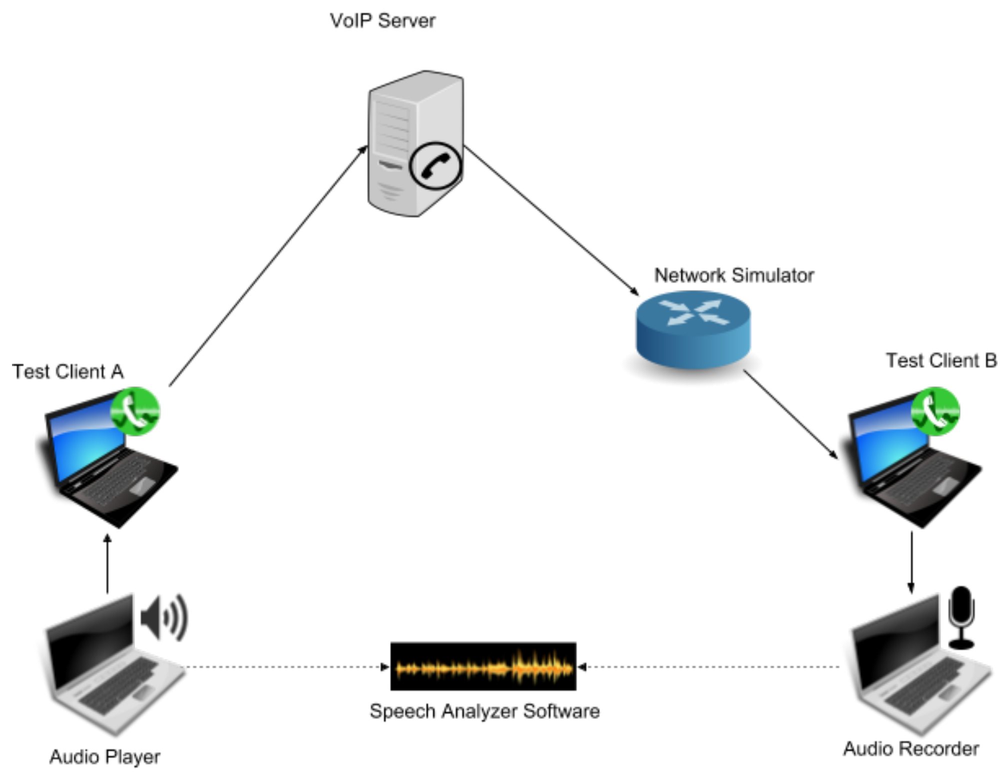
Android Phone: One Motorola Moto G (2nd generation) with Android 6.0 and one Google Nexus 6P with Android 7.1.1.

Microsoft Windows: Two Dell Latitude E6430 laptops with Microsoft Windows 7.

Apple Mac: One MacBook Pro (Retina, 15-inch, Late 2013) and one MacBook Pro (Retina, 15-inch, Mid 2015) with OSX 10.12.4.

All PSTN to VoIP tests were using one iPhone on the PSTN side as the Test Client A and the solution under test as the Test Client B. See Figure 9 for the test bed diagram.

Voice Quality Test Flow



Source: Tolly, May 2017 Figure 9

Test Methodology

In all tests, two Dell OptiPlex 390 desktop PCs were used as the Audio Player and the Audio Recorder (one for each). Audacity 2.1.0 applications installed on the two PCs were used to play and record audio. One audio cable was used between the line-out port of the Audio Player PC and the microphone/line-in port of the Test Client A. Another audio cable was used between the line-out/headphone port of the Test Client B to the line-in port of the Audio Recorder PC.

Operating system volume was set to maximum on each platform. The Audacity playing level was set to 0.25 for the Windows-to-Windows and Mac-to-Mac tests and 0.3 for the iOS-to-iOS, Android-to-Android, and for all PSTN tests the record level was set to 0.4. Those values were determined by calibration between the Audio Player and Test Client A as well as between Test Client B and the Audio Recorder.

PacketStorm400E IP network emulator with software version 15.7v3 was used for WAN emulation. The Perceptual Objective Listening Quality Analysis software developed by OPTICOM GmbH was used to compare the recorded audio file and the reference audio file. The LongRefSWB.wav file provided by the POLQA software was used as the audio file under test. Each test was run five times. The best and the worst results were excluded from the results and the remaining three results were averaged.

Network Impairment Information

The following details the impairments generated for these tests.

Packet Loss

Standard packet loss, randomly distributed - simple, independent probability assigned to dropping each packet (2%, 5%, 10%). Tests were also run with no (0%) packet loss.

Jitter

Absolute timing modification according to a Gaussian distribution, with averages of 20ms, 60ms, 100ms, 200ms.

Buffer-bloat

Frequent queuing and bursting of packets with varying burst sizes, according to the following four test sequences:

Flow 1 (Largest spike = 200ms)

- Queue packets for 50ms, then burst queued packets
- Normal flow for 1750ms
- Queue packets for 100ms, then burst queued packets
- Normal flow for 2100ms
- Queue packets for 200ms, then burst queued packets
- Normal flow for 2300ms
- Queue for 80ms, then burst queued packets
- Normal flow for 1420ms
- Loop back to step 1 and repeat for total loop time of 8000ms

Flow 2 (Largest spike = 400ms)

- Queue packets for 100ms, then burst queued packets
- Normal flow for 1700ms
- Queue packets for 200ms, then burst queued packets
- Normal flow for 2000ms
- Queue packets for 400ms, then burst queued packets
- Normal flow for 2100ms
- Queue for 150ms, then burst queued packets
- Normal flow for 1350ms

- Loop back to step 1 and repeat for total loop time of 8000ms

Flow 3 (Largest spike = 750ms)

- Queue packets for 200ms, then burst queued packets
- Normal flow for 1600ms
- Queue packets for 400ms, then burst queued packets
- Normal flow for 1800ms
- Queue packets for 750ms, then burst queued packets
- Normal flow for 1750ms
- Queue for 250ms, then burst queued packets
- Normal flow for 1250ms
- Loop back to step 1 and repeat for total loop time of 8000ms

Flow 4 (Largest spike = 1500ms)

- Queue packets for 400ms, then burst queued packets
- Normal flow for 1400ms
- Queue packets for 800ms, then burst queued packets
- Normal flow for 1400ms
- Queue packets for 1500ms, then burst queued packets
- Normal flow for 1000ms
- Queue for 500ms, then burst queued packets
- Normal flow for 1000ms
- Loop back to step 1 and repeat for total loop time of 8000ms

Call Continuity Tests

These tests are described in detail in the main text.

Appendix: Definitions and Issues Caused

Appendix Information provided by 8x8.

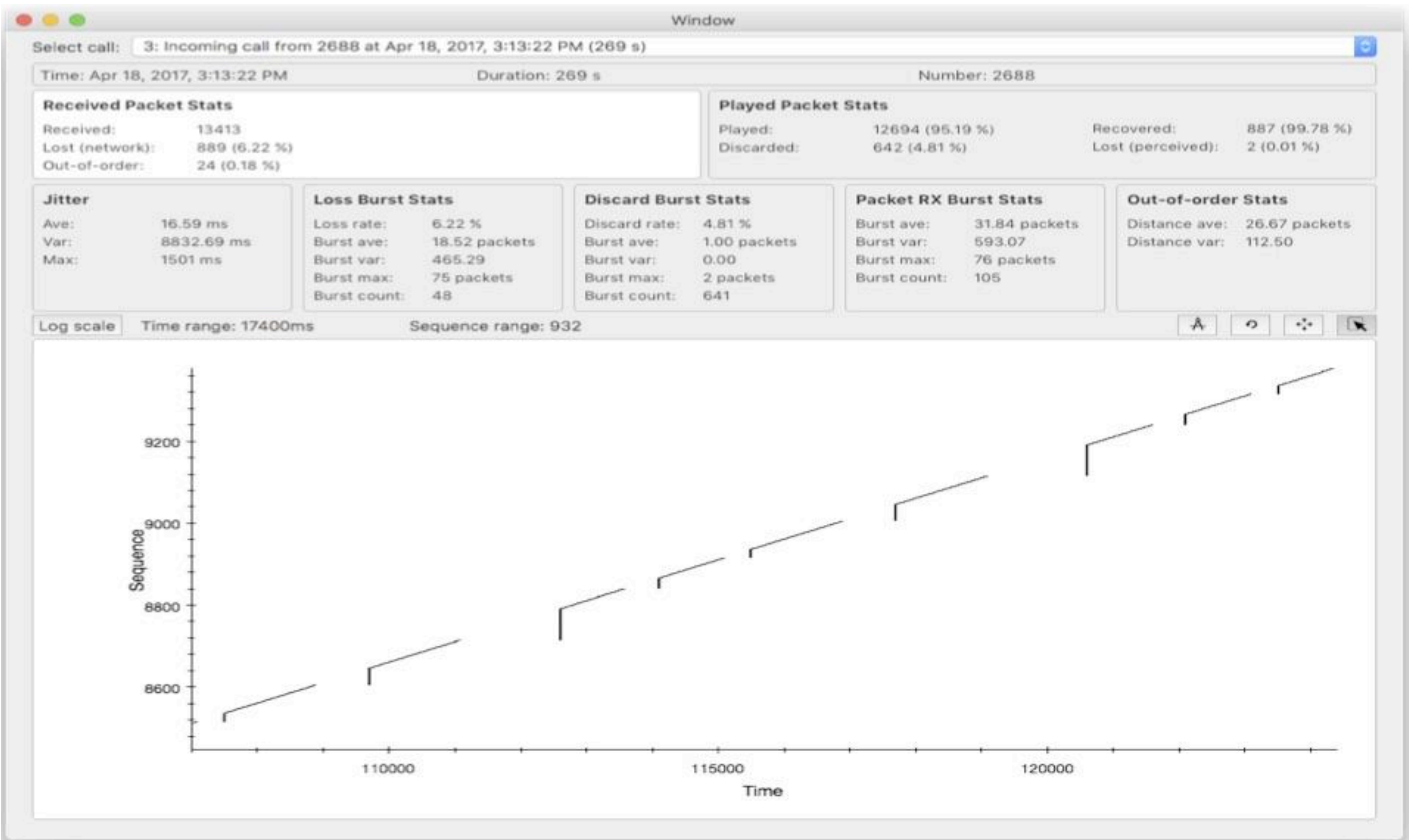
Bufferbloat

https://www.bufferbloat.net/projects/bloat/wiki/Introduction/

This is the undesirable latency that comes from a router or other network equipment buffering too much data. It is a huge drag on Internet performance created, ironically, by previous attempts to make it work better. The one-sentence summary is "Bloated buffers lead to network-crippling latency spikes." The bad news is that bufferbloat is everywhere, in more devices and programs than you can shake a stick at.

Unlike jitter where packets can arrive out of order, in the case of bufferbloat, there could be periods of time where there is no data, and all of a sudden a burst of data coming in, even in proper order. However, the effects are detrimental especially to many real-time applications including VoIP. The periods of silence (no data) and the burst of many packets arriving at the same time can completely distort and destroy the quality of a voice conversation.

Unfortunately in the recent years, due to the dropping price of memory and mis-conception of increasing performance with excessive buffering, too many network equipment including routers and WiFi devices have started to introduce more and more bufferbloat into the network paths. This phenomenon is still not as widely known or understood as jitter or packet loss, and not too many network tests reveal the level of



bufferbloat. Here is how bufferbloat looks like in case of packets related to the audio stream (this screen capture is for a call with severe bufferbloat measure by a tool created by 8x8 to visualize and analyze the effects of network conditions on audio streams):

Packet Loss

https://en.wikipedia.org/wiki/Packet_loss

This occurs when one or more packets cannot reach their destination. It is typically caused by network congestion. Packet loss is measured as a percentage of packets lost with respect to packets sent. Do you remember listening to a conversation and suddenly not hearing certain words? This is packet loss when you can't hear to the complete conversation.

Jitter

https://en.wikipedia.org/wiki/Jitter

This is the variation in the latency on a packet flow between two systems, when some packets take longer to travel from one system to the other. Jitter results from network congestion, timing drift and route changes. Have you experience distortion during a conversation? This can be caused by jitter. Also, excessive jitter can result in packet discards in real-time communications as packets arriving too late would be unusable.

POLQA

http://https://en.wikipedia.org/wiki/POLQA

Perceptual Objective Listening Quality Assessment, also known as ITU-T Rec. P.863[1] is an ITU-T Standard that covers a model to predict speech quality by means of digital speech signal.

- Successor of PESQ (ITU-T Rec. P.862)
- Can assess HD-Voice in wideband and super-wideband speech signals (50–14000 Hz)
- Score between 1.0 and 4.75

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Interaction with Competitors

In accordance with Tolly's Fair Testing Charter, Tolly personnel invited representatives from the competing companies to review the testing. No responses were received from any of the companies.

For more information on the Tolly Fair Testing Charter, visit:
http://www.tolly.com/FTC.aspx

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217121-WM-2-wt-yx-2017-05-24-VerK