

FEBRUARY 7, 2006

Motorola Canopy, Advantage WirelessBroadband Platform

Voice over IP (VoIP) over Canopy Advantage Platform



WEST MONROE PARTNERS

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Executive Summary

Motorola's Canopy® platform is a last-mile solution for wireless broadband. It has been accepted globally in areas where creating a wired infrastructure is either impractical or impossible. With the Canopy Advantage platform, Motorola has worked to improve on a technology that has emerged as a leader in the wireless broadband market. West Monroe Partners, an independent third party consulting organization, previously conducted testing on the original Motorola Canopy platform. Further tests were required to thoroughly evaluate the capabilities and limitations of the new Canopy Advantage platform, so Motorola retained West Monroe Partners to conduct this next phase of testing and document the findings and recommendations in this whitepaper.

A lab was created to simulate a number of different scenarios and determine factors that limit performance as well as the breaking points of the system. In addition to the lab, numerous customer interviews were conducted. The intention of these interviews was to help the team model their test scenarios based on the needs and experiences of customers in the field; they also gave insight into areas of concern for customers as well as how customers are using their Canopy Advantage systems. In addition to personal interviews, an online survey was conducted to poll Canopy customers that were deploying Voice Over IP (VoIP) in the field.

The lab testing consisted of three phases. The intention of the initial phase was to confirm that after completing software upgrades, the new Canopy Advantage hardware did not have any compatibility issues with the older Canopy equipment. Throughput, response time, and call quality were monitored to validate performance. The lab was configured with four Subscriber Modules (SMs) from each hardware platform (Canopy and Canopy Advantage platforms).

The second phase of testing was used to determine limitations and breaking points on the Canopy Advantage platform, as well as a subset of this platform Canopy Lite. Ixia's Chariot program was used to generate VoIP calls to and from the SMs and an Access Point (AP). Chariot also gathered data on a variety of variables including loss, delay and jitter. These data points were used to generate a computed voice quality score on the Mean Opinion Score (MOS) scale that is a generally accepted measure of voice quality in the industry. The MOS scale rates the quality of calls on a numeric scale. Customers score calls on a scale from "nearly all users dissatisfied" with a lower limit of 2.58 to "very satisfied" with a lower limit of 4.34. A number of configuration options were modified to determine the breaking points as well as optimal performance configurations of the Canopy Advantage platform.

The final phase of testing was focused around the improvements in the Canopy platform's Quality of Service (QoS) mechanism. Testing was conducted to validate the benefit of implementing QoS over Advantage. Ixia's Chariot program was used to generate the VoIP calls in addition to data streams for the testing scenarios. Data points were gathered on loss, jitter, delay, and MOS score. These scores were compared to the same scenarios run without QoS to determine and identify performance improvements by utilizing QoS. Motorola's improvement on the Canopy platform's QoS mechanism offers a higher level of interoperability with applications and other networks.

In conclusion, it is the finding of the independent third party, West Monroe Partners, that the Canopy Advantage platform is a stable and very viable option for the transmission of VoIP over wireless services.

Motorola's improvements in call scheduling, QoS and reduced system latency allow the call volume capacity of the system to sustain a high call quality level, while still allowing a provider to optimize their business plan and wireless network capacity for VoIP services.

The lab findings and recommendations of West Monroe Partners have also been further validated through a market research study of Motorola's customer success stories in the field with VoIP over Canopy deployments.

Testing has showed that the Advantage platform is capable of handling 26-28 simultaneous voice calls per AP with a 50% downlink configuration. These calls can be spread over numerous SMs, with each SM capable of handling at least 12 calls. This number was shown to be consistent regardless of the voice codec used. In a 75% downlink configuration, G.711U (a toll-quality voice codec) was able to handle 13-18 simultaneous calls per AP. These figures were verified through the customer interview process to confirm what customers are experiencing in their actual VoIP deployments in the field.

Additionally, testing has demonstrated that the Canopy Lite platform is a strong solution for low cost residential applications. The Canopy Lite SMs are well suited for transmitting 1 to 2 VoIP calls along with other data traffic. In this solution Canopy Lite SMs were found to provide toll quality VoIP phone calls.

It is expected that a Wireless Internet Service Provider (WISP) can provide a successful VoIP implementation over Canopy Advantage networks when considerations are given in the areas of network planning, network design, and voice network capacity planning. These are the biggest factors that will ultimately define the success of the implementation.



Field Tested vs. Theoretical Findings

Theoretical findings are based on calculations and extrapolations from gathered data. The calculations used in creating theoretical findings took into account a limited amount of known variables and extrapolated findings. While this approach provides a good idea of what a given system is capable of, it does not take into account any unknowns that occur in the field. In working with radio frequency (RF), it is important to do lab analysis of the area and the proposed solution, but it is just as important to test the solution in the field. Each environment will have a different effect on RF signals, and some of the traits that occur cannot be accounted for in a lab setting.

The testing completed for this white paper was completed in a lab, and was additionally validated by conversations with customers that are currently deploying VoIP over Canopy systems in the field. West Monroe Partners was able to achieve levels of service and performance in the lab that were consistent with what Canopy Advantage customers were seeing. These customers have gone into the field with the Canopy Advantage platform and completed testing. It is through their testimonials that we feel our results are consistent with what a customer can expect to see after deploying Canopy Advantage systems in a production environment.

What is the most successful part of your VoIP deployment?

- *"Versatility"*
- *"Cost effective to start..."*
- *"the fact that the customer can now get rid of copper wires from the big telcos in Norway"*
- Canopy Customer

The testing consisted of the use of nine SMs and one AP at our lab, all using the Canopy Advantage platform upgraded to the latest software release. The customers that were interviewed had up to 50 SMs with two six-AP clusters mounted on radio towers at distances of over two miles. The testing showed that the AP hits a real world limit. The more extensive testing in conjunction with the field testing completed by customers has given this testing a higher degree of accuracy with regard to expected performance levels in the field. The details of these results are located in the "Findings" section of this paper.



Improvements to the Canopy Platform

The original Canopy 5.7 GHz Access Point modules were designed to handle approximately 7 Mbps of aggregate throughput, with a latency range of 15 – 20 milliseconds. The Canopy Advantage platform Access Points are designed to handle up to 14 Mbps with a latency range of five to seven milliseconds. This reduction in latency and increase in bandwidth will allow for better performance with many broadband applications, including VoIP.

In addition to the improvements in bandwidth and latency, the Canopy Advantage platform also improved its offerings in terms of service levels and Quality of Service. Each subscriber module now has the ability to be configured to create a Committed Information Rate (CIR). This allows for a provider to have greater control over the network. A configurable CIR also allows for the creation of a tiered service offering. An administrator would be able to charge more for an increase in a subscriber's CIR.

One of the most significant changes that the Advantage platform offers is the hardware scheduler. The original Canopy platform used a software scheduler. Hardware scheduling allows for the dynamic use of control slots by the SMs and AP. With hardware scheduling, six configuration options are replaced by a single option. With software scheduling, the number of control slots allotted can be a factor in the performance of VoIP over Canopy systems.

As you increase control slots in software scheduling, you allow for more service

requests, but take away from overall bandwidth. With hardware scheduling, the user "reserves" a minimum number of control slots. These control slots are then always available for service requests. In addition, hardware scheduling allows for the use of non-reserved slots to send service requests. If a slot is not being used for data in a given frame, then it is available for use by a service request. This dynamic allocation of "reserved" slots is what allows for more efficient use of the available bandwidth.

"Canopy is the Cadillac of L2 wireless solutions!"
- **Canopy Customer**

Hardware scheduling has effects on the behavior of the high priority channel as well. Both the Canopy and Canopy Advantage platforms offer configurable high and low priority channels. In the Canopy platform, these channels are statically defined. These channels are always "on", regardless of traffic being passed on them or not. With the hardware scheduling improvements to the Canopy Advantage system, high priority traffic travels over "virtual channels". This allows for them to be used by low priority traffic if the high priority channels are not in use.

Another system improvement is the handling of Quality of Service (QoS). In the original Canopy platform, QoS was applied through the use of the Type of Service (TOS) field in the IP header. If the TOS bit is set for high priority the AP will prioritize this traffic in the queue and hold back any data that is not designated as such. The high priority designation in the original Canopy platform is a static allocation, meaning that when a number of slots are reserved for high priority they can only be used for this purpose. If no high priority traffic is being passed the designated high priority bandwidth will remain idle and unavailable for other traffic.

Canopy Advantage AP's using hardware scheduling handle the high priority traffic differently. The high priority channel is now dynamically allocated. This will allow for standard traffic to utilize the full bandwidth between AP and SM when no high priority traffic is being passed. As the need for high priority traffic arises, the AP will allocate the necessary amount of bandwidth in a separate

virtual channel. In addition to the changes in managing the high priority channel, Motorola has also moved away from using the TOS field. Differentiated Services (DiffServ) is the chosen replacement. DiffServ is becoming the industry standard in regards to QoS. It allows for the configuration of 6 bits (creating code points) in the TOS field that allows for 64 service level variations. DiffServ is also backwards compatible with the Type of Service bit settings. The DiffServ code points are then mapped to a priority within the Canopy Advantage network.

The Canopy Advantage platform also uses a software-based radio. This allows for the system to not be dependent on a specific chipset. This creates the ability to add additional features to the system in the future, without the need to retire hardware. The Canopy Advantage platform is also compatible with all current Canopy modules. During our interviews and surveys, we confirmed that many Canopy providers are running their networks with a mix of both Canopy and Canopy Advantage hardware platforms.

Motorola has also created a subset of the Canopy Advantage platform. This subset is called Canopy Lite. It works within the Canopy Advantage platform to provide a low cost alternative to Advantage SMS. These units feature a 512 Kbps throughput with a 768 Kbps burst rate. It additionally offers a 100 Kbps Committed Information Rate (CIR). These units have all the features and reliability that are incorporated into the Canopy Advantage platform. The only limitation comes from the throughput limits.



Testing Scenarios

A test lab was created to simulate VoIP calls to document and benchmark the quality of VoIP traffic over the Canopy Advantage network. In the lab the scalability of the Canopy Advantage platform was also tested, as well as the baseline differences/similarities between the Canopy and Canopy Advantage hardware.

The first phase of testing was to determine what differences existed between original Canopy hardware and Canopy Advantage hardware (please see Appendix for more detail on the hardware specifications). Canopy hardware Subscriber Modules (SM) were updated to the newest release of the Canopy Advantage software, 7.2.9. A series of simulated calls were run using Ixia's Chariot product between the four Canopy modules and the four Canopy Advantage modules. Results were gathered regarding throughput, delay, jitter, and a computed MOS score. The network was configured in a point to multipoint network with default configuration settings.

The second phase of testing was to determine the scalability of VoIP calls over an increasing amount of registered SMs. Ixia's Chariot product was again used to simulate VoIP traffic as well as data traffic between the SMs and the Access Point (AP). The network was configured to reproduce a data moving network. We tested all scenarios at both 50% downlink and 75% downlink. A number of elements in the network were varied to determine the different breaking points of the Canopy Advantage platform. The variables in the testing included:

- Number of Calls per SM
 - The number of calls originating from an SM was varied from 2 – 15. This was varied to discover the breaking point (where call quality began to degrade a considerable amount) of both the SM and the AP.
- Number of SMs passing traffic
 - The number of SMs passing VoIP traffic was varied from 2 – 9 SMs. This was to discover any issues that might occur as more SMs are added to the network.
- Compression Algorithm (Codec)
 - The codec was varied to show determine the difference in both call quality and total volume handled by the SMs and APs.
- Voice Traffic with and without data traffic
 - Data traffic was generated to determine its effects on call volume and call quality.
- Quality of Service – Differentiated Services Code points
 - Tests were run with QoS on and off with data to determine what effect it had on MOS score, delay, and jitter.

In the third phase of testing, Quality of Service tests were run to determine the overall effectiveness of implementing QoS on the Canopy Advantage platform. Code points were configured in the testing scenarios and then mapped to priority levels in the configuration of AP. There were two different scenarios for data transfer with voice calls. The first scenario was deemed a "typical web user". This scenario consisted of two constant data streams; a 64kbps download stream and a 10 kbps uplink stream. These were created per SM, in addition to the voice calls. The second scenario was an "aggressive" model. It consisted of a 64 kbps downlink and a 33.6 kbps uplink. Both streams were again constant and generated per SM in addition to the voice calls. The goal of the two testing scenarios was to simulate the different types of user traffic. A typical web use would consist of a high level of downlink traffic with very little uplink. This is consistent with web surfing, email use, and instant messaging. The "aggressive" model was

modeled around a higher rate of traffic, similar to what one would expect from a Peer-to-Peer file sharing user. This type of traffic requires a much higher uplink rate.

Additional testing was conducted utilizing two Canopy Lite SMs. The main goal of the testing was to determine the breaking point of the Lite SMs. Several variables were modified including:

- Number of Calls per SM
 - The number of calls originating from an SM was varied from 2 – 10. This was varied to discover the breaking point of the SM (where call quality began to degrade a considerable amount).
- Compression Algorithm (Codec)
 - The codec was varied to determine the difference in both call quality and total volume handled by the SMs.
- Voice Traffic with and without data traffic
 - Data traffic was generated to determine its effects on call volume and call quality.

Canopy Advantage Configuration

The Canopy Advantage network consisted of nine SMs setup in a point to multipoint network with a single AP. Testing consisted of creating traffic from a variable number of SMs, but all SMs were registered with the AP. All configuration options were left at factory default, except for the following changes:

- Uplink / Downlink percentage
 - Tests were conducted with both 50/50 configuration and 75/25 configuration (downlink/uplink).
- Quality of Service
 - Tests were conducted with DiffServ code points configured and the high priority channel on and off.
- All other configuration options were left at factory default.

Canopy Lite Configuration

The Canopy Lite network consisted of two SMs setup in a point to multipoint network with a single AP. Testing consisted of creating traffic from both SMs to the AP. All Canopy configuration options were left at factory default, except for the following changes:

- Uplink / Downlink percentage
 - Tests were conducted with both 50/50 configuration and 75/25 configuration (downlink/uplink).

Ixia Chariot Configuration

Ixia's Chariot is a Layer 4 through Layer 7 traffic generation program. It is designed to simulate traffic and load on a network without having to install all the desired services. The program consists of two pieces, a console and endpoints. The console is the driving force behind the traffic generation. Test scenarios were created, started, and monitored using the console. The endpoint software runs as a Windows service on the end devices. In our lab, the console software was connected to the AP and the endpoint software was installed on PCs connected to the SMs.

Lab Setup

The lab was configured using one Canopy Advantage Access Point and nine Subscriber Modules. Four of the Subscriber Modules were shipped with Canopy hardware and five were shipped with Canopy Advantage hardware. Ten laptops were configured with Windows XP Service Pack 2. The firewall was turned off and each was given a static IP address. A high level

configuration can be seen in the diagram below. All Canopy Advantage hardware was the 5.7 GHz model.

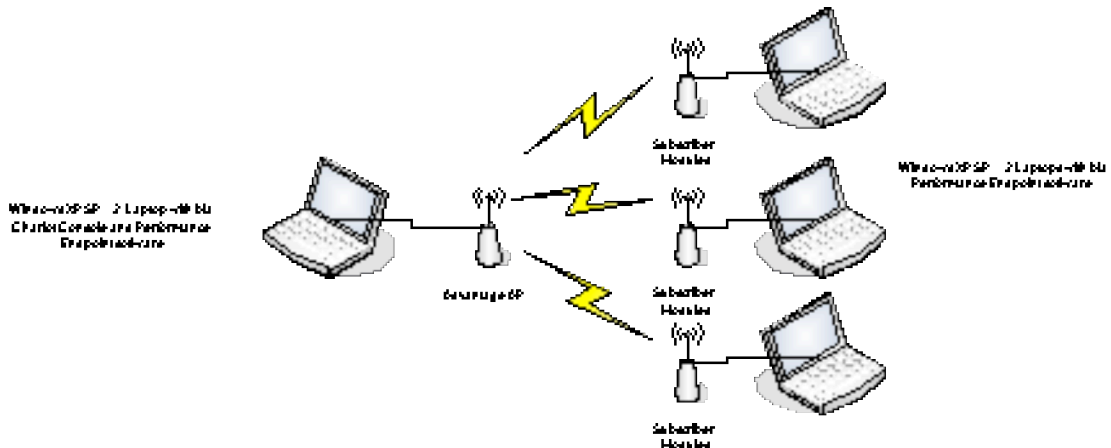


Figure 1: High Level Network Diagram

The Canopy Lite lab setup was modeled the same as the Canopy Advantage platforms lab setup. The lab was configured using two Canopy Lite SMs and one Canopy Advantage AP. Three laptops were configured with Windows XP Service Pack 2. The firewall was turned off and each was given a static IP address.



Findings

Canopy Hardware vs. Advantage Hardware

One of the benefits of creating a Canopy Advantage network is the ease of deployment. An all Canopy network can be updated to a Canopy Advantage network with a minimal amount of hands-on work needed.

All of the Canopy SMs that are currently deployed in the network can be upgraded to the Canopy Advantage platform via a software upgrade through the air. There is no on-site access needed for SM upgrades. Most work and configuration options can be done using Motorola's free Canopy Network Upgrade Tool (CNUT), available for download through the Canopy website. Once the tool is downloaded, follow the upgrade steps referenced in CNUT online help documentation. After upgrading the SM, the AP is the only piece of equipment that needs to be replaced, and even that can be converted into an SM.

Our lab was originally shipped four Canopy SMs and five Canopy Advantage SMs. After completing the update process, the entire network was using the Canopy Advantage platform. All test scenarios were run on the Canopy Advantage platform.

The upgraded Canopy Subscriber Modules provided equivalent results to the Canopy Advantage Subscriber Modules. The MOS score values were consistently in the same range and no trending could be found that would indicate an advantage of one set of hardware over the other. Values were also taken for throughput and response time when the two platforms were transferring data only. Response time is measured in seconds, and it represents the amount of time it took to complete the transaction. It is an end-to-end measure of the setup and tear-down of the file transfer. The results show that Canopy platform hardware, when updated to the software release 7.2.9 and being driven by a Canopy Advantage platform AP will provide nearly identical performance to the newer Canopy Advantage Subscriber Modules. The table below shows an average score over five separate tests for both throughput and response time.

Platform	Throughput Avg. (Mbps)	Response Time Avg. (s)
Advantage	4.251	1.7526
Canopy	4.294	1.7568

Table 1: Hardware Differences

It should be noted that all hardware used in this testing was the P9 model with software release 7.2.9. With software releases prior to 7.3.6, P8 models were unable to take advantage of the high-priority channel with hardware scheduling. This has been fixed with software revision 7.3.6. Additionally, P8 hardware is not capable of 2X modulation that reduces overall throughput capacity. Our testing was not affected by this limitation as we did not use P8 hardware.

Based on this testing we can conclude that when deploying new Canopy Advantage SMs in an existing network, the existing Canopy SMs do not need to be replaced. The only step that needs to take place is to update the Canopy SMs with the latest software release using the Canopy Network Update Tool, which can be centrally managed and pushed out to the network. This would allow for a provider with a Canopy Advantage AP to provide the same VoIP capabilities to

both Canopy and Canopy Advantage SMs. While the VoIP throughput would be equivalent, the data throughput available to the Canopy Advantage SMs would be greater.

Voice over IP Call Volume

West Monroe Partners found that an AP could handle approximately 26 to 28 calls in an all data environment, independent of codec. Subscriber Modules were found to individually be able to handle at least 13 simultaneous calls. Each call was observed to achieve a MOS score of above 4.0. The MOS score is a numerical value indicating the overall quality of a voice call. A score of 4.0 is the industry standard for a satisfactory quality level. Any MOS score below 4.0 was considered unacceptable.

The codecs used in this phase of testing were shown to be representative of what customers in the field were using. G.711U, G.726, and G.729 are ITU standard codecs. Through customer interviews and surveys, it was shown that when a Canopy network provider implements a VoIP deployment the choice of codec generally was one of the above three.

One of the most apparent distinctions made when modifying variables was the effect that downlink configuration had on VoIP call performance. At 75% downlink, tests were run with equal amounts of calls from varied number of SMs until the breaking point was found. The breaking point was considered reached when the average MOS score fell under 4.0. The breaking point was found to be varied amongst the different codecs. The codecs using the most bandwidth were able to handle the least amount of calls and the codecs using the least bandwidth were able to handle the highest amount of calls. The breaking point for these scenarios occurred due to bandwidth constraints on the uplink connection. As the breaking point was approached, jitter from SM to AP increased. Jitter from the AP to the SM stayed at a constant rate, verifying that the bottleneck was with the uplink connection. The table below shows the various breaking points among the different codecs with a 75% downlink configuration. As more calls were added to each scenario, the MOS score continued to drop at a fairly consistent rate. A more detailed view of the different breaking points can be found in the Appendix.

An “n/a” in the “Number of calls from each SM” column represents that an even number of calls was not placed from all SMs. For example, for the 9 SMs with 13 total calls, 4 SMs had two calls and 5 had only one call.

Codec	Number of SMs	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.711u	9	n/a	13	4.12	11	2.64	0.010
G.711u	8	n/a	13	4.11	13	2.85	0.078
G.711u	7	2	14	3.97	21	4.07	0.148
G.711u	6	n/a	15	4.04	32	3.56	0.226
G.711u	5	n/a	16	3.99	30	3.29	0.396
G.711u	4	n/a	17	4.07	32	2.42	1.125
G.711u	3	n/a	17	4.01	23	2.29	0.833
G.711u	2	9	18	4.19	12	1.75	0.328

Table 2: Various breaking points for G.711u at 75% Downlink Configuration

Codec	Number of SMs	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.726	9	n/a	15	4.02	12	2.40	0.019
G.726	8	n/a	15	4.06	10	1.92	0.015
G.726	7	n/a	16	4.04	10	2.58	0.047
G.726	6	n/a	17	4.00	11	2.93	0.092
G.726	5	n/a	18	4.04	16	2.85	0.061
G.726	4	5	20	4.09	21	2.67	0.059
G.726	3	n/a	20	4.08	12	2.31	0.131
G.726	2	n/a	21	4.04	19	2.10	0.214

Table 3: Various breaking points for G.726 at 75% Downlink Configuration

Codec	Number of SMs	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.729	9	2	18	3.98	10	2.56	0.020
G.729	8	n/a	20	3.96	15	3.16	0.039
G.729	7	3	21	3.95	21	3.74	0.171
G.729	6	n/a	22	3.97	21	3.07	0.129
G.729	5	n/a	26	3.97	63	2.76	0.020
G.729	4	n/a	26	3.93	76	3.05	0.100
G.729	3	n/a	23	4.00	32	2.40	0.019
G.729	2	12	24	3.98	54	2.16	0.000

Table 4: Various breaking points for G.729 at 75% Downlink Configuration

When the configuration was modified to test a 50% downlink configuration, the results changed dramatically. These tests showed that independent of codec, the AP could handle between 26-28 calls. As the breaking point was approached, the amount of lost bytes from AP to SM increased dramatically. This verifies that this is a constraint on the AP side of the connection. The following table provides a section of data to support the above theory. The table shows data results from the G.711U codec with seven SMs passing calls. Throughout testing, all data points showed a small gradual increase. As the breaking point was approached, the data shows a dramatic increase in percent bytes lost from the AP to the SM as well a significant reduction in call quality, indicated by the MOS score. This behavior was consistent across the different codecs and independent of the amount of SMs passing calls.

Traffic Flow	Codec	Downlink %	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
AP to SM	G.711U	50	26	4.24	78	2.229	0.113
SM to AP	G.711U			4.20	80	4.592	0.096
AP to SM	G.711U	50	27	2.08	83	2.734	4.099
SM to AP	G.711U			4.13	82	5.521	0.041

Table 5: 50% Downlink Configuration Breaking Point for 7 SMs.

The important distinction to note is that by simply adjusting the downlink percentage, a provider can quickly and easily increase the number of customers (or calls per customer) that their infrastructure can support. Close monitoring is needed to ensure that you have the link set properly. The majority of residential traffic heavily favors the downlink. If the network is configured with the uplink percentage unnecessarily high, bandwidth will be unavailable for use. On the other hand, VoIP traffic is symmetric by nature, favoring a 50% downlink configuration. The closer the configuration is to 50/50, the better performance of the VoIP traffic. Customers interviewed agreed that this was a very important factor in VoIP performance. Many customers have already begun to slowly shift the bandwidth configuration to meet the demands of their network.

Quality of Service

Quality of Service tests were run to determine the improvement gained when implementing QoS on the Canopy Advantage platform. Tests with two scenarios of data streams were run with QoS both on and off. The results of the “aggressive” model showed the greatest improvement. This was to be expected, as the more data pushed on a network, the greater the effect of shaping that traffic. The effects of QoS can be seen below. In a business setting, this shows that you may not see an immediate jump in performance on networks that are not saturated. This is due to the nature of the QoS. As the network approaches saturation, you will see a greater boost in performance when QoS is applied. When customers were asked in the field about QoS, often times they felt it was not necessary to implement. The network was not operating near saturation, and they were not pushing the limits of the AP or SM in terms of VoIP calls. However, in a more saturated network environment, the flexibility of the Diff Services configuration in the Advantage QoS mechanism provides a level of administrative control that was not previously available in the older version of the Canopy platform. The statistics below represent the average change when QoS is applied to the different testing scenarios. Statistics represented in green demonstrated an increase in value, and scores in red demonstrated a decrease in value.

Statistic	Scenario	
	Aggressive	Typical Web Use
MOS Avg.	.24	.06
One-Way Delay Avg. (ms)	9.89	4.55
Jitter (ms)	.74	.12
Percent Lost Bytes	.34	.15

Table 6: QoS Results

Canopy Lite

The technology behind Canopy Lite is similar to the technology in Canopy Advantage SMs. The main difference is the throughput limit. The limit on throughput was expected to be the limiting factor testing VoIP capabilities. Almost all codecs acted as expected with regards to call volume.

G.711U was able to place two simultaneous calls per SM with no quality degradation. When one third call was added, the number of dropped packets increased dramatically, causing call quality to suffer. G.726 was able to place 4 simultaneous calls per SM with no quality degradation. As a fifth call was added, an increase in dropped packets was recorded. This resulted in a lower MOS score. The G.729 codec supported twice the number of calls as the G.726 codec.

Codec	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.711U	2	4	4.37	5	.749	0.000
G.711U	3	6	2.80	6	.430	2.794
G.726	4	8	4.17	6	.376	0.000
G.726	5	10	1.91	5	.410	8.111
G.729	6	12	4.03	5	.770	0.000
G.729	8	16	4.03	8	1.120	1.720
G.729	9	18	1.77	7	.879	10.148

Table 7: Canopy Lite

G.711U and G.726 have known data rates of 64 Kbps and 32 Kbps each direction, respectively. This is shown in Table 19. Canopy Lite has a 512 Kbps throughput limit that appears to be evenly distributed between uplink and downlink. This allows for a max sustained rate of approximately 256 Kbps each direction. By looking at the compression rate for G.711U, we see a maximum of 2 calls can be placed before data begins to drop ($256 / 128 = 2$). Remember that each stream for the call is 64 Kbps of bandwidth, equaling 128 Kbps for a full duplex call. The calculations are similar for G.726.

The findings for the G.729 codec follow from the packet statistics as shown in Table 20. As indicated the G.726 and G.729 codecs produced the same number of packets per second, but the G.729 codec packet was about half the size due to unchanged packet overhead. So, even though the voice compression is four times better, the G.729 codec consumes half the channel bandwidth. Consequently G.729 can be expected to have only double the call capability as does G.726, and this was the result obtained.

Downlink percentages were modified and tested with both 50% and 75%. Tests were shown to have the same breaking points and similar statistics on jitter and bytes lost. This is to be expected due to the amount of bandwidth available from a Canopy Advantage AP. The 256 Kbps limit does not allow the Canopy Lite SMs to approach utilizing all of the AP's available bandwidth.

In aspects relating to QoS, Canopy Lite was shown to behave identically to the Canopy Advantage platform. However, due to the low amount of bandwidth, the increase in performance was negligible.

Overall Findings and Recommendations

Based on the testing completed the following Conclusions and Recommendations are being made:

- Canopy SM hardware performs equally with Canopy Advantage SM hardware when it has been upgraded to the latest software release and has hardware scheduling enabled.
- The number of calls an AP can handle is based upon a variety of factors. Downlink configuration will greatly affect the number of total simultaneous calls. At a 50% downlink configuration, the AP can handle 26-28 calls independent of codec. At a 75% downlink configuration, the constraint becomes the uplink bandwidth and codec choice becomes an issue.
- G.711U provided the highest quality calls, but allowed for the fewest total simultaneous calls. G.729 provided the lowest call quality, but allowed for the greatest number total simultaneous calls.
- The percentage of bandwidth configured for downlink traffic needs to be closely monitored. In an unsaturated data network with very few voice customers, 75% downlink would be sufficient to handle both data and voice traffic from customers. As the number of voice customers grows, the percentage should be shifted towards 50%. This will allow for an increased number of simultaneous voice calls with excellent quality.
- QoS using the high priority channel was shown to increase the MOS score average on a test scenario. QoS also decreased jitter, delay, and percentage of lost bytes over the duration of the call. The more data being passed in the background, the better effect QoS has on the quality of the call.
- Canopy Lite performed as expected in respects of VoIP and call quality. The platform is well suited for residential applications. It can easily handle 1-2 simultaneous calls while providing ample bandwidth for other applications.

Verifying lab results against a field deployment - Network Telephone

To validate whether or not our lab results were reasonable and on par with what was happening in the field, we reached out to several of Motorola's Canopy customers to find out what was their experience in the field.

Network Telephone, a carrier-grade telecommunications provider, is one of the customers that we spoke with during our research. Network Telephone saw the Canopy platform as a possible way to expand their voice services. They had looked at other wireless delivery methods, but were impressed by the Canopy platform's ability to handle interference. Network Telephone was targeting the small to mid-sized business market. A bulk of their target clients could have a need for between four and eight simultaneous calls. Network Telephone already has established VoIP and VoATM wired networks. They used data from these networks to build out on what they would be expecting to see from the new customers. Their data showed an oversubscription that varied from 6-1 to 10-1.

Network Telephone used a cautious approach when determining call volume through the Canopy Advantage platform. Their team went out and researched the Canopy Advantage platform using

Motorola documentation as well as earlier white papers regarding the Canopy platform. They also spoke with Canopy Advantage providers about other VoIP over Advantage deployments. In addition to looking at information regarding VoIP over Canopy Advantage, they also looked to VoIP over Wireless LAN information. Many of the concepts for planning and deployment hold true and are independent of the Layer 2 delivery method used. This research provided Network Telephone a baseline idea of what the system “should” handle. They felt that this theoretical baseline was good, but proving the data in a lab and more importantly the field was a critical success factor.

As with any carrier-grade provider, Network telephone ran extensive tests to determine the capacity a Canopy network would be able to provide. Network Telephone’s testing consisted of two phases. The first phase was a closed indoor environment and the results were encouraging. They decided to move to the second phase of testing which involved field deployment and testing in a real world scenario. The second phase involved fifty SMs and two clusters of six APs mounted on radio towers. The towers were approximately 2.4 miles apart with their office being off center between the two. The office is closer to the south tower, being approximately 2.2 miles. The setup can be seen in Figure 2.

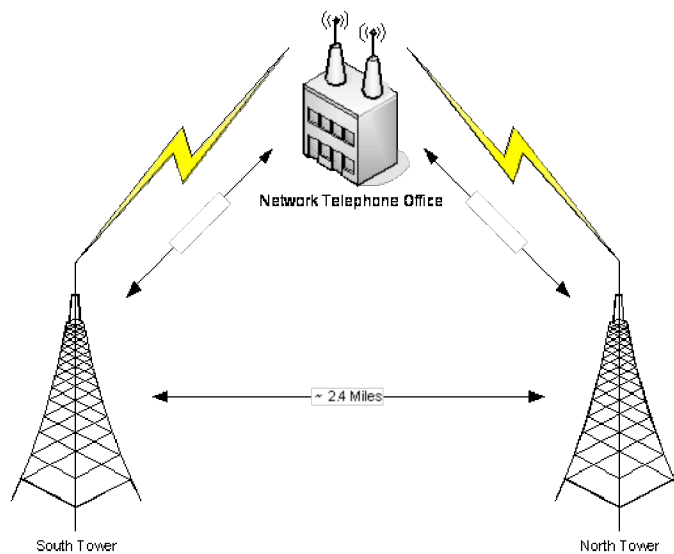


Figure 2: Phase Two Setup

Network Telephone then proceeded to setup an array of 50 SMs on the roof of their office. The SMs were all connected to a variety of CPEs (customer premise equipment) in their testing lab. Traffic flow from voice and data would be generated from the CPEs up to the roof array, out to the tower, and then back to the office via a back haul. The call would then reach the desired CPE. This traffic flow can be seen in Figure 3.

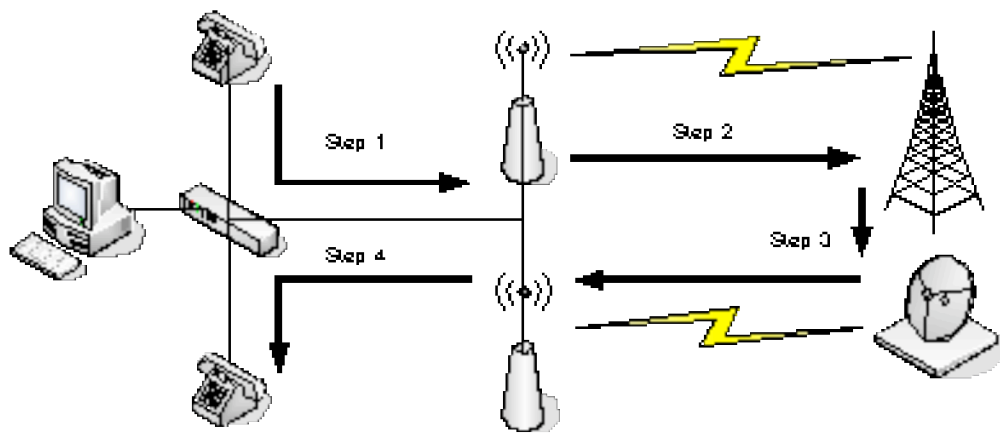


Figure 3: Traffic Flow

The traffic flow follows this general process:

1. A call is placed from a lab phone to another lab phone. The call is routed through the ATA and out to the roof array SM.
2. The SM transmits the data to the radio tower AP cluster.
3. The AP radio cluster transfers the data to the backhaul that in turn directs the call back to the lab.
4. The receiving phone's SM receives the data and passes it onto the ATA and eventually the phone.

In their field testing, Network Telephone found that a single AP could handle approximately 22 – 26 calls. They also stated that they feel comfortable running as many as 12 lines to a single SM. In their business model, Network telephone plans to use an oversubscription ratio of 6 to 1. This ratio was generated using research data and the data gathered from field testing. Using that ratio, they have made plans to handle over 100 VoIP customers per AP. Network Telephone's field testing factored in different real world factors such as propagation patterns and loss over distance, Fresnel zones, and other environmental factors. Despite including these real world factors, one can see the parallels of the lab testing conducted for this whitepaper and Network Telephone's field testing. The findings were consistent with the lab findings within a reasonable margin.

"We use Canopy exclusively and the performance has made us #1 in our area with VoIP being an offer we use to stand out from the competition who pushes Vonage over their WiFi. Canopy's reliability and stability have spoiled our customer base..."
- Canopy Customer



Network Planning Guide

Considerations for Network Capacity Planning

Planning is one of the biggest issues that a provider will face when looking to implement a VoIP offering on his network. Network planning will allow for a smooth transition into a VoIP offering, as well as an increased level of call quality. Network planning is not an exact science, there are trade-offs that must be made. Operating a VoIP network at perfect quality will decrease the amount of total calls available. Toll-quality calls can be made with a high number of simultaneous lines operating, but the system limitations as well as the appropriate over-subscription rate must be accounted for.

Oversubscription is a basic concept that has been studied for over 100 years with regards to the Public Switched Telephone Network (PSTN). Network usage on the PSTN has been monitored since its inception. Essentially, oversubscription is based on the fact that not every subscriber is using their phone line at all times. The public telephone network has been incorporating “statistical over-subscription” from the beginning. In the United States, most telecommunications providers plan for four to eight phones per service line². The more densely populated an area, the lower the oversubscription rate. Oversubscription ratios will vary according to the usage patterns of your customers. It is crucial to gather data on utilization rates and line usage as your network grows. With this data, a provider can take advantage of the individual usage patterns shown on his network. Network Telephone plans to initially release with a very conservative ratio of 6 to 1. As their network grows and more data is gathered, they plan on increasing their ratio. In their current networks, they have areas with a 10-1 oversubscription ratio. However, each WISPs over subscription rate can vary greatly depending on business plan (residential vs. business), geography, SLA, and current network.

Codec choice is another important factor when completing network planning. The three codecs that were tested provided a range in quality and call volume. The choice of codec will depend on the needs of the provider. If call quality is the number one issue, the G.711 codec provides the highest level of call quality. Customers interviewed found G.711U to be most compatible with different ATAs and hand-off equipment. The quality provided by G.711U comes at a trade off of bandwidth and call volume. G.711U operates using more bandwidth than G.729. G.729 provides fair call quality, but is able to handle a high number of simultaneous calls.

Bandwidth configuration plays a very important role in the amount of calls the AP can handle. Data traffic for a typical user is asymmetrical. This group of traffic includes web surfing, email, and instant messaging. Voice traffic is symmetrical by nature. A given conversation will provide almost equal levels from both sides. This presents a problem in terms of bandwidth configuration. As voice traffic increases on the network, the configuration should be adjusted. Customer interviews showed that 75% downlink was a standard configuration for a data traffic network that also supports VoIP calls. It was also indicated that this percentage is being shifted slowly as the need for VoIP services increases. The best performance from a VoIP perspective would be a 50% downlink configuration; however, this configuration would not be the most efficient for data. There is no perfect formula for deciding how to configure bandwidth for every network. Each network has different needs for their bandwidth, and the needs and requirements need to be analyzed and used to form guidelines for the network. The configuration flexibility inherent within The Canopy Advantage platform provides the administrator the ability to adjust their network settings based on the needs of their customers and the network saturation levels.

Equipment and cabling is another factor in dealing with call quality. There are a variety of different CPE devices that are available for Voice over IP. The quality of these devices can make a dramatic difference in overall call quality seen by the end-user. It is important to test out a variety of equipment to determine what works best in your environment. When using Canopy Advantage products, air-delay is a variable that must be accounted for. It can vary over the course of the day, and these changes must be handled by the CPE equipment. CPE becomes a focal point in the handling of jitter and delay. Depending on the anticipated capacity of the network, and the possible need to support Service Level Agreements, different level of CPE quality can be tolerated. Cabling issues are also important in call quality. Ethernet has a limitation of 328 ft. from transmitter to receiver. This becomes an important number when starting to run cable for homes, antenna towers, and roof arrays. Exceeding this length will introduce errors in the system unless Long Reach Ethernet technologies are deployed.

General considerations and recommendations can be made for a Canopy Advantage platform. With an oversubscription ratio of four to one and using the field tested number of approximately 25 calls per AP, the Canopy Advantage platform can handle 100 voice customers per AP. This number can be expected for 50% downlink configuration independent of the codec used. Using a different configuration, the number of simultaneous calls seen on any network will vary depending on many factors including codec, percent of downlink configured, QoS settings, and environmental factors in the Canopy network.

When a Canopy WISP is building a business plan they must take into account oversubscription rates, number of acceptable blocked calls, and quality of service to customers. All of these factors are important and must be considered before providing or advertising VoIP service. The WISP will have to make a choice on whether or not they will provide VoIP service through call manager equipment (i.e. Cisco, Nortel, etc.) or if customers will acquire service on their own (i.e. Vonage, AT&T CallVantage, etc.) and leverage the Canopy platform for broadband access where the third-party CPE device does the VoIP encapsulation.

Another important consideration in a VoIP network deployment is the ability to provide E-911 service. In May of 2005, the Federal Communications Commission (FCC) ordered that providers of VoIP service must certify that their customers were able to reach an emergency dispatcher when they called 911. Additionally, the emergency dispatcher must be able to identify the location and name of the caller. VoIP providers were given until late November of 2005 to comply with this order. At the time of writing, the FCC was relaxing penalties and enforcement on this issue, though it is an important issue to consider when implementing VoIP services.

As in any other bandwidth based network there is going to come a point when the network is at capacity. When this happens, there are a few options on how to handle it depending on the choice of service. In the case of VoIP service being provided through a call manager, the WISP has a couple of options. The WISP can either continue to allow calls to be added to the network and allocate less and less bandwidth per call or block the last call that pushes the network over its capacity. If the number of calls on the network is allowed to increase without limit, this will degrade the call quality of all calls on the network because less bandwidth will be available per call. This is not the recommended approach because this will cause an overall low quality of service resulting in jittering phone connections to many customers. However, if the last call is just blocked and given a "network is busy" signal, then this user can just try redialing in a few seconds when some capacity may have become available and all other calls will not be affected. Managing the bandwidth in this fashion will allow a higher level of service to be provided and managed across the network.

A WISP has very little control over their customers choosing a Vonage type of service. Generally, service providers will offer no service level agreements for third party software (such as Vonage or Skype). If a provider did wish to create a higher level of service for a Vonage type service, it is possible to configure some ATA's to use DiffServ code points. With the code points configured, the traffic can be placed on the high priority queue. In addition to the Canopy Advantage platform's traffic queuing, many providers are using a third party system to do bandwidth shaping and traffic monitoring. It should be noted that the Canopy Advantage platform or any other BWA system should not be handling admission control to the network. Throughout the interviews conducted during the drafting of this paper, customers consistently used a third party tool for admission control. The use of a third party tool will allow for the granularity that is needed to optimize network traffic.

To assist WISPs in this type of capacity planning it is recommended that they take into consideration Erlang tables which are discussed in the next section.

Erlang Tables

An *Erlang* is a unit of telecommunications traffic measurement. Strictly speaking, an Erlang represents the continuous use of one voice path. In practice, it is used to describe the total traffic volume of one hour. For example, if a group of users made 30 calls in one hour, and each call had an average call duration of 5 minutes, then the number of Erlangs this represents is worked out as follows:

$$\begin{aligned} \text{Minutes of traffic in the hour} &= \text{number of calls} \times \text{duration} = 30 \times 5 = 150 \\ \text{Hours of traffic in the hour} &= 150 / 60 \\ \text{Hours of traffic in the hour} &= 2.5 \\ \text{Traffic figure} &= \mathbf{2.5 \text{ Erlangs}} \end{aligned}$$

Erlang traffic measurements are made in order to help telecommunications network designers understand traffic patterns within their voice networks. This is essential if they are to successfully design their network topology. Erlang traffic measurements or estimates can be used to work out how many lines are required between a telephone system and a central office, or in the case of a Canopy network, given a level of available bandwidth, determine the acceptable amount of blocked calls between a SM and an AP. Blocked calls in this scenario do not represent that the far end of the call is busy, but that the medium to make the call is unavailable (a fast busy tone). Several traffic models exist which share their name with the Erlang unit of traffic. They are formulas that can be used to estimate the number of lines required in a network.

The main Erlang traffic models are listed below:

- **Erlang B**
This is the most commonly used traffic model and is used to work out how many lines are required if the traffic figure (in Erlangs) during the busiest hour is known. The model assumes that all blocked calls are immediately cleared.
- **Extended Erlang B**
This model is similar to Erlang B, but takes into account that a percentage of calls are immediately represented to the system if they encounter blocking (a busy signal).

The retry percentage can be specified.

- **Erlang C**
This model assumes that all blocked calls stay in the system until they can be handled. This model can be applied to the design of call center staffing arrangements where, if calls cannot be immediately answered, they enter a queue.

Erlang to VoIP Bandwidth Calculation

As explained above, the concepts of Erlang tables can be applied in a number of different ways to a voice telecommunications network. In the context of a Canopy network, the users will be applying these concepts to VoIP applications. There are a number of calculators that are available to assist Canopy users in developing an appropriate business plan for their network. Our testing showed that in lab conditions, the Canopy Advantage platform was able to handle between 26-28 calls. Table 8 shows the relationship between tested load, calls lost and Erlangs

Number of Simultaneous Calls	1 call lost in 100 load in <i>E</i>	2 calls lost in 100 load in <i>E</i>
23	14.5	13.4
24	15.3	14.2
25	16.1	15.0
26	16.9	15.8
27	17.2	16.6
28	18.6	17.4
29	19.5	18.2
30	20.4	19.0

Table 8: Load in Erlangs



Appendix

Network Telephone (<http://www.networktelephone.net/>)

Network Telephone is a carrier-grade provider of local and long distance telephone communications as well as high-speed Internet access to businesses in over 32 markets in 8 states. They target small to mid-sized companies that have a need for both voice and data communications.

Network telephone was founded in 1997 and has been growing steadily ever since. They are currently headquartered in Pensacola, Florida and have over 20 regional sales offices with an additional satellite headquarters in Atlanta, Georgia. They are staffed with over 500 employees, including a 24 x 7 x 365 Network Operations Center. This enables networking monitoring of Network Telephones ever growing data and voice network.

Despite the companies hold on its core business, Network Telephone was looking for a way to expand its carrier-grade voice services offerings. They began to test Motorola's Canopy system as a way to meet this need for expansion.

Canopy Hardware vs. Canopy Advantage Hardware

This section provides detailed results with regards to the Canopy Hardware vs. Canopy Advantage Hardware portion of testing. This phase of testing was completed to bring to light any differences between a Canopy SM that was updated to Canopy Advantage software, and a true Canopy Advantage SM. The details of the lab setup regarding these tests can be referenced on page 10 of this document.

Hardware	Throughput Avg. (Mbps)	Response Time Avg (s).
Advantage	4.2510	1.7526
Canopy	4.2938	1.7568

Table 9: Canopy vs. Canopy Advantage Throughput and Response Time

# of SM's of each type	# of Calls from Each SM	Total # of Calls	MOS Avg.		One-Way Delay (ms)		Jitter (ms)	
			Canopy	Advantage	Canopy	Advantage	Canopy	Advantage
1	1	2	4.37	4.37	6	5	0.131	0.041
1	2	4	4.37	4.37	4	4	0.301	0.02
2	1	4	4.35	4.37	5	7	0.928	0.51
2	2	8	4.37	4.37	7	7	0.548	0.666
3	1	6	4.37	4.37	7	6	0.51	0.529
3	2	12	4.24	4.27	9	9	2.315	2.391
4	1	8	4.36	4.35	5	5	0.432	0.401
4	2	16	3.89	3.76	43	45	3.925	4.441

Table 10: Canopy vs. Canopy Advantage with G.711u Codec

🔍 Detailed Data for Call Volume Tests

NOTE: The N/A value under “Number of calls from each SM” indicates that an inconsistent distribution of calls was tested across multiple SMs. E.g. five SMs were included in test and three SMs had two calls and two SMs had four calls.

G.711u with 75% Downlink

The following table displays the results for the G.711U codec with 75% downlink configuration.

Codec	Number of SMs	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.711u	9	2	18	3.49	64	5.00	1.688
G.711u	9	n/a	14	3.78	20	4.39	0.227
G.711u	9	n/a	13	4.12	11	2.64	0.010
G.711u	9	n/a	12	4.17	10	2.48	0.030
G.711u	9	1	9	4.35	6	0.52	0.000
G.711u	8	2	16	3.91	45	3.84	0.508
G.711u	8	n/a	14	3.94	16	3.72	0.023
G.711u	8	n/a	13	4.11	13	2.85	0.078
G.711u	8	n/a	12	4.22	9	2.13	0.083
G.711u	8	1	8	4.36	7	0.48	0.000
G.711u	7	3	21	3.10	79	5.13	4.540
G.711u	7	2	14	3.97	21	4.07	0.148
G.711u	7	n/a	13	4.09	14	3.41	0.126
G.711u	7	n/a	12	4.27	13	2.32	0.013
G.711u	7	1	7	4.36	5	0.67	0.001
G.711u	6	3	18	3.81	41	4.40	0.985
G.711u	6	n/a	16	3.93	35	3.96	0.567
G.711u	6	n/a	15	4.04	32	3.56	0.226
G.711u	6	2	12	4.29	8	1.85	0.036
G.711u	6	1	6	4.36	5	0.58	0.000
G.711u	5	5	25	3.30	77	4.27	4.083
G.711u	5	4	20	3.21	60	3.99	3.690
G.711u	5	n/a	17	3.77	41	3.51	1.235

Motorola
Voice over IP over Canopy Advantage

Codec	Number of SMs	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.711u	5	n/a	16	3.99	30	3.29	0.396
G.711u	5	3	15	4.14	20	3.25	0.131
G.711u	5	2	10	4.37	6	0.67	0.003
G.711u	5	1	5	4.37	5	0.31	0.000
G.711u	4	5	20	3.41	46	3.30	4.274
G.711u	4	n/a	18	3.79	37	2.84	2.284
G.711u	4	n/a	17	4.07	32	2.42	1.125
G.711u	4	4	16	4.30	21	2.40	0.025
G.711u	4	2	8	4.34	5	0.68	0.016
G.711u	4	1	4	4.37	5	0.64	0.002
G.711u	3	6	18	3.74	36	2.68	1.638
G.711u	3	n/a	17	4.01	23	2.29	0.833
G.711u	3	5	15	4.36	14	1.86	0.000
G.711u	3	2	6	4.35	6	0.28	0.001
G.711u	3	1	3	4.35	5	0.52	0.000
G.711u	2	10	20	3.41	31	2.53	4.825
G.711u	2	n/a	19	3.91	20	1.96	2.535
G.711u	2	9	18	4.19	12	1.75	0.328
G.711u	2	8	16	4.34	11	2.20	0.023
G.711u	2	2	4	4.37	5	0.47	0.000
G.711u	2	1	2	4.37	6	0.29	0.000

Table 11: G.711u at 75% Downlink

G.711u with 50% Downlink

The following table displays the results for the G.711codec with 50% downlink configuration.

Codec	Number of SMs	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.711u	9	n/a	26	3.64	83	3.625	0.774
G.711u	9	n/a	25	4.27	70	2.976	0.027
G.711u	9	2	18	4.36	9	1.946	0.000
G.711u	9	1	9	4.37	6	0.407	0.000
G.711u	8	n/a	26	3.86	80	3.460	0.049
G.711u	8	n/a	25	4.23	81	3.123	0.069
G.711u	8	3	24	4.32	52	3.025	0.000
G.711u	8	2	16	4.36	8	1.044	0.000
G.711u	8	1	8	4.37	5	0.175	0.000
G.711u	7	n/a	27	3.10	83	4.154	2.076
G.711u	7	n/a	26	4.22	79	3.421	0.101
G.711u	7	3	21	4.36	19	2.108	0.004
G.711u	7	2	14	4.36	7	1.039	0.000
G.711u	7	1	7	4.37	5	0.513	0.000
G.711u	6	n/a	27	3.37	80	3.699	1.398
G.711u	6	n/a	26	4.04	79	3.388	0.326
G.711u	6	4	24	4.30	58	2.894	0.044
G.711u	6	3	18	4.37	7	1.295	0.000
G.711u	6	2	12	4.37	5	0.603	0.001
G.711u	6	1	6	4.37	4	0.195	0.000
G.711u	5	n/a	28	2.68	81	4.241	4.098
G.711u	5	n/a	27	3.34	80	3.496	1.546
G.711u	5	n/a	26	4.22	78	3.246	0.103
G.711u	5	5	25	4.31	77	2.626	0.009
G.711u	5	4	20	4.36	12	1.949	0.003
G.711u	5	3	15	4.37	6	0.962	0.000
G.711u	5	2	10	4.36	4	0.519	0.024
G.711u	5	1	5	4.37	4	0.217	0.000
G.711u	4	7	28	3.12	80	3.599	2.321

Motorola
Voice over IP over Canopy Advantage

Codec	Number of SMs	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.711u	4	n/a	27	3.47	81	3.264	1.301
G.711u	4	n/a	26	4.19	78	2.899	0.159
G.711u	4	6	24	4.33	59	3.008	0.006
G.711u	4	5	20	4.37	11	1.748	0.001
G.711u	4	4	16	4.36	6	1.047	0.000
G.711u	4	2	8	4.37	5	0.336	0.003
G.711u	4	1	4	4.37	5	0.318	0.000
G.711u	3	9	27	3.92	78	2.737	0.472
G.711u	3	n/a	26	4.22	80	2.488	0.120
G.711u	3	8	24	4.34	48	2.370	0.005
G.711u	3	6	18	4.37	7	1.644	0.003
G.711u	3	5	15	4.36	6	1.278	0.002
G.711u	3	2	6	4.37	5	0.172	0.000
G.711u	3	1	3	4.37	4	0.280	0.000
G.711u	2	16	32	2.65	85	6.352	11.874
G.711u	2	14	28	3.09	79	3.061	2.201
G.711u	2	n/a	27	4.00	77	2.544	0.376
G.711u	2	13	26	4.08	70	2.125	0.290
G.711u	2	12	24	4.35	52	2.244	0.000
G.711u	2	10	20	4.37	8	1.248	0.000
G.711u	2	n/a	19	4.37	7	1.385	0.000
G.711u	2	9	18	4.37	7	0.922	0.003
G.711u	2	8	16	4.37	6	0.844	0.000
G.711u	2	2	4	4.37	4	0.036	0.000
G.711u	2	1	2	4.37	4	0.041	0.000

Table 12: G.711u with 50% downlink configuration

G.726 with 75% Downlink

The following table displays the results for the G.726 codec with 75% downlink configuration.

Codec	Number of SMs	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.726	9	2	18	3.73	15	3.92	0.156
G.726	9	n/a	16	3.93	12	2.87	0.057
G.726	9	n/a	15	4.02	12	2.40	0.019
G.726	9	n/a	14	4.08	10	1.24	0.013
G.726	9	1	9	4.17	5	0.33	0.000
G.726	8	2	16	3.93	13	3.13	0.020
G.726	8	n/a	15	4.06	10	1.92	0.015
G.726	8	1	8	4.17	8	0.29	0.000
G.726	7	n/a	18	3.83	18	3.87	0.179
G.726	7	n/a	17	3.97	12	2.93	0.046
G.726	7	n/a	16	4.04	10	2.58	0.047
G.726	7	n/a	15	4.12	9	1.50	0.001
G.726	7	2	14	4.15	10	1.49	0.000
G.726	7	1	7	4.17	8	0.28	0.000
G.726	6	n/a	19	3.80	24	3.77	0.358
G.726	6	3	18	3.94	15	3.65	0.187
G.726	6	n/a	17	4.00	11	2.93	0.092
G.726	6	n/a	16	4.06	9	2.58	0.035
G.726	6	n/a	15	4.12	10	2.08	0.027
G.726	6	2	12	4.15	9	0.81	0.000
G.726	6	1	6	4.17	4	0.33	0.000
G.726	5	4	20	3.71	115	3.45	0.205
G.726	5	n/a	19	3.97	20	3.13	0.198
G.726	5	n/a	18	4.04	16	2.85	0.061
G.726	5	n/a	17	4.11	12	2.47	0.028
G.726	5	n/a	16	4.10	11	2.65	0.037
G.726	5	3	15	4.12	15	2.08	0.007
G.726	5	2	10	4.16	4	0.71	0.014
G.726	5	1	5	4.17	3	0.31	0.000

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Codec	Number of SMs	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.726	4	6	24	3.27	69	3.44	2.959
G.726	4	n/a	22	3.68	30	2.96	1.153
G.726	4	n/a	21	3.78	22	3.00	1.061
G.726	4	5	20	4.09	21	2.67	0.059
G.726	4	4	16	4.14	9	2.09	0.025
G.726	4	2	8	4.17	4	0.32	0.003
G.726	4	1	4	4.17	4	0.44	0.000
G.726	3	7	21	3.95	26	2.58	0.322
G.726	3	n/a	20	4.08	12	2.31	0.131
G.726	3	6	18	4.16	17	2.12	0.000
G.726	3	2	6	4.15	3	0.22	0.050
G.726	3	1	3	4.17	3	0.05	0.000
G.726	2	11	22	3.92	23	2.31	0.408
G.726	2	n/a	21	4.04	19	2.10	0.214
G.726	2	10	20	4.18	10	1.79	0.009
G.726	2	2	4	4.17	4	0.25	0.002
G.726	2	1	2	4.17	4	0.05	0.000

Table 13: G.726 with 75% Downlink Configuration

G.726 with 50% Downlink

The following table displays the results for the G.726 codec with 75% downlink configuration.

Codec	Number of SMs	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.726	9	3	27	3.21	79	2.85	1.558
G.726	9	n/a	26	4.00	78	2.49	0.135
G.726	9	n/a	25	4.09	72	2.25	0.021
G.726	9	2	18	4.16	9	1.33	0.001
G.726	9	1	9	4.17	5	0.14	0.000
G.726	8	n/a	27	3.31	78	2.59	1.237
G.726	8	n/a	26	4.09	78	2.38	0.033
G.726	8	n/a	25	4.11	71	2.49	0.012
G.726	8	3	24	4.11	56	2.09	0.016
G.726	8	2	16	4.16	9	1.25	0.000
G.726	8	1	8	4.17	4	0.17	0.000
G.726	7	n/a	27	3.52	77	2.72	0.835
G.726	7	n/a	26	4.01	76	2.32	0.135
G.726	7	n/a	25	4.11	74	2.47	0.006
G.726	7	3	21	4.15	16	1.54	0.001
G.726	7	2	14	4.16	8	0.67	0.000
G.726	7	1	7	4.17	5	0.19	0.000
G.726	6	n/a	27	3.42	78	2.64	1.013
G.726	6	n/a	26	4.09	78	2.42	0.030
G.726	6	n/a	25	4.10	78	2.06	0.020
G.726	6	4	24	4.14	45	2.12	0.008
G.726	6	3	18	4.17	8	1.74	0.001
G.726	6	2	12	4.17	6	0.57	0.001
G.726	6	1	6	4.17	5	0.15	0.000
G.726	5	n/a	27	3.40	78	2.84	1.076
G.726	5	n/a	26	4.11	75	2.12	0.018
G.726	5	5	25	4.12	73	2.04	0.000
G.726	5	4	20	4.17	10	1.50	0.003
G.726	5	3	15	4.17	6	0.77	0.000

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Codec	Number of SMs	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.726	5	2	10	4.17	5	0.27	0.001
G.726	5	1	5	4.17	4	0.23	0.000
G.726	4	n/a	27	3.83	76	2.72	0.373
G.726	4	n/a	26	4.12	76	2.26	0.000
G.726	4	6	24	4.13	50	2.18	0.005
G.726	4	5	20	4.16	9	1.64	0.011
G.726	4	4	16	4.16	7	1.19	0.000
G.726	4	2	8	4.17	5	0.23	0.000
G.726	4	1	4	4.17	4	0.15	0.000
G.726	3	9	27	3.83	76	2.40	0.360
G.726	3	n/a	26	4.09	76	1.99	0.035
G.726	3	7	21	4.16	13	1.91	0.005
G.726	3	6	18	4.17	7	1.64	0.000
G.726	3	2	6	4.17	5	0.25	0.002
G.726	3	1	3	4.17	4	0.15	0.000
G.726	2	14	28	3.52	75	2.35	0.852
G.726	2	n/a	27	3.79	75	2.31	0.423
G.726	2	13	26	4.01	76	2.15	0.133
G.726	2	10	20	4.17	7	1.47	0.000
G.726	2	2	4	4.17	4	0.07	0.000
G.726	2	1	2	4.17	3	0.15	0.000

Table 14:G.26 with 50% Downlink Configuration

G.729 with 75% Downlink

The following table displays the results for the G.729 codec with 75% downlink configuration.

Codec	Number of SMs	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.729	9	3	27	3.51	92	6.09	1.141
G.729	9	2	18	3.98	10	2.56	0.020
G.729	9	n/a	17	3.99	13	2.38	0.009
G.729	9	n/a	16	4.00	12	2.28	0.017
G.729	9	n/a	15	4.01	12	2.01	0.000
G.729	9	1	9	4.03	5	0.87	0.000
G.729	8	3	24	3.86	56	4.69	0.167
G.729	8	n/a	21	3.94	19	3.41	0.076
G.729	8	n/a	20	3.96	15	3.16	0.039
G.729	8	n/a	19	3.99	13	2.59	0.003
G.729	8	n/a	18	3.99	13	2.59	0.010
G.729	8	n/a	17	4.00	11	2.34	0.024
G.729	8	2	16	4.00	9	2.47	0.007
G.729	7	3	21	3.95	21	3.74	0.171
G.729	7	n/a	19	3.98	12	2.65	0.046
G.729	7	n/a	18	4.01	11	2.06	0.008
G.729	7	n/a	17	4.00	10	2.28	0.028
G.729	7	2	14	4.02	6	1.15	0.002
G.729	6	4	24	3.92	58	3.77	0.128
G.729	6	n/a	23	3.92	47	3.99	0.150
G.729	6	n/a	22	3.97	21	3.07	0.129
G.729	6	n/a	22	3.97	21	3.07	0.129
G.729	6	n/a	21	3.98	17	3.24	0.054
G.729	6	n/a	19	4.00	11	2.19	0.014
G.729	6	3	18	3.99	11	2.41	0.057
G.729	6	n/a	17	3.98	11	2.39	0.096
G.729	6	2	12	4.02	8	1.17	0.000
G.729	6	1	6	4.02	8	0.72	0.000
G.729	5	n/a	27	3.84	79	3.22	0.744

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Codec	Number of SMs	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.729	5	n/a	26	3.97	63	2.76	0.020
G.729	5	5	25	3.99	53	2.18	0.000
G.729	5	n/a	23	3.98	53	2.62	0.022
G.729	5	4	20	4.01	14	2.15	0.017
G.729	5	3	15	4.00	9	1.79	0.053
G.729	5	2	10	4.02	8	0.63	0.001
G.729	5	1	5	4.03	8	0.70	0.000
G.729	4	n/a	27	3.87	77	3.49	0.357
G.729	4	n/a	26	3.93	76	3.05	0.100
G.729	4	n/a	25	3.97	58	3.39	0.053
G.729	4	6	24	3.97	54	2.94	0.060
G.729	4	n/a	23	3.98	49	3.06	0.013
G.729	4	n/a	22	3.97	96	2.64	0.159
G.729	4	n/a	21	4.02	10	1.97	0.000
G.729	4	5	20	4.01	11	2.02	0.031
G.729	4	4	16	4.01	8	1.27	0.011
G.729	4	2	8	4.02	9	0.47	0.002
G.729	4	1	4	4.02	8	0.52	0.000
G.729	3	8	24	3.95	58	2.66	0.089
G.729	3	n/a	23	4.00	32	2.40	0.019
G.729	3	n/a	22	4.02	14	1.70	0.000
G.729	3	7	21	4.02	11	2.25	0.000
G.729	3	2	6	4.02	9	0.56	0.000
G.729	3	1	3	4.02	9	0.54	0.000
G.729	2	12	24	3.98	54	2.16	0.000
G.729	2	n/a	23	4.02	17	1.87	0.005
G.729	2	11	22	4.02	14	2.22	0.000
G.729	2	2	4	4.02	9	0.55	0.000
G.729	2	1	2	4.02	8	0.68	0.000

Table 15: G.729 with 75% Downlink Configuration

G.729 with 50% Downlink

The following table displays the results for the G.729 codec with 50% downlink configuration.

Codec	Number of SMs	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.729	9	n/a	28	3.72	76	3.11	1.388
G.729	9	3	27	3.86	76	2.73	0.517
G.729	9	n/a	26	3.96	75	2.86	0.003
G.729	9	n/a	25	3.98	59	2.65	0.003
G.729	9	2	18	4.02	9	2.02	0.000
G.729	9	1	9	4.03	4	1.23	0.001
G.729	8	n/a	28	3.77	75	3.21	1.067
G.729	8	n/a	27	3.95	74	2.90	0.080
G.729	8	n/a	26	3.97	70	2.44	0.004
G.729	8	n/a	25	4.00	46	2.24	0.000
G.729	8	3	24	3.98	52	2.64	0.006
G.729	8	2	16	4.02	8	1.85	0.003
G.729	8	1	8	4.03	4	0.79	0.000
G.729	7	4	28	3.70	82	2.65	1.363
G.729	7	n/a	27	3.90	81	2.75	0.260
G.729	7	n/a	26	3.94	78	2.05	0.036
G.729	7	n/a	25	3.98	58	2.33	0.000
G.729	7	3	21	4.01	18	1.52	0.001
G.729	7	2	14	4.02	13	1.49	0.000
G.729	7	1	7	4.02	10	0.84	0.000
G.729	6	n/a	28	3.76	77	2.47	1.174
G.729	6	n/a	27	3.95	76	2.32	0.078
G.729	6	n/a	26	3.97	74	2.34	0.006
G.729	6	4	24	3.97	48	2.34	0.035
G.729	6	3	18	4.02	9	1.52	0.000
G.729	6	2	12	4.02	7	1.14	0.000
G.729	6	1	6	4.03	6	0.89	0.000
G.729	5	n/a	28	3.80	75	2.48	0.977
G.729	5	n/a	27	3.96	74	1.74	0.042

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Codec	Number of SMs	Number of calls from each SM	Total # of Calls	MOS Avg.	One-Way Delay Avg. (ms)	Jitter (ms)	Percent bytes lost
G.729	5	n/a	26	3.98	66	1.77	0.000
G.729	5	5	25	3.97	58	2.16	0.029
G.729	5	4	20	4.02	10	1.23	0.000
G.729	5	3	15	4.02	7	0.77	0.000
G.729	5	2	10	4.03	6	0.73	0.001
G.729	5	1	5	4.03	5	0.98	0.000
G.729	4	n/a	28	3.74	77	2.56	1.312
G.729	4	n/a	27	3.91	77	2.41	0.319
G.729	4	n/a	26	3.96	75	2.03	0.006
G.729	4	6	24	3.97	54	2.94	0.060
G.729	4	5	20	4.02	10	1.46	0.000
G.729	4	4	16	4.02	7	0.89	0.000
G.729	4	2	8	4.03	6	0.64	0.000
G.729	4	1	4	4.03	5	0.74	0.000
G.729	3	n/a	28	3.88	74	2.30	0.484
G.729	3	9	27	3.97	74	2.24	0.000
G.729	3	n/a	26	3.97	76	1.68	0.000
G.729	3	n/a	25	3.99	47	2.30	0.000
G.729	3	8	24	4.00	41	1.70	0.025
G.729	3	7	21	4.02	11	1.30	0.000
G.729	3	2	6	4.03	5	0.77	0.000
G.729	3	1	3	4.03	5	0.83	0.000
G.729	2	n/a	27	3.91	74	2.33	0.326
G.729	2	13	26	3.97	71	2.12	0.026
G.729	2	n/a	25	3.99	45	2.16	0.000
G.729	2	12	24	4.00	41	2.12	0.000
G.729	2	2	4	4.03	5	0.42	0.000
G.729	2	1	2	4.03	5	0.64	0.000

Table 16: G.729 with 50% Downlink configuration

Voice over IP Basics

Voice over IP (VoIP) permits the movement of voice traffic over Internet Protocol (IP)-based network. IP is a standard for data transmission based on packet-switching technology. Voice is broken into a series of packets at the transmitting end. The components are then reassembled and decoded at the receiving device.

Voice communications is both real time and mission-critical. Any delay can make a call prohibitive and lead to an undesired poor quality of service. Packet loss can be caused by router congestion that may lead to a loss of portions of words or sentences. Traffic can multiply as the number of routers is increased in the network leading to longer delays. Network jitter, where packets don't arrive in sequence, can lead to unavoidable delays and poor quality of service.

Coder/Decoder (Codec)

A voice coder is the device that converts an analog voice signal into a digital signal. The digital signal is also compressed to reduce bandwidth requirements. Using a hybrid coding technique with complex algorithms, the voice waveform is sampled and the speech parameters are extracted. Thus, in any predefined time period, the waveform is assembled by a synthesis technique to closely assemble the original waveform. The best way to reduce latency is to change the voice coding method; however, the trade-off is voice quality vs. bandwidth required. While there is a delay in the voice compression methods used, there is little further delay with decompression regardless of the algorithm used.

Compression Algorithms	
Algorithm	Description and Rates
G.711u	Pulse code modulation (PCM) specifies the initial analog-to-digital conversion of speech. Speech is transmitted at 64 kbps – which is considered to be toll quality. ITU standard for H.323-compliant codecs and most frequently used in the USA.
G.711a	Same as above, however it utilizes the A-law for companding, which is the most frequently used standard in Europe.
G.726	A waveform coder that uses Adaptive Differential Pulse Code Modulation (ADPCM) at 32 kbps. ADPCM is a variation of PCM, which only sends the difference between two adjacent samples, producing a lower bit rate.
G.729	High-performing codec; offers compression with high quality. Algorithm runs at 8.4 kbps with 10-ms delay and a compression ratio of 8-to-1.
G.723.1-MPMLQ	ITU algorithm that offers voice transmission with quality at a rate of 6.3 kbps with 30-ms delay. Uses the multi-pulse maximum likelihood quantization (MPMLQ) impression algorithm.
G.723.1-ACELP	ITU algorithm that offers voice transmission with quality at a rate of 5.3 Kbps with 30-ms delay. Uses the conjugate structure algebraic code excited linear predictive compression (ACELP) algorithm.

Source: Gartner and IXIA

Table 17: Compression Algorithms

📌 Standards for Measuring Call Quality

Call quality measurement has traditionally been subjective: picking up a telephone and listening to the quality of the voice. The leading subjective measurement of voice quality is the MOS (mean opinion score) as described in the ITU (International Telecommunications Union) recommendation.

In voice communications, particularly Internet telephony, the mean opinion score (MOS) provides a numerical measure of the quality of human speech at the destination end of the circuit. The scheme uses subjective tests (opinionated scores) that are mathematically averaged to obtain a quantitative indicator of the system performance.

Compressor/decompressor (codec) systems and digital signal processing (DSP) are commonly used in voice communications because they conserve bandwidth. But they also degrade voice fidelity. The best codecs provide the most bandwidth conservation while producing the least degradation of the signal. Bandwidth can be measured using laboratory instruments, but voice quality requires human interpretation.

To determine MOS, a number of listeners rate the quality of test sentences read aloud over the communications circuit by male and female speakers. A listener gives each sentence a rating as follows: (1) bad; (2) poor; (3) fair; (4) good; (5) excellent. The MOS is the arithmetic mean of all the individual scores, and can range from 1 (worst) to 5 (best).

Mean Opinion Score (lower limit)	User Satisfaction
4.34	Very satisfied
4.03	Satisfied
3.60	Some users dissatisfied
3.10	Many users dissatisfied
2.58	Nearly all users dissatisfied

Table 18: MOS Values

The E-model is a complex formula; the output of an E-model calculation is a single score, called an “R factor,” derived from delays and equipment impairment factors. Once an R factor is obtained, it can be mapped to an estimated MOS. R factor values range from 100 (excellent) down to 0 (poor). An estimated MOS can be directly calculated from the E model’s R factor.

📌 Ixia Chariot Testing Software

To determine values such as MOS and R-factor, it is not feasible to have human listeners to make these subjective judgments at all times. For the purposes of these tests and this paper, IXIA’s Chariot software product was used to determine these values and compile the data necessary. Chariot has the capability to provide a tremendous amount of data in a testing environment. The following is an example of the types of information that can be gathered. For the purposes of this whitepaper, the focus was to use the advanced call quality measurements to determine how VoIP traffic performs on the Canopy network.

Advanced call quality measurements

Predicts call quality by calculating a MOS based on the industry standard E-model specified in the ITU recommendation G.107. Improving on the base standard, the VoIP Test Module takes

into account additional network factors, such as jitter and consecutive lost datagrams, which can severely impact overall call quality

Tests VoIP-enabled network equipment

Examines the effectiveness and performance of VoIP-enabled network equipment. The VoIP Test Module enables the user to verify that prioritization techniques work as planned with a mixture of traffic and measure the performance impact of other network elements, such as VPNs, on delay-sensitive VoIP traffic. Enables the user to test the limits of the network by generating up to 10,000 VoIP sessions. By identifying the point where call quality begins to suffer, the VoIP Test Module empowers the user to make informed decisions about the implementation and expansion of VoIP in the network.

Emulates complex networks in test lab

Allows the user to emulate complex networks with a mixture of both VoIP and non-VoIP traffic by using Chariot and its VoIP Test Module. By using Chariot in the lab environment, the user can stress test network equipment, test network changes before deployment or replicate end-user environments and reported problems. Chariot evaluates the effectiveness of QoS. The user can ensure that voice traffic is receiving necessary resources at the proper time without starving other business-critical applications.

Optimizes network design

Supplies on-demand testing for tuning network to minimize delay, jitter and lost data.

Settings Used in VoIP Testing

The initial delay option was not used to introduce a standard distribution into the voice testing. Silence suppression was also not used. Additional settings can be seen in Table 19 and Table 20 below.

Codec	Data Rate	Default Datagram Size (ms)	Frame Size	Jitter Delay	Look Ahead Delay	Theoretical Max MOS
G.711U	64 kbps	20	1	2 datagrams (40 ms)	0 ms	4.41
G.726	32 kbps	20	10	2 datagrams (40 ms)	.125 ms	4.22
G.729	8 kbps	20	10	2 datagrams (40 ms)	5.0 ms	4.07

Table 19: Ixia Settings

Codec	Packets per Second	Packet Size (octets)	One-Way Throughput (Kbps)
G.711U	60.67	212	86
G.726	50.67	133	54
G.729	50.67	73	30

Table 20: Flow Statistics by Codec

🔗 Lab Specifications

Subscriber Module	
Device Type:	5.7 Multipoint - Subscriber Modem
Software Version:	CANOPY 7.2.9 (Jul 23 2005 01:49:03)
Software Boot Version:	CANOPYBOOT 3.0
FPGA Version:	070605 (DES Sched) P9
Scheduling Type:	Hardware
2x Rate:	Enabled

Access Point	
Device Type:	5.7 Multipoint - Access Point
Software Version:	CANOPY 7.2.9 (July 23 2005 01:49:03) AP-DES
Software Boot Version:	CANOPYBOOT 3.0
FPGA Version:	070605 (Single, 40 MHz ExtBus, Des, Sched)
Scheduling Type:	Hardware
MP Double Rate:	Enabled

IXIA Software	
Chariot Console	Version 6.10
Endpoint Software for WinXP	Version: 6.10

Endpoint and Console Laptops	
IBM T42 Laptop	Windows XP SP 2 Intel Pentium M 1.70 GHz Processor 512 MB RAM Intel PRO/1000 MT Mobile Connection

Glossary of Terms

Acronym	Meaning
AP	Access Point
Codec	Compressor/Decompressor
Committed Information Rate (CIR)	CIR is the bandwidth rate at which your service provider guarantees delivery. Data transmitted above this rate is a best effort delivery.
CPE	Customer Premise Equipment
Delay	The time, generally in ms, it takes to transmit a message from one endpoint to another
Differentiated Services (DiffServ)	Differentiated Services is scalable method of providing Quality of Service through many networks. DiffServ uses code points to define service levels or classes. DiffServ is defined in IETF RFC 2474 and 2475/.
DSP	Digital Signal Processing
ITU	International Telecommunications Union
Jitter	Jitter is the variation on time between packets at a destination arriving generally caused by network congestion or route changes.
kpbs	1000 bits per second
Kbps	1024 bits per second
Loss	Loss describes the amount of packets that are sent by an endpoint, that do not arrive at the destination.
Mbps	1024 Kbps = 1,048,576 bits
MOS	Mean Opinion Score - To determine MOS, a number of listeners rate the quality of test sentences read aloud over the communications circuit by male and female speakers. A listener gives each sentence a rating as follows: (1) bad; (2) poor; (3) fair; (4) good; (5) excellent. The MOS is the arithmetic mean of all the individual scores, and can range from 1 (worst) to 5 (best)
PSTN	Public Switched Telephone Network
QoS	Quality of Service
R-Value/Factor	The E-model is a complex formula; the output of an E-model calculation is a single score, called an "R factor," derived from delays and equipment impairment factors. R factor values range from 100 (excellent) down to 0 (poor).
SM	Subscriber Module
TOS	Type of Service
VoIP	Voice Over IP
VPN	Virtual Private Network
WISP	Wireless Internet Service Provider



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Disclaimer:

This whitepaper merely provides a starting point for planning and sizing hardware requirements for customers to deploy VoIP over Canopy Advantage platform. Because these tests were run in constrained environments, such as an isolated lab, they do not necessarily translate directly to deployable scenarios. Therefore, it is important to understand that while this whitepaper is meant to help customers prepare for a VoIP over Canopy Advantage network roll out and capacity-planning effort, any data generated contained in this whitepaper is only meant for general sizing, benchmarking, or deployment recommendations. Results may not be representative and may vary. Accordingly, neither Motorola nor West Monroe Partners can guarantee actual results in a real world deployment. In addition to these benchmarking results and recommendations, customers should also consider, but not limit, evaluation to point-to-point mileage, line of sight, network capacity, and expected peak call volume time (Erlang tables) when planning a VoIP over Canopy deployment.

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