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VoIP and Database Traffic Co-existence over IEEE 802.11b WLAN with Redundancy

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***Abstract*—**This paper presents the findings of two experiments that were performed on the Redundancy in Wireless Connection Model (RiWC) using the 802.11b standard. The experiments were simulated using OPNET 11.5 Modeler software. The first was aimed at finding the maximum number of simultaneous Voice over Internet Protocol (VoIP) users the model would support under the G.711 and G.729 codec standards when the packetization interval was 10 milliseconds (ms). The second experiment examined the model's VoIP user capacity using the G.729 codec standard along with background traffic using the same packetization interval as in the first experiment. To determine the capacity of the model under various experiments, we checked three metrics: jitter, delay and data loss. When background traffic was added, we checked the response time in addition to the previous three metrics. The findings of the first experiment indicated that the maximum number of simultaneous VoIP users the model was able to support was 5, which is consistent with recent research findings. When using the G.729 codec, the model was able to support up to 16 VoIP users; similar experiments in current literature have indicated a maximum of 7 users. The finding of the second experiment demonstrated that the maximum number of VoIP users the model was able to support was 12, with the existence of background traffic.

***Keywords*—**WLAN, IEEE 802.11b, Codec, VoIP, OPNET, Background traffic, and QoS.

I. Introduction

WIRELESS local-area networks (WLAN) have had a significant impact on the methods of data communication, as users are now demanding faster and more efficient ways of transferring data from one place to another. WLANs are relatively easy to install, stable, have acceptable speed, and are inexpensive due to high competition among manufacturers. WLANs allow mobile connections for any network that uses cables, help in supplying a backup replacement for any existing network, allow some network devices to move from one place to another, and add the ability to extend the network beyond the limits of the cables. Although WLANs are considered a good data-centric network choice in universities, airports, restaurants, and other enterprise markets, there is also a growing interest in using them for voice. Currently, Voice over Internet Protocol (VoIP) is gaining widespread popularity because it provides a low-cost medium of voice communication. However, data traffic and voice traffic have opposite requirements. Data traffic is asynchronous (delays are acceptable) and extremely sensitive to errors, while voice traffic is synchronous (significant delays are not acceptable) and more tolerant of errors.

Voice traffic is highly sensitive to the network Quality of Service (QoS) factors such as delay, jitter, packet loss, throughput, and Bit-Error-Rate (BER). In this paper, we measured delay, jitter and packet loss for voice traffic and response time for the database traffic.

Several wireless 802.11 technologies are now available. Our main focus in this paper, however, will be on the well-known and most commonly deployed one: the IEEE 802.11b. The theoretical bit rate for the 802.11b is 11 Mbps in the 2.4 GHz band. There are two MAC protocols covered in the IEEE 802.11 standard: the Distribution Coordination Function (DCF) and the Point Coordination Function (PCF). DCF is asynchronous by its very nature; it implements the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) basic access method to share medium. PCF provides synchronous service. It implements a polling access method in which an access point polls the stations in a cyclic manner to allow them to transmit data. The DCF is mandatory while the PCF is optional. Additionally, PCF relies on the asynchronous service provided by the DCF. Our simulations in this paper assume the DCF.

In DCF, a station senses the medium before transmitting a packet. If the medium is sensed to be idle for a time interval more than a Distributed Inter-Frame Space (DIFS), the station transmits the packets. Otherwise, the transmission process is deferred and the backoff process is started. In this case, the station computes a random time interval, the backoff interval, a value in the range zero and a maximum of Contention Window (CW). This interval is then used to initialize a backoff timer. This timer is decremented (the amount is called the slot-time or the maximum round-trip delay) only when the medium is idle; otherwise, decrementing is frozen during the transmission of another station. When the medium becomes idle, the station waits for a DIFS and then periodically decrements the backoff timer.

When the backoff timer expires, the station is allowed to transmit. When two or more stations start transmission at the same time, a collision occurs. In wireless networks, it is not possible to detect collision. To solve this, an acknowledgement is needed to inform the sending station that the transmitted frame is successfully received. The acknowledgement transmission is initiated at a time interval equal to the Short Inter-Frame Space (SIFS), after the end of the reception of the previous frame. By definition, the SIFS is less than the DIFS (DIFS = SIFS + 2 x slot-time). For this, there is no need for the receiving station to sense the medium before transmitting the acknowledgement.

When the acknowledgement is not received, the station assumes that the transmitted frame was not received, and the sender must retransmit. After each collision, the Contention Window is doubled until a predefined maximum (CWmax) is reached in order to reduce the probability of collisions. Following each transmission, and while the station still has frames to send, it enters a new backoff process.

Literature demonstrates a set of related work. In Reference [1] we introduced our RiWC model and showed that the model functions well under different types of traffic, namely: file transfer protocol (ftp), hyper-text transfer protocol (http), and database (db).

The VoIP capacity (number of users) of the 802.11b using different voice encoding schemes was studied in [3] and [6], an analytical model and simulation were used. In Reference [7], an upper bound model was used. These 3 references show that when the packetization interval is 10 ms, the G.711 allows up to 6 users, while the G.729 allows 7 users. The authors of Reference [4], among other tests, used the E-Model and simulation to show that when the G.711 codec is used, and assuming that all users are at a fixed distance from the access point (AP), only 5 VoIP simultaneous calls are possible. In Reference [8], the authors used commercial software and calculations to monitor the traffic at a 10 ms packetization interval and showed that the maximum is 6 simultaneous VoIP calls in a single cell when the G.711 codec is used, while the maximum is 7 simultaneous VoIP calls when the G.729 codec is used. We were not able to find in the available literature a clear upper bound for the capacity of the 802.11b having background traffic together with voice.

The rest of this paper is organized as follows: Section II describes the model and experiments, section III covers the scenarios we tested, section IV analyses the results, and the conclusion is drawn in section V.

II. Model & Experiments' Description

In this section, we introduce our model and, for the purpose of evaluation, we define some evaluation parameter terms and specify their acceptable standard values. Fig. 1 shows the RiWC model. In the wireless LAN part, we assumed 2 sites, each of which has two access points (AP), and each AP is connected to a layer 3 switch as shown. The 2 layer 3 switches are connected to an IP Gateway core switch. In the wired LAN part, we assumed that applications have servers and are connected through a firewall to the IP Cloud. There is no single point of failure in the WLAN; if an access point fails, all users stay connected through the other one. In addition, if a switch fails, all users stay connected through the other one. Observe that in some of our scenarios the server (e.g. DB) was purposely removed. In this case, only VoIP traffic will be available. The model is designed as such to allow maximum reliability.

Acceptable standards and their values were extracted from [2], [5], [9], [10], [11] and [12].

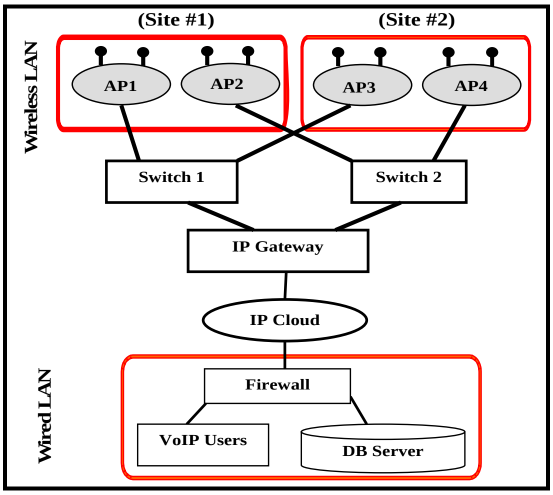

Fig. 1 RiWC Model

A. Delay

Delay is defined as the amount of time that a packet takes to travel from the sender's application to reach the receiver's destination application. It might be caused by the codec scheme and router queuing delays. Acceptable delay value is between 0 ms and 300 ms and unacceptable above this [5]. Reference [10], however, indicates that delay should be less than 200 ms. We will use the value 200 ms as our maximum acceptable delay value.

B. Jitter

Jitter is defined as the variation in delay of the packets arriving at the receiving end. It can be caused by: congestion, insufficient bandwidth, varying packet sizes in the network, and out of order packets. Acceptable jitter value is between 0 ms and 50 ms and unacceptable above this [11]. We will use the value 50 ms as the maximum acceptable jitter value.

C. Packet Loss

Packet Loss is defined as the packets discarded deliberately or non-deliberately by intermediate links, nodes and end-systems along a given transmission path. They are caused by line properties (Layer 1), full buffers (Layer 3) or late arrivals (at the application). For the purpose of this study we will use the equivalent term **Data Loss** which we define as the difference between the packets sent and the packets received. Acceptable data loss value is between 0% and 1.5 % and unacceptable above this [12]. Reference [9], however, accepts the data loss when it is less than 2 %. We will use the value 1.5% as the maximum acceptable data loss value.

D. Response Time

Response Time is defined as the length of time elapsed from a user's request and the system's first response. A value of 0.1 second gives the user an impression that the system is reacting instantaneously. A value of 1 second is the same as the user's flow of thoughts where no special feedback is necessary. A value of 10 seconds, however, will be enough to keep the user's attention provided that a "percent-done indicator" and the ability to interrupt the process are given to user [2]. We will use the value 1 second as our maximum acceptable response time measure to determine the capacity of our model.

III. Simulated Scenarios

Using OPNET Modeler 11.5, we tested the RiWC model under different scenarios. Our first experiment aimed at finding the maximum number of simultaneous Voice over Internet Protocol (VoIP) users the model would support under the G.711 and G.729 codec standards when the packetization interval was 10 ms. The second experiment examined the model's VoIP user capacity using the G.729 codec standard along with background traffic (DB) using the same packetization interval (i.e. 10 ms) as the first experiment. A summary of the experiments performed is shown in Table I. Observe that we have tested the model with different types of applications. We aimed at finding the maximum capacity of simultaneous voice users and the effect of voice application on the background traffic and vice versa.

In OPNET, we simulated each experiment's scenario for a duration of 30 minutes. Each VoIP user was to call continuously for 120 seconds; the inter-repetition time was constant and set to 120 seconds. The first call, however, was started after 100 seconds. The operation mode of all calls was set to simultaneous. When only VoIP calls were tested (in the existence of the codec schemes G.711 and G.729), we assumed the buffer size for both the callers and the APs to be 128000 bits (one of the default values in OPNET). Only when background traffic is added did we select the other default value, 1024000 bits. All VoIP calls were performed in both directions (i.e. from the caller in the WLAN part to the callee in the wired LAN part and vice versa). A summary of the parameters and their values is listed in Table II. Notice that all other simulation values that are not shown indicate that the default OPNET values were left the same.

TABLE I
Experiments' Specification

Experiment and Codec	Description	Tested Parameter
1 - G.711	VoIP traffic only; number of voice users varied from 2 to 7.	Throughput, Jitter, Delay, and Packet loss.
1 - G.729	VoIP traffic only; number of voice users varied from 2 to 16.	Throughput, Jitter, Delay, and Packet loss.
2 - G.729	VoIP traffic and background traffic (Heavy DB); number of voice users varied from 2 to 14; number of DB users is 22 in all scenarios.	Throughput, Jitter, Delay, Packet loss, and Response time.
3 – n/a	Background traffic (Heavy DB) only; number of DB users is 22 in this scenario.	Throughput and Response Time.

TABLE II
Experiments' Parameters

Profile Name	Parameter	Value
VoIP - G.711	Frame Size (seconds)	10 ms
	Lookahead Size (seconds)	0
	Coding Rate (bits/seconds)	64 Kbps
	Speech Activity Detection	Disabled
VoIP - G.729	Frame Size (seconds)	10 ms
	Lookahead Size (seconds)	0
	Coding Rate (bits/seconds)	8 Kbps
	Speech Activity Detection	Enabled
DB	Database Access	High Load
	Transaction Mix (Queries/Total Transactions)	50%

IV. Results Analysis

We start by looking at throughput, and imposed a strenuous set of parameters. Fig. 2 shows the average throughput when the number of VoIP users varied from 2 to 7. The throughput increased as the number of VoIP calls increased. The reason for the wave-like behavior is due to the fact that each call lasted for 2 minutes and the inter-repetition time is also 2 minutes, making each cycle 4 minutes. This behavior is almost the same even when we studied other codecs; the only difference is in the amount of the throughput achieved: in the G.711 it ranged from 100,000 bps (2 users) to about 550,000 bps for 7 users, while it ranged from 28,000 bps (2 users) to 235,000 bps (16 users) in the G.729 codec. When DB traffic is added in the G.729 (see Fig. 3), however, it ranged from 1,000,000 bps for 2 users to 1,250,000 bps when the number of users was 13.

Second, we look at the delay. Fig. 4 and Fig. 5 show how the delay changed as the number of VoIP calls changed. The delay stayed acceptable (less than or equal to 200 ms) as long as the number of users was 6 or less. When another user was added, however, the delay jumped to 360 ms. This helps in saying that the model can accommodate 6 or less VoIP calls, which is consistent with findings in related literature.

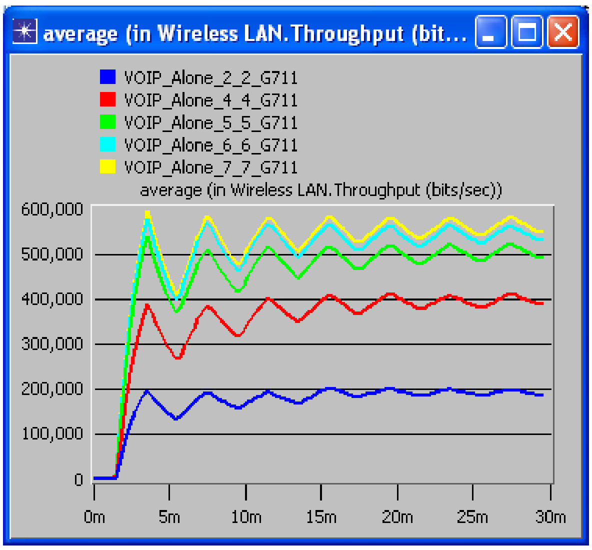

Fig. 2 G.711 Average Throughput

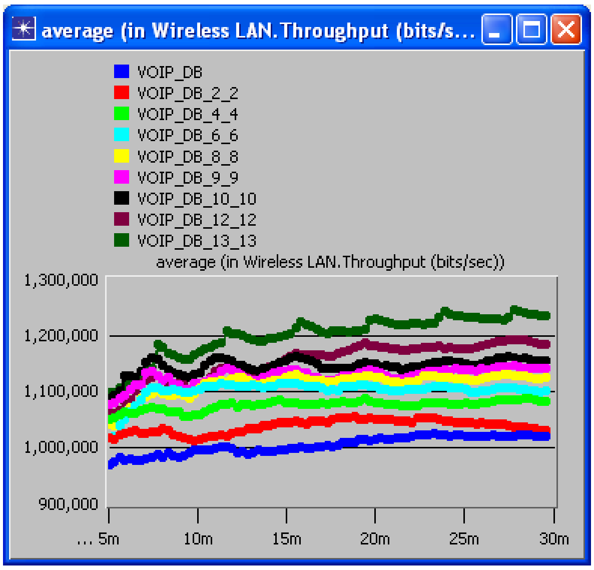

Fig. 3 DB and G.729 Average Throughput

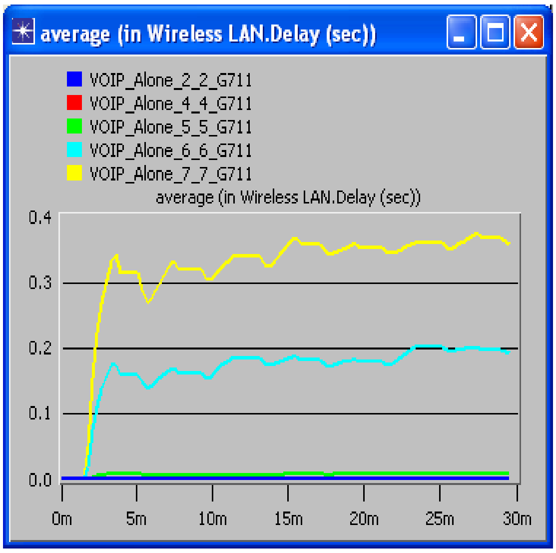

Fig. 4 G.711 Average Delay

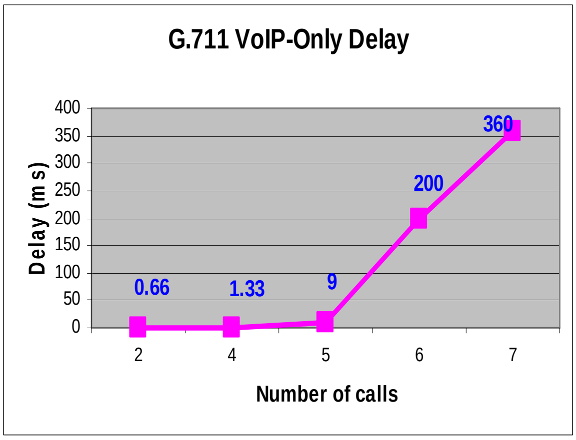

Fig. 5 G.711 Average Delay

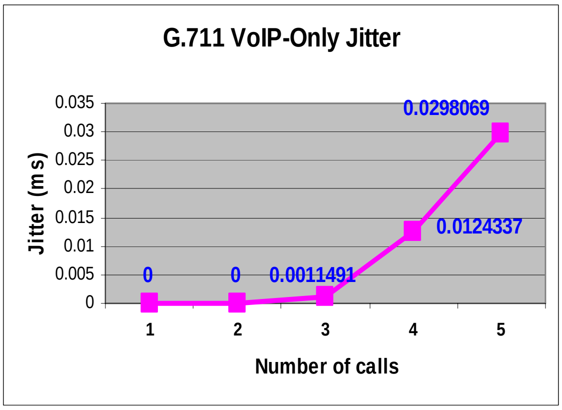

Fig. 6 G.711 Average Jitter

Third, we examined jitter. Fig. 6 shows that jitter stayed in the accepted range (less than50 ms) even when the number of VoIP calls reached 7. Therefore, jitter was not the main factor in determining the capacity of our model.

Table III summarizes the results achieved when the G.711 codec was used and no background traffic was added.

Fourth, we study the data loss. Fig. 7 shows the percentage of data loss calculated as the difference between the packets sent and the packets received. It can be concluded that when the number of VoIP calls is 5 or less, the data loss is acceptable (less than or equal to 1.5%). However, when the number of voice calls increases to 6 or more, the data loss becomes unacceptable. This result is consistent with [4].

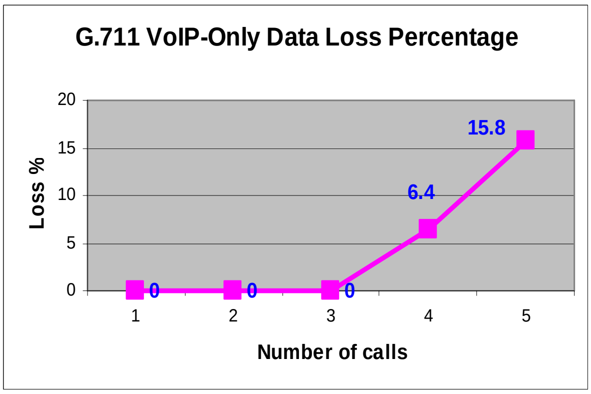

Fig. 7 G.711 Data Loss Percentage

TABLE III
G.711 VoIP-Only

Calls	Delay (ms)	Jitter (ms)	Data Loss %	Acceptable Overall QoS?
2	0..66	0	0.0	Yes
4	1.33	0.0011491	0.0	Yes
5	9.00	.0124227	0.0	Yes
6	200.00	0.02	6.4	No
7	360.00	0.0298069	15.8	No

TABLE IV
G.729 VOIP-ONLY

Calls	Delay (ms)	Jitter (ms)	Data Loss %	Acceptable Overall QoS?
2	0.33	0.00	0.000	Yes
4	0.44	0.00	0.000	Yes
6	0.60	0.00	0.000	Yes
12	2.50	0.02	0.000	Yes
14	15.00	0.22	0.000	Yes
15	70.00	0.60	0.195	Yes
16	150.00	1.11	0.600	Yes
17	401.00	1.60	4.316	No

TABLE V
G.729 VOIP AND DB TRAFFIC

# of VoIP Calls	DB Query Response Time (sec)	Delay (ms)	Jitter (ms)	Data Loss %	Acceptable Overall QoS?
0	0.460	n/a	n/a	n/a	Yes
2	0.460	2.10	0.000	0.00	Yes
4	0.460	2.25	0.030	0.00	Yes
6	0.475	2.75	0.048	0.00	Yes
8	0.500	4.00	0.050	0.00	Yes
9	0.515	6.20	0.075	0.00	Yes
10	0.550	11.00	0.225	0.00	Yes
12	1.000	65.00	0.800	0.00	Yes
13	7.000	600.00	1.600	0.23	No

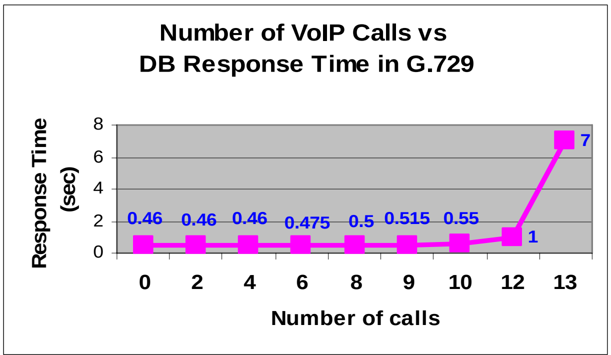

Fig. 8 G.729 Response Time

A similar set of scenarios was carried out in another experiment when we changed the codec scheme to G.729. The findings of the experiment for all scenarios are listed in Table IV. The delay changed from 0.33 ms for 2 VoIP calls to about 150 ms when the number of VoIP calls increased to 16. This means that by considering the delay, the capacity of the model is 16 VoIP users. The jitter on the other hand did not affect our conclusion since it is in the acceptable range for all scenarios; it ranges from 0 ms for 2 VoIP calls to 1.11 ms for 16 VoIP calls. The data loss value ranges from 0% for 2 VoIP calls to 0.60% for 16 VoIP calls. Notice that any extra voice call would affect both the delay and the data loss percentage and puts them outside the accepted range (see the last row of Table IV). This indicates that the capacity of our model is 16 simultaneous VoIP calls.

The final experiment involves a similar set of scenarios to the previous one with the addition of background traffic; namely, database traffic. Table V summarizes the results for all scenarios of this experiment. We introduce in the second column a new performance metric: response time. We examined the response time for database queries and compared it in various scenarios against the situation when there are no voice calls; i.e. when the number of VoIP calls is 0. Column 2 of Table V and Fig. 8 show that the response time was almost the same when the number of users was 0, 2, and 4. There was a small change in the response time, however, when the number of VoIP users increased to 10. When the number of VoIP calls increased to 12, the response time jumped to 1 second (acceptable values are 1 second or less). When the number of VoIP calls was set to 13, the response time jumped to 7 seconds; clearly, something affected the whole system. This makes us conclude that our model can accommodate 12 VoIP calls while 22 heavy load database users are accessing the model. By observing the delay column in Table V, we notice that the delay stayed in the acceptable range as long as the number of VoIP calls was below 13 and became unacceptable when the number of VoIP calls became 13; this result is consistent with the response time result. By looking at the jitter column, we find that the jitter is acceptable for all scenarios (0 ms to 1.6 ms). This shows that our model keeps the amount of jitter low; hence, it is not the dominant parameter for deciding the capacity of our model. The data loss percentage as shown in the table is zero for all acceptable numbers of VoIP calls (2 to 12); it slightly changes (0.23%) when the number increases to 13.

By comparing the findings in Table IV to those in Table V we can see that adding more voice calls affected the database response time and increased it. This increase reduces the capacity of the model from 16 users when no database traffic is available to 12 users when there is database traffic. A closer look at the jitter and delay columns in Table IV and Table V shows that the database traffic increased both the jitter and the delay, even though they both stayed in the acceptable range.

TABLE VI
SUMMARY OF RESULTS

	Experiment/Codec values		
	G.711	G.729	G.729
Packetization Interval	10 ms	10 ms	10 ms
Background traffic?	No	No	Yes
Maximum Capacity (number of voice calls)	5	16	12
Delay (ms)	9	200	65
Jitter (ms)	0.0011491	1.1	0.8
Data Loss%	0	1.48	0
Response Time (seconds)	n/a	n/a	1
Deciding Factor	Data Loss%	Delay and Data Loss%	Response Time and Delay

TABLE VII
FACTORS ACCEPTED VALUES

Factor	Accepted range	Unaccepted Range
Delay (ms)	0 to 200	Above 200
Jitter (ms)	0 to 50	Above 50
Data Loss %	0 to 1.5%	Above 1.5
Response Time (seconds)	0 to 1	Above 1

The change in the delay is greater than the change in the jitter. For example, when the number of VoIP calls was 12, the delay was 65 ms with the availability of database traffic, while it was only 2.5 ms when no database traffic was available. By comparing the equivalent jitter values at 12 VoIP calls, we see that it was 0.02 and 0.8 respectively.

V. CONCLUSION

In this paper, we showed that the RiWC model works well when VoIP and database traffic co-existed. The model's capacity is superior to the traditional one, which only uses one access point per site. By employing redundancy in the model, we were able to show that the model was able to support a greater number of users when VoIP traffic is used alone and when other traffic (database) co-existed. This outcome indicates that our model is performing well and the scale enhanced, especially when codec G.729 is used and the packetization interval was set to 10 ms.

We summarize our findings in Table VI. We based our observations on the values in Table VII.

The findings of the first experiment indicated that the maximum number of simultaneous VoIP users the model was able to support was 5. This result is consistent with recent research findings.

When using the G.729 codec, the model was able to support up to 16 VoIP users. Similar experiments in current literature indicate a maximum of only 7 users.

The findings of the second experiment demonstrate that the maximum number of VoIP users the model was able to support in the existence of background traffic was 12.

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