

Simulation Method to Evaluate the Performance of 802.11 b/g Wi-Fi Access Point for Exclusive VoIP Traffic

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Abstract—Since the costly and power hungry GSM infrastructure is not suitable for basic voice communication requirements in villages with thin traffic and solar power only, the alternative is to examine the use a small network of outdoor, weather proof and water proof, low bandwidth 802.11 b/g (11/54 Mbps) Wi-Fi Access Points (AP)with high gain omni antenna, earmarked for exclusive VoIP traffic, to enable smart phone clients to talk to each other in VoIP mode. In this scenario, it is necessary to examine what is the distance possible from smart phone to AP, between 2 APs and also how to maximize the number of simultaneous VoIP traffic handling capacity of the AP.

This paper examines this aspect of the AP by software simulation of multi channel voice traffic and analyses the results obtained, when tested with a Wi-Fi AP in 11 and 54 Mbps mode. Based on the field area distance trials conducted and the simulation results of the multi channel capability of the AP, it is projected that engineering solution is possible. Such a solution will have major social benefit in developing countries, to cater for basic voice communication needs of villages and rural areas.

Index Terms— VoIP, SIP server, 802.11 b/g Wi-Fi, AP.

I. INTRODUCTION

With a network of outdoor, low bandwidth 802.11 b/g (11/54 Mbps) Wi-Fi Access Points (AP) with high gain omni antenna, it is possible to cater for VoIP voice between smart phone users in a rural or village of limited area.

The traffic requirement being small, traditional GSM operators, with costly infrastructure of BTS, BSC and MSC, do not find it commercially attractive, to serve such rural environment. Also, a typical BTS require about 3 KW prime power, which is difficult to manage in such rural environment. On the other hand, about 20 watt solar power with battery back up, will be more than adequate for the Wi-Fi AP with active high gain antenna . Cost wise also, the low bandwidth 802.11 b/g Wi-Fi AP with high gain omni antenna, is only a fraction of a BTS cost of GSM network.

To engineer such a network of Wi-Fi APs as infrastructure, for this application, it is necessary to establish (a)

possible near LOS distance at 2.4 Ghz, between smart phone and Wi-Fi AP mounted on a short mast (about 10 meter height) (b) distance between 2 adjacent APs at 2.4 Ghz (c) backhaul connectivity between such VoIP domains (d) under roaming conditions, how many smart phones can be serviced by an AP, if they happen to be in that locality.

There are many reports to say that a Wi-Fi AP is very poor in handling voice traffic. But such evaluation has been made in mixed voice and high bandwidth data traffic environment, wherein it has been found that even a few channels of voice will drastically bring down the speed of the data traffic. This problem can be easily addressed, by reserving some low bandwidth exclusively for VoIP voice traffic.

To establish the simultaneous exclusive VoIP voice traffic handling capacity of a 11/54 Mbps Wi-Fi AP, this software simulation exercise is being undertaken, details of which is presented in this report, which describes, how the simulation software is prepared, how its correctness tested, how it is used in live testing in 802.11 b/g Wi-Fi AP link and how the performance results are analyzed.

Based on the results of the simulation software testing with a outdoor 11 / 54 Mbps Wi-Fi AP, to ascertain the number of simultaneous VoIP traffic from many smart phones to be serviced by the AP under roaming conditions, and based on the distance figures obtained from smart phone to AP and between 2 APs, the Bill of Materials (BOM) for catering for VoIP voice clients with smart phones, in a rural area or village of about 10 Sq Km has been ascertained. Also, the backhaul connectivity between such VoIP SIP domains has also been ascertained.

Section 2 gives a short list of Literature Survey. The authors have examined the functioning and performance of 802.11 b/g (11 / 54 Mbps) Wi-Fi Access Point (AP) from different perspective. Section 3 defines the problem in brief. Section 4 gives the motivation to carry this work. Section 5 describes the innovative content. The analytical aspects of the Wi-Fi AP protocol of packet string and its effect on handling number of channels simultaneously is covered in Section 6. Section 7 gives the proposed solution. Section 8 gives the design and implementation of the proposed solution.

II. LITERATURE SURVEY

The authors of [1] presented an analytical upper bound and using simulation, showed this bound to be constrained in scenarios, where channel quality is good and delay constraints are weak or absent. Then, through simulation, exhibit that capacity is highly sensitive to the delay budget allocated to packetization and wireless network delays and also show, how channel conditions and voice quality requirements affect the capacity.

Selecting the optimum amount of voice data per packet is shown to be a trade-off between throughput and delay constraints: by selecting the packet size appropriately given the delay budget and channel conditions, the capacity can be maximized.

Authors of [2] suggest that Voice over IP (VoIP) in 802.11 wireless networks (Wi-Fi) is an attractive alternative to cellular wireless telephony. Unfortunately, VoIP traffic is well known to make inefficient use of such networks. Experiments demonstrated that VoIP halves the available TCP capacity of an 802.11b hotspot when six to eight VoIP stations share the medium, and effectively extinguishes TCP connectivity when ten VoIP stations are present. Further, it is shown that neither the higher data rates of 802.11a/g nor the 802.11 standard for quality of service, 802.11e, fully ameliorate the problem. Instead, the problem is rooted in Wi-Fi's contention based medium-access control mechanism and considerable framing overhead.

The authors have proved that increasing handset deployment has the potential to cripple existing hotspot and enterprise Wi-Fi networks and have proposed SoftSpeak, a pair of backwards-compatible software extensions that enables VoIP traffic to share the channel in a more efficient, TDMA-like manner and addresses contention and framing overhead.

Authors in paper [3] have considered IEEE 802.11b, 802.11a and 802.11g systems and has provided analytical and simulation results for the conversational speech capacity of WLANs and compared the different WLAN technologies. They have showed that increasing the packetization interval increases channel efficiency by decreasing the number of packets generated per second, thereby reducing the number of times the high overhead is incurred. They have also assumed -Packets that are delivered past the application specific deadline normally comprise stale info and are generally dropped. Therefore, from the application perspective, there is no distinction between late arriving packets and lost packets. As such, the fraction of overdue packets need to also be accounted for in calculating voice potential.

The authors of [4] have presented an experimental evaluation of VoIP capacity in WLANs under various data and voice loads. They have described a quantifiable performance metric for voice sessions that accounts for both packet losses and delays. They have also presented experimental capacity measurements with and without

background data traffic. They have analyzed that to provide quality of service guarantee to voice traffic, an admission control mechanism must be administered to avoid overloading the network. An experimental evaluation of VoIP capacity in WLANs under various data and voice loads is presented here. The obtained capacity estimates could be used in admission control engine to make an admission or a rejection decision.

In the article[5], the author carried out analytical and simulation experiments to obtain the results for the conversational speech capacity of WLANs and compared the different WLAN technologies in this regard.

Considering IEEE 802.11b, 802.11a and 802.11g systems in the infrastructure mode authors found that voice capacity is a strong function of the channel bandwidth, codec packetization interval, data traffic and the packet size used by data. They conclude that the analytical results are in close agreement with simulations.

Real time QOS planning requires finding out the VoIP capacity in dynamic QOS conditions in VoIP network, reasonably accurately but in simple terms. [7,8]. Simple formulation for VoIP capacity is possible taking into account the timescale of typical transmission and reception of a IP / UDP payload frame.

In real-world deployment of Wi-Finetworks it can be anticipated that other applications will be running over the network. It is well known that VoIP capacity can be seriously degraded in such a scenario [9].

Priority services for VoIP in Wi-Fi, is a method to improve the QOS for voice, in practical networks with a mix of data and voice traffic. Some manufacturers have this feature in their APs, still satisfying 802.11 b standards.[10].

Important parameters which affect the VoIP traffic are RF bandwidth, codec rates, packet size. An empirical formula based on these parameters has been worked out and using OPNET simulator, experimental results obtained and compared with empirical formula results.

III. PROBLEM DEFINITION

GSM technology implementation with many analog components and sub systems is highly expensive to service thin, essentially voice, traffic areas in villages.

Non availability of prime power is also a major handicap for the GSM service provider.

Return on Investment (ROI) is not attractive for a GSM service provider to cater rural area. The traffic requirement being small, traditional GSM operators do not find it commercially attractive, to serve such rural environment.

Hence we are examining, how VoIP Wi-Fi network can provide a cost effective and low power (solar powered) solution at a fraction of cost of GSM infrastructure for a rural area or village.

IV. MOTIVATION

In spite of the fact that during the last few decades, traditional GSM network has reached many parts of the world, one cannot forget that GSM infrastructure BTS, BSC, MSC s highly expensive and power hungry for the rural area and village environment in many parts of the world. On the other hand, the Internet IP components of wired and wireless LANs, like Wi-Fi, routers and IP switches have come down drastically. It will benefit the society of weaker section of people if one can cater for basic voice communication in villages and rural areas, using Internet technology. There are service providers who cater for Internet voice in advanced countries and our effort is to examine such a solution suitable for villages at an affordable cost.

V. INNOVATIVE ASPECTS

Internet technology is used to bring out a cost effective and low prime power solution for basic voice communication need of the rural areas or villages. VoIP and outdoor Wi-Fi with high gain omni antenna based solution is the innovation in this paper.

VI. ANALYTICAL ASPECTS

First, it is required to determine a simple formulation for voice capacity, based on approximations for the timescale of VoIP packet transmission in Wi-Fi[6]

Figure 6.1 depicts data frame, as it handled by the AP, which illustrates the additional delays which restrict the flow of real time payload UDP voice packets.

<----- T ----->

| DIFS | Back Off | UDP VoIP data frame | SIFS |

Courtesy[6] Fig 6.1 . Data frame depicting the AP handling UDP VoIP payload.

(In addition in the case of IP data frame, there will be ACK data string also, in this frame).

DIFS stands for Distributed Inter Frame Spacing, and SIFS stands for Short Inter Frame Spacing, which in addition to Back Off are the safety time gaps given while handling the IP / UDP payload frame. In the case of IP frame, there will be further time taken due to ACK frame in the reverse direction. The values for these time delays in DIFS etc are specified in 802.11b DSSS (Direct Sequence Spread Spectrum).

It can be seen that Distributed Inter Frame Spacing (DIFS), back Off, Short Inter frame Spacing (SIFS) contribute delays in handling the VoIP payload data by the AP, even for the simple layer 2 function of receiving and retransmitting. This restricts the capacity of an AP in handling voice traffic.

Estimation of the capacity (N) the number of voice channels, taking into account the delay factors of the IP (UDP) frame handling by the AP, for factors of (a) RF bandwidth 11 / 54 Mbps (b) Packet size (MTU 800/640/480/ 400/ 320 etc) (c) Voice coding rate 64 Kbps (other lower coding rates are not considered, since the worst case is with highest rate of 64 Kbps) (d) voice traffic silent periods (3/4/5 %) of one second voice segment is made.

The following simple empirical formula adopted by many authors is used, for calculating the capacity N.

$$N = 500 * T_p / [774 + (592 + C * T_p / L]$$

Where T_p is frame size in msec (it is 100msec for 800 bytes voice frame), C is coding rate in Kbit per sec 64 for PCM g 711) and L is RF bandwidth in Mbps (11 for 802.11b).

The simulation software enables testing real link of 802.11 b/g (11 / 54 Mbps) Wi-Fi AP for varying values of packet sizes (800 to 320 bytes), RF bandwidth 11 and 54 Mbps and 64 Kbps coding rate. The simulation methodology and the results obtained when tested with typical outdoor Wi-Fi 802.11 b/g AP are presented in subsequent paragraphs.

VII. PROPOSED SOLUTION

The aim is to establish a mechanism to route voice traffic at specific areas with the condition that the traffic originating within the specified area does not cross the boundary at any point and has to end within the specified area(village), where the calls are not routed to PSTN that is they need not go to the Telecom operator. This works in dual mode that is if the traffic is outside the boundary then it is routed to GSM otherwise internally routed. Figure 7.1 depicts the VoIP communication

When the calls are to be established between smartphones connected to same network, the SIP server will handle the signalling and other aspects. There is no need to route the call via PSTN. But when the call has to be established with a smartphone in another IP network or any other PSTN number, it connects to the gateway and through FXO port, it is routed to callee number.

Figure 7.1 depicts a typical network of 802.11 b/g Wi-Fi Access Points serving as infrastructure for VoIP traffic from smart phone SIP clients. SIP server software is provided in one node to cater for call establishment between SIP clients.

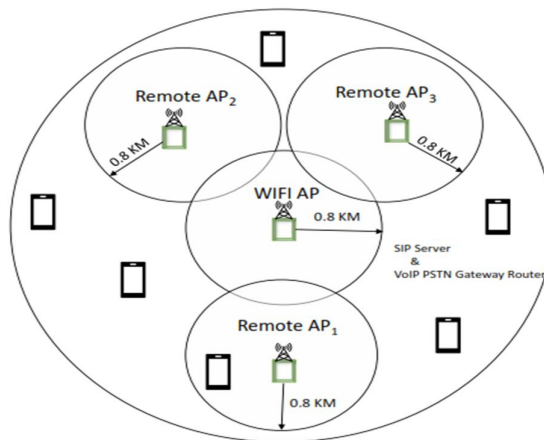


Fig 7.1 Block Diagram of VoIP Wi-Fi Topology

Figure 7.2 shows using a VoIP PSTN gateway router, how VoIP and PSTN domains can be integrated, which will also serve as backbone network for interconnecting 2 VoIP SIP domains in far away locations.

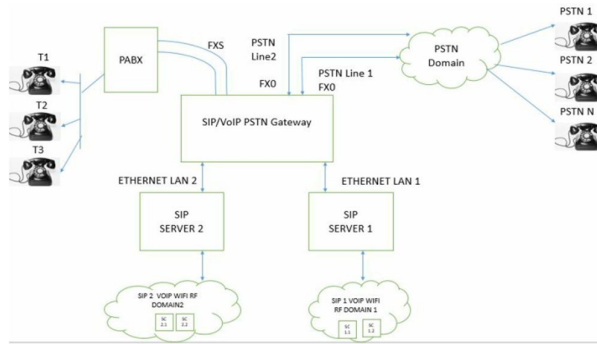


Figure 7.2: VoIP PSTN Gateway for interconnecting VoIP and PSTN domains

To establish the simultaneous VoIP call traffic handling capacity of a 11/54 Mbps Wi-Fi AP, software simulation exercise is being carried out. How the simulation software is prepared, how it is validated, how it is used in live testing in 802.11 b/g Wi-Fi AP link and how the performance results are analyzed are presented in the paper.

VIII. METHOD OF IMPLEMENTATION

This simulation is for emulating an environment, as if multiple smartphones are talking to each other simultaneously, using VoIP Wi-Fi mode and the link being setup via an exclusive 2.4 Ghz, 10/54 Mbps data rate of an 802.11b/g Wi-Fi access point. Figure 8.1 depicts experimental setup for simulation of multiple VoIP calls.

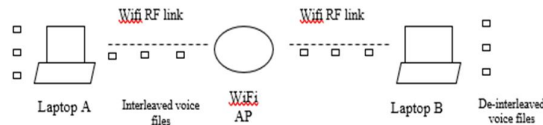


Fig 8.1 Simulation of multiple voice channels

8.1 Simulation software preparation :

Voice files are prepared apriori and saved in the transmitting end Laptop A. Laptop B is used as the receiver of the voice files, via the Wi-Fi AP RF link. The multiplexing method adopted to simulate voice traffic from multiple sources occurring simultaneously is as follows:

Each voice file is divided into several segments. Then these segments are interleaved and arranged in the order to simulate as if the traffic is occurring simultaneously from many sources. After arranging these voice segments, before transmission, delay is introduced between 2 consecutive voice segments to simulate voice activity silent period while talking. To find its effect on the channel capacity supported by the AP, varying time delays(30/40/50 msec) are introduced between 2 successive one second voice segments. This represents 3/4/5 % silent periods respectively(Generally, in duplex mode, while one is talking, other is listening and hence nearly 40 to 50 % silent period is taken care of, particularly in satcom links. However, this is not applicable in this simplex mode of testing). After arranging these segments, the UDP link is invoked to transmit from laptop A to be received by laptop B, via the target Wi-Fi AP. To find the effect of packet size (MTU) on the channel capacity supported by the AP, varying packet sizes 350 /400/450/480 bytes are used in the simulation. The above testing method follows the idea of one physical link supporting many virtual links. At the receiving end, after the transmission is over, the received UDP frames are used by receiver end software, to regroup the one second segments of respective files (de segmentation), saved and the quality of received voice files checked against the transmitted voice for all the channels. All these are carried out using socket programming in Python.

8.2 Verification of correctness of simulation software

Before using the simulation software in the target environment of Wi-Fi AP between the transmit and receive laptops, it is first verified that multiplexing of all the voice channels, for transmission via a 10 Mbps RF link to the receiving end and demultiplexing is done correctly. This is done by:

- o Software loop back and checking that transmit and receive file of same channel is same by decoding the received packets and listening to the voice and so is the case for all the voice channels.
- o By checking over an exclusive 10 Mbps Ethernet link between the transmit and receive laptops.

- o By checking over an exclusive Wi-Fi RF link between the 2 laptops. Peer to peer communication, Wi-Fi Direct utility.

The above will ensure that our simulation software emulates the environment of multiple VoIP frames being handled by the target Wi-Fi AP.

8.3 Link testing with target 802.11 b/g Wi-Fi AP :

- The number of channels supported by the 11 Mbps 802.11 b Wi-Fi AP will be established by varying the number of channels in the simulation s/w in steps of 10 ie: 10, 20, 30, 40 and 50. Analytical calculations indicate that a 10 Mbps 802.11 b Wi-Fi AP can handle about 30 to 35 (one way) 64 Kbps voice packets with 100 msec (800 byte) payload size for reasonable quality of voice.
- First at the course level of 10s of channels and then in finer single channel level.
- The voice channel support check is done under 12 different scenarios of (a) voice activity factor -- 3 types (with 3,4,5 % delay between one second voice segments) and (b) 4 different MTUs (350, 400, 450, 480 bytes). The following table depicts the results obtained for the 12 cases and also the packet loss in these cases.

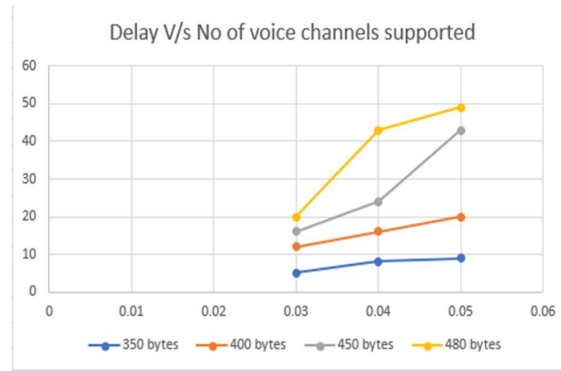
TABLE I. VOICE CHANNEL SUPPORT FOR 12 SCENARIOS

MTU	Voice activity silent p	E2Edelay(s)	Packet loss	%age of P	Voice ok
350	0.03	0.142152	797	8.764984	5
	0.04	0.106365	113	1.002662	8
	0.05	0.072919	21	0.169382	9
400	0.03	0.0663	298	2.152246	12
	0.04	0.05776	212	1.190877	16
	0.05	0.062969	93	0.427429	20
450	0.03	0.070005	136	0.826245	16
	0.04	0.072725	84	0.352778	24
	0.05	0.058586	44	0.111874	43
480	0.03	0.075495	106	0.61081	20
	0.04	0.075537	68	0.195864	43
	0.05	0.087482	50	0.115287	49

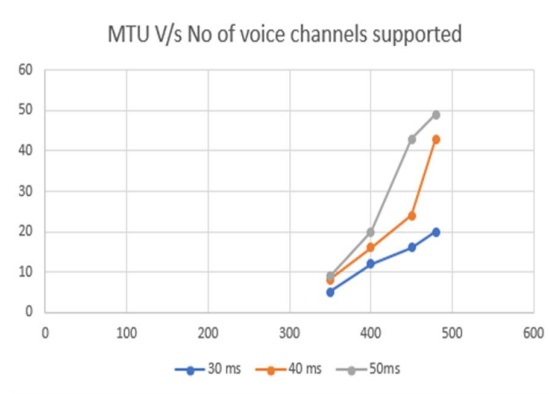
Wireshark is used to get details of the packets sent and packets received, from which Packet Loss (PL) % and End to End (E2E) delay are computed.

IX. ANALYSIS OF RESULTS AND DISCUSSION

- It can be seen from the table, that the 11 Mbps (802.11b) Wi-Fi AP can support varying number of simultaneous VoIP voice channels. It varies from as low as 5 to, as high as 49 one way 64 Kbps voice. Lowest capacity of 5 is for 350 byte MTU and 30 msec (3%) silent time and the highest capacity of 49 is for 480 byte MTU and 50 msec (5%) silent time.
- Graph 1 depicts the effect of voice activity (silent period) on number of channels supported. More time delay or silent period (50 msec) supports more channels and less the delay (30 msec), lesser the number of channel support.
- Graph 2 depicts the effect of MTU size on the number of channels supported. Larger MTU means more number of channels support.
- Packets that are delivered beyond the application specific deadline usually contain stale information and are generally dropped. Therefore only those packets that arrive prior to the deadlines are considered successfully delivered, while packets received afterwards are considered lost. We introduce a performance metric called the Packet Loss Ratio (PLR) that measures the percentage of voice packets that are failed to be delivered to a voice application on time.
- The following 4 tables 2 to 5 and the Graphs 3 to 6 depict End to End (E2E) delay and Packet Loss for 11 Mbps 802.11b and 54 Mbps 802.11g Wi-Fi APs for various MTUs and for 10 to 50 channels. It can be seen that larger the MTU, the E2E delay and Packet Loss performance is better and the corresponding increase in number of voice channel supported.
- Performance of 54 Mbps is better than 11 Mbps for all channel loadings and MTUs.



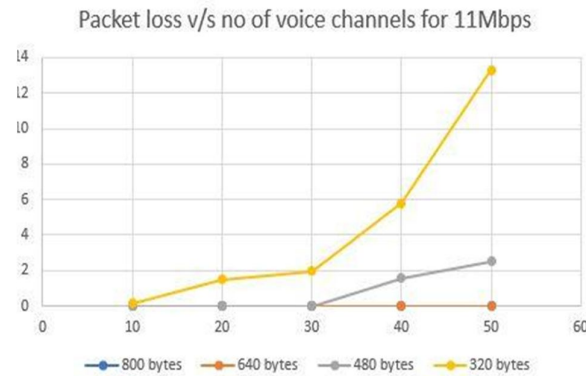
Graph 1 Voice activity silent period v/s number of voice channels supported for 64kbps encoded voice file in case of 11Mbps Wi-Fi RF link



Graph 2 Packet size (MTU) v/s number of voice channels supported for 64kbps encoded voice file in case of 11Mbps Wi-Fi RF link

TABLE II. PACKET LOSS PERCENTAGE FOR DIFFERENT MTUS AND NO. OF VOICE CHANNELS FOR 802.11b

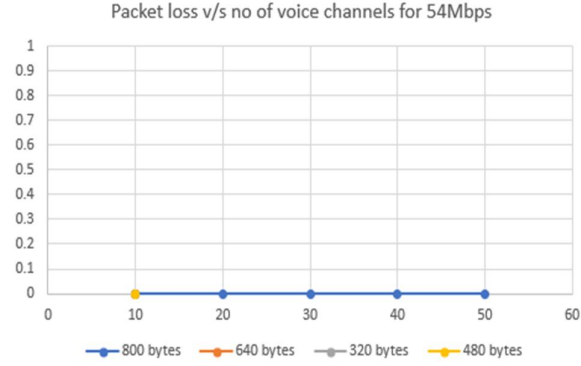
11Mbps					
MTU-No of voice files	10	20	30	40	50
800	0	0	0	0	0
640	0	0	0	0	0
480	0	0	0	1.584541	2.510682
320	0.158103	1.468775	1.949934	5.810277	13.28538



Graph 3 Packet loss percentage v/s number of voice channels supported for 64kbps encoded voice file in case of 802.11b 11Mbps Wi-Fi RF link

TABLE III. PACKET LOSS PERCENTAGE FOR DIFFERENT MTUS AND NO. OF VOICE CHANNELS FOR 802.11G

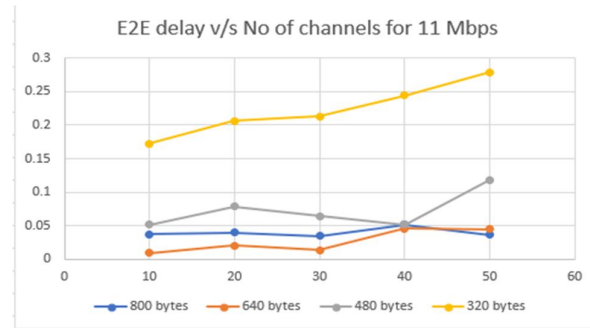
54Mbps					
MTU-No of voice files	10	20	30	40	50
800	0	0	0	0	0
640	0	0	0	0	0
480	0	0	0	0	0
320	0	0	0	0	0



Graph 4 Packet loss percentage v/s number of voice channels supported for 64kbps encoded voice file in case of 802.11g 54Mbps Wi-Fi RF link

TABLE IV. E2E DELAY FOR DIFFERENT MTUS AND NO. OF VOICE CHANNELS FOR 802.11B

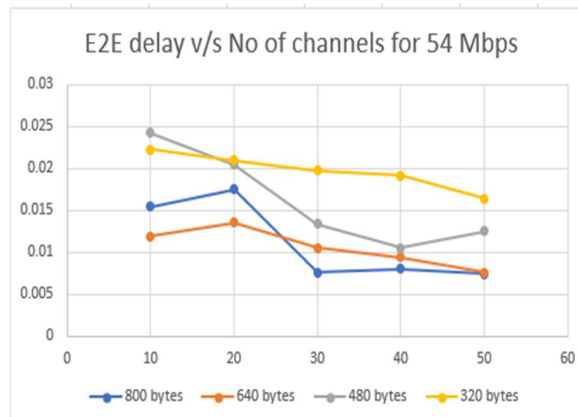
MTU\No of files	10	20	30	40	50
800	0.037142	0.039695	0.034588	0.051765	0.036506
640	0.009069	0.020006	0.013409	0.04507	0.045001
480	0.05147	0.078517	0.064171	0.051765	0.117848
320	0.172587	0.206054	0.212868	0.244536	0.279181



Graph 5 E2E delay v/s number of voice channels supported for 64kbps encoded voice file in case of 802.11b 11Mbps Wi-Fi RF link

TABLE V. E2E DELAY FOR DIFFERENT MTUS AND NO. OF VOICE CHANNELS FOR 802.11G

MTU\No of files	10	20	30	40	50
800	0.015438	0.017484	0.007555	0.007961	0.007391
640	0.011875	0.013479	0.010467	0.009369	0.007597
480	0.024185	0.020401	0.013382	0.010467	0.012519
320	0.022202	0.02091	0.019719	0.019106	0.016349



Graph 6 E2E delay v/s number of voice channels supported for 64kbps encoded voice file in case of 802.11 g 54Mbps Wi-Fi RF link

Inference from the work carried

It is observed that the 11 Mbps (802.11b) Wi-Fi AP can support varying number of simultaneous VoIP voice channels. It varies from as low as 5 to, as high as 49 one way 64 Kbps voice. Introducing more time delay or silent period supports more channels and less the delay, lesser the number of channels supported. Larger MTU gives a greater number of channels support because of reduced processing overhead. It can be seen that larger the MTU, the end to end delay and Packet Loss are minimal. Hence there is an increase in the number of voice channels supported.

X. CONCLUSION AND FUTURE WORK

The focus of this work was to establish a cost effective alternative to very expensive and power hungry GSM infrastructure. This simulation method obviates the need for large number of smart phones on VoIP Wi-Fi mode to be available for evaluating in the field, to determine the number of VoIP calls that can be handled simultaneously by a low bandwidth (11 Mbps) 802.11 b Wi-Fi AP.

From the experiment carried out, it is determined that as the number of channels increase, packet Loss and E2E delay also increases. With reasonable MTU size, of the order of 480 bytes, it is possible for the 11 Mbps 802.11b Wi-Fi AP to support about 40 simplex or 20 duplex exclusive VoIP voice channels. Simultaneous voice channel capacity of a Wi-Fi AP ensures that, when deployed in a cellular environment, in a peak hour scenario, the more number of smart phones could be serviced by the nearest AP for reasonable voice quality performance.

This work gives the confidence that a reasonable small network of about 16, 802.11 b -- 11 Mbps outdoor APs with high gain omni antenna, can serve well for voice and some data traffic also, in a cost-effective manner, compared to the costly and power-hungry GSM network of BTS, BSC, MSC for rural village of about 10 Sq. Km area (1.75 Km radius).

Future work

Field trials should be carried out for the results obtained. Ascertain maximum no. of smart phones to be serviced by a single AP which is of relevance under critical roaming conditions.

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