Voice Capacity Evaluation over Wi-Fi Network

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ABSTRACT

To satisfy customer and with good market benefits, voice over Wi-Fi networks can ensure voice quality depending on various network parameters, configurations and traffic conditions such as channel noise, capturing effects. Accuracy in voice capacity estimation model leads satisfaction to network designers. This paper reviews an analytical model which evaluate voice over Internet Protocol (VoIP) capacity and over Wi-Fi networks. ITU-T E-model is introduced to ensure good voice quality and VoIP call capacity.

Key words: Call capacity, E-model, VoIP, VoIP Network Topology, Multihop Wi-Fi Networks.

INTRODUCTION

VoIP is increasingly becoming the most popular means of voice communication. The packetized voice communication using VoIP is indifferent to physical, medium access, or routing layer protocols and can be used with a variety of network infrastructures. Customers can now use VoIP from anywhere using various types of communication devices, e.g., PCs, mobile phones, PDAs, etc. As the cost of Internet usage is decreasing fast with the availability of high-speed broadband Internet, the transport of voice packets is becoming cheaper. Low service cost is the primary reason behind the quick growth of VoIP market which is expected to reach $40 billion by 2015[1]. Acceptance of VoIP and user convenience will be even higher if calls from portable devices (e.g., mobile phone or iPhone) use Wi-Fi as low cost last mile network. Compared to other last mile and carrier networks, Wi-Fi does not require the use of cellular or other similar networks and, therefore, can deliver low cost service. Moreover, due to the ad hoc nature of operation, Wi-Fi is the easiest and low-cost solution to form temporary voice networks using VoIP in places like emergency road crashsite, disaster area, or war fields. Quality VoIP communication over Wi-Fi networks poses certain challenges. Maximum throughput in such wireless networks is much lower than their wired counterparts. The Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) based medium access mechanism used in Wi-Fi networks wastes considerable time in collision avoidance which increases end-to-end delay and leads to voice quality degradation. In multihop Wi-Fi networks, a critical zone is formed around the Access Point (AP) limiting channel utilization even more. Since the IEEE 802.11 standards do not guarantee an upper limit of packet delay or loss, call jitter and call drop can occur, specially under high traffic load, affecting voice quality. These necessitate the use of a thorough and precise VoIP capacity model that can be used in designing Wi-Fi networks so that call drops and call jitter can be reduced.

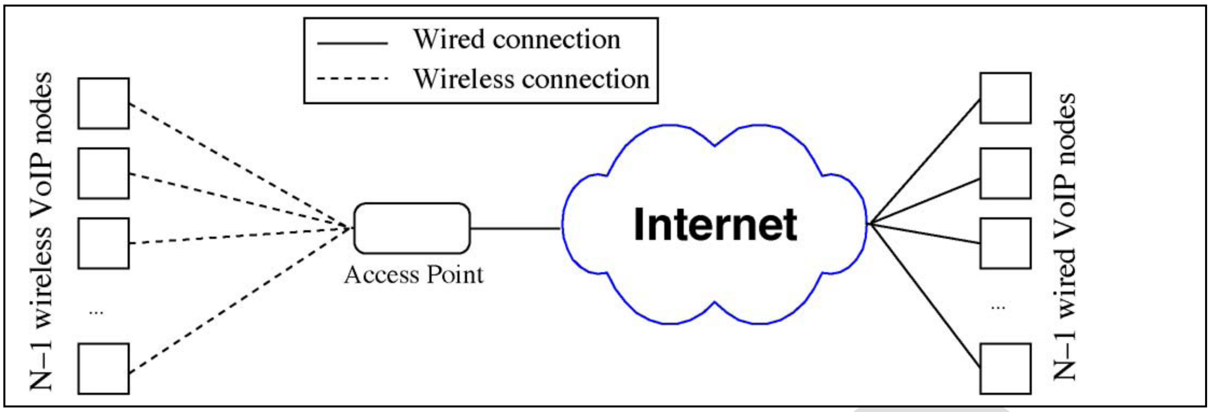

Fig 1. VoIP Network Topology

Fig 1. shows Voip network topology[5]. The capacity model will be very useful to network designers in (i) designing Wi-Fi networks to support call quality, (ii) determining preferable WLAN size and number and placement of APs, and (iii) selection of networking equipments based on their transmission and interference ranges and capture threshold, etc.

II. REQUIREMENTS

A. Voice Quality Requirement

Call quality has usually been subjective. The leading subjective measurement test is the mean opinion score where a subjective test is done by human experimenters to grade quality of a voice call in a range from 1 for Bad to 5 for Excellent. ITU-T defined an objective and computational method called E-model for use in transmission planning with the assumption that network impairment factors can be mapped to psychological satisfaction levels. E-model presents UPQ (User Perceived Quality) of transmitted voice in a rating of R-score in terms of network parameters. PSQM (Perceptual Speech Quality Measures) and PAMS (Perceptual Analysis/Measurement System) are two other objective methods developed by KPN (Koninklijke PTT Netherland) and British Telecom, respectively. Both of them are intrusive, i.e., they require a reference signal to be injected in one end of the network while in another end the received signal is compared to the original. ITU-T combined the best features of these two methods to introduce PESQ (Perceptual Evaluation of Speech Quality). MOS requires human experimenters and PESQ is intrusive. They cannot be used during the network designing phase. On the other hand, E-model is non-intrusive, represents voice quality in terms of network configuration, and does not require the network to exist beforehand. Therefore, E-model is the ideal tool in assessing voice quality during network designing and planning. E-Model represents UPQ of voice in score in a scale of 1 to 100. Since MOS is the most widely used voice quality assessment method, we start with levels and map them to score. While most studies focused in assessing VoIP capacity for medium quality calls, we investigate both high and medium quality calls[1].

B. Budget

Degradation in different network conditions incurs destructions to overall voice quality. ITU-T Recommendation G.107 defines as a weighted sum of basic signal-to-noise ratio, simultaneous impairments, delay and loss factors, as well as other compensation factor due to the advantages of access to the user. ITU-T defined the value of for a number of scenarios based on connectivity type and mobility.

C. End-to-End Delay

End-to-end delay is calculated as the delay from mouth (sender) to ear (receiver) and can be expressed as a sum of different delay factors in the end-to-end path, i.e., where different delay factors are described as follows. With aggregation level , the encoder puts number of voice frames in a single UDP packet where each voice frame is of length seconds. Therefore, an initial packetization delay of is introduced in the end-to-end path. Additionally, some encoders look into succeeding frames to improve compression efficiency for the current one. This incurs a look ahead delay. The delay in the Internet is denoted by which depends on the distance and route.

It is considered that the decoder is reasonably fast compared to the network and no queuing delay is experienced by a packet at the receiver node's interface queue before it is put into the dejitter buffer[1].

D. End-to-End Loss,

Packet loss depends on the underlying network and traffic pattern. The end-to-end loss has three components: queuing loss , channel access loss , and dejitter buffer loss. Out of all generated VoIP packets, portion (ratio of lost packets to total number of packets) is lost in the queue. The rest reaches the MAC layer of which is dropped due to contention. Finally, of packets reaches the dejitter buffer whose portion is dropped for being too late (otherwise would cause severe jitter). A larger buffer can reduce dejitter buffer loss at the cost of a higher dejitter buffer delay, and vice versa. However, using sufficiently large dejitter buffer, the dejitter buffer loss can be kept close to zero. This leaves the other two loss parameters and to be estimated for end-to-end loss, calculation[1].

III. NETWORK SCENARIO

Here, maximization of the number of calls which is the same as the number of stations is taken place, i.e., we add calls as long as the sum of the impairments due to the end-to-end delay and loss is less than the limit of total impairments. The impairments are calculated from codec, network parameters, and delay and loss factors. While VoIP applications can adjust codec configuration, other factors are highly dependent on the underlying network. Delay and loss in the queue are functions of queue length, arrival rate and departure rate, and the network can support the calls only if the queue does not grow continuously and the impairments due to expected delay and loss remain within the impairment budget. Routing and relaying are the two approaches to provide multihop connectivity in wireless networks. Routing is a widely researched and practiced solution, whereas relaying has not been adequately explored in wireless networks. Fig. 2 shows a multihop 802.11 network[1].

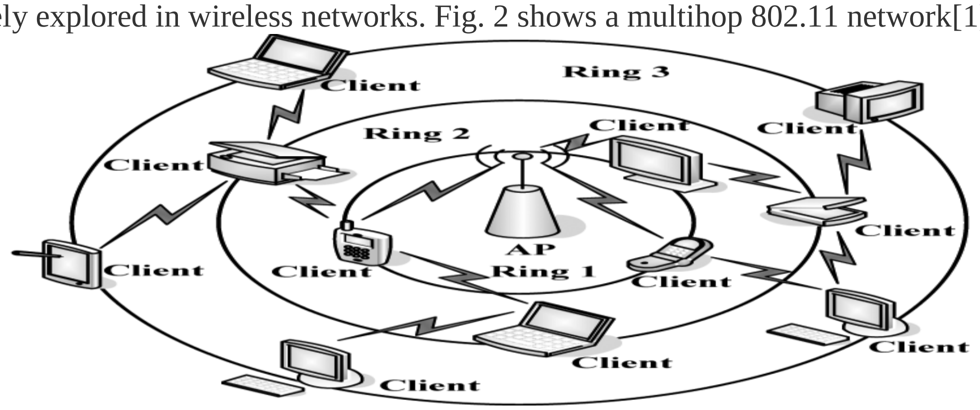

Fig. 2. A Multihop 802.11 network.

IV. *VoIP Call Capacity*

The effective packet arrival rate is much higher for nodes in lower indexed rings forming a critical zone in the immediate area of the AP and this limits capacity of the entire network. When the effective arrival rate for any node closer to the AP becomes greater than its maximum service rate, its queue starts growing and queue loss increases fast. This problem is most severe in ring 1 and to avoid such scenario the following condition must be satisfied. In ring 1, there are nodes, each of which has NIs. On the other hand, there are nodes in the WLAN, each of which initiates a two-way VoIP call with other nodes simultaneously is 1 for one-to-one call and higher for audio-conferencing). For each one-way VoIP packet stream, the encoder generates packets/sec which means a total of VoIP packets are generated per second in the whole WLAN. All of these packets must be forwarded through ring 1. The expected VoIP call capacity in a one hop IEEE 802.11b network with G.711 voice codec is about 85 simultaneous calls, but the actual observed capacity is only 5 calls even at the highest data rate and under zero loss conditions. In this paper we analyze the reasons behind this inferior performance of VoIP traffic. We also present algorithms at the medium access control layer to improve the observed call capacity. Finally, using ns-2 based simulations, we evaluate the algorithms and show that performance improvements of up to 300% can be achieved.

V. COMPARATIVE STUDY

Paper Title	*Voice Quality Rating/Bandwidth Usage*	*Budget/Impairment*	*Delay*	*VoIP Call Capacity*
An Analytical Approach for Voice Capacity Estimation Over WiFi Network Using ITU-T E-Model	Voice quality Rating, R≥70.07for medium quality and 80.16 for high quality	BudgetR=93.3553 $-I_d(d_e)-I_{e\_eff}(e_e)+A$ where $I_d(d_e)-I_{e\_eff}(e_e)$ denote impairments due to end-to-end delay(d_e) and end-to-end loss(e_e) , respectively.	$d_e= d_l+n_a d_f+d_i+d_q+d_c+d_j$ where n_a denote aggregation level, d_f denote voice frame length, d_l denote ahead delay, d_j denote static jitter buffer, d_q, d_c denote delay faced in queue	Voice call capacity n_s= 28.35 for medium quality and 13.19 for high quality
VoIP and Tracking Capacity over Wi-Fi Networks	Bandwidth Usage=57.72%	Impairment= 50%	UDP-1.86% TCP-0.28% for packetization interval 10.	With TCP Capacity-03 Without TCP Capacity- 06

Measurement and Analysis of the VoIP Capacity in IEEE 802.11 WLAN	$R=93.2-I_d-I_e$	Below 80% with 16 CBR calls and 40 VBR calls	$D = M \cdot \mu_{avg} = \frac{B}{S} \cdot \mu_{avg}$ Where, M denote buffer size in the number of packets, B denote buffer size in bytes, S denote IP packet size in bytes, μ_{avg} is the average transmission time	$N_{CBR}=P/(2.T_t)$ where, N_{CBR} denote maximum number of CBR calls, P denote packetization interval, T_t denote total transmission time for one packet

VoIP Call Capacity over Wireless Mesh Networks	G.729A performs better than G.711 in 11 Mbps case, it performs worse in 54 Mbps due to high value of *Ie* which is 11 and 0 for G.729A and G.711, respectively	28.3553 for medium 13.1952 for high	d = dL + nAlF + dI + dQ + dM + bg where dL denote head delay bg denote delay in dejitter buffer in the receiving end g denote packet inter arrival delay. dQ denote delays in protocol stack, the queue, and medium access, dM. dI denote the delay in the Internet and depends on the distance and route.	

CONCLUSION

The capacity of the network in terms of the number of simultaneous supported. Low bit rate codec's i.e., G.729 and G.723.1 provide higher VoIP capacity, at the expense of small quality reduction. The number of voice calls being supported simultaneously can be increased. The effect of packet loss on the VoIP capacity and model the delay of voice packets as function of the number of carried calls. A multi-hop WLAN architecture quantified its benefits. It is also concluded that the optimum packet size selection can be made without knowledge of the channel conditions.

FUTURE SCOPE

It is concluded that the capacity is highly sensitive to the delay budget allocated to packetization and wireless network delays. By selecting the packet size appropriately given the delay budget and channel conditions, the capacity can be maximized. The optimum packet size selection can be made without knowledge of the channel conditions.

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