Analysis of Voice Over Wi-Fi in a Wireless Lan with IEEE 802.11b Standard

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Abstract

Voice over Wi-Fi (VoWi-Fi) emerged after Voice over IP (VoIP) proved to be a useful technology to replace usual coaxial cable phone system. The breakthrough in VoWi-Fi may affect the cell phone business in the near future due to its lower cost, mobility and portability. In this paper, performance analysis of voice quality over the Wi-Fi network based on the IEEE 802.11b standard was done through simulation and the results analyzed in detail.

1. Introduction

The idea of VoWi-Fi actually comes from mobile applications where mobile terminals that were predominantly meant for voice only services can support other data related applications such as SMS (short message services) few years ago. The idea of VoWi-Fi was derived from the fact that the existing wireless local area networks, which were initially design to support data communications, can eventually support voice communication. However, there are issues pertaining to the voice quality that a wireless network can support due to its limited resources. This resulted in many manufacturers like Cisco, SpectraLink, Meru Networks and AirFlow Networks coming out with their own designs and implementations to support voice over wireless network. Even IEEE 802.11 Task Group E came out with the standard 802.11e which is dedicated for multimedia applications like voice and video over the wireless networks. In this paper, we attempted to see how voice can be delivered over WLAN. Services provided by MAC as well as Real Time Transport Protocol (RTP) and Real Time Transport Control Protocol (RCTP) will be discussed. The rest of the paper is organized as follows: Section 2 covers medium access control methods used in Wi-Fi, Section 3 coverers voice encoding and compression techniques used in wired and wireless networks, section 4 covers simulation methodology and results discussion and finally section 5 concludes the paper.

2. Medium Access control Methods for Transmitting Voice over Wi-Fi

The proliferation of internet protocol (IP) into wireless domain like GPRS, 3G and Wireless LAN has further increase the challenge of delivering real time services like voice and video over these bearers. Data is transmitted over IEEE 802.11 medium by using either the Distributed Coordination Function (DCF) or Point Coordination Function (PCF).

2.1 Distributed Coordination Function (DCF)

DCF is used as the core mode of operation for distributed infrastructure (star) network. It uses a contention based access method. A mobile node that is ready to transmit a frame will sense the medium. If the medium is busy, it will wait for an additional predetermined period of time of DIFS length. During that contention period, the mobile node will calculate the random back-off time by multiplying the time slot picked up in the window by a random number. The mobile number ticks down the random back-off time, checking to see if the medium is busy. The mobile

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node with the shortest time gain access to the medium first and transmits its frame. The collisions can now occur only when two or more mobile nodes select the same time slot to transmit. These mobile nodes will have to re-enter the contention procedure to select the same time slot to retransmit the collided frames [2].

2.2 Point Coordination Function (PCF)

PCF is specified in IEEE 802.11 as an optional protocol framing method. PCF was designed to accommodate those services requiring both voice and data transaction. PCF works in round robin fashion [2].

Version	P	CRC Count	Marker	Payload Type	Sequence	#
Timestamp						
Synchronization Source (SSRC) Identifier						
Contributing Source (CCRC) Identifier						
Payload						

Figure 1. The real-time transport protocol (RTP) used to deliver real-time data such as voice and video over the network whether it is wireless or wired networks [2]

3. Voice Encoding and Compression Techniques

The real key to sending voice over any packet data network (wireless or wired) is encoding and compression. Encoding digitizes analog signal like voice. Compression offers several advantages, one of which is the reduction of raw bandwidth required to support the information transfer [2]. VoWi-Fi like VoIP makes use of Digital Signal Processors (DSP) which is the engine for voice coders to compress as well as convert analog voice signal into data packet (RTP packet) so that they can be transported over an IP-based network [2]. The term DSP refers to the combined effort of DSPs and codec (compression and decompression) to perform the conversion of analog and digital signals into IP communication flows. There are a few voice encoding and compression technique like PCM (G.711), ADPCM (G.726) and many others.

3.1 Pulse Code Modulation

Pulse Code Modulation or PCM is a digital scheme for transmitting analog signal. The signals in PCM are binary, where there are two possible states: logic 1 (high) and logic 0 (low). This is always true no matter how complex the waveform happens to be. PCM is possible to digitize all form of analog data, including motion video, voice, music, telemetry and virtual reality (VR). To obtain PCM from analog waveform at the transmitting end of a communication system, the analog amplitude is sampled at regular intervals. The sampling rate is several times the maximum frequency of the analog waveform in cycles per second or Hertz. The instantaneous amplitude of the analog waveform at each sampling is round off to the nearest level. This process is known as quantization. The number of levels is always a power of 2 such as 4, 8, 16, 32, and 64.

3.3 Real Time Transport Protocol

Real-time Transport Protocol (RTP) is an application layer protocol. It is an IP-based protocol providing support for the transport of real-time data such as video and voice. It used lover level protocol such as User Data Protocol (UDP) and Transport Control Protocol (TCP) for the transport across the network [2]. See figure 1 for details.

In most situations UDP is used instead of TCP especially for transmitting multimedia applications like voice and video across a wireless network. UDP has less overhead since it does not provide several functions such as sequencing the datagrams. It sends, packet receipt verification, missing packet retransmission and other flow control services. Regardless of the underlying network protocol used, RTP provides data transport for real-time data like voice. It provides several functions to ensure data is synchronized for all users and will be recombined correctly at the receiving end by using the information contained in the RTP packet [2].

However, an RTP packet is still just a packet and there are several problems packets have that are magnified when dealing with real-time information. First, packets may or may not be received when they are, they may not be in the same order as when they were sent. This makes reconstructing the packet stream in the proper order and requesting missing packets very important [2]. This resulted in the design of another protocol known as Real-time Transport Control Protocol which works along side with RTP to provide feedback for flow control to manage several aspects of the delivery of real-time content [2].

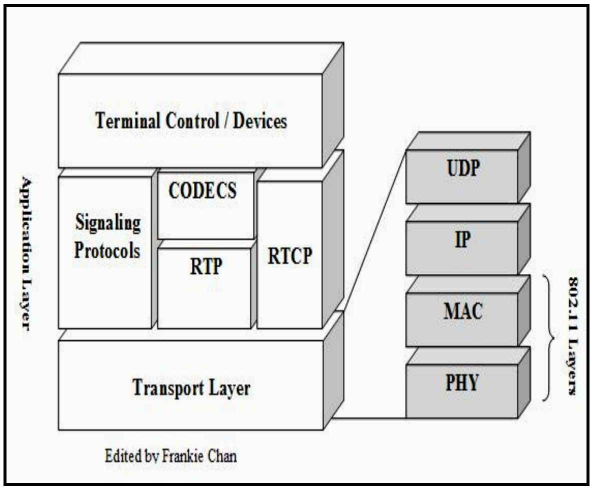

Fig. 2. The Voice over Wi-Fi (VoWi-Fi) protocol [2].

3.4 Problems Faced by Voice over Wi-Fi

Voice over Wi-Fi has its shortcomings. The 802.11 WLAN standard was originally design for data services and not for voice. Therefore it faces problems such as scalability, quality of services (QoS), delay impairments etc.

3.4.1 Scalability Problem

The IEEE 802.11 standard uses a mechanism known as Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) which was based on Ethernet wired network which uses Carrier Sense Multiple Access with Collision Detection (CSMA/CD) mechanism. Since the CSMA/CA is also based on a binary exponential back-off algorithm that only grant user access whenever there is no other user transmitting in the network. Stations essentially do their best to listen for transmissions and avoid collisions. The problem is the scheme is largely ineffective applied in situations where multiple clients are active such as applying in VoWi-Fi network. Like the Ethernet hub, which will cause performance to degrade considerably in populated and crowded networks, the wireless LAN is subject to prohibitive scalability issues. When a large number of nodes are active, typically six to ten, running either voice applications (voice calls) or data transmission, the network slows down as the nodes or stations constantly collide, back-off and wait [1].

3.4.2 Quality of Services (QoS) Problem

Furthermore, the 802.11 standard lacks the mechanism to be able to recognize voice and apply QoS at layer 2 in order to avoid packet delay and loss. Voice packets are treated identical to any other data packet. Wired networks today have the ability to control QoS at the Internet Protocol, IP layer (layer 3) because Ethernet switching capacity (layer 2) is high enough to ensure packets will be transmitted with minimal loss and with minimal retransmission. By contrast, WLAN today lack the ability to control contention with the efficiency of wired Ethernet switches. Therefore QoS suffers “over-the-air” (at layer 2) before any IP (layer 3) QoS mechanisms have a chance to make any effect [1].

3.4.3 Delay Impairments

Like any network, Voice over Wi-Fi also cannot escape from delay problems. Basically there are a few major delays that will affect the voice quality over the Wi-Fi network. They are propagation delay, packetization delay, medium access control delay, jitter buffer delay and transport delay.

4. Simulation Results and Discussion OPNET Modeler was used to simulate the voice performance over the IEEE 802.11b Wi-Fi network which has a maximum data rate of 11 Mbps. An office environment Wi-Fi network with 2 mobile subnets (2 AP domains) which can cover up to an area of 300m x 300m was designed. We started from 2 mobile nodes per AP domain, increased it to 4 and 6 mobile nodes per AP domain to observe the voice performance. The statistics that we obtained are voice packet end-to-end delay and packet delay variation as follow.

4.1 Voice Packet End-to-End Delay

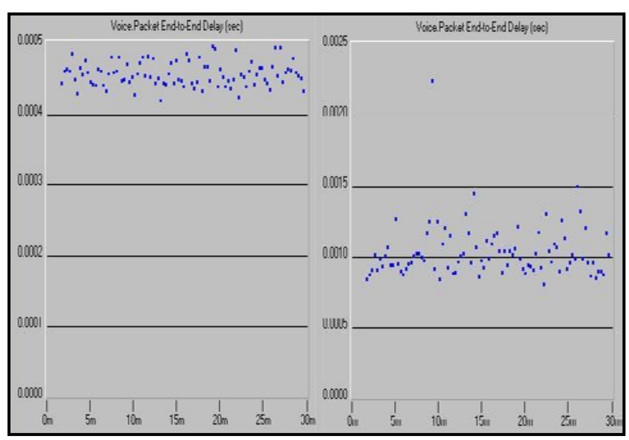

Fig. 3. Comparing voice packet end-to-end delay between 2 Mobile Nodes per AP Domain (left figure) and 4 Mobile Nodes per AP Domain (right figure).

Voice packet end-to-end delay is the time a voice packet takes to travel from the sender process (calling mobile node) to the receiver process (called mobile node). It is actually the sum of the processing, queuing, transmission and propagation delay. It can clearly be observed from figures 3 and 4 that when more active mobile nodes running voice applications (voice calls) simultaneously are added into a Wi-Fi network, the voice packet end-to-end delay increases significantly. From 0.0004s-0.0005s range for 2 mobile nodes per AP domain, it increases to 0.0005s-0.0025s range for 4 mobile nodes per AP domain. It later increases to a range of 0s-3s for 6 mobile nodes per AP domain.

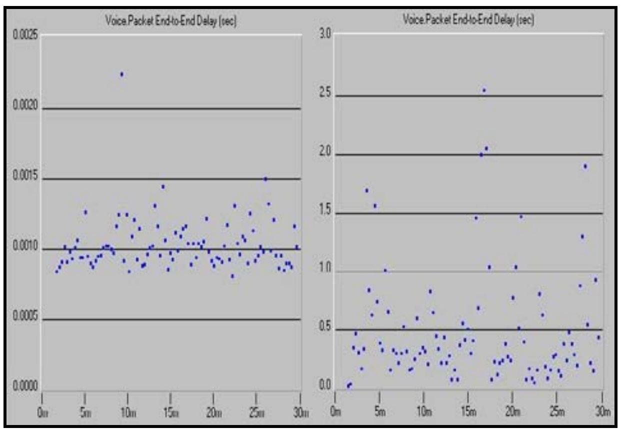

Figure 4. Comparing voice packet end-to-end delay between 4 Mobile Nodes per Access Point (left figure) and 6 Mobile Nodes per Access Point (right figure).

According to International Telecommunication Union (ITU) standard, the packet end-to-end delay has to be less than 0.4s in order to protect and preserve the voice quality over the wireless network. Therefore, the design of Wi-Fi network to have 2 and 4 mobile nodes per AP domain is therefore ideal to implement VoWi-Fi applications. However, for more mobile nodes per AP domain, the contention to use the scarce network resources certainly increases. This will eventually cause the network or voice packet end-to-end delay to rise which will result in poor voice quality over the Wi-Fi network.

5. Conclusions

Simulation results and statistics have shown that VoWi-Fi applications are applicable to Wi-Fi network with less mobile networks (less than 6). When active mobile nodes are running voice applications such as voice calls and audio streaming over the Wi-Fi network, the contention for network resources will increase and this eventually causes increase in voice packet end-to-end delay in Wi-Fi network. This results in poor voice quality over the Wi-Fi networks. In order to support voice applications over the Wi-Fi network, it is recommended that Wi-Fi network with higher data rates such as IEEE 802.11g which supports up to 54 Mbps has to be used. Our next research work will address this issue. It expected that 802.11g will provide better performance for voice applications over the Wi-Fi network in terms of voice quality as well as support more concurrent voice calls over the Wi-Fi networks. IEEE 802.11g standard can support up to approximately 4 to 5 times the amount of voice calls as compared to IEEE 802.11b which was investigated in this study.

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