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Enhanced channel access mechanism forvoip services in ieee 802.11 Networks

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Abstract:

Transmitting voice through IP data network can provide significant cost savings. However if not managed properly, voice quality can degrade due to data network congestion. VoIP is voice over an Internet Protocol (IP) based network.

In this paper we will investigate QoS indications (for voice packets), Improvement of NS2 network simulator in order to simulate statistical QoS in 802.11 MAC. The IEEE 802.11e Standard has been introduced recently for providing Quality of Service (QoS) capabilities in the emerging wireless local area networks. This 802.11e introduces a contention window based that is Enhanced Distribution Channel Access (EDCA) technique that provides a prioritized traffic to guarantee minimum bandwidth needed for time critical applications. However this EDCA technique resets statistically the contention window of the mobile station after each successful transmission. This static behavior does not adapt to the network state hence reduces the network usage and results in bad performance and poor link utilization whenever the demand for link utilization increases. For that purpose a new adaptive differentiation technique has been proposed for IEEE 802.11e wireless local area networks that take into account the network state before resetting the contention window.

To improve the QoS of Voice over Internet Protocol services we proposed a new traffic for VoIP. The performance of the proposed technique and proposed traffic is evaluated and compared with the original IEEE802.11a technique. Preliminary results show that the proposed adaptive technique enhances the channel utilization and increases throughput

Keywords: Voice over Internet Protocol (VoIP), QoS, WLANs, DCF, enhanced DCF.

1.Introduction

VoIP stands for Voice over Internet Protocol. As the term says VoIP tries to transfer voice (mainly human) using IP packets over the Internet.

Voices over IP (VoIP) applications are gaining an ever increasing popularity in the Internet community, favored by the massive deployment of wireless access technologies. For instance, more than eighty million users have already subscribed to Skype, the most popular VoIP commercial application for personal use, roughly 10% of which are estimated to be simultaneously online at any time. While it is not clear whether VoIP will ultimately replace traditional telephony, its massive diffusion may act as the main driving factor for the actual deployment of Quality of Service (QoS), both in the Internet backbone and in the (wired or wireless) access segments. For this reason, using VoIP as a test case in the performance evaluation of new QoS components, such as (to name a few) scheduling, resource reservation, admission control, traffic policing, traffic engineering, etc., has become a common practice. Unlike classic data applications, in which easily quantifiable, data-related performance metrics (e.g., throughput and mean packet delay) most often represent meaningful evaluations, the actual performance of VoIP applications depends on user perception (a concept often referred to as Quality of Experience, QoE). For this reason, the ITU-T has established a computational model, called the Emodel, which defines a quality factor - the so-called R score — to capture the effect of mouth-to-ear delay and losses in packet-switched networks. The R score can then be mapped to the Mean Opinion Score (MOS), which in turn can be converted into subjective quality levels (e.g. "good", "poor"). Despite this, assessing the VoIP performance through measures taken at the IP level – rather than taking into account the user perception – is often the norm in QoS literature. However, it can be shown that a sound assessment of VoIP quality has to take into account several factors which extend beyond the IP level. For instance, playout buffers, which come as part of a VoIP application, play a crucial role: packets that are successfully delivered within a given deadline at the IP level can in fact be delayed or dropped at the playout buffer.

How does VoIP work? Before sending the voice across the network, VoIP digitalizes it in data packets, sends them and reconverts them to voice at destination. Why do we convert it to the digital format? Digital format can be better controlled: we can compress it, route it and convert it to a new better format, and so on. In addition, digital signals are more noise tolerant than analog ones.

Overview on a VoIP connection:

- First, we use ADC to convert analog voice to digital signals (bits) - This is made by hardware, typically by card integrated ADC.
- Now the bits have to be compressed in a good format for transmission: there are a number of protocols, for example PCM, Pulse Code Modulation, Standard ITU-T G.711. The most important demand from such protocols is to convert digital data to a standard format that could be quickly transmitted.
- Here we have to insert our voice packets in data packets using a real-time protocol. VoIP data packets are packed in RTP (Real-Time Transport Protocol) packets, which are inside UDP-IP packets. VoIP doesn't use TCP because it is too heavy for real time applications, so instead a UDP datagram is used.
- However, UDP has no control over the order in which packets arrive at the destination or how long it takes them to get there. Both of these are very important to overall voice quality and conversation quality. RTP solves the problem enabling the receiver to put the packets back into the correct order and not wait too long for packets that have either lost their way or are taking too long to arrive (we don't need every single voice packet, but we need a continuous flow of many of them and ordered).
- We need a signaling protocol to call users: ITU-T H323 does that. This protocol allows a variety of elements talking each other: terminals, clients that initialize VoIP connection, Multipoint Control Units (MCUs) to provide conference and more… This protocol allows not only VoIP but also video and data communications.
- At the receiver we have to disassemble packets, extract data, then convert it to analog voice signals and send it to sound card (or phone).
- All that must be done in a real time fashion because we cannot wait for too long for a vocal answer.

2. Background

VoIP is voice over an Internet Protocol (IP) based network. All networks will be supporting IP. There are two ways of looking at VoIP: regulatory/business and technical. We are going to address the technology. The regulatory and business perspective will provide a framework by which VoIP will be provided. However, the regulatory and business view is far too complex to discuss in a white paper. As a service, voice is a basic necessity. Despite the preponderance of email, people prefer to talk to one another rather than email one another. Declining minutes of use in the wireline network is due to the existence of wireless communications and email. As a mass market service, voice is the basic service of all services. Without voice a telecommunications service provider is not meeting the needs of all of its customers.

The Internet was not originally designed to carry audio communications. In fact the Internet protocol could not meet the exacting requirements of the voice service customer. Once an ISP is capable of providing voice it will be able to take advantage of its position as an information services provider to the user and provide all services (including voice) to the user. At one time, VoIP was provided as a best effort service just as other Internet services had been. The Internet Protocol is a "best effort" protocol.

In general there are business and technical benefits to deploying an IP network. The business benefits are:

- ❖ Reduced long distance costs
- ❖ Lower network costs
- ❖ More enhanced services – Voice over IP is just one of the services.

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The technical benefits are:

- Less bandwidth for more calls
- More efficient use of network resources
- Distributed network intelligence

The network signaling protocol of the Internet is TCP/IP. The Transmission Control Protocol/Internet Protocol (TCP/IP) protocol suite was originally used for and still is used for the internetworking of Local Area Networks (LANs). All of the signaling protocols used in the Internet are part of the TCP/IP protocol suite. The Transport layer in the TCP/IP suite is comprised of two protocols; the TCP and the UDP. TCP stands for Transmission Control Protocol. UDP stands for User Datagram Protocol. The TCP (Transmission Control Protocol) performs the transport layer functions of the Internet Protocol. The UDP (User Datagram Protocol) is a connectionless function that is normally used by database lookup applications.

Although originally designed for data services, the Internet can also support real-time traffic such as voice and video. The technology of voice over Internet Protocol (VoIP), also known as Internet telephony, IP telephony, or packet voice, enables real-time voice conversations over the Internet. It has attracted much interest from academia and industry because of the following facts:

• VoIP has much lower cost than traditional telephone service.

• The universal presence of IP makes it convenient to launch VoIP applications.

• There is increasing demand for networks to interact with end users having real-time data, voice, and video images, leading to the requirement for integrated voice, data, and video services.

• The emerging digital signal processing (DSP) and voice coding/decoding techniques make VoIP more and more mature and feasible. Therefore, VoIP is anticipated to offer a viable alternative to traditional public switched telephone network (PSTN).

To provide person-to-person (instead of place-to-place) connections anywhere and anytime, the Internet is expected to penetrate the wireless domain. One very promising wireless network is the wireless local area network (WLAN), which has shown the potential to provide high-rate data services at low cost over local area coverage. Working in the license-exempt 2.4 GHz industrial, scientific, and medical (ISM) frequency band, the IEEE 802.11b WLAN offers a data rate up to 11 Mb/s, while IEEE 802.11a WLAN and European Telecommunications Standard Institute (ETSI) HIPERLAN/2 can support data rates up to 54 Mb/s at the 5 GHz frequency band. As a wireless extension to the wired Ethernet, WLANs typically cover a small geographic area, in hotspot local areas where the traffic intensity is usually much higher than in other areas. The promising VoIP technology and wide deployment of WLANs are expected to drive the application of voice over WLAN (VoWLAN), which will experience a dramatic increase in the near future. Figure 1 shows a typical VoWLAN system where voice conversation happens through the access point (AP). At the sender, the analog voice signal is compressed and encoded by a codec. After inclusion of the Real-Time Transport Protocol (RTP)/User Datagram Protocol (UDP)/IP headers during the packetization procedure at the transport and network layers, voice packets are transmitted over the networks and finally to the receiver end. At the receiver, a playout buffer is usually used to alleviate the effect of delay jitter. Then the receiver applies depacketization and decoding to recover the original voice signal. One major challenge for VoWLAN is quality of service (QoS) provisioning. Originally designed for high-rate data traffic, WLANs may experience bandwidth inefficiency when supporting delay-sensitive and low-rate voice traffic. Hence, it is essential to enhance the QoS support capability of current WLAN standards, such as the most popular IEEE 802.11 standard.

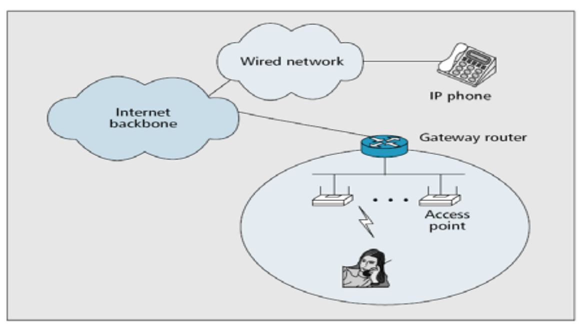

Figure 1. The architecture for VoIP over WLAN

3.Overview of 802.11 - 802.11a, 802.11b and 802.11e

802.11 -- refers to a family of specifications developed by the IEEE for wireless LAN technology. 802.11 specify an over-the-air interface between a wireless client and a base station or between two wireless clients. The IEEE accepted the specification in 1997. 802.11 defines physical and MAC layers and

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provides 1 or 2 Mbps transmission in the 2.4 GHz band using either frequency hopping spread spectrum (FHSS) or direct sequence spread spectrum (DSSS).

802.11b (also referred to as 802.11 High Rate or Wi-Fi) -- an extension to 802.11 that applies to wireless LANS. 802.11b was a 1999's ratification to the original 802.11 standard, allowing wireless functionality comparable to Ethernet. Most WLANs deployed today use 802.11b technology, which operates in the 2.4 GHz band and supports a maximum theoretical data rate of 11 Mbps, with average throughput falling in the 4 Mbps to 6 Mbps range. In a typical office environment, its maximum range is 75 meters (250 feet) at the lowest speed, but at higher speed its range is about 30 meters (100 feet). Minimizing interference can be difficult because 802.11b uses only three non-overlapping channels. 802.11b uses only DSSS. So, its advantages are: it allows multiple connections to a remote network, data transfer and mobility. But, as we can see here, 802.11b isn't perfect at all. Actually, 802.11b has three major problems: limited bandwidth, interference from other devices and also it doesn't allow any Quality of Service (QoS).

802.11a -- an extension to 802.11 that applies to wireless LANs. Operating in the 5 GHz band, 802.11a supports a maximum theoretical data rate of 54 Mbps, but more realistically it will achieve throughput somewhere between 20 Mbps to 25 Mbps in normal traffic conditions. In a typical office environment, its maximum range is 50 meters (150 feet) at the lowest speed, but at higher speed, the range is less than 25 meters (75 feet). 802.11a has four, eight, or more channels, depending on the country. 802.11a uses an orthogonal frequency division multiplexing encoding (OFDM) scheme rather than FHSS or DSSS. In general, if we want high performance and minimal radio frequency interference, then 802.11a is the way to go (but without QoS!)

4.Channel accessing mechanisms in 802.11 a and 802.11b.

The IEEE 802.11 WLAN (both a and b) have two different channel accessing mechanisms, namely, the distributed coordination function (DCF) and point coordination function (PCF). DCF is based on the carrier sense multiple access with collision avoidance (CSMA/CA) channel accessing mechanism, while PCF is based on the polling technique. The DCF operation mode consists of two techniques for packet transmission. The default scheme is a two-way handshaking technique where a positive acknowledgement is transmitted by the destination station upon successful reception of a packet from a sender station. Another scheme involves a four-way handshaking technique known as request to send/ clear to send mechanism (RTS/CTS). By this scheme, the sender first sends RTS to reserve the channel before its transmission, and upon receiving CTS from the receiver, the normal packet transmission and the ACK response proceeds. On the other hand, for the PCF operation mode, stations are polled in turn, and the station with a packet pending for transmission sends the packet upon being polled. In IEEE 802.11 networks, the DCF mode is the fundamental channel access method and coexistence between DCF and PCF is required. The period in which the system operates in PCF mode is called contention free period (CFP), while the period in which the system operates in DCF mode is called contention period (CP). Moreover, using just PCF presents the following inefficiency: If every wireless station connected to an AP are polled regardless of whether it has data to transmit or not may result in considerable polling overhead. This overhead may be reduced by maintaining a dynamic polling list at the AP. A station with data to transmit asks the AP to enroll to this list and after some idle time the AP deletes it from the list. In this sense, DCF is still needed in addition to PCF, in order to provide the stations a way to send the enrollment requests to the AP.

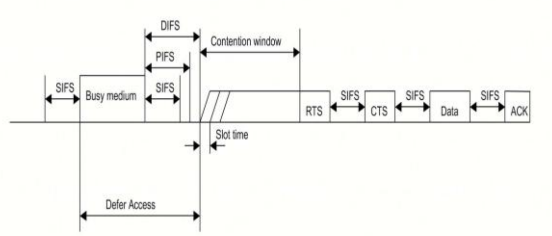

Figure 2. MAC Layer

802.11e – Is an enhanced version of 802.11, currently under IEEE development. It keeps most of technical parameters of its predecessor, but has very significant quality: it provides Quality of Service (QoS) support for LAN applications, which will be critical for

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delay-sensitive applications such as Voice over Wireless IP (VoWIP). The standard will provide classes of service with managed levels of QoS for data, voice, and video applications. It introduces the concept of hybrid coordination function (HCF) for the MAC mechanism. HCF is upward compatible DCF and PCF, in the same time providing QoS stations with prioritized and parameterized QoS access to the wireless medium.

EDCF and EPCF -- HCF provides two different means of supporting QoS. First there is the extension of the widely deployed distributed coordination function (DCF) that makes use of CSMA/CA. DCF provides coordination, but it doesn't support any type of priority access of the wireless medium. The enhanced DCF (EDCF) mechanism adds four levels of statistical access priority, enabling the separation of frames into different priority levels. Each level corresponds to an individual prioritized output queue. Each output queue contends for a transmission opportunity (TXOP). The minimal specified idle duration before starting a frame transmission (ICF – inter frame spaces) is different for each specific queue: SIFS (short IFS) is used by acknowledgement packets. PIFS (point coordination function IFS) is used by the AP to take control of the channel and start CFP. DIFS is used by data packets and so on. The backoff computation is also different for the individual queues. Contention window is increased after each collision. EDCF provides statistical priority only. It does not guarantee that low priority frames will always wait until all higher priority frames are transmitted. The second 802.11e QoS mechanism is an extension of PCF of the original 802.11 standard. This method uses a QoS-aware point coordinator, called hybrid coordinator (HC). The AP usually plays this role. The HC uses its higher channel access priority to allocate transmission rights (TXOPs) to wireless stations to transmit QoS data so that the predefined delivery priority, service rate and delay are satisfied. The wireless station may sent a TXOP request to the HC either while in EDCF mode, or during another TXOP granted to it or in a special CCI interval (controlled contention interval) when contention occurs only among QoS stations wishing to get a TXOP. During a TXOP the station may initiate multiple frame exchange sequences. This gives EPCF the flexibility to support bursty QoS traffic. EPCF inherently provides hard QoS guaranties.

802.11e allows Quality of Service (QoS), while the original 802.11a and 802.11b do not .

The proposed approach is based on adapting the values of CW depending on the channel congestion level. In IEEE 802.11e the value of CW is incremented whenever a station fails to transmit due to a collision. This would imply that when the channel is highly congested CW would acquire values distant from CWmin and close to CWmax. Similarly, when the channel is free, CW values would be close to CWmin and distant from CW_{max}. Hence, it is feasible to estimate the channel congestion level by taking into consideration the current value of CW. We use a very simple approach to estimate this level. In this approach, we start from the fact that CW value ranges in the interval [CW_{min}, CW_{max}], then we compute its relative distance ($CW_{current}$ – CW_{min}) compared to the maximum distance ($CW_{current}$ – CW_{min}) as an indication for channel congestion level. It follows that the estimated link congestion ratio in the proposed Adaptive scheme can be written as:

$$Ratio = \frac{(cw_{current} - cw_{\min})}{(cw_{\max} - cw_{\min})}$$

In the proposed scheme, the ratio is weighted as follows.

$$ratio = weight \times \frac{(cw_{current} - cw_{\min})}{(cw_{\max} - cw_{\min})}$$

For instance, the weight of the ratio would be very small if current channel estimate is used in a transmission that occurred several minutes ago. However the ratio would be highly weighted if the difference in time between estimation and transmission is of the order of milliseconds. To obtain some preliminary simulation results, the weight was fixed in this paper to a value of 0.9. Indeed, the weight converged to this value after several tests. This is due to fact that video streaming is characterized by transmission occurring at very small time intervals

The CW value of the proposed adaptive scheme, CW_{new} can be given then as follows

$$cw_{new} = weight \times \frac{(cw_{current} - cw_{\min})^2}{(cw_{\max} - cw_{\min})} + cw_{\min}$$

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The ratio is a normalized value ranging from 0 to 1 that reflects the weighted degree of channel contention. This ratio would take a value close to 0 whenever the channel is free. Therefore CWcurrent would have a value close to CW_{min} and distant from CW_{max}. The value of this ratio would be close to 1 whenever the channel is congested. Therefore CWcurrent would have a value distant from CWmin and close to CW_{max}. Multiplying this ratio by the factor ($CW_{current} - CW_{min}$) and adding the result to CW_{min} would result in a value bounded by [CW_{min}, CW_{max}]. This value of CWnew would be a good representation of the backoff timer value needed for transmission for the current traffic priority taking into account the current network conditions.

5. Simulation scenario

The simulation topology of this scenario is simple. It consists of 8 mobile nodes: 4 source nodes and 4 destination nodes. Each node is transmitting with a different priority. Node 1 is given a higher priority than Node 2, which is given also a higher priority than Node 3. Node 3, in its turn, is given a higher priority than Node 4. Each source is a Constant Bit Rate source over UDP (User Datagram Protocol). The size of a transmitted packet is 512 bytes. Transmission rate of a node is 600Kbps. We assumed that the nodes are in transmission range at a constant distance of 195 m. The simulation time lasted for 80 sec.

To model voice traffic the simulations use ITU-T G.729 standard. G.729 is supported widely in VoIP products. In G.729, the voice is encoded at the rate of 8 kbps and with 20 or 40 bytes payload size in a packet. The voice quality can be degraded compared to another widely used standard, G.711, because the compression in G.729 can be lossy. However G.729 requires less bandwidth. The payload size is 20 bytes. With packet overhead, the rate required is 26.4 kbits/s

.6. Simulation results

In this section we present simulation results, which are meant as a proof of concept of how the contributed simulation framework can be exploited for a sound and simple performance evaluation of VoIP applications in ns-2. We therefore purposefully set up a very simple networking environment.

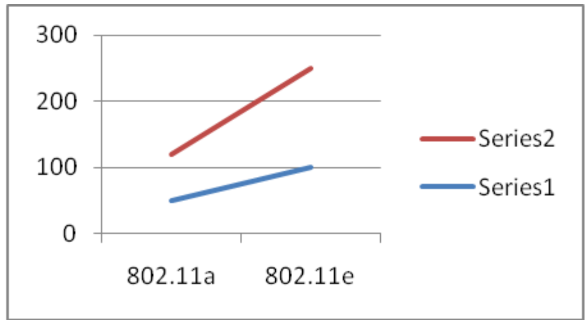

Figure 4.packet transfer rate

Bit rate

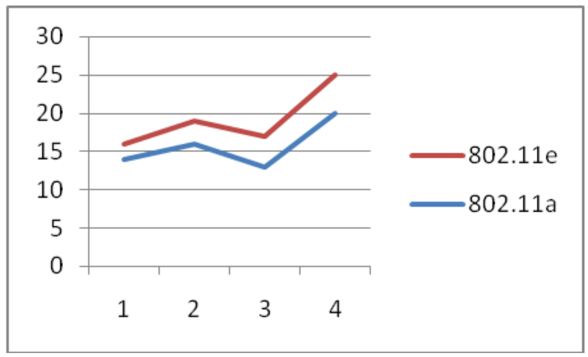

Figure 4.packet loss rate

7. Conclusion

In this project the performance of the IEEE 802.11a and IEEE 802.11e systems have been evaluated. We have also proposed a new adaptive differentiation technique for resetting the value of the contention window after each successful transmission. The proposed adaptive technique takes into account the current level of link utilization when resetting such value. We have performed several simulations, for different Scenarios, using NS-2, to evaluate the proposed technique compared to IEEE 802.11a and IEEE 802.11e. During the evaluation, we have focused on three parameters: bit rate, end-to- end packet delay. And the results reveal that the proposed traffic gives better results than IEEE802.11a.

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