Quality of Service (QoS) of Voice over MAC Protocol 802.11 using NS-2

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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. Voice over IP (VoIP) is gaining an ever increasing popularity. As such, it nowadays represents one of the most studied test applications in the performance evaluation of wireline and wireless networks in which the IEEE 802.11 distributed coordination function protocol or enhanced DCF protocol is used. However, since both DCF and EDCF are contention-based medium access control protocols, it is difficult for them to support the strict QoS requirement for VoIP.

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 and Show that Uplink/Downlink problem exists in EDCF and suggest different solutions using QoS parameters (DIFS, CWmin and PF).

Keywords

NS-2, QoS, VOIP, IEEE 802.11, MAC Protocol.

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. On the other hand, playout buffers dampen the jitter, so that evaluating the jitter at the IP level (rather than after the playout buffer) often verestimates it. Let us consider, for instance, the works presented at the 26th IEEE International Conference on Computer Communications (IEEE INFOCOM 2007), which are focused on the performance analysis of VoIP traffic.

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:

1) First, we use ADC to convert analog voice to digital signals (bits) - This is made by hardware, typically by card integrated ADC.
2) 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.
3) 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.
4) 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).
5) 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.
6) 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).
7) All that must be done in a real time fashion because we cannot wait for too long for a vocal answer.

2. Background

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.

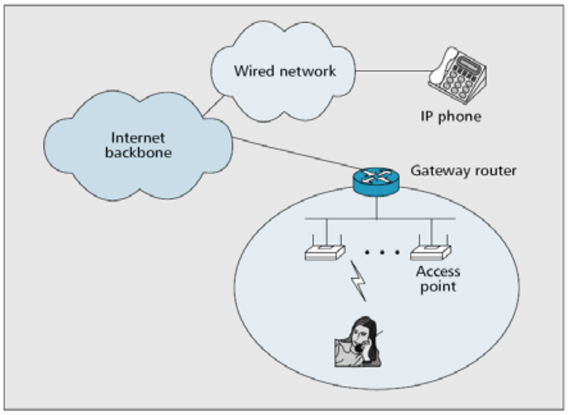


Figure 1. The architecture for VoIP over WLAN.

3. Network Simulator (Ns-2)

The Network Simulator (ns-2) is the de-facto standard simulation tool for the networking community, as shown by recent surveys led on the proceedings of important networking conferences. Ns-2 is an open-source simulator, continuously enhanced and extended thanks to the contribution of a large community of researchers. It includes a large number of network protocols, applications, algorithms, devised for wired and wireless networks. A new version of the simulator (known as ns-3) is currently being developed at the time of writing. However, ns-2 lacks a sound and flexible simulation model of a VoIP application, as well as routines for processing events so as to evaluate VoIP performance according to the Emodel. The contribution of this paper is therefore a software framework that enhances ns-2, thus allowing a sound simulation and performance evaluation of VoIP applications. The simulator is based on two languages: an object oriented simulator written in C++ and an Otcl (an object oriented extension of Tcl) interpreter used to execute user's command scripts. In these scripts, the user can define a particular network topology, the specific protocols and applications that he wishes to simulate and the form of output he wishes to obtain from the simulator. NS is a discrete event based simulator. The Tcl script defines when an event should occur. For example: $ns at 1.0 "$ftp start". When $ns is a Simulator instance (an event scheduler) and $ftp is a FTP application that was set above lower layers (TCP, physical network – topology, delays, etc.) earlier defined by the user in the script. The simulator can produce both the visualization trace (Nam) as well as an ASCII file trace corresponding to the events registered at the network. In addition, the user can define a custom trace format to track system parameters that interest him. For example the queues at the input node of links or the window sizes of tcp at some node. It is also possible to make computations in place and output them to the trace file with the data derived from the simulated objects. Random variables with different distribution can be created in NS. It is also possible to generate the same random sequence of random numbers in different runs in order to generate the same behavior. For example, it becomes of interest to choose connection's parameters (such as time of beginning or end of activity) at a random way. For example, a typical distribution for generating packets with different sizes is Pareto. The distribution interarrival times of new connections is frequently taken to be exponential.

After tracing the simulation the user can use tools such as awk or perl to process the output files and then gnu plot or Xgraph to show graphs of the interesting parameters.

<u>Interesting parameters for comparison may be:</u>

- When using PCF, influence of alternations of CP/CFP time on end to end delay and throughput.
- Influence of the addition of QoS (HCF) on the number of possible simultaneous VoIP connections, end to end delay and throughput.
- End to end delay and throughput in constant bit rate (CBR) VoIP traffic versus variable bit rate (VBR) VoIP traffic.
- Effect of VoIP traffic burstiness on the average end to end delay.
- Throughput of VoIP connections under heavy load with and without non-VoIP connections (possibly with different network configuration – few APs etc…)
- Throughput of VoIP connections without heavy load
- Influence of packet size on throughput
- Influence of different voice encoding techniques – G711, G729 and G723.1.
- Performance under "bad conditions" – environment with high error rate (burst)
- Impact of number of VoIP connections on throughput, delay.

The parameters actually chosen are highly dependent on the nature of simulator extensions given by Alvarion.

4. 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 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!)

DCF and PCF -- 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.

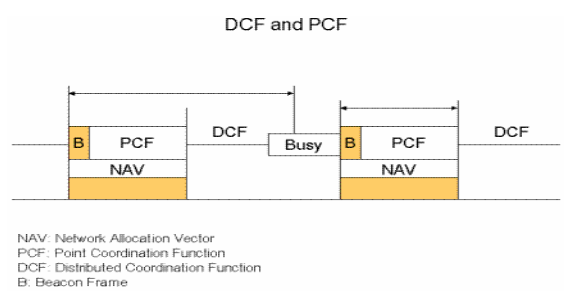


Figure 2: DCF and PCF.

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 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.

<u>EDCF and EPCF</u> -- 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. (See Fig. 3) 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.

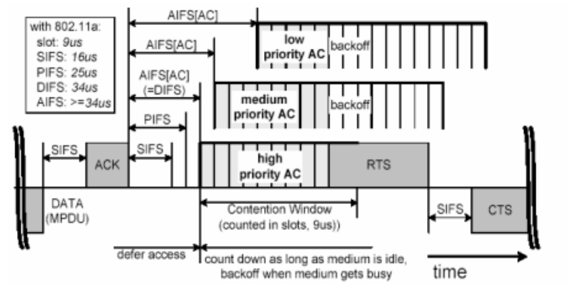


Figure 3: EDCF.

5. Comparison between 802.11a and 802.11b

- By far the top reason for choosing 802.11a is the need to support higher end applications involving video, voice and the transmission of large images and files. In addition, 802.11a does a superior job of supporting densely populated areas of users having lower bandwidth needs, such as surfing the Internet. 802.11a can deliver data rates up to 54Mbps and there's enough room in the 5GHz spectrum to support up to 12 access points operating in the same area without causing interference between access points. This equates to 432Mbps (12 X 54Mbps) total data rate performance. 802.11b doesn't come close to the performance of 802.11a. There are only three non-overlapping channels for setting access point frequencies, which severely limits capacity. So, the major advantage of 802.11a compared to 802.11b includes higher throughput rates and increased channel support.
- Like Ethernet and Fast Ethernet, 802.11b and 802.11a use an identical MAC (Media Access Control). However, while Fast Ethernet uses the same physical-layer encoding scheme as Ethernet (only faster), 802.11a uses an entirely different encoding scheme, called OFDM (orthogonal frequency division multiplexing). OFDM is not a form of spread spectrum. Instead, OFDM divides a data signal across 48 separate sub-carriers within a 20MHz channel to provide transmissions of 6, 9, 12, 18, 24, 36, 48, or 54Mbps. Data rates of 6Mbps, 12Mbps, and 24Mbps are mandatory for all 802.11-compliant products. OFDM is extremely efficient, which enables it to provide the higher data rates and minimize multi-path propagation problems.
- 802.11a doesn't talk to 802.11b. Because 802.11a and 802.11b operate in different frequencies, there's no chance they'll be interoperable. The 802.11a and 802.11b technologies can coexist, however, because there is no signal overlap. The 802.11 standard offers

no provisions for interoperability between the different physical layers. The solution to this problem is multimode radio cards that support multiple 802.11 PHYs, such as 802.11a/b. As a result, an 802.11a/b radio within an end user device will automatically sense whether the access point is 802.11a or 802.11b and then communicate accordingly.

- A very significant detail is, that 802.11e allows Quality of Service (QoS), while the original 802.11a and 802.11b do not.

6. Simulated Network

Actually, all wireline networks are not simulated, since it is not interesting for our goal. Instead we let the access point to produce packets on all connections that go from wireline to wireless.

Setting Connections

- Each wireless station has a different wireline station it "talks" to.
- There are N + 4 connections: half from a wireless node to a wireline node and half the other way (the connections are in one-direction).
- There are N voice connections. The parameters we investigate are taken only from these connections.
- Voice Connections: - CBR over UDP.

\- Packet size (with all overhead): 180 bytes as in G711 voice codec.

\- Packet inter-arrival time – 20ms

- Best Effort Connections: DATA - FTP over TCP.

\- Packet size (with all overhead) : 1560 bytes

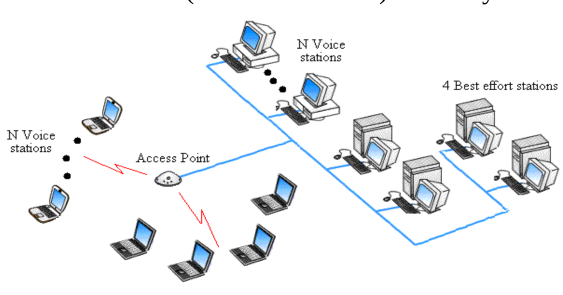
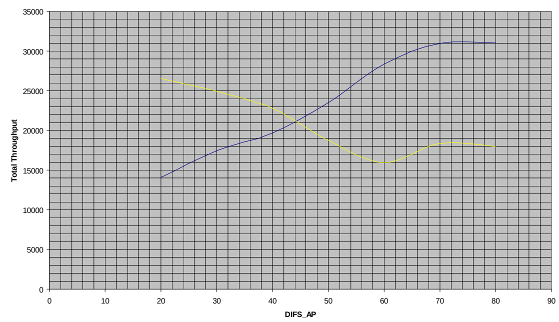


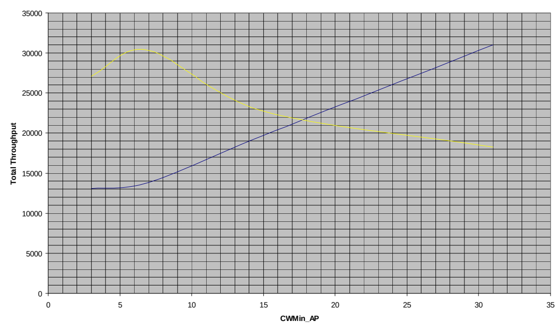
Figure 4. Simulated Network.

7. Performance Evaluation

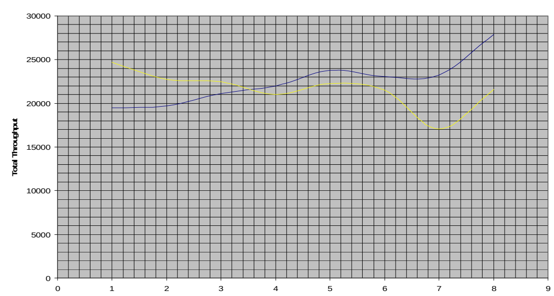
In this section we present simulation results, which are meant as a proof of concept of how the contributed simulation framework can be exploited for a sound and simple performance evaluation of VoIP applications in ns-2. We therefore purposefully set up a very simple networking environment.



Uplink/Downlink Throughput as function of DIFS



Uplink/Downlink Throughput as function of CWMin



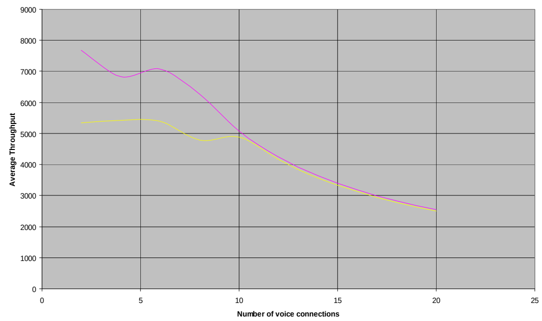
Uplink/Downlink Throughput as function of PF

Figure 5. Simulated Results for Uplink/Downlink Throughput

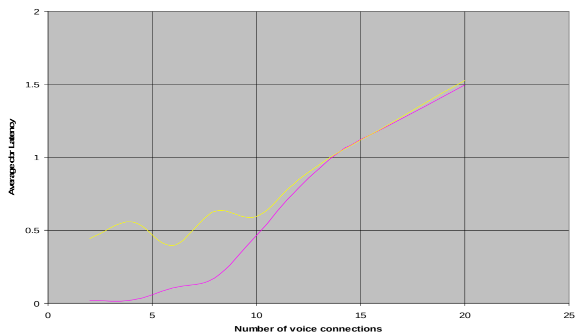
Other Parameters:
PF of AP = 2, DIFS of AP = 40, CWMin of AP = 15
DIFS of Voice = 50, CWMin of Voice = 31, PF of Voice = 2
DIFS of BG = 80, CWMin of BG = 63, PF of BG = 2
CWMax – 1023,

---------------- **Uplink Throughput**

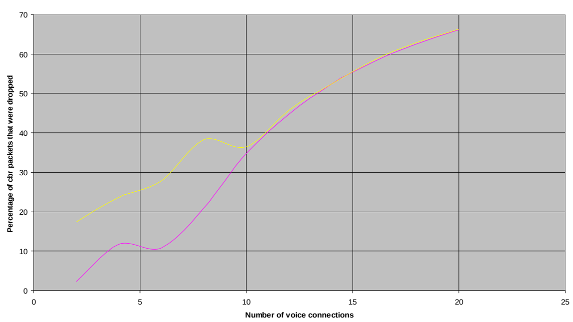
---------------- **Downlink Throughput**



Average Throughput as function of VOIP Calls



Average cbr Latency as function of VOIP Calls



Percentage of dropped cbr packets as function of VOIP Calls

Figure 6. Simulated Results.

Other Parameters:
Voice: DIFS - 0.000020, CWMin - 15, PF - 2
Data: DIFS - 0.000080, CWMin - 63, PF – 2
CWMin of 802.11b – 63, CWMax – 1023

--------------- **802.11e**
--------------- **802.11b**

8. Conclusions

In this paper we presented extensions to ns-2 that allow a sound and efficient simulation of VoIP applications. Access Point should **dynamically** adjust its medium access parameters (DIFS, CWmin and PF) depending on number of mobile stations and network load. It should aspire to get at least equal service as all the mobile stations it serves (together).

We have shown that as the number of stations (and thus network load) increases, the AP has to get higher priority (lower QoS parameters) to reach the uplink = downlink point. AP should also consider the type of data (priority classes) that passes through it when adjusting its own medium access parameters. When adjusting AP’s priority, it is better to use DIFS and CWMin rather than PF, and when PF is used, it should be accompanied with changes in CWMin, since sometimes the changes PF inflicts are too strong and not very predictable unlike the other two parameters.

The Voice class, to which we gave lower parameters (higher priority), got better service than in 802.11b => QoS. The improvement of service to Voice could be clearly seen when number of voice connections is relatively small (up to 3 times more than number of background connections). When the number of voice connection becomes large, the performance of 802.11b and 802.11e (EDCF) is very much alike, since giving priority has meaning only when there are many connections with lower priority. If there are many voice connections, they compete mainly among them and not so much with the Data connections. Channel capacity limits the number of possible connections through the channel. The channel contention method doesn’t help either.

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