

Mandating QoS in Wireless LANs

Introduction

Tripleplay – Data, voice and video services over an integrated or converged network is much talked about these days. However, most existing networks have been designed with data applications in mind. Data applications are non-real time. It makes little difference to the user if the data is minimally delayed, or is received with varying latencies, or was dropped due to congestion or corruption. A frame which has been dropped can be retransmitted later. Similarly, the wireless LAN MAC protocol defined in the standard IEEE 802.11 was designed for data networks. The basic access mechanism remained the same in the 802.11a/b/g protocol, which is also referred to as the legacy 802.11 protocol.

Networks designed for non-real time traffic, like data, are today being used to support real-time applications like VOIP¹ (or VoWLAN²) and streaming video, which are inherently different from data traffic. Real-time applications have very different requirements and characteristics compared to data traffic. *Packet loss* affects the quality of the voice or the video and degrades the user experience. Minimum *delay* is an important requirement, as delayed frames are as good as dropped frames for real-time traffic. *Jitter* is another important parameter, since it indicates the variation in the delay and requires buffering to smooth the jitter. Finally, *throughput* or bandwidth requirements have to be kept in mind; unlike data which can use the available bandwidth at any instant, real-time applications require guaranteed and periodic bandwidth. The legacy 802.11 protocol was enhanced with various Quality of Service (QoS) mechanisms to serve the requirements of real-time traffic like voice and video.

Characteristics of Voice and Video traffic

Voice traffic is characterized with small packet size (from 80 to 256 bytes), frame periodicity leading to deterministic or constant bit rate traffic (21 to 320 Kbps).

There are three key parameters in which voice differs from data traffic: low tolerance for delay, even lower tolerance for jitter and low tolerance for packet loss. These three parameters influence the two factors which determine the user experience for voice traffic: Voice Delay and Voice Clarity. Voice Delay leads to many problems. When there is an imperceptible time lag between the speaker and the listener, conversations can appear “cold” or uncomfortable. When delay is large, the rhythm of conversations is disrupted, resulting in excessive interruptions and misunderstandings. A delay of 100-150 ms is considered acceptable for conversation. However, delay above 200 ms is detectable by humans and can adversely affect the conversation quality. Voice Clarity determines the quality of the voice signal, and is affected by packet loss and jitter. Compared to jitter which can totally destroy voice quality, voice is relatively tolerant to errors or loss of frames

¹ Voice Over Internet Protocol

² Voice Over Wireless LAN

since the algorithms used can hide the packet loss (PLC³). The average one-way jitter should be less than 30ms.

Streaming video traffic has a highly variable frame size which is much larger than voice. Packets are fairly large in size, ranging from 65 bytes to 1500 bytes and the traffic is usually Variable Bit Rate. There are two key parameters in which video traffic differ from data traffic: low tolerance for delay and packet loss. Compared to voice traffic, streaming video applications require higher bandwidth (depending on the encoding protocol, between 10s of Kbps to 10s of Mbps), but are not highly delay or jitter sensitive. They can buffer 4-5 seconds of traffic which can smoothen out jittery traffic considerably. They can also tolerate more losses than voice traffic.

QoS is essential for 802.11

For voice or video traffic to be successful over the WLAN networks, Quality of Service measures are required to solve the problems of *delay*, *jitter* and *packet loss*. Fortunately, the IEEE 802.11e specification provides for many mechanisms to deal with these.

The WLAN QoS consists of two parts: EDCA and HCCA. EDCA is an enhancement to the DCF⁴ protocol mentioned earlier, while HCCA is an enhancement to the erstwhile PCF⁵ protocol. EDCA prioritizes between different classes of traffic using queues called Access Categories (AC).

In the legacy 802.11 protocol, all traffic is treated with equal priority. Hence, a voice frame can be stuck behind some large video or data frame and get delayed until the other frames are delivered. EDCA overcomes this problem by defining different traffic classes and prioritizing between them using queues called Access Categories (AC). In this scheme, voice is given higher priority than video, which in turn is given high priority than data. Hence, voice gets faster access to the medium by contenting for shorter durations than video. Similarly, video gets faster access to the medium than data. Congestion in one traffic class does not affect any other traffic classes, and hence real-time data does not get affected by congestion in non-real time traffic.

The EDCA access mechanism increases the protocol efficiency using a feature called *frame bursting*. It allows each class of traffic to have control of the medium for varying periods of times (called *Transmit Opportunity*). During a Transmit Opportunity, a particular AC can transmit frames one after the other without contending for the medium between frames. The Transmit Opportunities are allocated so that video traffic gets the highest amount of bandwidth, followed by voice and finally by data.

EDCA also allows for minimal guarantee of bandwidth. It supports *admission control* mechanisms, which controls the amount of bandwidth available to a particular access category. Admission control is extremely important to guarantee the QoS of the network by disallowing a particular AC to use the medium when the

³ Packet Loss Concealment: an algorithm used to mask the effect of lost or discarded frames.

⁴ Distribution Coordination Functions

⁵ Point Coordination Function

allocated bandwidth for that priority traffic is exhausted. This feature is useful when multiple users use the network for high priority access like voice and video. It control the bandwidth allocated to each user. The admission control mechanisms are particularly useful in guaranteeing the QoS of the network by denying access to the medium to new users when the existing capacity of the channel is used up.

EDCA mechanisms can minimize the problem of delay and guaranteed bandwidth, but does not guarantee minimal jitter and periodic bandwidth. HCCA deals with parameterized traffic flows and offers the perfect mechanism to deal with jitter and periodicity, in addition to delay and guaranteed bandwidth. It is a TDM⁶-like scheme and makes use of sophisticated scheduling to minimize jitter and guarantee periodic bandwidth for real-time applications. It also increases the protocol efficiency by removing the time spent in contending for the medium. Each application registers its bandwidth and polling interval requirements with the central scheduler at the Access Point. Strict admission control protocol ensure that users do no exceed the available bandwidth of the medium. Once a user traffic flow is admitted by the scheduler, it exercises fine grained control over the bandwidth, jitter and periodicity of the traffic.

In addition to enhanced channel access mechanisms, 802.11e also provides for a block acknowledgement scheme (similar to the TCP/IP Sliding Window Protocol) where the legacy 802.11 per-frame acknowledgement⁷ is removed, and a consolidated acknowledgement frame is transmitted to indicate the reception status of multiple frames. It also provides for an acknowledgement-less scheme, which increases the utilization of the medium and is more useful for real-time data that do not require retransmission on failure. Since the per-frame acknowledgement is removed in both these schemes, the effective throughput of the channel increases and allows more voice and video applications to be used simultaneously.

Unlike the legacy protocol, it is possible to exchange frames directly between devices without passing them through the AP. This feature is called *sidelinks*, or DLS. Sidelinks are very useful for high bandwidth video streams. A 10 Mbps video stream being exchanged between two devices actually use 20 Mbps of medium bandwidth. Using the DLS⁸ protocol, the effective bandwidth utilization for the video stream is only 10 Mbps.

Finally, power save techniques are of paramount importance since they allow for increased standby time in handsets. Since a handset is in standby mode 95% of the time, the QoS power save mechanism like APSD⁹ helps increase power efficiency drastically as compared to legacy power save modes.

Enabling 802.11n using QoS mechanisms

The existing mechanisms of 802.11e are being reused in the 802.11n proposals being offered by the various groups in the 802.11n Working Group (WG). The

⁶ Time Division Multiplexing, where each device is given a slice of the air time for data transfer.

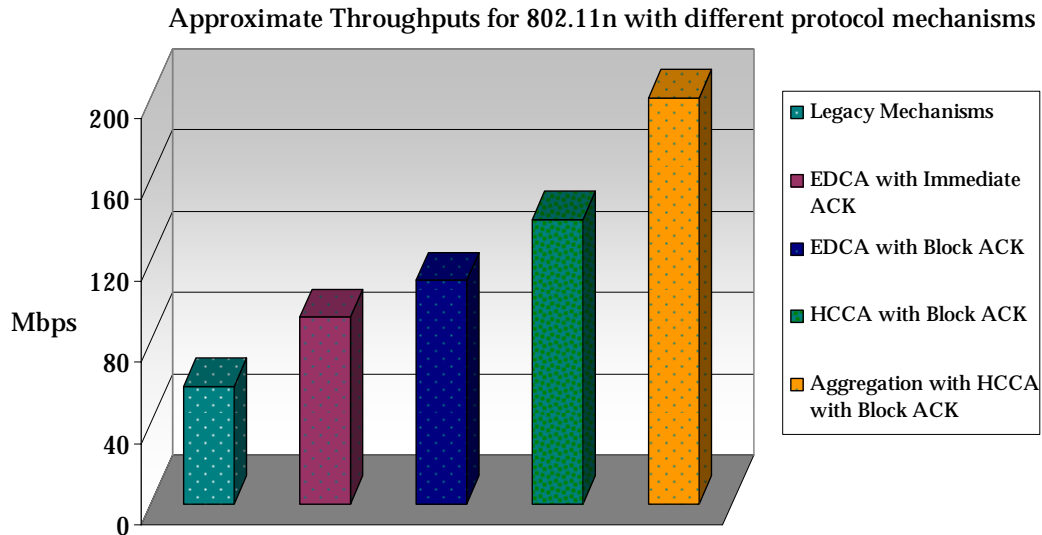
⁷ Called Immediate Acknowledgement Policy

⁸ Direct Link Set-up

⁹ Automatic Power Save Delivery, a power save technique using in 802.11e.

802.11n protocol essentially deals with goodput¹⁰ above 100 Mbps. The high throughput of 802.11n can only be achieved by leveraging the mechanisms that are used in the 802.11e standard. The QoS mechanisms make the protocol more efficient by getting rid of some of the protocol overheads.

Assuming an 802.11n PHY which is capable of transmitting data at 240 Mbps:



Legacy mechanisms can achieve a maximum goodput of ~58 Mbps using a frame size of 1500 bytes.

802.11e EDCA mechanisms with immediate acknowledgement policy¹¹ can achieve a maximum goodput of ~92 Mbps using a frame size of 1500 bytes.

802.11e EDCA mechanisms with block acknowledgement scheme can achieve a maximum goodput of ~110 Mbps using a frame size of 1500 bytes.

802.11e HCCA mechanisms with block acknowledgement scheme can achieve a maximum goodput of ~140 Mbps using a frame size of 1500 bytes.

By using standard QoS mechanisms with an 802.11n capable PHY operating at 240 Mbps, the goodput is raised from 58 Mbps to 140 Mbps, and the MAC protocol efficiency is raised from 24% to 58%.

We can take this illustration one step further and compute the goodput using the HCCA mechanism described above, but this time with a feature of 802.11n MACs called *frame aggregation*.

Thus, using 802.11e HCCA mechanisms, block acknowledgement scheme and frame aggregation, the maximum throughput that can be achieved using a

¹⁰ Throughput at the MAC-SAP, or the user throughput.

¹¹ Per-frame acknowledgement response.

aggregated frame size of 6000 bytes is ~200 Mbps! Therefore a MAC protocol efficiency of 80% can be achieved by using a combination of 802.11e and 802.11n.

In addition to the increase in goodput and efficiency, other advantages of different QoS mechanisms have also prevailed in 802.11n.

Current Industry Trends

The 802.11e specification has been ratified recently and turned into a standard, after many years of work by the members of the IEEE 802.11e WG. Variants of the two 802.11e mechanisms are being offered as specifications by the Wi-Fi Alliance. These are called WMM (Wireless Multi Media) and WMM-SA (Wireless Multi Media Scheduled Access) corresponding to EDCA and HCCA mechanisms respectively of the IEEE 802.11e standard. The Wi-Fi Alliance offers certification for WMM. Many vendors have got their products certified for WMM, and these devices are today available in the market and are beginning to gain acceptance by users. The Wi-Fi Alliance is currently performing plug-fests between WMM-SA implementations of various vendors, and will soon begin to offer WMM-SA certification. However, it remains to be seen whether WMM-SA capable devices will be successful in the market.

The 802.11n protocol is in its very early stages of development. There are competing proposals before the IEEE 802.11n WG, supported by two groups, TGnSync and Wwise. A few months ago the two groups decided to come up with a joint proposal and present it as the first draft to the 802.11n WG. Meanwhile, a consortium of vendors called EWC has released their version of the 802.11n specification with features from the TGnSync and Wwise proposals. Current expectation in the 802.11n community is that IEEE can complete the standard only in 2007.

Irrespective of how the final proposal shapes up, there are certain features that are understood to be a basic minimum for an 802.11n system:

1. *Frame aggregation*, a mechanism to join together smaller frames and create a single jumbo frame
2. *Block acknowledgement*, a mechanism where a single consolidated acknowledgement frame is returned from the receiver to the transmitter indicating the reception status of multiple frames
3. *MIMO - Multiple In Multiple Out*, a technology using multiple antennas between the transmitter and receiver in order to take advantage of multi-path signals.
4. *40 MHz channelization*, a feature in which two adjacent 20 MHz channels are used to provide a wider bandwidth.

The raw throughput for the final 802.11n standard is expected to be in the region of 600 Mbps. Assuming a MAC efficiency of 80%, the goodput is expected to be around 480 Mbps. This puts 802.11n in direct competition with PAN protocols like UWB¹² in the home connectivity segment.

¹² Ultra Wide Band

The 802.11n standard is expected to open up a new market for faster wireless connectivity in enterprises and hotspots, as well as personal networking and multimedia based applications like video phones, HDTVs with wireless connections, etc.

About the Author

Amit Bansal is a HW Architect with the Semiconductor Intellectual Property group of Wipro. His domain of expertise is wireless protocols. He has designed system solutions around the 802.11 suite of protocols for the last few years. He attends the IEEE 802.11 working group meetings and has been closely following all the WLAN standards. He can be reached at amit.bansal@wipro.com.



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