

An Intelligent Method for Bandwidth Utilization in IEEE 802.11e Wireless LAN

S. Sarmah, S.K. Sarma, M. Rahman



Abstract: IEEE 802.11 Wireless LAN, popularly known as WiFi, has become the admired source of internet connectivity for most of the offices as well as organizations. Due to the rapid growth of multimedia data and VoIP and also to provide better quality of service (QoS), bandwidth and management of bandwidth have become important factors in 802.11 wireless LAN. In 802.11 wireless LAN, Enhanced Distributed Coordination Access (EDCA) mechanism provides QoS with service differentiation by setting different priority to different traffic types. Some of the available methods provide priority based on user category also. In this paper we have proposed an intelligent bandwidth management technique based on the user priority value, traffic priority value and the bandwidth requirement value. By combining these three values, a new metric called request weight value is calculated. Based on this request weight value, dynamically bandwidth is allocated to the users. From the simulation results, we have seen that the proposed method performs well in proper utilization of bandwidth among the available users.

Keywords: Bandwidth, IEEE 802.11, Traffic sent, Traffic received, Load, Throughput, Delay.

I. INTRODUCTION

Due to increasing popularity and demand of VoIP and multimedia data, bandwidth management has become a complex and serious concern for most of the available network. IEEE 802.11 Wireless LAN has emerged as a default internet technology for most of the end users and therefore management of available bandwidth in wireless network is also an important issue. In recent years, many bandwidth management algorithms have been proposed by many researchers and their main goal is to achieve the better quality of Service (QoS) by managing the available bandwidth resources. In 802.11 wireless LAN, admission control mechanism also plays an important role by restricting admission of the users based on the available bandwidth. For proper managing and utilization of the available bandwidth, load balancing mechanism can be used. By using the load balancing technique, load can be distributed among the available access points. In 802.11 wireless LAN, EDCA categories the traffic in to four access categories. For better quality of service, different priority has been set for different access category in EDCA [1] [2].

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In IEEE 802.11 wireless LAN, we have a bandwidth management technique based on user priority without considering the traffic or application. For every user some priority value is assigned.

This bandwidth management technique can provide service to all the users depending upon the priority value assigned to the users. In that method, three types of users are considered: high priority, medium priority and the low priority. Based on the priority value, access priority is given to a user. From the analysis, it is seen that the high priority users always get the priority irrespective of the traffic type whereas lower priority users are always deprived from the service. To get rid of this problem, a dynamic bandwidth management system has been proposed in this paper. The proposed method allocates the bandwidth to the users based on the request weight value. The request weight value is calculated from the user priority value, traffic priority value and bandwidth requirement. Bandwidth requirement can be calculated for each of the traffic using some formula. In the following section we have discussed how bandwidth consumption is calculated for various applications. In section III, we have discussed the bandwidth calculation formula, in section IV, proposed method for bandwidth management has been explained, in section V, experiment and results are given and finally in section VI, we have provided the concluding remarks of the work.

II. CALCULATING BANDWIDTH CONSUMPTIONS FOR VARIOUS APPLICATIONS

Bandwidth is defined as the capacity to transfer data from one station to another station in a given amount of time. Bandwidth is usually measured in terms bits per second (bps) or byte per second (Bps). It is the carrying capacity of a particular network. There are two ways to calculate the bandwidth requirement.

- First we have to calculate the available bandwidth of a particular network.
- Secondly we have to calculate the bandwidth required for a particular traffic or application.

For calculating the available bandwidth and the link capacity two methods are applied by measuring the network characteristics. The methods are: active method and the passive method. Passive measurement technique observes the network internally and it will not hamper the other traffics. This method needs the access to all the access point, routers and other servers for some administrative purpose.

There are some tools available through which we can measure the bandwidth. MRTG [3] is one of the tools that can be used in passive measurement. This tool calculates the traffic load of all outgoing and incoming link to a router.



If we can calculate the accurate load of a link and link capacity is known we can calculate the available bandwidth. IPMON [4] is another technique of passive method which gathers packet traces at different points of network.

The packet traces can be used for analysis of different behaviours such as packet size, protocol type etc which helps to calculate the available bandwidth. In active measurement method, some kind of probe traffic is injected in to the network at the source of the traffic and determines the statistics at the receiver. This active method accesses both the source and receiver in end to end fashion and also affects the other internal traffic. By using probing technique, information about the capacity of the link and available bandwidth is gained. Now a days we have some available tools through which we can compute the total bandwidth of a network. LinkWidth is a method through which we can calculate the link capacity and the available bandwidth [5]. LinkWidth method also uses probe packet between two ends for gathering the information.

For computing the bandwidth requirement for a particular traffic or application, we have to calculate the bandwidth separately for each of the traffic. IEEE 802.11 Enhanced Distributed Coordination Channel Access (EDCA) provides service differentiation by classifying the traffic in to four access categories. They are Voice, Video, best effort and the back ground traffic. The bandwidth requirement for each of the traffic is different. There are some factors based on which we can calculate the bandwidth requirement for a particular application. Following are the factors based on which we can calculate the bandwidth requirement for Voice over IP (VoIP) [6][9][10].

- Codec. The different codec are: G.711, G.729, G.723, and G.722.
- Sample Period
- IP Header
- Header Compression, which is known known as Compressed RTP (cRTP – Compressed Real-Time Transmission Protocol – RTP + UDP + IP)
- Transmission Medium, and layer two protocol such as point-to-point protocol (PPP), Frame Relay, Ethernet.
- Silence Suppression

A. Codec

A codec is a computer program that encodes or decodes a digital data stream or signal. It also compresses data which can help in faster transmission and decompresses received data. Codec can be used in videoconferencing, streaming media, and VoIP applications. Digital Signal Processor (DSP) is the hardware device through which codec is performed. The main job of the codec is to convert analog waveforms into digital waveforms and it is constructed in terms of packet and transmitted as an IP packet. The codec samples the analog wave form typically at 8,000 times per second for a fixed time period to create a frame of data. The time period is usually 10ms, 20ms, or 30ms, and depends on the codec deployed. There are various types of codec techniques [7][9].

- G.711- It is a Pulse Code Modulation (PCM) based codec which sends 8000 samples per second rate and 8 bits per sample, i.e. PCM generates 64,000 bits per second, or 64 Kbps. In this codec no any compression is performed.

- G.726- It is based on adaptive differential pulse code modulation (ADPCM). This codec does not send 8 bits per sample like G.71, instead it sends 4, 3 and 2 bits per sample. It generates 32 Kbps, 24 Kbps, or 16 Kbps respectively.
- G.722- It is a wideband speech encoding standard. This codec divides the input signal into two subbands and both the subbands are encoded using ADPCM. G.722 supports a bit rate of 64 Kbps, 56 Kbps, or 48 Kbps.
- G.728- It is based on low delay code excited linear prediction (LDCELP). G.728 codec only supports 16 Kbps.
- G.729- It is based on conjugate structure algebraic code excited linear prediction (CS-ACELP). It is a popular Cisco voice-enabled device to put two digital voice samples in one IP packet. It supports 8 Kbps bit rate.

B. Sample Period

For better quality, sample period should be less. Longer sample period produces higher latency which can affect the quality of traffic.

C. IP Header

An IP header comprised of three fields: Internet Protocol (IP), User datagram Protocol (UDP) and Real-Time transport Protocol (RTP). The Size of IP is 20 bytes which is responsible for delivering the data at the destination in a proper order. The size of UDP is 8 bytes and is responsible for routing. The size of RTP is 12 bytes and RTP is responsible for reconstructing the sample accurately and provides a technique for estimating delay and jitter with the help the RTCP (Real-Time Transport Control Protocol).

D. Header Compression

By compressing the IP Overhead with compressed Real-Time Transport Protocol (cRTP), the 40 byte IP overhead is reduced to just 2 or 4 bytes.

E. Layer 2 Overhead

When a packet moves from upper layer to layer 2 of OSI model, it wraps the layer 3 IP packet with layer 2. Layer 2 can be Ethernet, ATM, Frame Relay, PPP, VPN or other transmission types. For each type of layer 2, overhead is different.

F. Silent Compression

It is also called VAD (Voice Activity Detection) which suppresses the transmission of data during times of silence when one person is talking at a time. Silence suppressions reduce the bandwidth consumption from 30% to 50%.

III. BANDWIDTH CALCULATION FORMULA

For calculating the bandwidth, we have to know about codec of the application, codec bit rate and voice payload. Packet per second (PPS) can be calculated from the codec bit rate and voice payload size. The following table shows the codec bit rate, Voice Payload, IP overhead and the PPS for each of the codec [12][13].



Table 1. Packet Size and Packet Rate Variation for Different Codec

| Codec and Packetization Period | Codec Bandwidth (Kbps) | Voice Payload (Bytes) | IP Overhead (Bytes) | PPS |
|---------------------------------------|------------------------|-----------------------|---------------------|-------|
| G.711 with 20-ms packetization period | 64 | 160 | 40 | 50 |
| G.711 with 30-ms packetization period | 64 | 240 | 40 | 33.33 |
| G.729 with 20-ms packetization period | 8 | 20 | 40 | 50 |
| G.729 with 40-ms packetization period | 8 | 40 | 40 | 25 |
| G.711 with 30-ms packetization period | 64 | 240 | 40 | 33.33 |

A. Calculation of Bandwidth requirement for VoIP

The following formula can be used for calculating the bandwidth requirement.

$$\text{Bandwidth}(BW) = \text{TotalPacketSize} \times \text{PacketPerSecond}(PPS) \tag{1}$$

Here Total packet size = (Layer 2 header: MP or FRF.12 or Ethernet) + (IP/UDP/RTP header) + (voice payload size).
PPS = (codec bit rate) / (voice payload size).
For each of the codec, codec bit rate is different.

B. Calculation of bandwidth requirement for video steaming

Bandwidth of a video stream depends on the resolution and the number of frames. There are some other factors such as audio bandwidth and communication and compression overhead, these are not added most of the cases.

$$\text{FrameSize} = \text{Resolution} \times \text{ColourDepth} \tag{2}$$

$$\text{BitRate} = \text{FrameSize} \times \text{FrameRate}(fps) \tag{3}$$

Here ColourDepth is also known as bit depth. It is the number of bits used to indicate the colour of a single pixel, in a video stream. Frame rate is the no of frames transmitted per second. Based on the resolution, frame rate is different. For example the frame rate of D1 resolution (720X480) is 30 fps. The following table gives the list of resolution and their Frame Rate, which can be used for bandwidth calculation.

Table 2. Resolution Vs Frame Rate

| Resolution | Frame Rate |
|------------|------------|
| CIF | 30 fps |
| 4 CIF | 15 fps+ |
| 4 CIF | 30 fps |
| HD 720 | 15 fps+ |
| HD 720 | 30 fps |
| HD 720 | 60 fps |

| | |
|---------|--------|
| HD 1080 | 30 fps |
| HD 1080 | 60 fps |

There are different techniques available through which bandwidth requirement can be calculated. By applying this technique, we can calculate bandwidth requirement for best effort and background traffic also.

IV. PROPOSED DYNAMIC BANDWIDTH MANAGEMENT SYSTEM

To properly manage and utilize the available bandwidth in IEEE 802.11 wireless LAN, a hybrid method has been proposed here. The proposed method is the combination of bandwidth requirement, traffic priority value and user priority value. We have already discussed the Enhanced Distributed Coordination Channel Access (EDCA) mechanism which provides different priority to different traffic. In the above sections, we also mentioned about a method which provides different priority to different users. In our proposed method we calculate the bandwidth requirement (BR) for a special application. By combining the value of user priority (UP) value, traffic priority (TP) and bandwidth requirement (BR), Request Weight (RW) is calculated which is shown in Table 3. The formula for calculating the request weight is given below:

$$\text{RequestWeight}(RW) = \text{UP_value} \times \text{TP_value} \times \text{BR} \tag{4}$$

Table 3. Request Weight Calculations

| User Priority Value | Traffic Priority Value | Bandwidth Requirement (BW) | Request Weight (RW) |
|---------------------|------------------------|----------------------------|---------------------|
| 3 | 1 (Background) | 1 | 3 |
| 3 | 2 (Best Effort) | 1 | 6 |
| 3 | 3 (Video) | 3 | 27 |
| 3 | 4 (Voice) | 4 | 48 |
| 2 | 1 (Background) | 1 | 2 |
| 2 | 2 (Best Effort) | 1 | 4 |
| 2 | 3 (Video) | 3 | 18 |
| 2 | 4 (Voice) | 4 | 32 |
| 1 | 1 (Background) | 1 | 1 |
| 1 | 2 (Best Effort) | 1 | 2 |
| 1 | 3 (Video) | 3 | 9 |
| 1 | 4 (Voice) | 4 | 16 |

Here we have taken three types of users. High priority users have the priority value 3, medium priority users with priority value 2 and lower priority users have the priority value 1. According to IEEE 802.11 EDCA, voice traffic has the highest priority value 4, and then the video traffic has the priority value 3, best effort with priority value 2 and back ground traffic with priority value 1. Bandwidth requirement for each of the traffic type is different. But for our simulation convenient, we have assumed voice; video, best effort and background traffic requires 4, 3, 1 and 1 unit bandwidth respectively.



In this proposed, by using the UP value, TP value and BR, admission controller computes the request weight (RW). Based on the Request Weight (RW), bandwidth allocation priority will be given.

A. Steps for Proposed Bandwidth Allocation Approach

- Maintain UP, TP BR.
- If any new connection request comes, Admission Controller will calculate the Request Weight (RW) by applying the UP value, TP value and the BR value.
- If RW value of a new request is higher, assign lesser AIFS or CW.
- If the RW value is low for a new request, assign higher AIFS or CW.

Higher the Arbitrary Inter Frame Space or Contention Window (CW) value, priority will be less since higher AIFS or higher CW value lead to increase in backoff time. If back off time increases user has to wait more amount of time. So user with higher RW value will have the higher priority and will able to access with preference. In the table III, we have seen RW of user with priority 2 and user with priority 1 for video and voice traffic is more than the user with priority 3 for background and best effort traffic, so in that case based on RW value user with priority 2 and 1 will have the preference over user with priority 3 to access the wireless LAN.

V. RESULTS AND DISCUSSIONS

For simulation of our proposed method, we have used OPNET Simulator (Academic Version). We have considered different traffic like best effort, back ground, video and voice for simulation. Following parameters have been taken for analysis the performance of the proposed mechanism which is shown in Table IV. To evaluate the performance of the proposed method, we have taken three Request Weight (RW) value: 3, 18 and 48. By taking the three RW value, we have compared the performance by taking the following parameters. From the simulation results we have plotted some graphs. Here green colour represents the RW value 48, which is the highest, red colour represents the RW value 18, which is a medium value and blue colour represents the RW value 3, which is the lowest value of RW.

Table 4. Parameters units

| Application | Parameters | Units |
|-------------|-----------------------|-------------|
| WLAN | Data Traffic Received | Packets/Sec |
| | Data Traffic Sent | Bits/Sec |
| | Dropped Data Packet | Bits/Sec |
| | Delay | Sec |
| | Throughput | Bits/ Sec |

A. Traffic Received

It is defined as the data traffic received in the network at a given point of time. In Fig. 1, we have clearly noticed that the traffic data received value of the user with RW 48 is better than the users with RW 18 and RW 3.

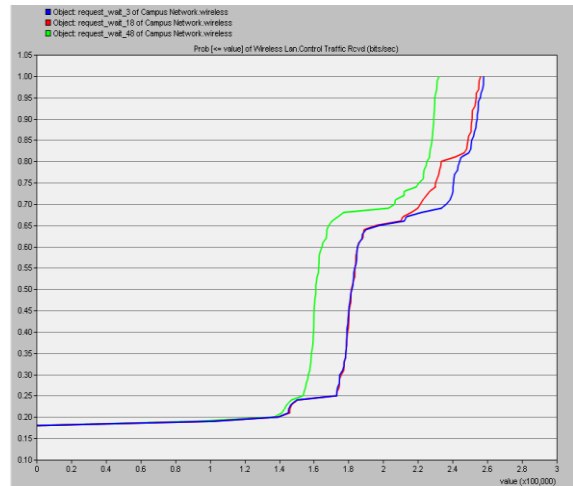


Fig. 1. Wireless LAN Traffic Received

B. Traffic Sent

Traffic sent is defined as the data traffic sent from the network at a given point of time. In Fig. 2, it is also clear that the traffic data sent value of user with RW 48 is better than the user with RW 18 and RW 3.

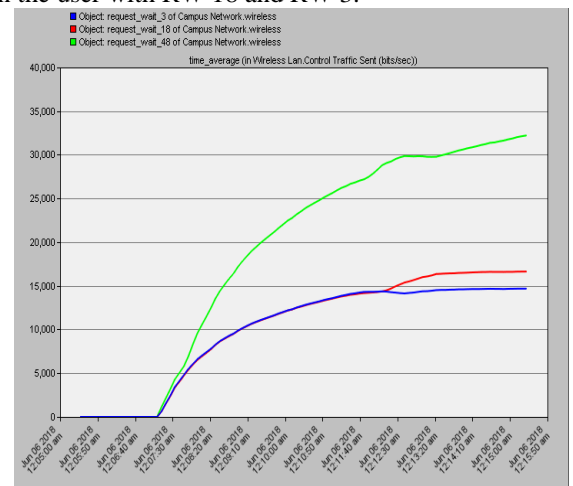


Fig. 2. Wireless LAN Traffic Sent

C. Data Dropped

In Fig. 3, it has been noticed that for the first 90 seconds there is no data dropped for user with RW 48, after that user with RW 48 has comparatively minimal data dropped than users with RW 18 and RW 3. In case of user with RW 18 also, there is no data drop up to first 90 seconds. But in case of user with RW 3, data dropped after 15 seconds of simulation and data dropped continuously increases for some time period. In the entire simulation user with higher RW performs better in terms data dropped compared to other two user types.

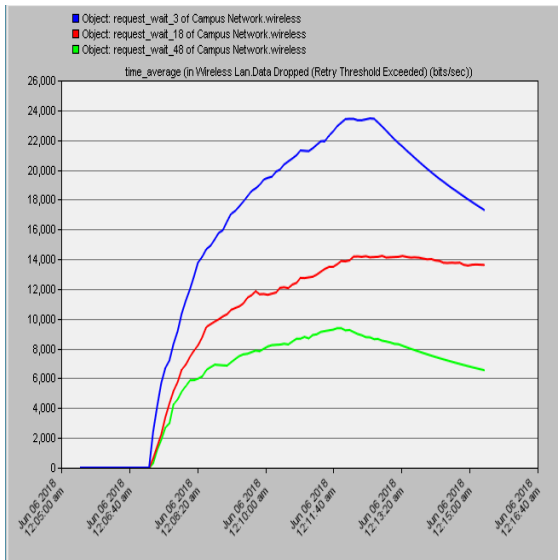


Fig. 3. Wireless LAN Data Dropped

D. Delay

Here, delay represents the end to end delay of all the packets received by the MACs of all the nodes of Wireless LAN in the network and forwarded to the higher layer. In Fig. 4, we have observed that user with RW value 48 has less delay compared to users with RW 18 and RW 3.

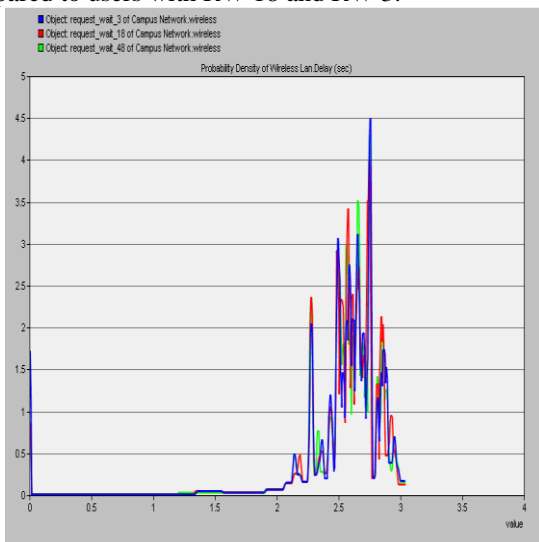


Fig. 4. Wireless LAN Delay

E. Throughput

Throughput is how much data actually travel through channel successfully from one station to another in a given point of time. In Fig. 5, we have seen that user with RW 48 have very higher throughput than the user with RW 18 and RW3.

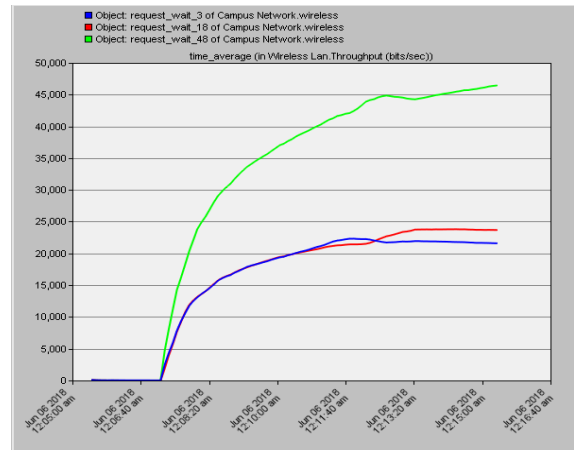


Fig. 5. Wireless LAN Throughput

VI. CONCLUSION AND FUTURE WORK

In this paper, we have proposed a dynamic bandwidth management scheme for achieving better QoS in IEEE 802.11 Wireless LAN. The proposed method is a hybrid method which combines the user priority value, traffic priority and bandwidth requirement value. Here we have taken three types of users, highest priority value is set to user 3 and then medium priority to user 2 and lowest priority is set to user 1. Traffic priority is defined according to the IEEE 802.11 EDCA mechanism. For each of the traffic type bandwidth is calculated mathematically. The method computes the request weight value from the above parameters. Allocation of bandwidth is done based on the request value. The proposed method has been simulated using OPNET academic version. From the simulation results, we have seen that the proposed method can allocate the bandwidth dynamically to all users. Main advantage of this algorithm is: it considers both the priority, traffic priority and user priority to allocate the bandwidth. Simulation has been carried out by taking five parameters. From the comparative analysis by taking the five parameters, we have seen that the proposed method performs well in managing and utilizing the available bandwidth properly. Finally we can conclude that the proposed scheme can give fair bandwidth allocation to all its users dynamically. In future we can calculate the total available bandwidth and distribute the available bandwidth to the entire user based on the request weight value.

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