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# Wireless Measurement Scheme for Bandwidth Estimation in Multihop Wireless Adhoc Network

By GK Srinivasa Gowda, CV Srikrishna & Kashyap D Dhruve

*SSET, Ernakulum, Kerala*

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*GJCST-E Classification : C.2.1*

WIRELESS MEASUREMENT SCHEME FOR BANDWIDTH ESTIMATION IN MULTIHOP WIRELESS ADHOC NETWORK

*Strictly as per the compliance and regulations of:*

# Wireless Measurement Scheme for Bandwidth Estimation in Multihop Wireless Adhoc Network

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## I. Introduction

A decentralized multi hop wireless network and is referred as Adhoc network since every node is entrusted in forwarding its data, and so the strength of the node which forwards the data is made dynamically that is based on the connectivity of the network. Mobile *ADHOC* Networks *(MANETs)* facilitates composite distributed systems which include wireless mobile nodes that can autonomously and vigorously self-compose into erratic and temporary, "*ADHOC*" network topologies. In *MANETs* if a user wishes to use multimedia applications like audio and video conferencing, live streaming of audio and video files it requires an efficient *QoS* multicast method to be in place. *MAC* Layer performance and routing techniques generally provide for the *QOS* in *MANETs*. To formulate a trustworthy *QOS* in *MANET*, *QOS MAC* protocol, resource reservation techniques and QoS routing protocol a proper cooperation technique is to be adopted to achieve it [1]. Since the network topology in *MANET* changes continuously so, it is difficult to attain a good *QOS* routing. Best effort distributed *MAC* controllers are preferably used in accessible wireless ad hoc networks to attain a respectable *QOS* for real time application which is connected with the design of decentralized media access control *(MAC)* model.

Available bandwidth estimation is a most important part of the admission control for *QOS* in wired and wireless networks both. In wireless networks channel fading takes place persistently and there is also error induced from the physical obstacles due to which the available bandwidth endures rapid time In addition to this in the wireless network a shared-access medium exists because of which the available bandwidth changes with the number of hosts competing for the network. Wireless last-hop networks containing the *IEEE 802.11* protocol in Distributed Co-ordination Function *(DCF)* mode are becoming popular at dependent variations but these effects do not persist in the case of wired networks and thus makes the variable bandwidth measurement or estimation a challenging task.

A rapid rate. In DCF mode, the *802.11* protocol does not involve any centralized unit to co-ordinate user's transmissions. The *MAC* layer generally uses an *CSMA/CA* algorithm for common use of the medium. Bandwidth is generally related to spectral width of electromagnetic signals or propagation characteristic of communication system in physical layer communications, whereas in term of data networks, bandwidth refers to the data rate that a network link or a network path transfers. In this article we lay emphasis on estimation of bandwidth metrics in the later data network

*Author α : Professor, SSET, Ernakulum, Kerala, India.*
*E-mail : srinivasgowdasset@gmail.com*
*Author σ : Professor, MCA, PESIT, Bangalore, India.*
*E-mail : cvsrikrishna@yahoo.co.in*
*Author ρ : Technical Director, Planet-I Technologies, Bangalore, India.*
*E-mail : kashyapdhruve@hotmail.com*

frame. Especially to packet networks, wherein the bandwidth refers to the amount of data/information that a link can deliver per unit time considered. In applications such as live streaming of audio or video data, file transfer the availability of bandwidth impacts the performance of the application directly. Multimedia based applications which are generally more responsive in networks exhibiting lower latency. Network latency minimization can be achieved through lower end-to-end delays, high bandwidth links and rather low packet transmission latencies. Bandwidth plays a very important role in various network technologies. Various applications can get benefited by knowing the characteristics of bandwidth in the network path. If we take the example of *P2P* applications we can clearly see that it creates variable user-level networks which based on the present bandwidth available between peers. Overlay network can organize their routing tables based on the availability of the bandwidth of overlay links. Network providers provide links to their customers and generally charges according to the bandwidth purchased. Service-Level-Agreements *(SLAs)* between providers and customers mainly define service in terms of availability of the bandwidth at key interconnection point. Network carriers generally plan capacity upgrade in their own network which is based on the rate of increase of bandwidth utilization of their users. Bandwidth is also is a main notion in content distribution networks, intelligent routing system, end-to-end admission control, and audio-video streaming. The presented research work presents an available bandwidth estimation method for *IEEE 802.11*-based wireless *AdHoc* network; this work is specially created for decentralized network. The presented research work employs the enhancements made to the *MAC* layer and then the data rate has been increased. The splitting of the *MAC* layer and then increasing data bit strength will enable achieving higher data transmission rates and reduced network congestion. Estimation of bandwidth which is available for a wireless host to each of its neighbor solely depends on the effect of the phenomena on the working of the medium access method.

The research paper has been organized in a way that the second section discusses some of the dominant literatures researches done for estimating and managing bandwidth in ad-hoc network. The third section of the paper represents some dominant key theoretical backgrounds of bandwidth estimation technique which is followed by next section that states our research contribution and techniques being implemented to attain the proposed measurement goals. The experimental study and results are discussed in the fifth section. The conclusion is discussed in the last section.

## II. Related Work

Bandwidth estimation and management mechanisms in networks have been researched upon for quite some time now. The swift growth in increasing requirement of the *QOS* oriented architectures that appropriately optimize the system functionalities and its overall performance are being invented and developed. Some of the dominating researches which are conducted for *QOS* optimization by implementing Bandwidth estimation are mentioned in this section. Research work [1] presents a protocol approach for access and routing facility, where the access is arbitrated by implementing synchronous signaling mechanism and the topology has been resolved by performing dissemination of node state. This work facilitates instinctive framework for providing arbitrating Radio frequency use and it employs the traffic mechanism to deliver *QOS*. *SWAN,* which is a stateless network model and uses a distributed control algorithm to bring service separation in mobile wireless ad hoc networks in a robust, simple and scalable manner, has been proposed in the research work [2]. An admission control and vibrant bandwidth management method which provides equality in the lack of distributed link level weighted fair scheduling is proposed in reference [3]. H. Luo et.al [4] projected a new scheme for packet scheduling which addresses this conflict. The significant contributions of this bandwidth estimation oriented work were, (a) a two tire service model, (b) an optimized algorithm for centralized scheduling, (c) a practical distributed back off based channel-estimation technique. A new distributive, localized, efficient and scalable solution to this problem has been in paper [5]. Ideal centralized fair queuing algorithm developed for ad hoc networks is being first analyzed and the desired global properties extracted. Then three localized fair queuing scheme has been proposed by the researchers. The work [6] represented various *QOS* requirements and elaborates the limitations and the advantages of the existing *QOS* routing protocol and comes with a QoS multicast Routing Protocol *(QMR)* with a variable hybrid method for *QOS* multicast routing. Literature [7] focused on multicast communication in ad-hoc networks and offered a simplification of routing trees into graphs that have more connectivity than trees and yet avoid long-term or enduring routing loops from happening.

For improving the Quality of Service *(QOS)* for multicast communication in *MANETs* some work has been presented in [8] *QAMNet* which propose the same.

Literature presented in [9] proposed a scalable *QOS* architecture in order to enhance the overall *QOS* for such networks. This method draws upon the positive aspects of both *IntServ* and *DiffServ*, and mainly extends upon the scalable *LANMAR* routing protocol to support *QOS*. A new *QOS*-aware medium access control *(MAC)* protocol that takes the above requirements into thought is presented in research work [10] whereas [11] gives an outline of cross-layer design which approaches for resource allocation in 3G *CDMA* networks, summarizes state-of-the-art research results, and gives more research issues. A *QOS*-aware routing protocol that contains an admission control scheme and a feedback scheme to set up the *QOS* requirements of real-time applications has been proposed in [12]. A deterministic scheme of packet delay and how to use it to derive both the packet pair property of *FIFO*-queuing networks and a new system (packet tailgating) for dynamically measuring link bandwidths has been proposed in the paper [13]. The time scales of significance range from a few milliseconds to a few minutes are presented in paper [14]. They also examine a phenomenon of compression (or clustering) of the probe packets comparable to the acknowledgement compression phenomenon recently experimented in *TCP*. The investigation of cause of these errors has been presented by Prasad [15], and showed that the presence of Layer-2 $L_2$ store-and-forward devices, which include Ethernet switches, have a nun-favorable effect on the correctness of VPS tools. Generally, each $L_2$ store-and-forward device adds additional serialization latency in a packet's delay, which results in constant underestimation of that L3 hop's capacity. The findings from a large-scale study of Internet packet dynamics observed by tracing 20,000 *TCP* bulk transfers between 35 Internet sites has been presented in paper [16]. An end-to-end scheme, called Self-Loading Periodic Streams *(SLoPS)*, for measuring available bandwidth presented by Jain [17]. *SLoPS* implemented in the tool known as path load. Two available bandwidth measurement methods, first one is the initial gap increasing *(IGI)* method and the other is packet transmission rate *(PTR)* method presented in paper [18].

## III. Theoretical Background

This presented section describes the theoretical background and few key factors in estimating bandwidth in Ad-Hoc network.

### *a) Bandwidth-Related Metrics*

We introduce three bandwidth metrics: available bandwidth, capacity, and Bulk-Transfer-Capacity *(BTC)* in this section, the first two of which are defined for both individual links as well as end-to-end paths, and *BTC* is generally defined for end-to-end path. The next section discusses the differences between links at the data link layer and links at *IP* layer. We consider the *1st* as segments and *2nd* as hops. A segment generally represents a physical point-to-point link, to a shared access local area network, or to a virtual circuit. But, a hop consists of a sequence of one or more segments, which can be connected through bridges, switches, or some other layer-2 devices. Generally we classify and end-to-end path P from a *IP* host S which serves as source to a host V which serves as sink as the order of hops which connects S to V.

### *b) Capacity*

A segment or a layer 2 link in normal circumstances transfers data at a constant bit rate, which is the broadcast rate of the segment. This rate is *10Mbps* on a 10 Base *T* Ethernet segment, and a rate of *1.544 Mbps* on a $T_1$ segment. The broadcast rate of a segment is restricted by the physical bandwidth of the fundamental propagation medium and its optical or electronic transceiver hardware. Due to its overhead of layer-2 encapsulation and framing a hop delivers a lower transmission rate then its normal rate at the *IP* layer. Let us consider that the nominal capacity of a segment is $C_{L2}$, the transmission time for an *IP* packet of size $L_{L2}$ bytes is

$$_{L3} = \frac{L_{L3} + H_{L2}}{C_{L2}} \dots\dots\dots (1)$$

Here, $H_{L2}$ represents the total layer-2 overhead that is needed to summarize the IP packet. So the capacity $C_{L3}$ of that segment at the *IP* layer can be defined as

$$C_{L3} = \frac{L_{L3}}{_{L3}} = \frac{L_{L3}}{\frac{L_{L3} + H_{L2}}{C_{L2}}}$$

$$= C_{L2} \frac{1}{1 + \frac{H_{L2}}{L_{L3}}} \dots\dots\dots (2)$$

The *IP* layer capacity mainly depends on the size of the *IP* packet which is comparative to the layer-2 overhead. If we consider the *10BaseT* Ethernet, $H_{L2}$ is *38 bytes (12* bytes for the inter frame gap, 18 bytes for the Ethernet header, and the equivalent of 8 bytes for the frame preamble) and $C_{L2}$ is *10Mbps*. So the hop can deliver with a capacity of *7.24Mbps* for *100 – bytes* packets, and *9.75 Mbps* for *1500 – bytes* packets to the *IP* layer. Let us assume that the Maximum Transmission Unit *(MTU)* is *1500 bytes* whereas the layer-2 overhead (without any additional data-link encapsulation) is *8 bytes* for *PPP* transmissions.

We describe capacity $C_i$ of the hop $i$ to be the maximum possible *IP* layer broadcast rate at the same hop. We can see from equation 2 that the maximum transmit rate at the *IP* layer result from *MTU*-sized packets. So, we can define the capacity of the hop the bit rate which is measured at the *IP* layer, at which the hop can transmit *MTU*-sized *IP* packets. If we extend the previous definition to a network route, capacity *C* of an end-to-end path is maximum *IP* layer rate that the path can transmit from source to sink or we can say that the capacity of a path provides an upper bund on the *IP* layer throughput that a user can get assured to get from that path. The end-to-end capacity $C$ is determined by the maximum link capacity in the path, i.e.

$$C = \min_{i=1,\ldots,H} C_i \quad \ldots\ldots\ldots (3)$$

Here, $C_i$ is defined as the capacity of the $i^{th}$ hop and number of hops in the path is determined by H. the narrow link on the path is the hop with minimum capacity. Some of the paths contain rate limiters or the traffic shapers which make the definition of capacity complicated. Considerably a rate limiter at a link can transfer a "peak" rate $P$ for a definite burst length $B$, and a comparative lower "sustained" rate S for longer bursts. As we consider the capacity as an upper bound on the rate a path can transmit, it is beneficial to define the capacity of such a link to be based on peak rate S relative to the sustained rate $P$. Where as, a rate limiter produces only a part of its basic segment capacity to an *IP* layer hop. We can see the example like ISP's often use rate limiters to distribute the capacity of a *OC - 3* link between different customers, taking charge from each customer which is based upon the magnitude of the bandwidth distribution. The capacity of that hop is defined as the *IP* layer rate limit of that hop. Some layer-2 scheme do not function with a constant broadcast rate, we can see as instance we can see, *IEEE 802.11b* wireless *LANs* has a transmission rate of *11, 5.5, 2, or 1Mbps,* which depends on the bit error rate of the wireless medium. During the time gaps in which the capacity remains constant we can use the previous elaboration of link capacity for such technologies.

*c) Available Bandwidth*

Existing bandwidth of a link or end-to-end path is another important metric. The existing bandwidth of a link is related to the unused or free capacity of the link for a definite time period. Even if the capacity of the link depends on the underlying transmission scheme and propagation medium, the existing bandwidth of a link moreover depends on the network load at that link, and is usually a time-varying metric. At any given instant in time, a link is either transferring a packet at the full link capacity or it is idle so, the immediate utilization of a link can only be either "0" or "1" thus a significant definition of available bandwidth requires time averaging of the immediate utilization over the time burst of interest. We can represent the average utilization $\overline{\mu}(t-\tau,t)$ for a time period $(t-\tau,t)$ is given by

$$\overline{u}\,(t-\tau,t) = \frac{1}{\tau}\int_{\tau-\tau}^{t} u(x)\,dx \quad \ldots\ldots\ldots (4)$$

Here, $u(x)$ represents the instantaneous available bandwidth of the link at time $\tau$, we can refer time length as the average timescale of the existing bandwidth. Figure 2 will illustrate this averaging effect.

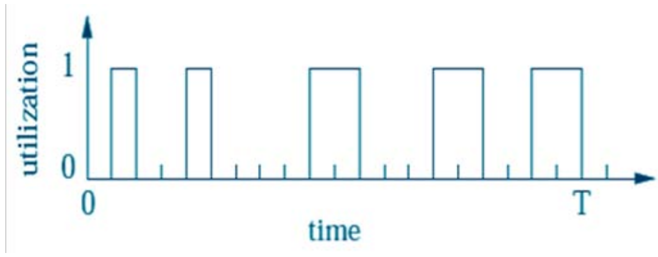


*Figure 1 :* Instantaneous utilization for a link during a time period *(0, T)*

The use of link is restricted to *8 out of 20* in this example between *0* and *T*, which yields an average use of *40%* now we will define the available bandwidth of a hop *i* over a fixed time interval. The average available bandwidth *Ai* of hop *i* can be represented by utilizing fraction of capacity,

$$A_i = (1 - u_i)C_i \quad \ldots\ldots\ldots (5)$$

Here, $C_i$ represents the capacity of hop $i$, and $u_i$ represents the average utilization of that hop in a certain given time interval. If we extend the definition what we have studied previously to a $H$ hop path, the bandwidth which will be available then is the minimum available bandwidth of all $H$ hops,

$$A = \min_{i=1,\ldots,H} A_i \quad \ldots\ldots\ldots (6)$$

The hop which has the minimum existing bandwidth is known as the tight link *1* of the end-to-end path. A *"pipe model with fiuld traffic"* presentation of of a network path, where a pipe represents each link is represented in figure 3. The width of the each pipe corresponds to the comparative capacity of their corresponding link. The utilized part of the link's capacity has been represented by shaded area, whereas spared capacity is represented by unshaded area. The end-to-end capacity is determined by the minimum link capacity $C_i$, whereas the end-to-end available bandwidth is determined by the minimum available bandwidth $A_i$. As shown in the figure 3, there is a difference between the path and the tight link.

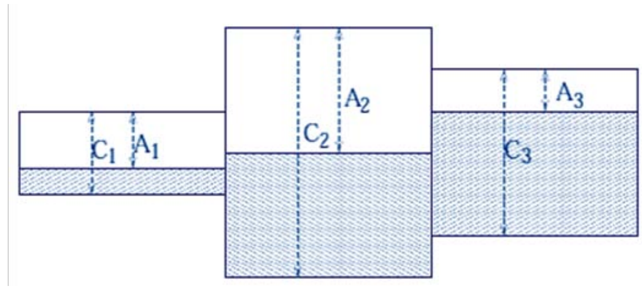


*Figure 2 :* Pipe model with fluid traffic for 3-hop network path

Various methodologies to measure existing bandwidth make the supposition that the link utilization remains stable when averaged over time, i.e. they suppose a stable network load on the network path. As this hypothesis is logically over moderately short time bursts, diurnal load variations will make a change in measurements made over longer time bursts.

We can note that traffic variability or long-range dependency effects cannot be prevented by constant average utilization. As the average available bandwidth changes frequently with a certain time period so, it is needed to measure it quickly. This is very much useful for the application that uses the available bandwidth measurement to adapt their transfer rate. If we look at the capacity of a path it remains constant for a long interval of time so, the capacity of path need not to be measured in a hurry as compared to the available bandwidth.

*d) TCP Throughput & Bulk transfer capacity*

Throughput of a *TCP* connection is a very important bandwidth-related metric in *TCP/IP* networks as, *TCP* is the most important transfer protocol in the Internet which carries almost the *90%* of the network load [19]. So, a *TCP* throughput metric will be of a great significance to the end users. It is not an easy task to define the throughput of a *TCP* connection exactly. Various factors may cause change in *TCP* throughput which includes type of the cross network load such as *UDP* or *TCP*, the numbers of the connecting *TCP* connections, transfer size, the buffer size of the *TCP* socket at both sender and receiver's end, congestion along the acknowledgement path, router buffer's size, capacity and load of every link in the network route. *TCP* Throughput can also be affected by the selection of the primary window size [21], change in the requirement and implementation of *TCP*, as New Reno [20], Reno, or Tahoe, use of *SACK's* [22] versus *CACK's* and numerous other parameters. If we take the example of a typical web page, throughput of small transfer mainly depends on the Round-Trip Time *(RTT)*, congestion time, and slow-start scheme of *TCP*, rather than on the bandwidth which exists on the path. In addition to this when we use diverse versions of *TCP*, the throughput of a large *TCP* transfers over a fixed network path can vary drastically even when the available bandwidth is similar.

The Bulk-Transfer-Capacity *(BTC)* [21] usually defines a metric which represents the attainable throughput by a *TCP* connection. *BTC* is the highest throughput which can be obtained by a single *TCP* connection. *TCP* Congestion control algorithm which is specified in *RFC* 2581 [22] must be implemented in the connection. A *BTC* measurement must give in detail various other important parameters about the accurate implementation of *TCP* at the end hosts [21] because, *RFC* 2581 leaves some performance details open. Here we can note that the available bandwidth and *BTC* are essentially different metrics. Available bandwidth does not depend on the particular transport protocol whereas the *BTC* is specified by *TCP*. The *BTC* depends fully upon the how *TCP* shares bandwidth with other *TCP* flows, whereas available bandwidth metric normally assumes that the average network load remains fixed and estimates the extra bandwidth that a path generally offers before its tight link gets drenched. We elaborate this point by supposing a single-link path with capacity is being saturated by single *TCP* connection. Due to path saturation, we know that the available bandwidth in this part will be zero, but the *BTC* comes around *C/2* if the *BTC* connection uses the same *RTT* as the previous *TCP* connection.

## IV. Our Contribution

In our presented research work the bandwidth estimation scheme has been implemented as an essential component in the construction of: (a) a dynamic bandwidth management scheme for single-hop mobile ad hoc networks, and (b) an explicit rate-based flow control scheme for multi-hop mobile ad hoc networks.

In this proposed technique the *MAC* layer has been divided into two sub layers, in which the common sub layer transmits the data bits while another one is responsible for transmitting the control signals. In spite of the sub-layering *MAC* the *PHY* has been optimized for higher data rate and with optimal power efficiency. The data rate has been increased at this layer thereby increasing network throughput with minimized network congestion, maintenance count, networks overheads, and end to end delay with highly monitored data transmission.

*a) Available Bandwidth Estimation*

Stages in the transfer of a single packet using the *IEEE 802.11 DCF MAC* protocol has been shown in figure 4. Throughput of transferring a packet is measured as

$$TP = \frac{S}{T_r\,T_s} \dots\dots\dots (7)$$

Here the size of packet is represented by *S*, time when *ACK* is received is *tr* and *ts* represents the time when the packet is ready at the *MAC* layer. $T_r$-$T_s$ Is the time interval which includes channel busy and contention time. Since the channel condition is different to each one so the throughput estimates kept separate to different neighbors.

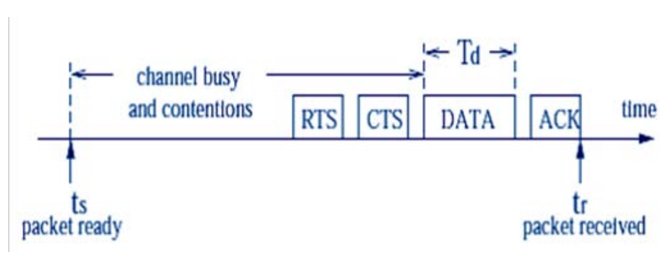


*Figure 4 :* IEEE 802.11 Unicast packet transmission sequence

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Any other wireless host which is well within the reach of its broadcasting range is the neighbor of a wireless host. The effect of the contention on available bandwidth is captured by the link layer measurement method. $T_r$-$T_s$ Will increase if the contention is high and similarly the throughput *TP* will decrease. This scheme also facilitates the capture of effect of fading and interference error as, if the *RTS* or data packets get affected by this then they have to be re transmitted, this then increases the $T_r$-$T_s$ and subsequently decreases the available bandwidth. Thus our available bandwidth measurement scheme takes into an account by the phenomena causing it to diminish from the theoretical maximum channel capacity. We should take this into consideration that we can measure the available bandwidth only by using the successful link layer broadcasting of an ongoing data flow.

The measured throughput of the packet generally depends on the size of the packet and is directly proportional to it as the large packet sends more data once it grab the channel therefore it has more throughput. For the throughput to be not depends upon the packet size, we generally normalize the throughput to a pre defined packet size. In Figure 4, the actual time for the channel to transmit the data packet is represented by

$$Td = \frac{S}{BW_{ch}} \;\ldots\ldots\ldots(8)$$

Where *BWch* represents the bit rate of the channel. We take the channel bit rate as pre defined. The broadcasting time of the two packets should be differ only in their times to transfer the *DATA* packets. Therefore, we have:

$$(T_{r1} - T_{s1}) - \frac{S_1}{BW_{ch}} = (T_{r2} - T_{s2}) - \frac{S_2}{BW_{ch}}$$

$$= \frac{S_2}{TP_2} - \frac{S_2}{BW_{ch}} \;\ldots\ldots\ldots(9)$$

Here, $S_1$ represents the actual data packet size, and *Pre-defined* standard packet size is represented by $S_2$. During the course of simulation we generally varies the packet size from 64 bytes to 640 bytes and send the network load from one host to another. The raw throughput which we measured is normalized a standard packet of size 512 bytes. The raw throughput which generally depends upon the size of the packet is directly proportional to it whereas, normalized throughput does not depend upon it. Therefore to remove the disturbance introduced by the considered raw throughput from packets of different sizes we use the normalized throughput to be represented as the bandwidth of the wireless link. Robustness of the *MAC* layer bandwidth estimation is the one another main issue. To eliminate the bandwidth in the current time window we use the measurement of the bandwidth of a link in discrete time intervals by taking out the average of the throughputs in recent packets in the past time window.

Since the channel condition changes at time intervals so this estimation may not be perfect. Now, to evaluate the estimation error, we run a *CBR* flow with the use of *UDP* having data rate 160 kbps from one node to another in a 10 node one hop environment. The background network load contains one *TCP* flow in the light channel contention case, and seven *TCP* flows in the heavy contention case. The main to reason to use the *TCP* only is to generate a cross-traffic with bursts to the *UDP* flow. By using the average of packet throughput in the past time window we can measure and normalize the throughput of the *CBR* flow in every 2 seconds.

*b) Channel Time Proportion and Admission Control*

Bandwidth estimation scheme has been used in the admission control in single- and multi-hop wireless networks which was also used in the previous section. The concept of channel time proportion *(CTP)*, using a simple example is introduced. Let us assume that the throughput *TP* over a specific wireless link is 10 *MAC* frames of a specific size *S* per second, based on the point of argument and substantial error experienced on this link. Suppose a specific flow requires 3 frames over this link between the neighbors. Thus it need to be active must on the sending host's interface for 30% of unit time, on an average therefore, this leaves only 70% of unit time existing to other flows out of the interface, which affects their admission directly, we can also extend this logic to bits per second. Suppose K bits is being transmitted over a wireless link in a second, where a specific level of contention and physical error is present, and a user requires a minimum throughput of *E* bits per second, then from this effect user needs *1/k* of unit time on the source interface. Basically by dividing bandwidth requirement in bits per second by the available bandwidth which is estimated we can obtain the *CTP* requirement of a flow. The *CTP* obligation is a portion. Generally admission control divides up to 100% channel time on an edge among the various flows based on their requirement and some fixed fairness standard.

i. *Dynamic Bandwidth Management in Single-hop Ad hoc Networks*

Admission control and dynamic bandwidth management method which we represented in the paper give fairness and rate guarantee even in the absence of distributed link layer fair scheduling. The methodology is normally applicable where peer-to-peer multimedia broadcasting which need to adapt their broadcasting rate co-operatively such as smart-rooms. With particular *CTP* requirement we generally map minimum and maximum bandwidth needed of a flow.

The main part of the methodology, a bandwidth manager *(BM)*, a share of channel is allotted by the *BM* to each flow depending upon the need relative to the other flows. To obtain minimum *CTP* the *BM* uses an algorithm known as max-min fair algorithm. To make the admitted flow only occupy the channel for a burst of time allotted to them flows control their transfer rate co-operatively. Since, the existing bandwidth changes thus the network load change, then the channel access time for each individual flow has been re-allocated by the *BM*. By doing the simulation we can see that every flow in the network receives at least minimum requested share of the network bandwidth at a very low cost and with greater probability.

ii. *EXACT*

It is a rate based explicit flow control which is planned for the multi-hop ad hoc network situation. In this, data transfer rate of the flow, which is passing the router is determined by each router, which is generally based on the measurement of the bandwidth of the outgoing wireless links. First the request of every flow is transformed into a request for channel time fraction, and using the max-min fairness standard the total channel time is allocated to the contending flows.

## V. Results Obtained

The presented research work has been implemented with Dot net tool with provided operating conditions. In this research work the developed architecture for enhanced packetization scheme at *MAC* layer has provided a significant system enhancement and yielded much better results. Here in this research paper the comparison for both centralized and decentralized *AdHoc* network has been done and the graphs illustrating network congestion, throughput, transmission error rates and the utilization of bandwidth has been obtained. In spite of these dominating parameters a crisp and alarming facts has been observed that justifies the robustness and high performance of the presented system architecture for bandwidth estimation.

Figure 6 represents the comparative result illustrating the average congestion measured for both centralized as well as decentralized network topology. From figure it can be stated that the proposed technique has exhibited a very uniform with minimum congestion as compared to the centralized topology which is having much higher congestion load that decreases as per the increase in network size. The result data states that the proposed system has reduced the congestion by approximately two times.

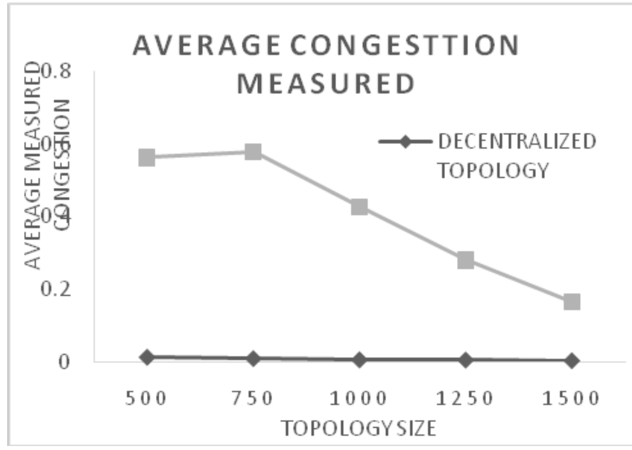


*Figure 6 :* Congestion measured for centralized and decentralized network

The ascending graph (Figure 6) for network throughput states that the overall network throughput is also higher by *0.01%* as compared to existing centralized system. Meanwhile Figure 7 illustrates the transmission error rates and the proposed system is found to be more productive as compared to the existing systems. One of the striking results has been found to for network congestion rate. The proposed scheme of bandwidth estimation has illustrated the congestion rate reduction five times better as compared to existing one. Thus the developed system architecture has presented a highly potential solution for *QOS* oriented bandwidth estimation technique.

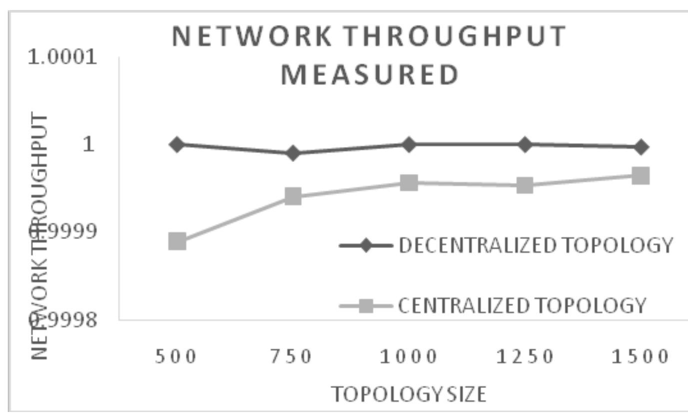


*Figure 7 :* Network throughput measured

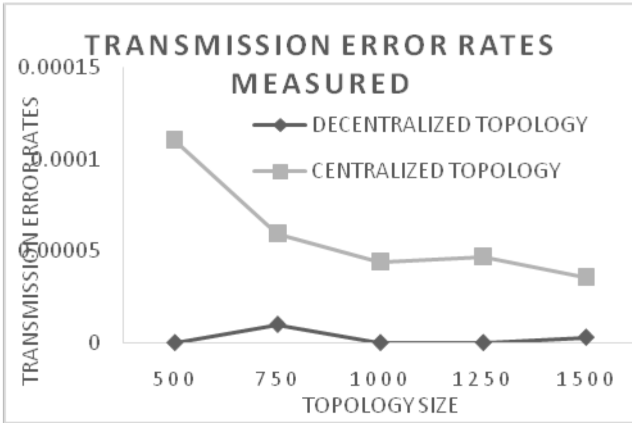


*Figure 8 :* Transmission error rate measured for centralized and decentralized AdHoc network

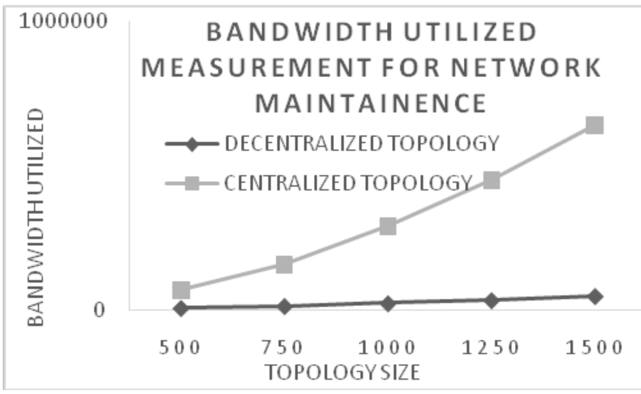


*Figure 9 :* Bandwidth utilized in centralized and proposed topology

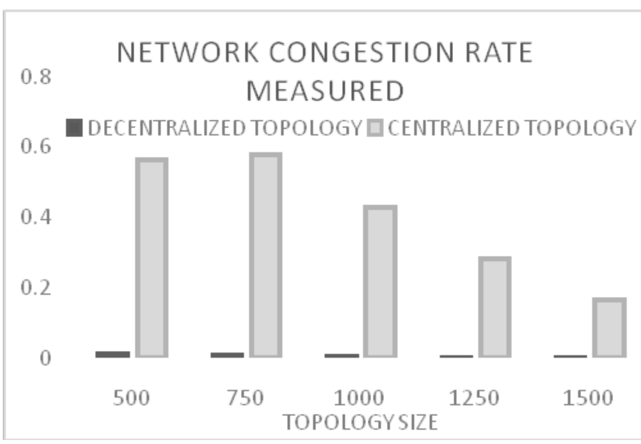


*Figure 10 :* Network Congestion rate for

## VI. Conclusion

In this research work effective and optimized system architecture for *QOS* oriented bandwidth estimation has been proposed where the scheduling at the *MAC* layer has been modified and the slots has been prepared to transmit the data, thus by increasing the data rate as well as increasing higher bandwidth utilization. The overall system architecture developed on Microsoft Visual Studio 2010 platform has exhibited highly optimized results based on quality oriented parameters like network throughput, network congestion rate, transmission error rate, bandwidth utilization, maintenance count packets, and end to end delay, monitoring overheads etc. the developed system architecture has illustrated the reduction of *160%* in network congestion and the overall network throughput has also increased by *0.01%.* The number of maintenance count has been reduced drastically. Thus the proposed system has exhibited a highly potent mechanism for bandwidth estimation.

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