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ESTIMATION-BASED QUEUE SCHEDULING MODEL TO IMPROVE QOS FOR END USERS IN MANETS

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Abstract. Using MANETs for real time applications is always a challenge as the network is extremely dynamic with brisk topology changes. Despite this, several real time schedulers have been developed that aimed at providing QoS to ad hoc nodes. The quality of service (QoS) is standardized in terms of capacity, reliability, link quality, delays/jitters, and network cost. Thus, for QoS, the better transmission should be maintained at end user as well as at the transmitting unit. QoS of a network is affected by delays and bandwidth allocated for transmission. For an efficient network, it is required to predict these metrics during transmission. For this, in this paper, integration of quaternion-based Kalman filter is performed that predicts the required bandwidth and the network delays with higher accuracy. From the analysis, it is shown that bandwidth can be optimized but it is not possible to aloof delays in the network. Thus, while implementing such admission control procedures, estimation process allows control over delays and sustain them from going beyond a certain threshold value. The model proposed is a novel approach and has not been formulated in any of previous work related to QoS in MANETs. The effectiveness of model is demonstrated using both simulation and real time results.

Keywords: Bandwidth, end to end delays, MANETs, quaternion Kalman filter, quality of service, queue-scheduling, throughput

Mathematics Subject Classification 2010: 16-Hxx

1 INTRODUCTION

MANETs are small networks that operate without usage of any central notifying and controlling device. Each device in mobile ad hoc network has the capability of receiving, transmitting, as well as forwarding. Nowadays, each device in network tends to transmit multimedia data that requires more bandwidth, and better transmission services as compared to networks that are used to transmit only small sized packets or messages that mostly are dependent upon the energy usage rather than the effective bandwidth. MANETs have a mobility constraint as a barrier to their performance. For this, network management is performed to lower the delays during handoffs during data transmission [1]. Another important aspect of MANETs' performance is topology management that allows stabilized network for data transfer [2]. Traditionally, MANETs are used for some critical situations such as disaster management, airplane breakage exhaustion system, enemy tracking system; but in scenario that requires transmission of multimedia data directly between the devices without any centralized node, effective transmission is of utmost importance. This transmission can be defined in terms of bandwidth efficiency and QoS to end users. Improved QoS widens the application area of MANETs. For enhanced, optimized and correct transmission, bandwidth efficiency needs to be increased. Also, it is to be noticed that for effective transmission, the amount of energy consumption by the network also increases. Therefore, it is required to use such models that perform a lesser number of computations for transmitting multimedia data over the network [3]. It is to be noted that as the number of intermediate nodes used for relaying purposes increases, the path tends to increase, thus, overall delay increases, which can be controlled by following proper topology or identifying the type of data to be transmitted, and time required to transmit them over a particular channel. This estimation process should be based upon real time network data identification. Various models have been developed for QoS improvement but they lack in network estimation procedure that decreases their versatility. In proposed model, work has been carried on the identification of the type of data and selecting optimized transmission strategy between the mobile nodes based upon Kalman estimation filters. The mode of transmission can control the transmission barriers. Here, modes of transmission mean the number of channels used to transmit data over to another relaying node. Thus, the data division and packet relaying plays a vital role in channel-based transmission. Hence, the overall scenario for real-time applications must be capable of differentiating the normal, regular and priority type of data that needs to be transmitted over the channel. For this, hop to hop transmission, packet priority allocation, packet reordering and packet arrangement system are managed. Also, for this, admission control system that has relative differential service system for performing these tasks is required. The differentiation is random due to rapid changes in bandwidth, energy utilization, link capacity and mobility concerned. However, network estimation performed during transmission allows better selection differential services by providing logs of required bandwidth and delay estimation. In case, nodes are to be treated static, then these differential services should be absolute. In the scenario considered, data is to be transmitted effectively over mobile nodes, for this, the absolute model can only be used at transmitting unit. The analysis shows that estimation-based queue scheduling allows correct selection of queue and allows efficient allocation of bandwidth, thus improving QoS to end users. The effectiveness of model is demonstrated based on both simulation and real time results.

The rest of the paper is structured as follows: Section 2 highlights the related research work. Section 3 describes the system equations and traffic prediction based upon quaternion-based Kalman filters. Section 4 illustrates the formulation of proposed model and its integration with the system equation to form an efficient queue scheduler. Delay, bandwidth and throughput evaluations are provided in Section 5. Section 6 presents the simulations and result analysis. Section 7 demonstrates the real time application analysis of proposed model. Conclusion based upon final observations is presented in Section 8.

2 RELATED WORK

Various models developed for real-time usage of mobile ad hoc networks have been reviewed. These models have been modified as per usage of the network, but the major drawback that appears is the weakening of quality of services at the end user. Many traffic scheduling algorithms have been developed which combines the features of admission control and traffic control algorithm. But, these use absolute differential service-based methods of data transmission for real time packets, thus, cannot be relied upon in terms of performance under extensive conditions such as dynamic topology.

Geng et al. [4] have designed a protocol over MAC layer that ensures QoS over ad hoc networks. The protocol operates over IEEE standards. It considers reservation system for QoS enhancement based on channelization. The protocol controls the functionality of network nodes depending upon the requirement of reserved slots for transmission. It is capable of solving hidden terminal problem as well as fairness queue issues over ad hoc networks. The protocol is analyzed over network throughput and delays w.r.t. network load.

Another model is given by McLaughlin et al. [5], as shown in Figure 1 and Figure 2, which is based upon IP packet scheduler. The paper includes the circuit implementation for weighted fairness model for improving network QoS. The model utilizes the capabilities of general purpose sharing for efficient and fair allocation of bandwidth. Also, the bandwidth consumption was very high that resulted into increase in overall delays, thus, lowering the channel capacity and decreasing the performance. The paper also includes the hardware implementation of a circuit that uses weighted fairness scheme for high-speed packet sorting. The architecture is efficient for a network with pre-defined metrics but for highly dynamic network, performance degrades as transmission progresses. Shreedhar et al. [6] have designed another model which is termed as stochastic fairness queue model and is based upon round robin technique shown in Figure 3, which is only applicable for a lesser number of packets and low traffic networks. This model, as shown in Figure 4, is capable of providing quality of service for end-users and is based upon hashing technique; this model classifies the packets in the queue on the basis of a probabilistic approach. This model also increases the overheads as hashing technique would account for residing the packet coming from the same source into different queue depending upon the generated hashing function, thus, for real-time applications, it becomes complex to allow intolerant delays. The model is applicable to routers and gateways, however, it increases redundancy if applied to each node in the network. Also, when the number of nodes has increased in the network, lack of queue prediction causes it to operate as FCFS; thus, lowering the network performance.

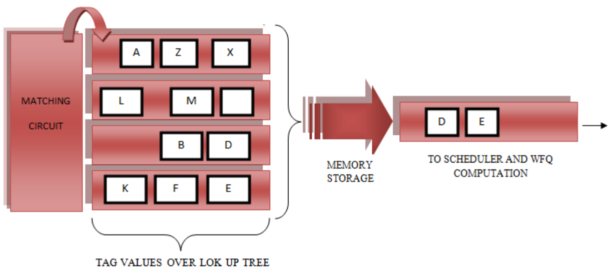


Figure 1. McLaughlin's weighted fairness queue model [5]

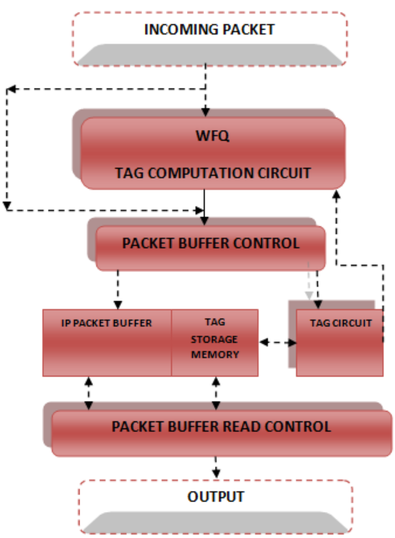


Figure 2. McLaughlin's weighted fairness queue architecture [5]

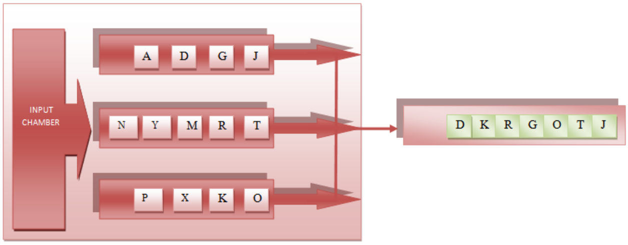


Figure 3. Round robin model [6]

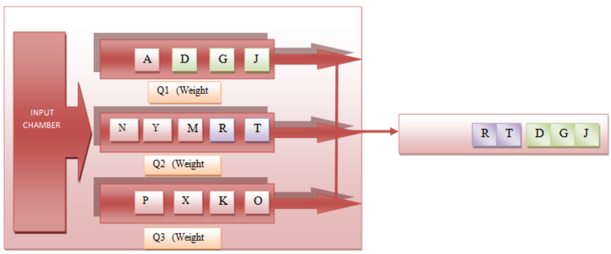


Figure 4. Deficit round robin model [6]

Further a model for improving the end user-based quality of service is given by Varaprasad et al. [7] and is termed as weighted fair queue model (WFQ). This model, as shown in Figure 5, serves the queue with lower weight. Hence, weight was assigned on the basis of presentation units (multimedia datagram or packets). It reserves bandwidth for particular users and divides the working of the MANETs transmission into the set of classes with weights assigned to them. The priority-based approach was used to identify the high priority class. However, it provides the minimum bandwidth to each class and also reserves bandwidth for a particular class. This improves the quality of service for the end user, however, bandwidth pre-allocation decreases the performance when there is more traffic and number of classes is also higher as compared to available bandwidth.

Sun et al. [8] have designed a modified proportional fairness model with multi-hop queue scheduling for cognitive radio-based MANETs. The model is cross layer-based model that restricts QoS for end users in MANETs. The model is MAC layer-based service distributor that ensures that QoS is not affected during transmission. It computes the availability of channels and buffer. Then, it uses MAC analyzer to compute packet priority and packet scheduling before selecting the transmission strategy. Along with it, performance measurement is also carried during the end to end transmission. The model is demonstrated in Figure 6. The major drawback in the QoS scheduler is that no estimation is carried out during transmission, also in the case of addition of intermediate nodes, it requires regular updating of channel list that causes increased overheads.

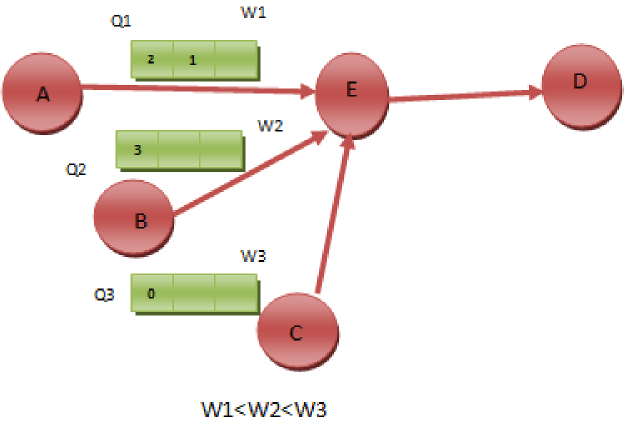


Figure 5. Weighted fair queue model [7]

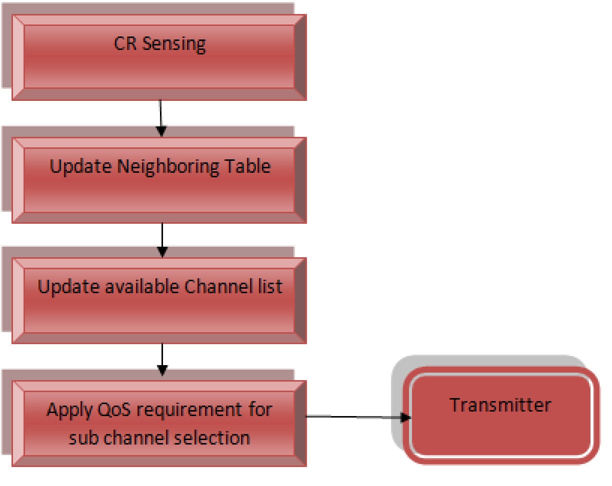


Figure 6. Sub channel selector and transmitter [8]

Peng et al. [9] have designed a virtual routing protocol that offers QoS in terms of bandwidth and topology management over OLSR protocol [9]. Authors implemented a virtual cross layer approach to share information between the nodes of the network and used it as a decision parameter during routing, thus offering QoS over the wireless mesh network. Multiple channels are incorporated for performance evaluation of this protocol. It estimates the network bandwidth depending upon the information passed on to a node during topology management.

Sondi et al. [10] have designed QoS over OLSR as guidelines and operated it as an integrated approach rather than a fixed underlying algorithm. Authors designed governing rules for MANETs like networks that can improve their QoS without using particular algorithms. The approach used in their paper demonstrates the identification of factors based on heuristics that optimizes the performance of the network.

Kuo et al. [11] have implemented a cross-layer approach to offer multimedia services over MANETs. It is link to link connectivity-based implementation. Authors integrated dynamic hash table and IPV6 to provide multimedia transmission over a network that has frequent topology changes and unreliable wireless links.

In an another approach by Bouk et al. [12], authors emphasized on selection of appropriate gateway node to provide seamless QoS to end users over MANETs. Taking into account the path dynamics, a feedback system is incorporated into the network that is capable enough to handle end to end delays. Authors focused on link expiration time to evaluate the gateway during path planning on basis of system dynamics in the network.

Yu et al. [13] have used game theory approach providing security along with QoS over MANETs. Bayesian theoretic approach is incorporated to identify the winning paradigm for transmission. Collaborative communications are provided over MANETs by selection of appropriate relaying mechanism based on the network inputs in terms of throughput and security analysis performed for the system. The approach focuses on attaining equipoise over the network to improve its performance.

Wen et al. [14] derived a queuing theory-based approach for service discovery in MANETs. The approach includes a $M|M+|1$ model for service discovery on MANETs nodes. The model uses control variables that predict the system throughput to analyze the system behaviour. The approach is adaptive to variations in a network environment. The process defined in model operates at server level, thus allows facilitating the local variables. The whole network is divided into zones comprising backbone and mobile nodes with mobile nodes operating on queuing theory, as shown in Figure 7. Thus, it was analyzed that performance depends on admission control algorithm as well as scheduling technique for a network that encounters congestion. Also, the QoS scheduler should be efficient enough to be incorporated into system model without causing any additional overheads.

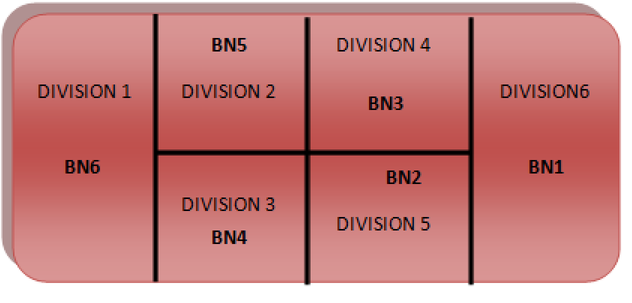


Figure 7. Wen's BN P2P overlay [14]

3 SYSTEM EQUATION AND TRAFFIC PREDICTION

An efficient network can be described as a combination of models that have the sensing capabilities regarding the data flow over the network. Sensing capabilities to a network can be provided through estimation process. For a model ensuring QoS to end users, it is required to estimate the traffic flow and the actual bandwidth requirement. The gap between the actual and allocated resource can cause an increase in network overheads, thus lowering the QoS to end users. Therefore, an efficient network model is required that can observe and predict the network conditions and then take decisions for correct allocation of resources such as required bandwidth for queue transmission. This estimation of network state can be achieved using quaternion Kalman filter. Quaternion filter analyzes the prior and posterior network states to form the prediction equation. This prediction equation allows to evaluate the next required resource. Therefore, it is required to form system equation and then perform traffic prediction before implementing the QoS scheduling model. The equation formulated to represent the system behaviour is derived using existing quaternion algebra over Kalman filter [24, 25, 28] and integrating it with network parameters and standard network equations. However, no such approach exists in literature that has highlighted such intensive application of quaternion Kalman filter over mobile ad hoc network for queue scheduling predictions.

3.1 Quaternion Filter Based Network Analysis

Let $\overline{w_1}$, $\overline{w_2}$, $\overline{w_3}$ be the weighing factors for link prediction, traffic prediction and queue prediction, respectively, that are defined over quaternion [24] as:

$$\overline{w_1} = X_1 M + X_2 L + X_3 D + X_4, \tag{1}$$
$$\overline{w_2} = Y_1 N_p + Y_2 Nrt + Y_3 T_{pkl} + Y_4, \tag{2}$$
$$\overline{w_3} = Z_1 QMS + Z_2 TQS + Z_3 WPU + Z_4 \tag{3}$$

where $X_1, X_2, X_3, Y_1, Y_2, Y_3, Z_1, Z_2, Z_3$ are the hyperimaginary numbers [24] denoting bandwidth, delay estimators, M is node mobility, L is link state, D is the relying distance between any two nodes, N_p is the number of presentation units, Nrt is the number of re-transmission, T_{pkl} is the total transmission time, QMS is the memory slots occupied in queue, TQS is total available memory slots in queue, WPU is total weight of presentation unit and X_4, Y_4, Z_4 are the scalars that define the measurable quantities. Quaternion defined in the Equations (1)–(3) allows application of vector spaces with complex as well as real numbers over 4-dimensional parameters defined over the network. It also provides the values that can be easily fed as an input to other analysis system such as Gauss-Markov model or Euler model. From definition of quaternion [24], $\overline{w_1}$, $\overline{w_2}$, $\overline{w_3}$ can be expressed as:

$$\overline{w_1} = \begin{bmatrix} w_1 \\ X_4 \end{bmatrix} = \begin{bmatrix} M & L & D & X_4 \end{bmatrix}^T, \tag{4}$$

$$\overline{w_2} = \begin{bmatrix} w_2 \\ Y_4 \end{bmatrix} = \begin{bmatrix} N_p & Nrt & T_{pkl} & Y_4 \end{bmatrix}^T, \tag{5}$$

$$\overline{w_3} = \begin{bmatrix} w_3 \\ Z_4 \end{bmatrix} = \begin{bmatrix} MQS & TQS & WPU & Z_4 \end{bmatrix}^T. \tag{6}$$

Let $\overline{w}$ be the overall network prediction weighing factor quaternion defined over $\overline{w_1}$, $\overline{w_2}$, $\overline{w_3}$ as:

$$\overline{w} = A_1\overline{w_1} + A_2\overline{w_2} + A_3\overline{w_3} + A_4, \tag{7}$$

such that

$$\overline{w} = \begin{bmatrix} \mathbf{w} \\ A_4 \end{bmatrix} = \begin{bmatrix} \overline{w_1} & \overline{w_2} & \overline{w_3} & A_4 \end{bmatrix}^T. \tag{8}$$

Quaternion defined in Equation (7) provides the aggregated equation of multiple parameters defined as individual entity over single quaternion. This provides a single solution that can be used for further analysis using less computational time. Let α be the difference in queue rate due to mobility for each node (assuming α to be constant and same for each node), $\mathbf{w}$ is defined using quaternion [24] as:

$$\mathbf{w} = \begin{bmatrix} k_x^{'}\mathrm{Sin}(\frac{\alpha}{2}) & k_x^{''}\mathrm{Sin}(\frac{\alpha}{2}) & k_x^{'''}\mathrm{Sin}(\frac{\alpha}{2}) & 1 \\ k_y^{'}\mathrm{Sin}(\frac{\alpha}{2}) & k_y^{''}\mathrm{Sin}(\frac{\alpha}{2}) & k_y^{'''}\mathrm{Sin}(\frac{\alpha}{2}) & 1 \\ k_z^{'}\mathrm{Sin}(\frac{\alpha}{2}) & k_z^{''}\mathrm{Sin}(\frac{\alpha}{2}) & k_z^{'''}\mathrm{Sin}(\frac{\alpha}{2}) & 1 \end{bmatrix}. \tag{9}$$

Note that for MANETs, topology is flat grid, i.e. the axis of movement will be defined for (x, y) only with $z = 0$. Thus, $\mathbf{w}$ becomes:

$$\mathbf{w} = \begin{bmatrix} k_x^{'}\mathrm{Sin}(\frac{\alpha}{2}) & k_x^{''}\mathrm{Sin}(\frac{\alpha}{2}) & k_x^{'''}\mathrm{Sin}(\frac{\alpha}{2}) & 1 \\ k_y^{'}\mathrm{Sin}(\frac{\alpha}{2}) & k_y^{''}\mathrm{Sin}(\frac{\alpha}{2}) & k_y^{'''}\mathrm{Sin}(\frac{\alpha}{2}) & 1 \\ 0 & 0 & 0 & 1 \end{bmatrix} = \begin{bmatrix} k^{'} & k^{''} & k^{'''} & 1 \end{bmatrix} \tag{10}$$

where $k^{'}, k^{''}, k^{'''}$ are the unit vectors along the axis of movement defined for $\overline{w_1}$, $\overline{w_2}$, $\overline{w_3}$ [24]. This can be noted that in actual mobile ad hoc networks, there might be situations where no LOS (line of sight) is available due to blocks of terrestrial objects such as buildings. Thus, it is required to include the area-based state equations and also form prediction equation for geographical set up of the area. However, in this case, pre-defined geographical constraints are provided, thus neglecting any flaws due to geographical bounds. Also, for an efficient network, interference model is required to address the activity involved due to wireless signals. However, the model considered in this paper operates at Layer 2 of ad hoc network, thus neglecting any parameters of underlying Layer 1. Also, interference is not considered due to reason that separate channels are used for transmission analysis that does not allow any conflict between signals that may arise if transmitted over the same channel. It is to be noted that a network model is never fully efficient, thus, for prediction equations, it is necessary to model network errors for providing effective QoS to end users. Let β_m be the measured bandwidth requirement for queue transmission such that

$$\beta_m = \beta_r + R + N_e + E_c \tag{11}$$

where β_r is the actual bandwidth allocated, R is the routing overheads, N_e is average network error and E_c is error due to queue crashing. Queue crashing is defined as a failure in queue formation and transmission due to the unambiguous behaviour of priority queue selector. Note that R, N_e and E_c follow Gaussian distribution. These variables can be estimated using regression models such as Gauss-Markov model. Routing overheads (R) can be estimated using the theory of least squares, i.e. R for i^{th} queue of j^{th} node is defined as:

$$R_{j,i} = \lambda_0 + \lambda_1 t_{j,i1} + \lambda_2 t_{j,i2} + \ldots + \lambda_n t_{j,in} + \epsilon_{j,i} \tag{12}$$

where λ denotes the average network error rate at active time slot $t_{j,in}$ for j^{th} node and ϵ_i is error on the i^{th} queue observation. Also, ϵ is absolutely random and is normally distributive with mean 0 and variance σ^2. Application of least squares over Gauss-Markov model allows conjecture for various parameters as a number of equations formulated for the system overpowers the number of real time values available for the underlying system. Note that the $\lambda_0 + \lambda_1 + \lambda_2 + \ldots + \lambda_n$ are the sum of a set of finite but undefined value since error rate can be affected by random and any of the network parameter. In general, these can be expressed as matrices by taking expectation of above equation as:

$$E[R_j] = R^{'}T^{'} \tag{13}$$

where $R^{'}$ is matrix of network errors and $T^{'}$ is matrix of network time slot. Applying Gauss-Markov model, $R^{'}$ can be expressed over f(R_1, R_2 … R_n) which is an estimator function. Sum of unknown parameters $\lambda_1 + \lambda_2 + \ldots + \lambda_n$ can be expressed as estimator function $h(\lambda_1, \lambda_2, \ldots, \lambda_n)$ such that

$$E[f(R_1, R_2 \ldots R_n)] = h(\lambda_1, \lambda_2 \ldots \lambda_n) \tag{14}$$

where $h(\lambda_1, \lambda_2, \ldots, \lambda_n)$ can also be evaluated using linear estimator as:

$$h(\lambda_1, \lambda_2, \ldots, \lambda_n) = L^{'}R^{'} \tag{15}$$

where $L^{'}$ is the link speed matrix at time slot t_i. Thus, the whole system of multiple quaternion can be expressed as:

$$y(t) = \begin{bmatrix} \overline{w_1}(t) & \overline{w_2}(t) & \overline{w_3}(t) \\ N_e(t) & R(t) & E_c(t) \end{bmatrix}, \tag{16}$$

using definition of quaternion derivation [24] and error model [24], the state governing equation is given as:

$${}_G\overline{\mathbf{w}}^{'}(t) = \frac{1}{2}\Omega y(t)\ \beta_r\ {}_G\overline{\mathbf{w}}(t) \tag{17}$$

where Ω is the vector product matrix [24]. It is to be noted that from actual definition, only global frame is considered. This is specified by the reason that GPS is used for node positioning and no local positioning reference frame is used in this case. Let β_t, β_{t-1}, …, β_1 be the vector that denotes the bandwidth allocation for time t, $t-1$, $t-2$, …, 1 and D_t, D_{t-1}, …, D_1 be the vector that denotes end to end delay for same time slot. Applying Kalman filter [25], following equations are formed:

$$\beta_t = G_t\overline{w}_1(t) + O_t, \tag{18}$$
$$D_t = H_t\overline{w}_2(t) + O_t^{'} \tag{19}$$

where

$$\overline{w}_1(t) = G_t^{'}\overline{w}_1(t-1) + S_t, \tag{20}$$
$$\overline{w}_2(t) = H_t^{'}\overline{w}_2(t-1) + S_t^{'}. \tag{21}$$

Here G_t, $G_t^{'}$, H_t, $H_t^{'}$ denote the set of known parameters that can be obtained from values defining changes in network models over time. These values can be considered constant over time. O_t, $O_t^{'}$ are the observation errors and S_t, $S_t^{'}$ are the system errors that provide values for any kind of network delays. Equations (18)–(21) provide state equations over which Kalman filter can be applied for estimations. It can be noted that these equations are generalized from standard equation of line with slope k, i.e. $y = mx + k$. Thus, using quaternion and then feeding the results into linear estimation system would simplify the prediction process. These predictions can be carried out using simple prediction theory approaches such as Bayes' formulations for probability [30, 25]. Therefore, using similar approach, following state equations can be used for traffic and network state prediction.

$$P(\overline{w}_1(t) \mid \beta_t, \beta_{t-1}) = \frac{P(N_e\text{avg} \mid \overline{w}_1(t), \beta_{t-1}) \times P(\overline{w}_1(t) \mid \beta_{t-1})}{P_{total}(N_e\text{avg} \mid \overline{w}_1(t), \beta_{t-1})}, \tag{22}$$

similarly,

$$P(\overline{w}_2(t) \mid D_t, D_{t-1}) = \frac{P(R_e\text{avg} \mid \overline{w}_2(t), D_{t-1}) \times P(\overline{w}_2(t) \mid D_{t-1})}{P_{total}(R_e\text{avg} \mid \overline{w}_2(t), D_{t-1})}. \tag{23}$$

For overall network estimation, state knowledge equation is given by:

$$P({}_G\overline{\mathbf{w}}'(t) \mid F_t, F_{t-1}) = \frac{P(S_e\text{avg} \mid {}_G\overline{\mathbf{w}}'(t), F_{t-1}) \times P({}_G\overline{\mathbf{w}}'(t) \mid F_{t-1})}{P_{total}(S_e\text{avg} \mid {}_G\overline{\mathbf{w}}'(t), F_{t-1})} \tag{24}$$

where

$$F_t = M_t(\beta_t, D_t){}_G\overline{\mathbf{w}}'(t) + K_t(O_t, O_t') \tag{25}$$

and

$${}_G\overline{\mathbf{w}}'(t) = M_t'(G_t, G_t'){}_G\overline{\mathbf{w}}'(t-1) + K_t'(S_t, S_t'). \tag{26}$$

$N_e$avg, $R_e$avg and $S_e$avg are the average network error, average routing overheads and average system error, respectively. The Equations (24)–(26) allow Kalman filter estimator to be integrated with queue scheduler for efficient allocation of QoS to end users. Let $X_{\beta(t)}$, $X_{D(t)}$ denote the observation matrix for bandwidth and delay estimation, respectively. Over the network with global frame of reference, $X_{\beta(t)}$, $X_{D(t)}$ will be defined using [24] as:

$$X_{\beta(t)} = [\Pi\lfloor {}_G\overline{\mathbf{w}}'(t) f_0(\beta_t)\times\rfloor L'] \tag{27}$$

and

$$X_{D(t)} = [\Pi\lfloor {}_G\overline{\mathbf{w}}'(t) f_0(D_t)\times\rfloor L'] \tag{28}$$

where f_0 denotes the estimator function. The noise error N_0 is defined as:

$$N_0 = (\Omega\; y(t)\; \beta_r) - P_e + \text{Avg}(N_e\text{avg}, R_e\text{avg}, S_e\text{avg}) \tag{29}$$

where P_e is the process error. Let P be the covariance matrix over y(t) and $P'_{\beta(t)}$ and $P'_{D(t)}$ be the covariance over N_0 such that:

$$P'_{\beta(t)} = X_{\beta(t)} P X^T_{\beta(t)} + N_0 \tag{30}$$

and

$$P'_{D(t)} = X_{D(t)} P X^T_{D(t)} + N_0. \tag{31}$$

Hence, using indirect Kalman filter [28], Kalman gain K_β and K_D will be given as:

$$K_\beta = P X^T_{\beta(t)} P'_{\beta(t)}, \tag{32}$$

$$K_D = P X^T_{D(t)} P'_{D(t)}. \tag{33}$$

Therefore, using property of Kalman filter [24, 28], corrections $C_\beta(e)$ and $C_D(e)$ for bandwidth and delay, respectively, will be defined as:

$$C_\beta(e) = K_\beta N_0, \tag{34}$$

$$C_D(e) = K_D N_0. \tag{35}$$

4 PROPOSED MODEL

In the proposed model, admission control and scheduling process were integrated with estimation-based quaternion Kalman filters that served the purpose of successful allocation of the quality of services to end users, thus improving the network quality and sustaining high link speed. The proposed model is based on two major factors: number of presentations units (PU) and maximum number of handover possible in the network. The technique used in the proposed model divides the nodes into a group of transceiver units, each with defined number of channels available for transmission. The presentation units were divided into three categories: High Priority, Regular, and Normal PUs. The packet priority was noticed on type of presentation unit. The receiver at each device arranged the packets on the basis of PUs and fed them into the transmitting queue that was maintained for each device which was using it either for direct transmission or relaying with it for multi-hop transmission. A timely window was deployed that controlled the entry of packets after fixed interval; a weight was assigned to each queue based on priority and size. The queue with higher number of priority packets was assigned more negative weight. For this, the variables of Kalman filters provided the state observation values that were used to predict the network behaviour before queue selection with higher accuracy. Thus, the queue with lowest weight was transferred first followed by another similar process between the next intervals using network prediction chart. The interval can be managed on the basis of bandwidth allocated or available at that instant of transmission time. Another major concern was number of handovers which includes the time to switch between nodes or devices when a particular node goes beyond the range and another node is used for relaying purpose. This has to be maintained to know actual bandwidth available as change in path may cause variation in bandwidth available for allocation to particular channel used by transmitter to transmit data of particular queue. Also, it is required to compute overall delays that may degrade the performance at later stage of transmission. Thus, packets with high priority get transmitted at once followed by those that are regular packets. With extreme high bandwidth, the interval between packet transmissions decreases, hence lowering the amount of memory units to be used for maintaining queue. This benefits during the peak transmission of presentation units. The proposed model, as shown in Figure 8, can ensure following key points:

1. It serves queue with higher number of priority packets as well as with lower weight.

2. Bandwidth allocation is dynamic and therefore, no need to reserve bandwidth, hence, improving the quality of each presentation unit arriving at the end user.
3. Each queue is handled with set transmission criteria. This maintains the link quality, and reliability of the network also stabilizes.
4. It reduces the delays effect due to pre-calculation of queue size and estimation of transmission slot available on particular node.
5. The estimation process provides the self regulation and optimization in bandwidth selection and delay control.

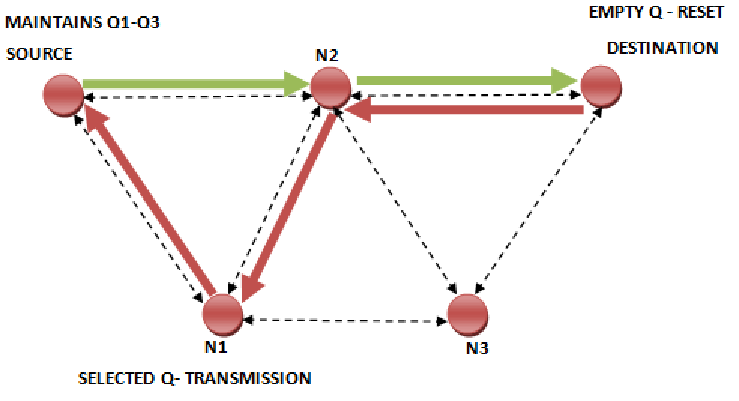


Figure 8. Queue transmission – proposed model

For analysis three queues, namely Q1, Q2, and Q3, for transmission of packets over the network were considered. The configuration of these queues is shown in Table 1.

The weight is assigned on the basis of packet priority. Q1 gets transmitted first followed by Q2 and Q3. From Table 1, it is clear that final weight calculated is used to compare the transmission criteria for queues, however, if the weight would have been same for any of the queues, then the number of high priority packets would have been considered for deciding the transmission order. The pseudo code for queue formation and transmission is given below:

This algorithm is handled by two thread operations, one for queue formation and updating of table while the other handles queue selection and transmission process. Thus, the overall scenario does not lead to any overheads that might have occurred due to single thread handling all operations serially. In general, parallel run of the algorithm saves the time required for queue generation and selection of packets for transmission. The algorithm works by computing number of presentation units to be transferred that are fed into quaternion Kalman filter-based priority checker module which computes the priority of these packets based on configuration as shown in Table 1. This whole process is carried out in $O(k)$ time, where k denotes the

CONFIGURATION	**Q1**	**Q2**	**Q3**
Number of memory slots	8	8	8
Extended memory slots	+5	+5	+5
Transmission limit*	8	8	8
Memory slots occupied	7	8	5
Weight for high priority bit	−3	−3	−3
Weight for regular bit	−2	−2	−2
Weight for normal bit	−1	−1	−1
Number of high priority PU	3	1	1
Number of normal PU	3	4	4
Number of regular PU	1	3	0
Weight calculated	−14	−13	−7

* the transmission limit is defined in terms of PU transferred in unit time slot

Table 1. Queue configuration

Algorithm 1 Queue formation and transmission algorithm

Require: $k \geq 0 \vee N$
  while $i < k$ **do**
    Priority_Checker(PU) ← Kalman_Filter_Estimate(PU)
    Maintain Priority Table(Weight)
    Select Queue with empty slots
    if SelectedQueue > 1 **then**
      Compare Queue with lower weight
      Insert PU ← Kalman_Filter_QueueEstimate(LowerWeightQ)
      Update Priority Table(Weight)
      Transmission Scenario ← Kalman_Filter_Bandwidth(PU)
      Check Network State ← Kalman_Filter_NetworkStatus(PU)
      Time count(T) $= t + \delta t$, where T is total time including reset time t
      Transmit Queue
    end if
  end while
  Reset

number of presentation units. The queue selection for lower weight and inserting presentation unit into queue are done in $O(1)$ time. Thus, for a thread, handling the queue formation, the overall complexity remains to be $O(k)$. For other thread, operating simultaneously, the complexity again remains the same, depending upon the number of presentation units fetched and transmitted over the network. Also, if more such units are used on a single node the complexity becomes $O(n * k)$, where n denotes number of such units on a single node. This can be controlled by running multiple threads for alternative units. But, this may lead to more computation and higher consumption of processing energy. The operational units required for performing the above task are shown in Figure 9.

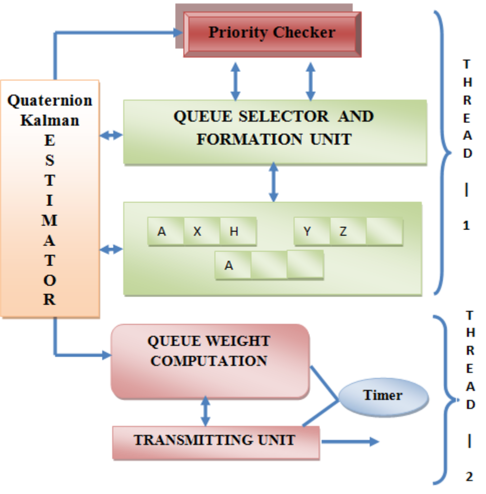


Figure 9. Operational view of proposed scheme

5 BANDWIDTH AND DELAY ANALYSIS

In the proposed model, queue scheduling has been optimized at the node end, but there is always an issue when the transmission is to be considered due to mobile nodes that may vary the transmission process due to various parameters such as delay and bandwidth. Delays can never be removed completely from real-time application, however, delays can be lowered to such a level that these can be neglected. Bandwidth offered to a node also controls the amount of delay in the network that might have occurred due to low performance at a node because of various computations being performed simultaneously. On evaluation, it was analyzed that for better performance the overall delays in the network should not be more than 400 ms as delays beyond this degrades the network performance.

5.1 Bandwidth Evaluation

Bandwidth depends on the number of channels used per node in the overall network. For real-time applications, the number of channels may vary from node to node. The overall network can be classified by the number of channel variants in the network structure. For N number of hops, if the same number of channels are indulged, then the overall channels used will be computed as $N * C$, where C denotes the number of channels used by a node. The numbers of channels to be used are pre-determined and pre-configured. Also, $N * C_i = N$, i.e. $C_i = 1$ where i varies from 1 to N, if single channel is used on each node. Thus, bandwidth will depend on the minimum rate that each node can follow to transmit presentation units and the number of channels used for transmission process. Therefore, required network bandwidth will be calculated as:

$$\beta = \frac{\text{Min}_{rate} + \frac{20\,\%\ \text{of Routing Overheads}}{\text{Loss } Time}}{\text{No. of channel used}} \tag{36}$$

$$= \frac{\text{Min}_{rate} + \frac{20\,\%\ \text{of}\ R_H}{L_T}}{\sum_{i=1}^{N} C_i} \tag{37}$$

$$= \frac{\text{Min}_{rate} + \frac{20\,\%\ \text{of}\ R_H}{L_T}}{N} \tag{38}$$

$$\text{for } C_i = 1, \forall i.$$

Using Equation (34), the estimated bandwidth will be given as:

$$\beta = \frac{\text{Min}_{rate} + \frac{20\% of R_H}{L_T}}{N} + C_\beta(e). \tag{39}$$

The approach will be better for a network with smaller data applications to be handled between the hops, however, in an extensive network, hybrid nodes are present. The set of these nodes differs in configuration, and the type of data that can be transmitted over them depends on the number of channels used and link capacity of each node. The minimum rate of transmission will vary as per the transmission policy of the network. By transmission policy we mean that whether all hops are transmitting at same rate or variable rate. For the same rate, the number of channels used per node will remain the same. Therefore, the power requirement for computation will remain the same for all hops. For network operating on multimedia type of applications, constant rate for each hop can be used. For them, the minimum transmission rate will be selected as follows:

$$\text{Min}_{rate} = \text{MIN}\left(\sum_{k=1}^{N} R_k\right)$$

where R_k denotes achievable rate for each node. Min_{rate} is the minimum rate feasible to each node in the network for transmission process [3].

5.2 Delay Evaluation

Delays in a network are that time slot during which the transmission rate is lower than the certain threshold value defined for successful transmission to the receiver. Delay in a network cannot be aloof completely, however, for enhanced performance, delays should be controlled under certain threshold value. Thus, it is required to identify those factors that will extensively affect the network in terms of performance failing which can cause various network overheads. Considering the same network model with N number of nodes, the mobility can be denoted as $m_1, m_2, \ldots, m_k$, $\forall k \in N$ where k denotes the active number of connections. Also, $d_1, d_2, \ldots, d_k$, $\forall d \in D$ is the distance covered by each node in defined range where D is the end to end distance. The active number of connections are considered keeping in mind that the network delays will be considered only for those nodes that are configured to be active during scenario generation of the network. Let $T_{arrival}$ be the time taken by node to arrive in range, I_t be the network idle time, D_p be the propagation delays, T_{pkl} be the transmission delays, such that

$$\text{Delays}_{(t)} = T_{arrival} + I_t + D_p + T_{pkl}, \tag{40}$$

$$D_p = \frac{Nrt(N-1)N_p}{\beta}, \tag{41}$$

$$T_{arrival} = \sum_{i=1}^{k} \frac{d_i}{m_i}. \tag{42}$$

Therefore,

$$\text{Delays}_{(t)} = \sum_{i=1}^{k} \frac{d_i}{m_i} + I_t + \frac{Nrt(N-1)N_p}{\beta} + T_{pkl}. \tag{43}$$

Equation (41) computes delays related to transmission process. It is the idle time for which no transmission occurs in the network. This can be computed as average wait time of the network structure. Average wait time can be defined as the number of cycles that were utilized without any data transfer. Also, T_{Pkl} is the time loss due to packet loss. It can also include various losses such as fading loss, vegetation loss. For a controlled network, i.e. network in which mobility of node is strictly defined and is manageable, all active nodes cover equal distance during the transmission process. For such network scenario, node approaching time changes as:

$$\sum_{i=1}^{k} \frac{d_i}{m_i} = \frac{d_{avg}}{m_{avg}}. \tag{44}$$

Therefore, in general for this type of mobile node network, the delays will be defined as:

$$\text{Delays}_{(t)} = \frac{d_{avg}}{m_{avg}} + I_t + \frac{Nrt(N-1)N_p}{\beta} + T_{pkl}. \tag{45}$$

Using Equation (35), estimated end to end delay will be given as:

$$\text{Delays}_{(t)} = \frac{d_{avg}}{m_{avg}} + I_t + \frac{Nrt(N-1)N_p}{\beta} + T_{pkl} + C_D(e). \tag{46}$$

Nrt is the number of re-transmissions carried when a packet is dropped or when no ACK is received from the receiving unit. The model proposed in this paper is capable of handling packet admission as well as scheduling tasks. Hence, avoid any congestion in the network. It ensures QoS to end users with improved efficiency over traffic differentiation services. The performance of the model is measured over delays/jitters, average network throughput, link capacity, and variable routing overheads. The working of algorithm shows that bandwidth is optimized and is not wasted as it would have been during pre-reservation techniques. The slots left idle during the transmission process are also utilized. Packet categorization and resource allocation improved the network survivability assuring the successful transmission of packets up to end receiver.

5.3 Network Throughput

In mobile ad hoc network, network throughput varies due to continuous link switching between the nodes. Ideally, network throughput is one of the important parameter that needs to be improved for better resultant transmission in a network. In the network where latency is involved, such ideal transmission is practically not possible. Throughput will improve only if the round trip latency is decreased or maintained at certain limited value. This can be computed either by pre-deciding the limit of the maximum threshold value for round trip latency or by decreasing the transmission delays. If ACK is not a concern for protocol chosen for transmission, i.e. three-way-handshake is not used for transmission process, latency does not need to be doubled as no round trips are involved. Thus, the transmission process is of concern in throughput evaluation. In proposed model, ACK is not a bounded requirement for calculations. Therefore, the evaluation will depend on the delays evaluated for the overall network. Here, the boundary conditions for throughput calculations remain the same for queue formation and transmission. The dependent factors include latency in terms of node selection or transmission handovers over the link, the type and the number of channels incorporated over the links active at transmission time. Thus, network throughput can be evaluated for two types of scenarios:

5.3.1 Common Channel-Based Network

In this scenario, common channel-based network model is considered, which operates on a single type of channel with same active frequency range over each node. Also, the number of channels used on each node is the same. For such type of network model, throughput (Th) can be computed as:

$$Th = \frac{PU}{N_{latency}} \tag{47}$$

where $N_{latency}$ is the network latency computed as:

$$N_{latency} = \frac{(PU - Nrt)}{link_{capacity}} + Cmp_{delays}. \tag{48}$$

Here, Cmp_{delays} are the computational delays for overall transmission process and $\text{link}_{capacity}$ is the maximum traffic a network can support such that:

$$\text{link}_{capacity} = \frac{\text{Min}_{rate}}{\text{Number\_of\_channels\_used}}. \tag{49}$$

Cmp_{delays} are bound to occur even if the network is performing to its capacity. The value of Cmp_{delays} ranges from 8 ms to 10 ms. $N_{latency}$ is the transmission delay that is computed over the link capacity and number of bits to be transferred between the particular link. The throughput computed will be defined only for single link active at time of transmission process. Also, for the network that relies on ACK for error detection and correction, the $N_{latency}$ will be twice for round trip. Since the network model is similar, i.e. each node has the same active channel frequency with the equal number of channels usage. Thus, it is efficient to compute overall throughput of the network as $N_{latency}$ will be same for each link that becomes active during transmission. Therefore, for overall network,

$$N_{latency} = \frac{(PU - Nrt)k}{\text{link}_{capacity}} + \text{Cmp}_{delays}, \forall k \in N \tag{50}$$

$$\text{link}_{capacity} = \sum_{i=1}^{k} \frac{Min_{rate}}{C_i}. \tag{51}$$

The performance of each link in such type of network remains the same. However, it is to be noticed that for such type of network model, even if the achievable rate for some nodes is larger than the minimum achievable rate, still transmission is carried out at minimum rate. This decreases the performance of the network. This type of problem can be overcame by variable channel-based network.

5.3.2 Variable Channel-Based Network

In this scenario, the variable channel-based network model is considered. This chooses transmission rate depending upon the amount of available transmission energy rather than operating at a similar rate. The power consumption for such type of network is higher but the performance is better than for those operating at similar rate. $N_{latency}$ for such type of network will vary from link to link and will be computed as:

$$N_{latency} = \frac{\sum_{i=1}^{k}(PU_i - Nrt_i)}{\text{link}_{capacity}} + \text{Cmp}_{delays}, \forall k \in N \tag{52}$$

where

$$\text{link}_{capacity} = \frac{\frac{1}{R_1} + \frac{1}{R_2} + \ldots + \frac{1}{R_k}}{C_1 + C_2 + \ldots + C_k} = \sum_{i=1}^{k} \frac{\frac{1}{R_i}}{C_i}. \quad (53)$$

With the use of variable channels for transmission process, the power consumption of the network increases, but it can be optimized either by reducing the packet size or by decreasing the number of re-transmissions (*Nrt*). This will allow the node to perform to maximum capacity but it may also decrease the fault tolerance of the network. Thus, energy always remains a constraint for network when QoS is to be improved.

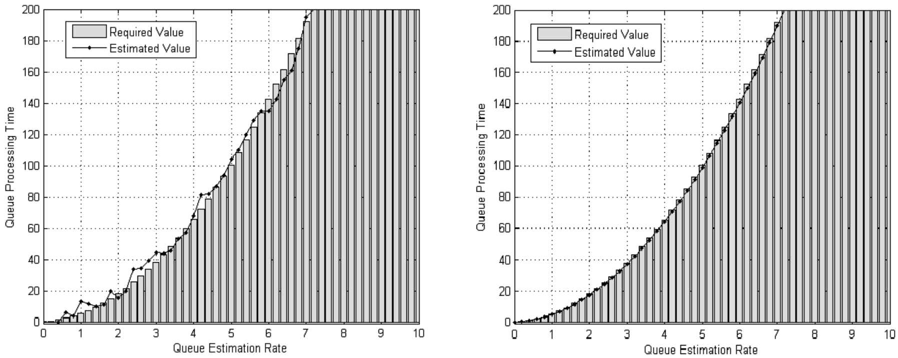
PARAMETER	**VALUE**	**DESCRIPTION**
Dimensions	$1\,500 \times 1\,500$ sq. m.	Area for node placement
Number of nodes	5, 15, 25	Node sets for evaluation
Simulation time	1 000 s	Total evaluation time
Traffic type	CBR-VBR	Traffic over TCP
Number of connections	4, 10, 14	Active number of connections
Packet size	1 024 bytes	Average packet size
Mac layer	802.11 b	IEEE standard for wireless
Buffer size	5 000, 10 000, 15 000	Buffer capacity of each node
Propagation radio model	Two ray ground	Topological propagation model
Maximum speed	30 m/s	Node mobility
Minimum speed	10 m/s	Node mobility
Interval time to send	2 p/s	Time to wait before sending
Number of users	20	Active users across each node
Transmission range	250 m	Transmission distance of node
Node mobility range	$0-2.5$ m	Node steps during movement
Max. channel capacity	8	Queue capacity of each channel
Rate	256 kbps	Initial transmission rate

Table 2. Parameter configuration

6 SIMULATIONS AND RESULT ANALYSIS

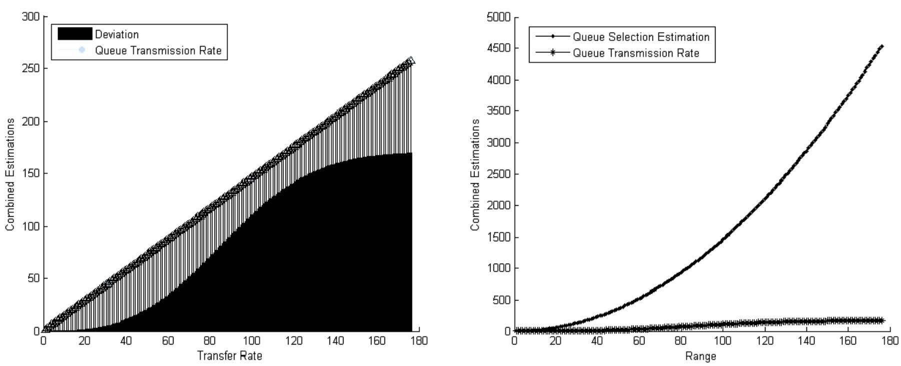
The simulation for proposed model was performed using *NS-2.35* simulator and the result analysis were evaluated using MATLAB. The configuration of the system model, as shown in Table 2, consists of a maximum of 25 nodes with 14 active numbers of connections. Initial bandwidth (β) is set at 256 Kbps. With packet size of 1 024 bytes, a total of 50 simulation runs are made to verify the results and analyze the behavior of the network model. During the first phase of simulations, queue model was evaluated using MATLAB to check the estimation accuracy of quaternion-based Kalman filter. The result was evaluated for queue estimation, bandwidth estimation and the end to end delay estimation over the ad hoc network.

Figures 10 a), 10 b) and 11 provide the graphical analysis of queue estimation. Kalman filter gives more accurate results for queue selection. Similarly, evaluation were carried out for bandwidth and the end to end delay, as shown in Figure 12 a) to Figure 15. The results for estimations were realized and verified by transforming the existing Kalman filter [25, 32] into the quaternion Kalman-based estimator.



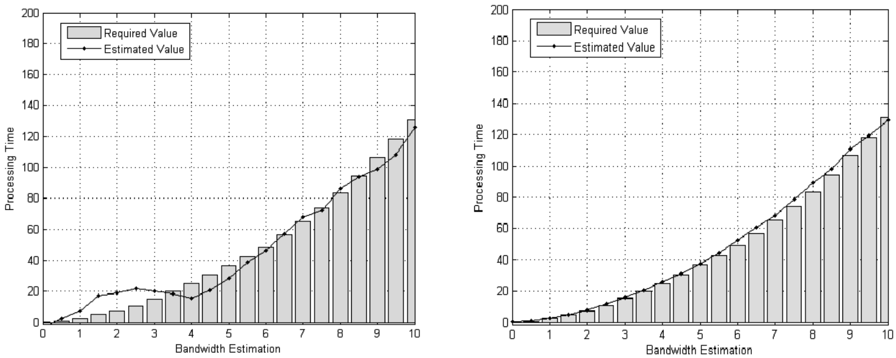
a) b)

Figure 10. Queue estimation comparison, a) Queue estimation without Kalman filter, b) Queue estimation using quaternion based Kalman filter



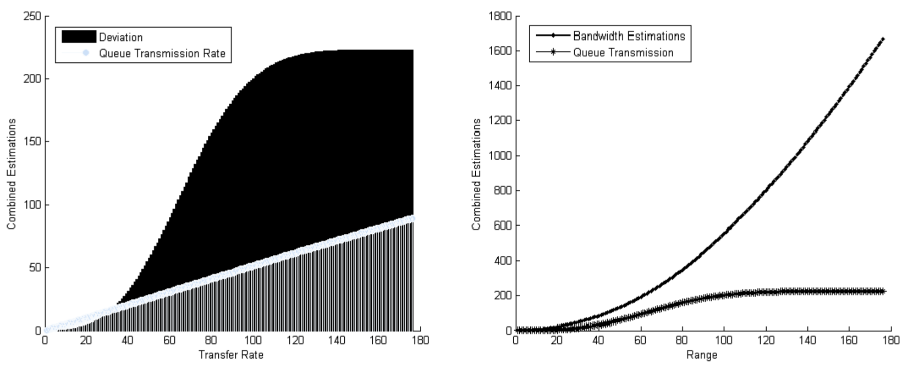
a) b)

Figure 11. Statistical analysis of queue estimation using quaternion Kalman filter, a) Deviation during estimations, b) Estimation improvement during queue transmission



a) b)

Figure 12. Bandwidth estimation comparison, a) Bandwidth estimation without Kalman filter, b) Bandwidth estimation using quaternion based Kalman filter

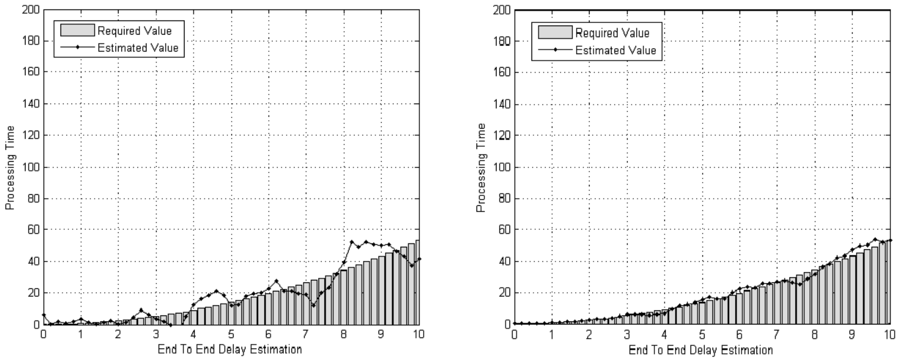


a) b)

Figure 13. Statistical analysis of bandwidth estimation using quaternion Kalman filter, a) Deviation during estimations, b) Estimation improvement during queue transmission

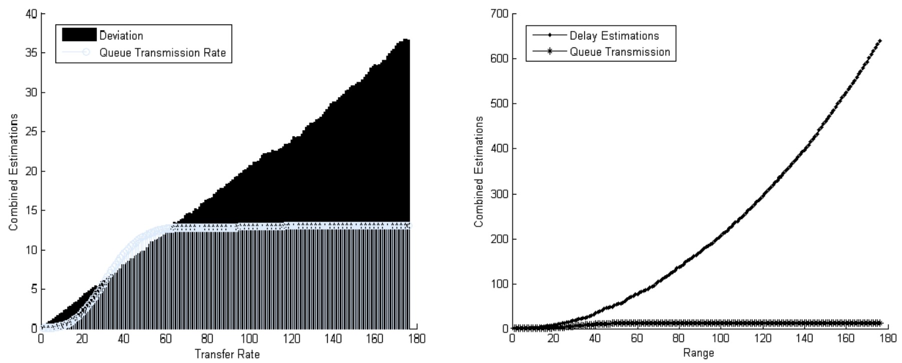
Bandwidth and delay predictions are difficult to estimate as dynamic changes in the network directly affect these parameters. Despite of that, quaternion-based Kalman filter was able to estimate these values with lesser error corrections.

Comparative analysis has been carried out on the basis of end to end delay and bandwidth consumption for queue selection and transmission of presentation unit over the configured network with given initial transmission rate. The variation in the class used for transmission is carried out that helps in understanding the network performance. The graphical representation of bandwidth, end to end delays, throughput and link capacity for McLaughlin, weighted fairness queue model, $M|M+|1$, proportional fairness queue model and our proposed model is shown



a) b)

Figure 14. Delay estimation comparison, a) Delay estimation without Kalman filter, b) Delay estimation using quaternion based Kalman filter



a) b)

Figure 15. Statistical analysis of end to end delay estimation using Kalman filter, a) Deviation during estimations, b) Estimation improvement during queue transmission

in Figure 16, Figure 17, Figure 18 and Figure 19, respectively. For comparison, PU ranging from 5 000–50 000 was transmitted over the network. It was concluded that for McLaughlin model, average bandwidth wastage (no transmission usage) was 3.75 Mb/s, WFQ consumed 3.09 Mb/s, $M|M+|1$ consumed 2.78 Mb/s, proportional fairness consumed 2.30 Mb/s and proposed model consumed 1.25 Mb/s, also, the same input and scenario were considered for comparing end to end delay in the network models.

Average end to end delay for McLaughlin model was 491.5 ms, for WFQ was 388.5 ms, for $M|M+|1$ was 378.3 ms, for proportional fairness queue modes was 343.2 ms and for proposed model was 315.1 ms. Average throughput computed for

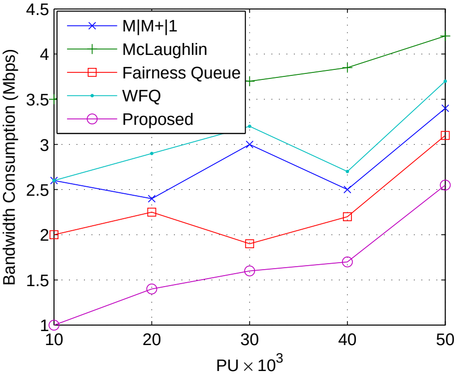


Figure 16. Bandwidth consumption vs. PU

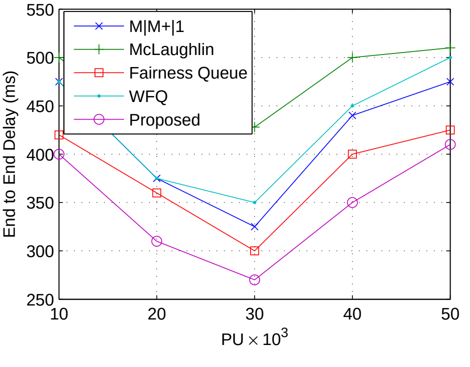


Figure 17. End to end delays vs. PU

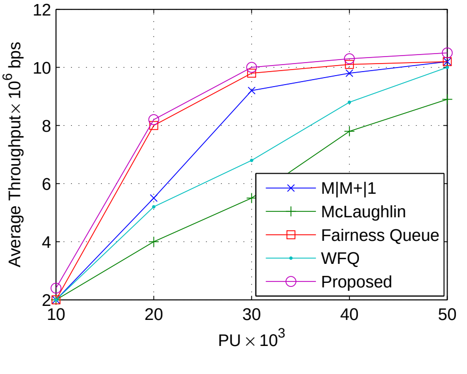


Figure 18. Throughput vs. PU

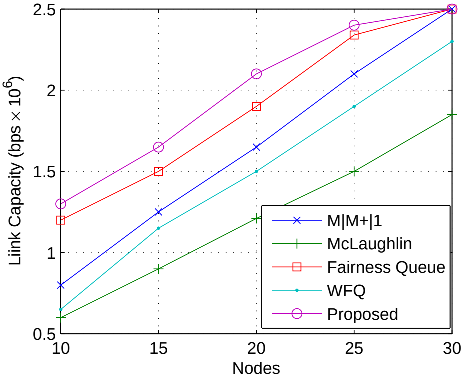


Figure 19. Link capacity vs. nodes

McLaughlin model was 5 332 Kb/s, for WFQ was 6 351 Kb/s, for $M|M+|1$ was 7 095.7 Kb/s, for proportional fairness queue was 7 688.735 Kb/s and for proposed model was 8 130.64 Kb/s. For bandwidth consumption, proposed model shows improvement of 66 %, 59 %, 55 %, 45 % approximately, as compared to McLaughlin, WFQ, $M|M+|1$ and proportional fairness model, respectively.

COMPARED WITH	**Bandwidth**	**End to End Delay**	**Throughput**
McLaughlin's model	66 %	34 %	35 %
WFQ model	59 %	24 %	23 %
$M\|M+\|1$ model	55 %	14 %	19 %
Proportional fairness model	45 %	12 %	15 %

Table 3. Percentage improvement in the proposed model

For end to end delays, proposed model shows improvement of 34 %, 24 %, 14 %, 12 % approximately, as compared to McLaughlin, WFQ, $M|M+|1$ and proportional fairness model, respectively.

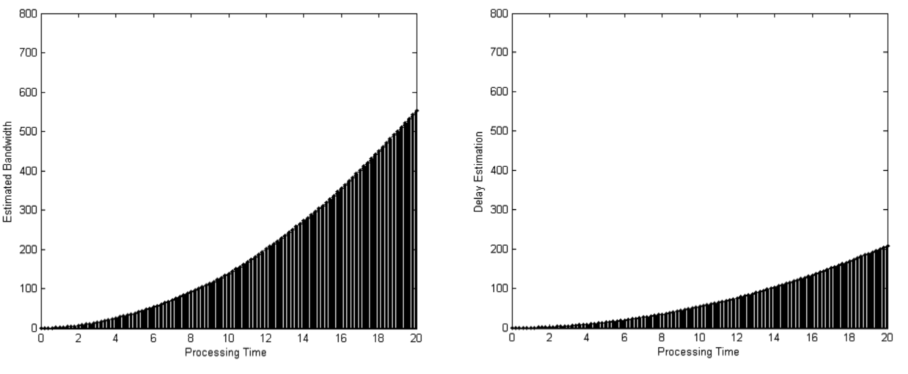
For throughput, proposed model shows improvement of 35 %, 23 %, 19 %, 15 % approximately, as compared to McLaughlin, WFQ, $M|M+|1$ and proportional fairness model, respectively. From the analysis recorded, as shown in Table 3, it can be noticed that our model consumes less bandwidth in queue formation and transmission, also, delays are maintained at certain threshold value such that network performance is not much affected.

7 REAL TIME ANALYSIS FOR QUEUE SCHEDULING

Real time analysis for proposed model were carried in order to demonstrate its effectiveness. The multiple system-based ad hoc modules were programmed and interfaces for systems were developed to evaluate the proposed model. The configuration details of the real time analyzer are shown in Table 4.

INTERFACE	**DESCRIPTION/VALUE**
Host	Back track 192.168.153.159
Intermediate nodes	Linux kernel, virtual machines
Total intermediate node available	8
Maximum hop distance	5
Number of programmable modules	7
Routing protocol	AODV
Threads generated	10 each node
User requests	10 each node
Monitoring interval	5 ms
Simultaneous threads for estimation	10

Table 4. Parameter configuration



a) b)

Figure 20. Bandwidth and end to end delay estimation comparison, a) Bandwidth estimation, b) End to end delay estimation

The results were recorded for bandwidth and delay estimation. Figures 20 a) and 20 b) present the estimation process for bandwidth and end to end delay during real-time analysis. The red line represents the estimated values over black coloured transmission values. It was observed that transmission rate was sufficient enough using quaternion Kalman filter to sustain with network delays. No transmission halts were observed during the transmission. The network was able to sustain topology changes which were created by turning off the intermediate systems and switching to virtual machines in between the transmission process. The scenario changes caused drops in bandwidth, but it was able to continue transmission without affecting the queue scheduling process.

8 CONCLUSION

Estimation-based queue scheduling proposed in the paper is capable of handling large traffic and ensuring QoS to end users. It is noticed that completely lowering the delays is not practically possible, however, delays can be controlled to go beyond certain threshold range. The transmission scheme was integrated with quaternion-based Kalman filter that estimated the queue selection, performed traffic prediction and delays estimations with higher accuracy. QoS oriented queue scheduling always remains an issue to optimize as more and more real-time applications are being developed for ad hoc networks. The proposed model offered relatively high data transfer rate with delays controlled within a certain range. The approach proposed can be utilized for improving data rate for several applications in MANETs. Also, the accuracy with which prediction is performed can allow MANETs to be used under crucial circumstances. A real-time analyzer was also created to perform network analysis for prediction over real traffic. The analysis shown in paper justifies that estimation-based queue scheduling is versatile and can adopt to any hard bound multimedia applications.

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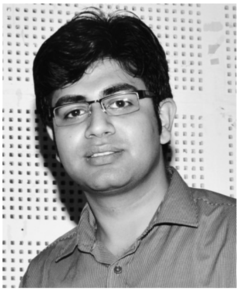

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