# Cross-Layer Optimization for Video Streaming Applications over IEEE 802.11 Mesh Networks

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**Abstract—In recent years, multihop wireless mesh networks have emerged as the next evolution in wireless network technology. The primary reason for this sudden emergence emanates from the advantages provided by the mesh networking technology. Wireless mesh networks provides ease of installation, cost effective deployments, high level of scalability in a coverage area and capacity, network flexibility and self-configuration capabilities. Furthermore, wireless mesh networks are expected to provide seamless and ubiquitous wireless communication. However, despite all these advantages, many research challenges still remain in wireless mesh networks. One such challenge is the support of real time applications such as video streaming. This paper presents a mechanism for the transport of real-time video using a cross-layer optimization technique. In this mechanism, rate adaptation is implemented in the data link layer for channel error control, link stability and reliability. In the network layer, the routing protocol is optimized for congestion control and optimal route selection by using congestion information from the data link layer and link quality metric from the network layer. Simulation results show that the proposed mechanism improves the performance of multihop wireless mesh networks when UDP is used as the transport protocol.**

**Index Terms—Cross-Layer, Video Streaming, Rate Adaptation, Routing Protocol.**

## I. Introduction

Wireless Mesh Networks (WMNs) have gained immense research interest from the wireless networks research society in recent years. This sudden interest emanates from developments which indicate that WMNs can offer ubiquitous communication and seamless broadband applications. WMNs are hybrid networks considered to be variants of Wireless Ad Hoc Networks (WANETs) as they are built based on a mixture of fixed and mobile nodes interconnected via wireless links to form a multihop WANET [1],[2]. Thus, WMNs inherit most, if not all of the multihop WANETs characteristics. There are numerous possible video streaming applications over multihop wireless mesh networks. Source of the applications that can be supported include spontaneous video conferencing in a place without wireless infrastructure, Internet Protocol Television (IPTV), Video on Demand (VoD), video transmission on the battlefield, search and rescue operations, security and surveillance systems [3]. However, there are still several research challenges that need to be addressed in all protocol layers for WMNs to support video streaming applications.

Transmission Control Protocol (TCP) and User Datagram Protocol (UDP) are two de facto transport layer protocols used within the wireless network technology. The majority of research studies based on these protocols reveal that TCP is unsuitable for real-time applications [4],[5]. TCP incorporates retransmissions and congestion control mechanisms. These mechanisms results in additional delays and jitter. Video streaming applications can tolerate a certain level of packet losses. However, they are very sensitive to delays and jitter. Moreover, congestion window based transport protocols are typically related to bursty data rates. This characteristic makes them unsuitable for video streaming applications which normally prefer steady data rates. For these reasons, UDP is more preferred as a transport layer protocol for real time applications.

However, a significant amount of packet loss in wireless networks is due to wireless channel errors and network congestion. UDP does not incorporate any error recovery mechanisms utilized by TCP. Thus, video streaming in wireless networks using UDP protocol can result in unpredictable degradation and poor video quality. In addition, UDP throughput degrades in wireless multihop networks with an increased hop count from the sender to the receiver. This behaviour is expected since packet error probability is maximized when more hops are encountered from source to destination [6]. To improve the performance of IEEE 802.11 multihop wireless mesh networks for video streaming applications, a mechanism that optimizes throughput, minimizes packet loss and improves end-to-end delays for UDP protocol is needed. In this paper, a cross-layer optimization technique based on the data link layer and network layer for UDP optimization is proposed. IEEE 802.11b is used as the benchmark standard to evaluate the proposed cross-layer mechanism. However, the cross-layer design can be easily adapted to IEEE 802.11a/g standards with slight modifications.

The rest of this paper is organized as follows. In Section II, related work is briefly discussed. Section III presents the rate adaptation algorithm. Section IV, presents the routing protocol optimizations. In Section V, simulation results are shown. Section VI concludes the paper and discusses future work.

## II. Related Work

Several approaches have been proposed to improve the performance of UDP in wireless multihop networks. Bansal et al. [6] proposed a WANET routing protocol that selects optimal routes based on the link bandwidth. The proposed protocol exploits the multi-rate capability of IEEE 802.11 wireless cards to select high bandwidth routes. For instance, if two routes exist between source and destination, if one route has a total bandwidth of 5.5Mbps and the other route 11Mbps, then the latter would be selected. The rationale behind this mechanism is that minimum hop count is not sufficient as a routing metric. The inefficiencies of minimum hop count are indicated in [1], [7]. Larzon et al. [8] adopted a different approach in which they propose a modified version of the original UDP protocol termed UDP-Lite. UDP-Lite uses a partial checksum by dividing packets into sensitive and insensitive parts in the UDP header. Errors in the sensitive part results in dropped packets, while errors in the insensitive part results in packets not being dropped. This mechanism has two major drawbacks. Firstly, UDP-Lite is backward incompatible with conventional UDP in that it requires modifications to be made on the conventional UDP to inter-work with it. Thus, UDP-Lite can only inter-work with UDP-Lite capable applications. Secondly, due to the partial checksum, application instabilities may result as certain applications may be incapable of handling erroneous packets.

Boyce et al. [5] proposed a UDP protocol known as Complete User Datagram Protocol (CUDP). CUDP uses channel error information obtained from the Physical (PHY) and link layers in the protocol stack to assist with error recovery at packet level. The disadvantage of this approach is that it requires information exchange from Radio Link Protocol (RLP) layer to Point-to-Point Protocol (PPP) / Internet Protocol (IP) / UDP and application layer. However, currently, information exchange from RLP to PPP/IP/UDP is not supported by the protocol stack. Thus, a redesign of the protocol stack is required. Lin [21] proposed an adaptive video transmission mechanism to alleviate the inefficiencies presented by unreliable transport protocols. The proposed mechanism makes use of the loss-rate and Round-Trip Time (RTT) metrics to determine if packet loss is caused by congestion or unreliable medium in the wireless network. The sending rate of the source is then adjusted accordingly based on the thresholds that are established for the metrics. The sending rate adjustment is accomplished by selectively dropping less important frames according to H.263 codec specification. The work that is most closely related to this study is Yang [9] in which the author adopts a cross-layer design approach similar to this study. However, in contrast to [9], rate adaptation is implemented without any modifications to the IEEE 802.11 Medium Access Control (MAC) layer design in the protocol stack in this study. In addition, the routing protocol in [9] uses minimum hop count metric which has been shown by [1],[7] to be inefficient. In this study, link quality metric which optimizes throughput, reduces packet loss and improves end-to-end delays is integrated into the routing protocol for optimal path selection.

## III. Rate Adaptation

The cross-layer optimization technique in this study comprises of two components: *rate adaptation* and *routing optimizations*. In this section, the technique that is used for rate adaptation and its construction is described. In the following section, the routing protocol optimizations and implementations are described. The illustration of the proposed cross-layer technique is shown in Figure 1.

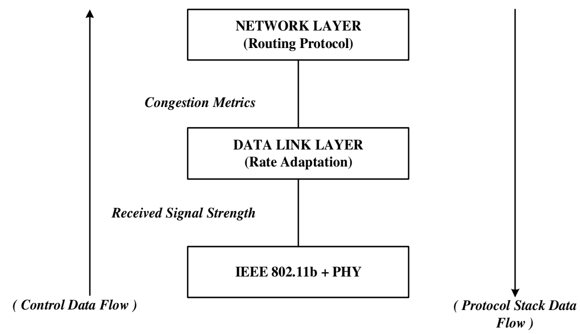


**Figure 1: Implementation of the Cross-Layer Design in the Protocol Stack**

The rate adaptation technique used in this study is based on the Signal-to-Noise ratio (SNR) automatic rate control technique discussed in [10]. The principle of this rate adaptation technique is to dynamically change the data transmission rate based on the SNR which is used as the channel quality estimation parameter. Bit-Error Rates (BER) and Frame Error Rates (FER) are dependent on the SNR in IEEE 802.11 networks. Consequently, throughput is dependent on FER and BER. Thus, an appropriate transmission rate (modulation scheme) for specific throughput can be selected based on the SNR. This is achieved by establishing specific SNR thresholds. These thresholds enable the rate controller to dynamically select an appropriate transmission rate for an optimal throughput

The rate adaptation scheme comprises of three components: *channel model, throughput model* and *automatic rate controller*. The channel model is utilized for channel quality estimations based on channel conditions. The throughput model estimates the attainable throughput based on the channel quality estimate. The rate controller automatically selects an appropriate transmission rate based on the channel quality estimation and the maximum attainable throughput.

### A. Channel Model

The channel model selection is based on the practical experimental evaluations of the log-distance model conducted in [11],[12]. Practical measurements were conducted using off-the-shelf IEEE 802.11 hardware equipment to validate the use of the log-distance model in simulations. The experimental results showed that the log-distance model can be used successfully in simulations to yield accurate results. The log-distance channel model is utilized in the rate adaptation scheme. It is expressed as follows.

$$\overline{P_{Loss}(dB)} = \overline{P_{Loss}(D_0)} + 10 \cdot \alpha \cdot \log\left(\frac{D}{D_0}\right) , \quad (1)$$

where $D_0$ is the close-in reference distance, $D$ is the distance between the source node and the destination node, and $\alpha$ the path loss exponent specified in [13].

### B. Throughput Model

The throughput model for the rate adaptation is based on [10],[14],[15]. Throughput is a function of packet length and the average time taken to transmit that packet. In IEEE 802.11, a packet is enclosed in a MAC Service Data Unit (MSDU) from the network layer to the MAC layer as shown in Figure 2 which illustrates the IEEE 802.11 transmitted frame format. Thus, the packet length refers to the MSDU length and the average time taken to transmit a packet refers to the time taken to transmit a Physical Layer Convergence Procedure Protocol Data Unit (PPDU).

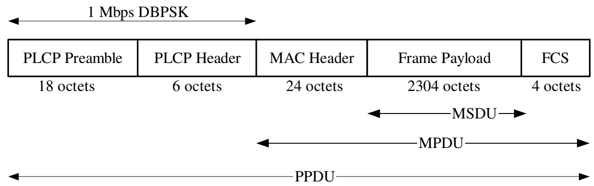


**Figure 2: IEEE 802.11 Transmitted Frame [15]**

$T_{PPDU\_frame}$, which is the average time taken to transmit a frame including retransmissions is modeled under the MAC protocol Distributed Coordination Function (DCF) mechanism with Request to Send (RTS) and Clear to Send (CTS) disabled [9]. To model $T_{PPDU\_frame}$, the probability of a frame error needs to be known. This is given by the following expression.

$$P_{PPDU\_Error} = 1-\left(1-P_{Overhead\_Error}\right)\cdot\left(1-P_{MPDU\_Error}\right) \quad (2)$$

$P_{Overhead\_error}$ is the physical layer overhead error probability and $P_{MPDU\_Error}$ is the MAC Protocol Data Unit (MPDU) error probability. The physical layer overhead error probability is given by the following expression.

$$P_{Overhead\_Error} = 1-\left(1-P_{b\_err1}\right)^{(18+6)\cdot 8}, \quad (3)$$

where $P_{b\_err1}$ is the *bit error probability* when transmitting the physical layer overhead. The physical layer overhead is always transmitted on 1Mbps Differential Binary Phase Shift Keying (DBPSK). The MPDU error probability is given by the following expression.

$$P_{MPDU\_Error} = 1-\left(1-P_{b\_err2}\right)^{(24+4+MSDU)\cdot 8}, \quad (4)$$

where $P_{b\_err2}$ is the *bit error probability* when transmitting the MPDU. $T_{PPDU\_frame}$ is then expressed as follows.

$$T_{PPDU\_frame} = T_S(0)+\sum_{i=1}^{+\infty}\left(1-p_{PPDU\_Error}\right)\cdot\left(P_{PPDU\_Error}\right)^{i}\left[\sum_{j=0}^{i-1}T_{\varphi}(j)+T_{\psi}(i)\right]\mu s, \quad (5)$$

where $T_{\varphi}(i)$ is the time taken to successfully transmit a frame including retransmissions and $T_{\psi}(j)$ is time between two consecutive frame transmissions after a frame transmission failure. Throughput is then expressed as follows.

$$Throughput(L_{MSDU}, T_{PPDU\_frame}) = \left(\frac{L_{MSDU}\cdot 8}{T_{PPDU\_frame}}\right)Mbps, \quad (6)$$

where $L_{MSDU}$ is the length of the MSDU in bytes

### C. Automatic Rate Controller

For the automatic rate controller, a simple threshold based technique was used. In this technique, the transmission rate is chosen based on the channel quality estimation and the maximum attainable throughput under the estimated channel quality conditions. The Receive Signal-Strength (RSS) is used as the channel quality estimator since it is a more practical parameter that is available than SNR in the simulations. The transmission power of each MSDU is known. Therefore, the receive power (RSS) is estimated using the channel model indicated in equation (1). The RSS is then mapped to a transmission data rate based on the estimated maximum attainable throughput modeled in equation (6). The mapping of the channel conditions to a transmission rate is achieved by obtaining thresholds from the throughput vs. RSS curves. These curves are plotted and discussed in section IV.B.1.

## IV. ROUTING OPTIMIZATIONS

To optimize the routing protocol in the network layer for congestion and optimal path selection, the Dynamic Source Routing (DSR) protocol [16] is modified to utilize congestion information from MAC layer and link quality metric in the network layer.

### A. Congestion Metric

To obtain congestion information at each node a method presented in [17] is used. In this method, two congestion metrics are used. The node's MAC layer utilization and the instantaneous transmission queue length. MAC layer utilization indicates the node's usage of the medium in the surrounding area. The usage of the medium is established by monitoring the operation state of the node. In this study, two operation states are used for the node, the busy state and the idle state. The instantaneous MAC layer usage is considered 0 when the node is idle and 1 when the node is busy. To indicate the usage of the wireless medium in this study, it is averaged within a run time period of 10ms in the simulations.

Sometimes congestion occurs due to queue length packet backlog, and this may result in increased packet delays or dropped packets due to capacity limitations imposed on the queue length. Thus, instantaneous transmission queue length is used to indicate congestion triggered by queue length capacity.

### B. DSR Congestion Based Modification

To enable the DSR protocol to use the congestion metrics discussed in the previous subsection, the route discovery mechanism of the protocol is modified as discussed in [9],[17]. On reception of the Route Request (RREQ) packet, the router determines the packets intended destination. If it is the destination, it processes the packet and returns a Route Reply (RREP) as in normal DSR operation. However, if RREQ is not directed to the node, it then checks the congestion metrics. If the congestion metrics indicates that there is a high level of congestion around the node, it silently discards the packet instead of forwarding it. This is because an additional flood of RREQ packets in a highly busy medium can only result in congestion aggravation. Thus, it is desirable for the discovered routes to bypass the congestive nodes to improve the performance of the network.

### C. Link Quality Metric

Most existing WANETS network layer routing protocols commonly use minimum hop count (shortest route) as a routing metric. However, the minimum hop count metric presents several challenges for the wireless network [1],[7]. Firstly, it assumes that links are binary (meaning that they either exist or do not exist). Thus, a link that is not performing reasonably well for data is not ideal to broadcast packets. Secondly, the shortest route increases the hop range. Hence, the power received is degraded and packet loss is increased. Thirdly, several best shortest routes with equivalent distances but dissimilar characteristics may exist in a compact wireless network. Thus, a proper selection of the appropriate route is unlikely to be made. To alleviate inefficiencies provided by the minimum hop count metric, the Expected Transmission Count (ETX) metric was proposed in [7]. The ETX metric is a link quality metric that utilizes delivery ratios to overcome shortest route inefficiencies. It is calculated using the following expression.

$$ETX = \frac{1}{d_{forward} \text{ X } d_{reverse}}, \tag{7}$$

where $d_{forward}$ is the forward delivery ratio and $d_{reverse}$ is the reverse delivery ratio. The $d_{forward}$ is acquired by evaluating the probability of successful transmissions and the $d_{reverse}$ is acquired by evaluating the probability of acknowledged transmissions.

The two ratios $d_{forward}$ and $d_{reverse}$ are acquired using dedicated link probe packets of fixed size sent over a period $\tau$. The source ratio at any given instance $T$ is expressed as follows.

$$r(t) = \frac{Lp_{Count}(T - \lambda, T)}{\lambda / \tau}, \tag{8}$$

where $Lp_{Count}$ $(T-\lambda,T)$ is the count of fixed size packet probes received within the interval $\lambda$. The probe packet comprises of a count of probe packets received by the source from the surrounding nodes within the last $\lambda$ period. Thus, the receiver can calculate the $d_{forward}$ to the node which sent the probe.

### *D. DSR Link Quality Based Modification*

To overcome the minimum hop count problem and enable the routing protocol to incorporate the ETX metric, the DSR Route Discovery mechanism is modified. RREP packets which emanate from RREQ's are alternatively stored in a link cache rather than a route cache that is utilized by the basic DSR. The route cache stores information about the entire route from source to destination, while the link cache enables the separate storage of data about each connection in a source to destination path.

To deal with the link asymmetry problem, the nodes within the wireless network keep a list that contains nodes with one-way flows to unlisted nodes. If an RREP is unable to reach node B from node A, node B is registered on the list with a status indicating one-way flow probability. Thus, node A will no longer route on RREQ's sent from node B. If one-way flow is not ascertained between node A and node B for a significant period, the status is changed to questionable one-way flow. Thus, if node B sends an RREQ to node A, node A will maintain the RREQ and reply to node B with its own RREQ. If node B acknowledged receipt of the RREQ, then the original RREQ is sent onwards by node A, and node B is erased from the list. However, if the RREQ from node A is not acknowledged, node A discards the original RREQ from node B. Nodes are erased from the list when two-way flow is established.

To enable DSR to incorporate the ETX metric, link loss ratios are measured as indicated in sub-section C. In the simulations, IEEE 802.11b probe packets comprising of a payload with 134 bytes is used. In addition, a probe packet is sent every millisecond and $\lambda$ is set at 10 milliseconds. If an RREQ originating from node A is routed onwards to node C by node B, node B's address and the link quality metric evaluated over the link between node A and B are attached to the RREQ forwarded to node C. The link cache is then updated. This information will be added to the RREP which is sent back to node A. If the same request is initiated, the RREQ will be rerouted again if the link quality metric is better than the previous one. A scheme known as the Djikstra algorithm is used to select the best path that minimizes ETX from source to destination.

## V. PERFORMANCE EVALUATION

In this section, the performance of the cross-layer system is evaluated and validated through simulations. MATLAB 7.1 [19] is used for rate adaptation evaluations and the *NS-2* simulator version 2.29 [20] is used for routing optimization evaluations.

### *A. Simulation Methodology*

In order to effectively evaluate the performance of the cross-layer system, the rate adaptation simulation is conducted separately to obtain the rate controller thresholds. These thresholds are then incorporated into the *NS-2* simulator to evaluate the performance of the routing protocol cross-layer optimizations. In the rate adaptation simulations, the nodes operate under the IEEE 802.11b DCF MAC layer scheme. For routing optimizations, an IEEE 802.11b-based *NS-2* WMN network simulation model is created by randomly placing 4 nodes in a 500m x 500m square area as shown in Figure 3. In the network model, node 0 is the video source, node 3 is the destination node and the rest of the nodes are intermediate nodes. The transmission range of each node is set to 100m. In this paper, a fixed WMN is considered. As a result, all the nodes were stationary during the simulations. A UDP agent is used to stream the video frame packets from the video source to the destination node. The maximum transmission packet size is 1024 bytes. A high quality video trace file called Verbose StarWarsIV available in [18] is used. The video trace file is MPEG4 encoded and contains 89998 frames. The frames comprise of 7500 I-frames, 22500 P-frames and 59998 B-frames. The frame rate was set to 30 fps.

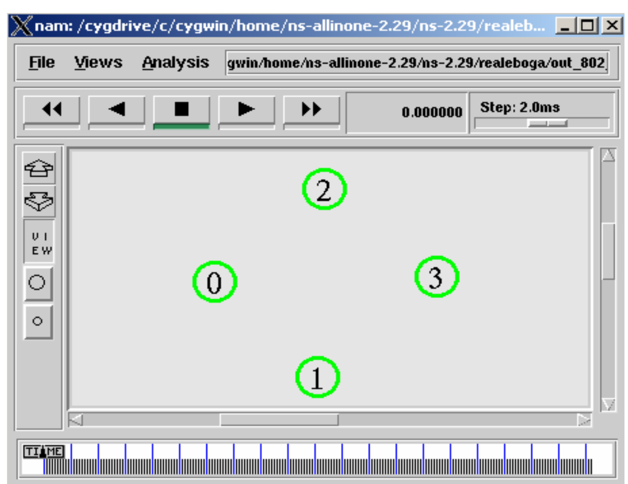


**Figure 3: Network Simulation Model**

## B. Simulation Results

This subsection presents the cross-layer system simulation results based on the simulation methodology discussed previously.

### 1. Rate adaptation

For rate adaptation, the throughput vs. RSS curves is implemented using the channel model and the throughput model discussed in section III. The values of the parameters used are shown in Table 1.

**Table 1: IEEE 802.11b Simulation Parameters [20]**

| Parameter | Value |
|---|---|
| SLOT | 20µs |
| SIFS | 10µs |
| DIFS | 50µs |
| PLCP Preamble | 144µs |
| PLCP Header | 48µs |
| CWmin | 31 |
| CWmax | 1023 |
| MSDU Length | 1500 bytes |
| Tx Power | 100 mW |
| Tx Distance | 500 m |
| Path Loss Exponent | 2.8 |

The throughput vs. RSS curves are shown in Figure 4. The results shown in Figure 4 were obtained using Equations (2), (3) and (4) shown in section III and Equation (13)-(25) found in [15] which give the bit error probability when transmitting an MPDU for each modulation scheme. The attainable throughput for different transmission rates based on the channel quality conditions (RSS) is shown. It is also shown in Figure 4 that the actual expected throughput for each transmission rate is significantly reduced. This is induced by the MAC and PHY layer overheads such as MAC header, PLCP preamble and PLCP header.

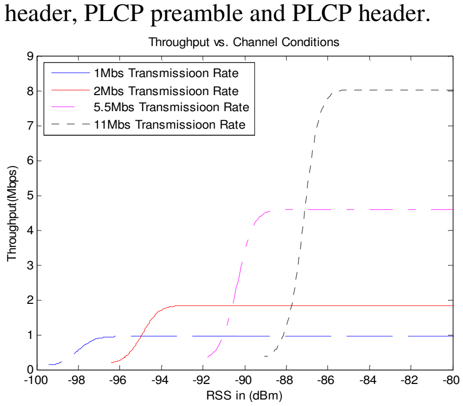


**Figure 4: Throughput vs. RSS Curves for IEEE 802.11b**

The rate controller thresholds extracted from Figure 4 are indicated in Table 2. These thresholds are used in *NS-2* simulations indicated in Figure 4.

**Table 2: Rate Controller Thresholds**

| Rate (Mbps) | Rx Signal Strength (RSS) |
|---|---|
| 1 | -95 dBm > RSS |
| 2 | -95 dBm < RSS < -90.5 dBm |
| 5.5 | -90.5 dBm < RSS < -88.25 dBm |
| 11 | -88.25 dBm < RSS |

### 2. Routing Protocol

For routing protocol optimizations, the DSR protocol discussed in sections IV is modified and implemented in *NS-2* simulator. For congestion based modifications, the MAC layer utilization threshold is set at 30% and the interface queue length threshold is set at 50. These congestion metric thresholds values are used to initiate protocol optimization.

### 3. Results and Discussions

The performance comparison of a network that does not implement the proposed cross-layer system and a network that incorporates the proposed cross-layer system is conducted to evaluate the performance of the proposed cross-layer system. The performance evaluation is carried out under high level congestion indicated by the congestion metrics thresholds values shown in the previous subsection. Throughput, delay and jitter are used as network performance metrics.

The graphs are plotted using the video source trace file and the destination node trace file generated by *NS-2* simulator after 80ms run time.

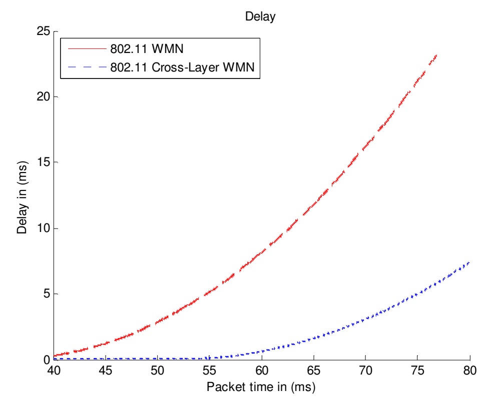


**Figure 5: End-to-end Delay vs. Packet time**

Figure 5 illustrates the achieved average delay. At 75ms packet time an average delay of 5ms was achieved for the cross-layer optimized network whilst 21ms was achieved for the network without cross-layer optimization. This results in a difference of 16 ms which is a considerable amount of delay in video streaming applications.

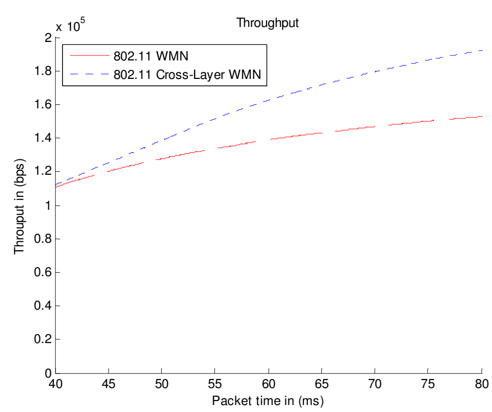


**Figure 6: Throughput vs. Packet time**

In Figure 6, at 75 ms packet time, 150 Kbps throughput for WMN without cross-layer optimization is achieved whilst 170 Kbps throughput for cross-layer optimized network is achieved. Thus, a 13% improvement is achieved by the cross-layer technique.

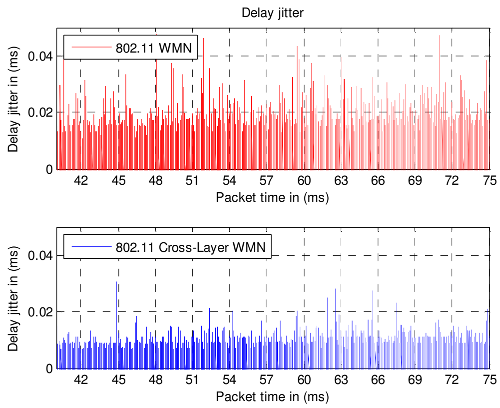


**Figure 7: Delay Jitter vs. Packet time**

In Figure 7, it is shown that the average jitter for the cross-layer optimized system is significantly lower when compared to the network without cross-layer optimization. For a network without cross-layer design an average peak of 0.016 ms is achieved. This is significantly higher when compared to an average peak of 0.01ms achieved in cross-layer optimized system. It is shown that the overall network performance is improved when cross-layer design is incorporated into a WMN.

## VI. Conclusion

In this paper, rate adaptation, and congestion aware and link optimized routing are leveraged to implement a cross-layer technique that optimizes the UDP protocol for video streaming applications in wireless mesh networks. Congestion metrics and link quality metrics are integrated into the DSR protocol for optimal route selection in the network layer. Furthermore, rate adaptation is established in the data link layer for link reliability. Simulation results show that cross-layer techniques focused on the data link layer and network layer can be efficiently used to improve unreliable and congestion prone transport protocols such as UDP for delay sensitive and throughput constraint applications such as video streaming in wireless mesh networks. The results achieved show that better performance metrics need to be developed for routing protocols to be optimized for IEEE 802.11 multihop wireless mesh networks. They also show that multiple performance metrics needs to be integrated in routing protocols to facilitate the alleviation of video streaming challenges associated with WMNs.

## Acknowledgements

The authors would like to thank the Telkom Centre of Excellence Program and F'SATIE at Tshwane University of Technology for making this research possible.

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