QoS Support for Video Transmission in Wireless Mesh Networks

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Abstract

To satisfy the quality of service (QoS) needs of video transmission in wireless mesh networks remains a significant research challenge, due primarily to link bandwidth limitations. This paper focuses on multi-channel multi-radio network technology and presents a comparative evaluation of two transmission algorithms that represent two alternative approaches to achieve QoS support. The Weighted Cumulative Expected Transmission Time (WCETT) algorithm aims to select an end-to-end path that offers low delay. The Adaptive Split Transmission (AST) reduces delay by distributing a video flow across several sub-flows that are transmitted simultaneously using different radios. Our study is based on simulations using *ns-2* and provides insight into the comparative performance of the two approaches. The main conclusion is that WCETT is better suited when offered traffic load is relatively low, whereas AST can work well even in highly congested situations.

Keywords: wireless mesh networks, QoS, WCETT, AST, multi-channel, multi-radio

1 Introduction

The deployment of wireless mesh networks (WMNs) has continued to gather pace in recent years. This is driven primarily by the desire to quickly increase penetration of residential broadband access and also in response to demands from large cities to provide robust multimedia networking for emergency services. It is supported by availability of WMN equipment from several well-established network equipment vendors. A parallel development is the increasing popularity of video applications on the Internet, and consequently as deployment of WMNs gathers pace, so too will the demand to provide video transmission with good quality of service (QoS). For a survey of WMNs see [1].

In comparison to wired networks, in a wireless environment the bandwidth is always more scarce. Moreover, the link quality can be easily compromised by interference and fading. However, the high rate traffic and stringent requirement for short delay are the main QoS features needed for video applications, raising significant research challenges. The multi-hop nature of WMNs accentuates these QoS problems, but recently, many new wireless technologies has been implemented on WMNs to improve the QoS solution, such as non-overlapping radio interfaces (e.g., IEEE 802.11a, 802.11g) [6], multi-channel devices and multi-radio devices and the use of them in WMNs has been proposed as the basis for QoS solutions [2-5]. In this paper we present a comparative evaluation of two transmission algorithms that represent two alternative approaches for QoS in WMNs.

This paper is organized as follows. Section 2 presents the two different algorithms. Section 3 evaluates these two algorithms through simulation results. Section 4 concludes the paper.

2 Transmission Algorithms

2.1 Weighted Cumulative Expected Transmission Time (WCETT)

WCETT [6] was proposed by Richard Draves, Jitendra Padhye and Brian Zill in 2004. This algorithm assigns a weight for each link according to the link's *expected transmission time* (ETT). ETT is calculated by

$$ETT = ETX * \frac{S}{B} \tag{1}$$

where S is the packet size, B is the link bandwidth and ETX is the number of expected transmissions. After collecting each link's ETT, the algorithm assigns a weight for each path based on the ETT value of each link on the path. The assigned path weight is called *weighted cumulative expected transmission time* (WCETT). WCETT is calculated by

$$WCETT = (1-\beta) * \sum_{i=1}^{n} ETT_i + \beta * \max_{1 \le j \le k} X_j \, , \tag{2}$$

where β is a tunable parameter $0 \leqslant \beta \leqslant 1$.

We now give some explanation for the equation (2). Assume the path has n hops, ETT_i is the ith link's ETT weight, and the network totally has k different available channels. The first term in (2) is

$$\sum_{i=1}^{n} ETT. \tag{3}$$

It represents the sum of transmission time along all hops in the route. This term makes sure that the path metric, which combines the ETTs of individual links, will have an increasing tendency if the number of hops is increased. The second term in (2) is

$$\max_{1 \le j \le k} X_j . \tag{4}$$

It denotes the worst transmission time at each channel. X_j is the sum of the transmission time of hops on channel j. Its expression is

$$X_j = \sum_{Hop\_i\_is\_on\_channel\_j} ETT_i \quad , \qquad 1 \le j \le k \tag{5}$$

Because the total path throughput is determined by the channel with the longest transmitted delay (i.e. the bottleneck channel), (5) actually shows the bottleneck channel capability and therefore guarantees that the WCETT metric is good for channel diversity.

2.2 Adaptive Split Transmission (AST)

AST [2] was proposed by Wanqing Tu and Cormac J. Sreenan in 2008. This algorithm utilizes the unused capacities in all of the available channels to transmit a video flow when the original channel has been overloaded. The algorithm collects the information about the available capacities of all channels and splits the video packet flow if necessary.

AST defines a new measurement to evaluate channels

$$\eta = \hat{C}(i) A(i), i \in [0, N-3], \tag{6}$$

where $\hat{C}(i)$ is the ith channel's unused capacity, and $A(i)$ is the availability of the ith channel. The larger η means more capability of the channel to transmit data.

For occupied channel, if we use $l(i', t)$ to denote the queue length of channel i' at time t, $\hat{C}(i', t)$ denotes the unused capacity of channel i' at time t, and r_T denotes the total capacity of channel i', we can calculate the channel's unused capacity below:

$$\hat{C}(i',t) = r_T - \sum_{j=0}^{F-1} r_j + \frac{d(l(i',t))}{dt}, i' \in [0, N-3], \quad (7)$$

where F is the number of existing flows which have being transmitted in this channel, r_j is the occupied capacity by the jth flow.
For unoccupied channels, the unused capacities can be collected through information exchange with its neighbours. This method means every node in the network should calculate his occupied channels' unused capacities, and send this information to his neighbours. Therefore, neighbouring nodes can retrieve the unused capacities from the received information. Moreover, both node's occupied channels capacities and unoccupied channels' capacities are included.

The availability *A(i)* is used to make sure that the channel will not incur congestion. It can be assigned as 0 or 1 to denote the availability. Firstly, the occupied channels can always be employed without incurring confliction. So, their *A(i)s* equal to *1*. Secondly, for unoccupied channels, the node *s* sends CONFLICTION DETECTION to its neighbouring nodes. After receiving this request, every neighbour replies with an acknowledgment that includes the information of the neighbour's channels status. After receiving these replies, the node *s* calculates the availability of each channel. The unoccupied channel's *A(i)* would be set as 1 if there are no neighbours receiving data through this channel. Oppositely, the *A(i)* would be set as 0 if any neighbouring node is using this channel to receive data.

3 Performance Comparison

3.1 Simulation Environment

We use the popular *ns-2* network simulator and as illustrated in Figure 1, the simulation topology is a 2-D mesh topology with the size increasing from 5*5 to 15*15. In 5*5 size, Node 0 is the sender and nodes 6, 12, 18 and 24 are receivers which are located in the diagonal.

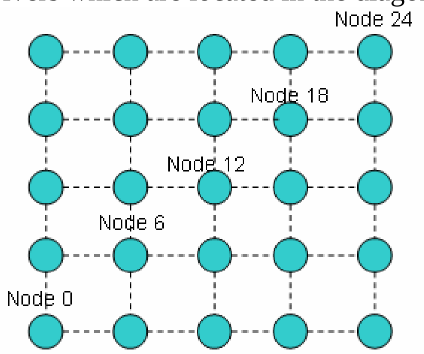

Figure 1. Example of Simulation topology in size of 5*5

Channel capacities vary at different nodes. The channels of Node 0 have 3 Mb/s transmit capacities, and all other nodes have 6 Mb/s capacities. There are two flows generated by the sender. First one is interference flow which has 1000 Kbit/s rate to introduce congestion. The second is the video flow (target flow) which has 2000 Kbit/s rate. By tracing the target flow, we can compare the performance result in both WCETT and AST simulations.

3.2 Simulation Results

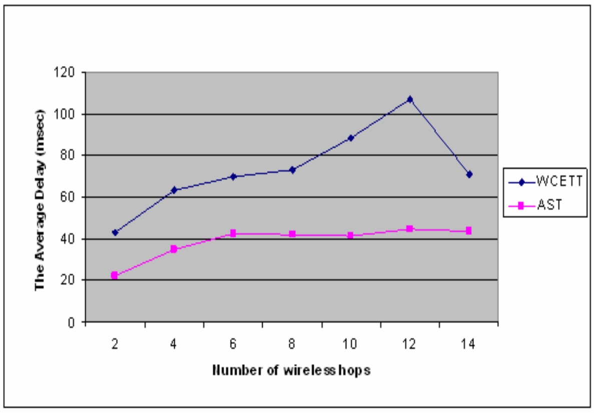

Figure 2 Average packet delay versus hop number

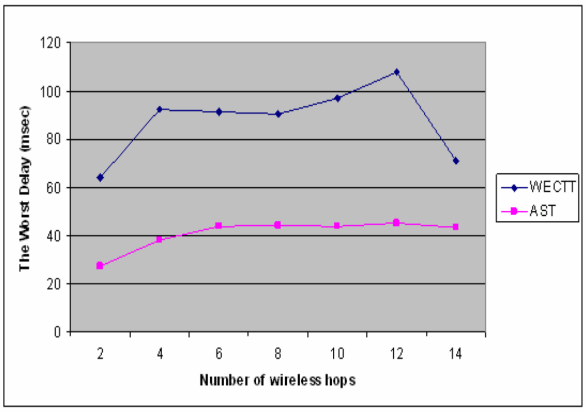

Figure 3 Worst-case packet delay versus hop number

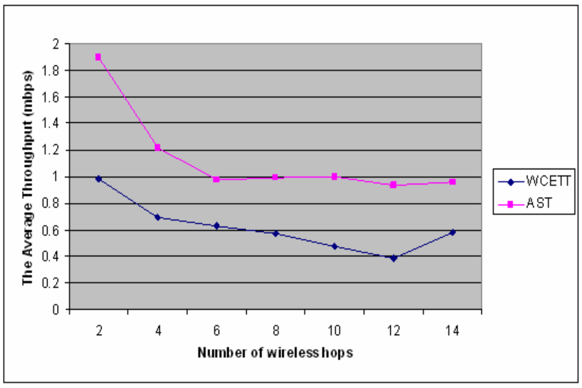

Figure 4 Average throughput versus hop number

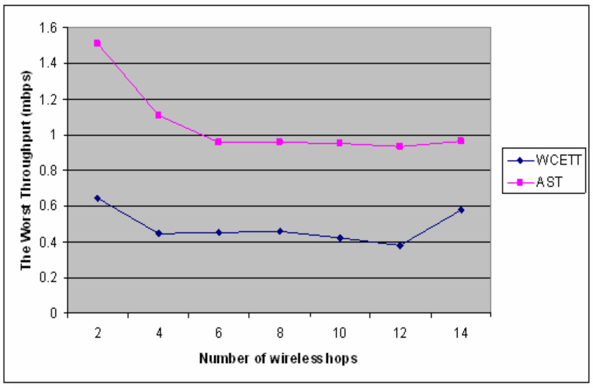

Figure 5 Worst-case throughput versus hop number

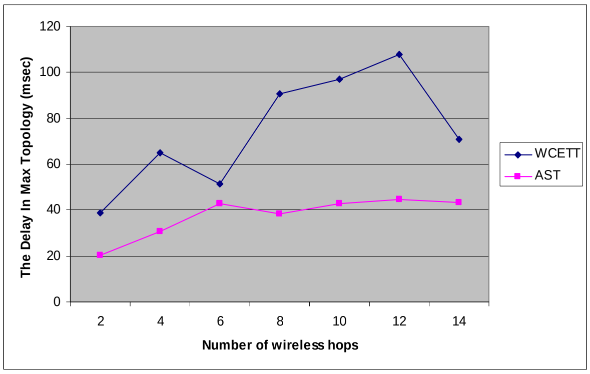

Figure 6 Delays in the maximum-size network (15*15) versus hop number

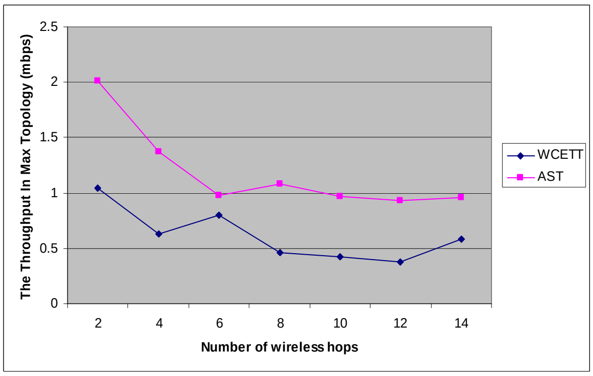

Figure 7 Throughput in the maximum-size network (15*15) versus hop number

3.3 Discussion

3.3.1 Channel Collision

With increasing hop number, throughput does not follow a strictly decreasing tendency and the delay does not follow a strictly increasing tendency but both fluctuate as shown in the result graphs. It is because of the random occurrence of channel collision. That means, in WCETT, two adjacent nodes may use the same channel to transmit the same video flow. Therefore, the node must wait for the channel resource available before transmitting data. It incurs longer delays and lower throughput. However, the amplitude of fluctuation of AST is much smaller than WCETT in the simulation result.

Therefore, the AST avoids the channel collision very well because it disperses the flow to several available channels.

3.3.2 Throughput and Delay

Due to the likely use of different video formats, the perceived qualities will be different. But for example, with an MPEG-4 video flow, 128 kbps bandwidth and the relatively low levels of delays (in 10s of milliseconds for the simulation scenarios) are certainly sufficient.

Comparing these two algorithms, both average packet delay and worst packet delay of WCETT algorithm are twice the values for the AST algorithm. Concerning the throughput, the Improved Quality (IQ) value is defined to evaluate these two algorithms. IQ is calculated by

$$IQ = \frac{\tilde{Q} - Q}{Q},$$

(8)

where $\tilde{Q}$ and Q are the highest the throughputs in AST and WCETT respectively. For one pair of sender and destination, the best video quality that the network transmission can guarantee is measured by the maximum video rate with acceptable delays (i.e. the highest throughput).

The average IQ of average throughput situation is calculated as $IQ_a = 84.8\%$.

The average IQ of worst-case throughput situation is calculated as $IQ_w = 118.4\%$.

3.3.3 WCETT *vs.* AST

As we have compared and evaluated in the simulations, WCETT suits to WMN video transmission in the network with lower traffic load, while AST is efficient in controlling traffic overload in heavy network conditions. However, the traffic control of AST has implications:

- It requires more channel resources for one flow. In simulation, 2.8 channels in average.
- ACT needs extra data in packet head to indicate sub-flow ID.
- It needs higher (but still moderate) computational complexity at intermediate nodes due to splitting and merging operation on sub-flows.

Although the modern technology is able to provide the required hardware for AST, the substantial increase in wireless traffic load requires a more efficient utilization of multiple radios and multiple channels in AST. This is a continued focus for our future research.

4 Conclusion

This paper compares and evaluates the performance of two representative algorithms: WCETT and AST in video transmission through using a set of simulations. In general, AST presents better performance in regard to the channel collision tolerance, the throughput and the transmission delay. Our comparison simulations show that AST holds the promise to implement real-time and QoS-controlled video transmission when the network traffic load is heavy.

In the near future, we are going to continue studying further details about video transmission performance such as delay jitter, resource utilization and etc. We also hope to set up a real-world test bed and continue to improve the AST algorithm.

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