Efficient Resource Utilization for Multi-flow Wireless Multicasting Transmissions

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***Abstract*— Wireless multimedia services are major applications of next generation wireless networks. This paper is one of the first to study the efficient utilization of network resources for increasing the number of concurrent multimedia flows when a channel becomes saturated. We theoretically study the *flow scheduling* policy and the *channel aggregation* policy in both single-hop and multi-hop wireless networks with the motivation of ameliorating the trade-off between limited channel resources and multiple flow transmission. To increase the number of performance guaranteed multimedia flows, based on the dynamic states of wireless channels and the profiles of multimedia flows, the two policies fully utilize the *performance gap* to schedule concurrent flows for transmission in turn and aggregate multiple channels' residual capacities for useful flow transmissions. We then design a novel algorithm - *efficient multi-flow multicast transmission* (EMMT) - to apply the proposed policies to practical wireless multimedia multicast applications. At last, we use ns2 simulations to evaluate the studied policies and the EMMT algorithm. Our simulation results prove the effectiveness of our schemes in improving network ability to admit more multimedia flows.**

***Index Terms*— Multimedia multicasting, resource awareness, flow scheduling, flow splitting, multiple flows, wireless networks.**

I. Introduction

One of the key characteristics of next generation wireless networks is the implementation of high-quality multimedia applications. High-quality multimedia applications require ample network capacity for real-time, continuous, and high-resolution communications. This challenges the reality that the radio frequency spectrum for wireless communications offers limited channel resources and the reuse of wireless channels is highly restricted due to the need to avoid interference. The result is a conflict between increasing multimedia applications in modern wireless networks and the limitations of channel resources.

A simple and direct way to improve this conflict is to utilize channel diversity to gain enough capacity. Wireless nodes employ more than one channel in parallel to increase their ability to accommodate more multimedia flows with guaranteed performance. However, there are limited non-overlapping wireless channels that can provide concurrent non-interfering transmission capacity. Earlier study [10] uses simulations to investigate whether channels with low overlap can be used for concurrent transmission without degrading performance. With this approach, a few more channels may be additionally employed to increase the transmission capacity of wireless nodes. However, multimedia applications become popular very quickly in modern wireless networks and users' requirements for communication quality increase all the time. Hence, before resorting to channel diversity, it is necessary to study efficient resource utilization for multi-flow wireless transmission.

Multi-flow transmission in wireless networks has been studied in recent research. A line of study [1-4] put efforts on the theoretical analysis of the maximum number of concurrent flows that a wireless network can accommodate under different interference influence. Jain *et al.* [1] proved that it is NP-hard to find such a maximum number of flows and therefore provided methods to compute the upper and lower bounds for this problem. Kodialam *et al.* [2-3] analyzed the maximum number of concurrent flows in a half-duplex wireless network. Wan [4] conducted this studies for single-radio single-channel wireless networks, subject to both bandwidth and interference constraints. In general, these studies commonly acknowledged that it is NP-hard to search such a maximum value of flow numbers due to complex wireless interference and transmission bandwidth. Hence, in these prior research, polynomial algorithms were also developed to achieve approximation bounds for the maximum values of flow numbers.

Another line of active research in wireless multi-flow transmission is to search for efficient resource allocation techniques that improve the ability of a wireless network to accommodate more non-interfering flow transmission. Cruz *et al.* [5] used primal-dual methods to compute long-term resource allocation policies. In [6], El-Batt *et al.* proposed an iterative scheduling and power-control scheme based on time slots. Middleton *et al.* [7] developed a polynomial-time algorithm for scheduling-routing power control in interfering half-duplex networks. This study was subsequently extended in [8] to allocate resources scheduling, routing, and power control for networks with streaming-packet data flows. M. Baghaie *et al.* [15] studied multi-flow cooperative transmissions in wireless multi-hop networks by exploiting delay constraints to minimize power consumption for prolonging network life. W. Tu *et al.* [11,14] designed the LCRT wireless multicast algorithm that admits multiple flows by selecting the minimum number of on-tree forwarders to reduce the intra-flow interference. In order to transmit multiple flows without causing interference/contention and high power consumption, these proposed techniques schedule wireless links and wireless nodes for improving a network's admission ability.

While it is important to study the influence of wireless interference on the number of admitted flows, the transmission opportunity created by efficiently utilizing allocated channel capacity should not be overlooked when exploring concurrent multimedia transmissions. By the allocated capacity[1], we mean the transmission capacity that a wireless node can achieve from a channel after counting the influence of different interference/conflict. The motivation of this paper is to investigate appropriate schemes that fully

[1]We will briefly introduce how an allocated capacity is achieved in literature in Section III.

utilizes a wireless node's allocated channel capacity to extend the ability of a network to admit performance guaranteed concurrent multimedia flows. In this paper, we tackle this problem by exploring the advantages of the performance gap which is defined as the difference between the acceptable performance bounds and the performance achieved when a channel is saturated. We present two novel flow management policies. The flow scheduling policy increases a wireless node's ability to admit more flows by intelligently scheduling flows to transmit in turn. The channel aggregation policy accumulates the residual capacities of multiple channels of a wireless node for transmitting a flow via multiple channels in parallel, when the residual capacity of no channel can accommodate such a full multimedia flow. The two policies are supported by a set of theoretical studies. In detail, for a wireless node achieving an allocated capacity of C from its output channel, the flow scheduling policy for single-hop wireless networks is based on the following findings.

- If the node is transmitting $(F-1)$ flows via this channel, it can admit a new flow f if f's average transmission rate is $\rho \leq \frac{C^2 \bar{D}}{F \bar{L}} - \sum_{i=1}^{F-1} \rho_i$, where $\bar{D}$ is the delay bound required by the receiver, $\bar{L}$ is the average packet size of f, and ρ_i is the average transmission rate of the ith $(i \in [1, F-1])$ existing flow;
- The F flows can be transmitted with guaranteed performance if and only if they takes turn to use the channel and flow i $(i \in [1, F])$ occupies the channel a time period of $\frac{T(\sigma_i + \rho_i T)}{C(T+\bar{D})}$ in its turn, where $T = \frac{C\bar{D} - F\bar{\sigma}}{F\bar{\rho} - C}$, $\bar{\rho} = \frac{\sum_{i=1}^{F} \rho_i}{F}$, ρ_i and σ_i are the average transmission rate and the burstiness of flow i respectively.

The theoretical findings of the channel aggregation policy suppose that a wireless node is equipped with K $(K > 1)$ channels and each channel has an allocated capacity of C_j $(j \in [1, K])$. Then, if channel j is transmitting F_j flows that have an average burstiness of $\bar{\sigma_j}$ and an average transmission rate of $\bar{\rho_j}$,

- the K channels can admit a new flow f together when the average transmission rate of f is $\rho \leq \frac{\sum_{j=1}^{K} C_j(\bar{D} - T_j)}{T_{max}}$, where $T_j = \frac{C_j \bar{D} - F_j \bar{\sigma_j}}{F_j \bar{\rho_j} - C_j}$, and $T_{max} = \max\{T_j, j \in [1, K]\}$;
- the performance of f is guaranteed if and only if using such a transmission way: f is split into K' $(1 < K' \leq K)$ subflows with the volume of $C_j(\bar{D} - T_j)$ $(j \in [1, K'])$ for subflow j; and the transmission of subflow j (via channel j) should start at kT_j $(k \in N)$ and lasts a time period of $(\bar{D} - T_j)$.

The above results are extensively developed for the use of the two policies in multi-hop wireless networks. By this study, we find that the flow scheduling policy or the channel aggregation policy designed for single-hop wireless networks cause desynchronization between the transmissions of different nodes on a multi-hop path. To solve this problem, the channel states of the "busies" node (i.e., the node with the heaviest traffic load) is used to develop our flow management policies for multi-hop wireless networks. We then design a novel algorithm *efficient multi-flow multicast transmission* (EMMT) which applies the above theoretical studies to practical network operations for improving a multicasting network's ability to admit performance guaranteed concurrent multimedia communications.

Finally, we use ns2 simulations to evaluate the proposed two policies and the EMMT algorithm against some related studies. The simulation results prove the effectiveness of our policies and algorithm in increasing the number of performance guaranteed flows: around 50% more flows can be admitted with acceptable performance by the EMMT algorithm as compared to some related work.

This paper is organized as follows. Section II analyzes the problems studied in this paper. Section III theoretically discusses the flow scheduling policy. Section IV analyzes the channel aggregation policy. Section V designs the efficient multi-flow multicasting transmission (EMMT) algorithm. Section VI presents our simulation observations and evaluations. At last, Section VII concludes this paper.

II. Motivation and Problem Formulation

As introduced, the major goal of our study is to propose schemes/algorithms that can increase the number of performance guaranteed concurrent multimedia flows by efficiently utilizing the capacity of each wireless channel. It is common throughout the literature that a wireless node will no longer admit more flows when its allocated capacity of output channels is saturated because of transmitting the existing flows. For the transmission illustrated in Fig. 1 (a), suppose that s can only transmit two flows (labeled blue and green) to r in order to meet r's performance requirements. If the third flow (labeled red) comes, as an example, we plot input and output graphs for s showing data amount vs. communication time in Fig. 1 (a)[2]. In the figure, the slope $m_{in} > m_{out}$ due to $\theta > \phi$, which indicates that the total transmission rate of the three flows exceeds s's output capacity for performance guaranteed transmissions. Hence, the performance of this transmission system keeps degrading during the communication because the backlog traffic keeps increasing. This explains why previous studies stop admitting flows when a channel is saturated.

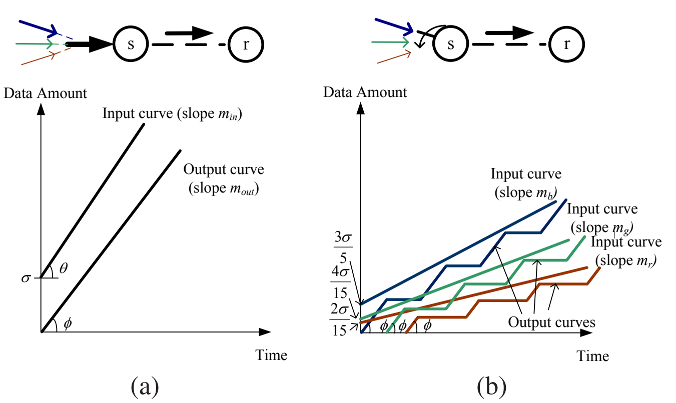


Fig. 1. An example comparing different methods used for transmitting three concurrent flows.

However, this insight is questioned by the observations from Fig. 1 (b). Instead of transmitting the three flows simultaneously, s separates the sending of the three flows by allocating different time slots. In this example, we suppose that the blue, green, and red flows have the burstiness of $\frac{3\sigma}{5}$, $\frac{4\sigma}{15}$, and $\frac{2\sigma}{15}$ respectively. The slopes of the straight line graphs of the input of the three flows are assumed to be m_b, m_g, and m_r respectively, which indicate the average transmission rates of the three flows respectively. Obviously, $m_b + m_g + m_r = m_{in}$. The three flows take turns to use the channel, as shown by the zigzag curves. Such scheduling

[2]The plotted input and output curves in this figure are based on the input and output functions of multimedia flows studied in [9].

manages to send the flows to r potentially with acceptable performance, since the output (i.e., the zigzag graphs) can quickly outpace the input (i.e., the straight graphs) because the channel's output capacity is sufficiently greater than the input transmission rate when the flows are managed by this example scheduling policy. These observations help to explain the extra transmission opportunity provided by efficiently using network resources.

In order to develop this study, it is important to find a metric that can provide extra transmission opportunity. Since multimedia flows with the quality not worse than network users' performance requirements are acceptable, if the performance of current flow transmission is within the required performance bounds, the performance gap[3] may enable a wireless channel to be reused for admitting more traffic until users' performance bounds are reached. As illustrated in Fig. 1 (b), it is an effective approach to utilize the transmission opportunity provided by the performance gap through scheduling the transmissions of concurrent flows. This finding raises an interesting problem: how to develop an appropriate flow scheduling policy that can be generally used to increase the number of performance guaranteed multimedia flows. Therefore, our first objective is to study a sound flow scheduling policy that should

1) correctly judge whether a channel is able to admit a new multimedia flow f without degrading the performance of current flow transmissions, and then
2) schedule multiple concurrent multimedia flows with guaranteed performance.

On the other hand, performance gaps are not always large enough for a wireless channel to admit a full multimedia flow. Instead of wasting the residual capacities created by small performance gaps, efficient resource utilization schemes may take the method of capacity aggregation into account, given that multiple channels/interfaces are popular for nodes in modern wireless networks. This encourages our second objective which is to investigate an appropriate channel aggregation policy that can be generally used to transmit a flow with guaranteed performance via multiple channels in parallel. A sound channel aggregation policy should

1) correctly judge whether the aggregated residual capacity of channels at a wireless node is enough for transmitting a flow f, and then
2) provide an appropriate method to transmit f over multiple channels without affecting other existing transmissions.

As we will present in the following sections, these studies are studied based on flow profiles, users' experience, and current network conditions to make full of the performance gap for admitting more flow traffic.

III. Flow Scheduling Policy

We first briefly introduce how to achieve an allocated capacity because this paper studies the efficient utilization of an allocated capacity for useful flow transmission. For a wireless node using IEEE 802.11 standard, it can occupy an output channel only when its attempt to transmit data under the idle channel state is successful. This indicates that the allocated capacity of this node is $C = p\hat{C}$, where p is the rate at which the node successfully sends flows after detecting an idle state, and $\hat{C}$ is the total capacity of the output channel. For the value of p, expressions has been proposed to obtain p in literature ([e.g., [13]). Hence, in this paper, we use C as an known value to develop the following policies.

Our analysis uses the theoretical results in [9] to model multimedia flows. According to [9], given $\sigma > 0$ and $\rho > 0$, for a multimedia flow, if its transmission rate at time t is given by the function $R(t)$, the following inequality exists if and only if $y \geq x$ for all x and y

$$\int_x^y R(t)dt \leq \sigma + \rho(y-x), \tag{1}$$

where σ and ρ are the burstiness and the average transmission rate of the multimedia flow respectively.

A. Flow Scheduling for Single-hop Wireless Transmissions

Our study is developed based on the performance of transmission delays and transmission throughput because they are two important metrics for multimedia users. Lemma 1 presents the *scheduling condition*.

Lemma 1. *Suppose a wireless node s achieves an allocated capacity C from its output wireless channel I. If this node is currently transmitting $(F-1)$ flows, it can admit a new multimedia flow f when the average transmission rate of f is $\rho_F \leq \frac{(C\bar{D}}{F\bar{L}} - \sum_{j=1}^{F-1} \rho_j$, where $\bar{D}$ is the delay bound required by users, $\bar{L}$ is the average packet size of the F flows, and ρ_j is the average transmission rate of flow j $(j \in [1, F-1])$.*

Proof. According to (1), for any flow i among the F flows (including the new flow f), within a time period τ, the expression $\int_t^{t+\tau} R_i(t)dt \leq \sigma_i + \rho_i\tau$ $(i \in [1, F])$ holds, where σ_i and ρ_i are the burstiness and the average transmission rate of the ith flow. Then if the wireless node s admits all the F multimedia flows, the total number of input data during a time period τ is $\sum_{i=1}^F \int_t^{t+\tau} R_i(t)dt$. Denote $\bar{\sigma} = \frac{\sum_{i=1}^F \sigma_i}{F}$ and $\bar{\rho} = \frac{\sum_{i=1}^F \rho_i}{F}$. We have $\sum_{i=1}^F \int_t^{t+\tau} R_i(t)dt \leq F(\bar{\sigma} + \bar{\rho}\tau)$.

In order to guarantee the full reception of these flows and therefore guarantee the throughput performance, the backlog traffic that cannot be transmitted immediately after being received by s should be put into a queue instead of being dropped. To make sure that the backlog traffic arrives at receivers in real time, it should be guaranteed that

$$\sum_{i=1}^F \int_t^{t+\tau} R_i(t)dt \leq C(\tau + \bar{D}), \tag{2}$$

where $\bar{D}$ is the end-to-end delay bound required by users. To ensure that expression (2) holds, $F(\bar{\sigma} + \bar{\rho}\tau) \leq C(\tau + \bar{D})$ should hold.

Since the time period τ should at least guarantee the amount of $F\bar{L}$ data to be transmitted, where $\bar{L} = \frac{\sum_{i=1}^F L_i}{F}$ and L_i is the average packet size of flow i, we have $\tau \geq \frac{F\bar{L}}{C}$. Moreover, based on [9], the F flows' average burstiness should be $\bar{\sigma} \geq \bar{L}(1 - \frac{\bar{\rho}}{C})$. If input these two conditions into the expression $F(\bar{\sigma} + \bar{\rho}\tau) \leq C(\tau + \bar{D})$, we obtain that f's average transmission rate should meet

$$\rho \leq \frac{C^2\bar{D}}{F\bar{L}} - \sum_{j=1}^{F-1} \rho_j .^4 \tag{3}$$

[3]The difference between the required performance bounds and the currently achieved performance, as we defined in Section I.

[4]From $F[\bar{L}(1 - \frac{\bar{\rho}}{C})] + F\bar{\rho}\tau \leq (\bar{D} + \tau)C$, we have $\bar{\rho} \leq \frac{\bar{D}C + C\tau - F\bar{L}}{F\tau - F\frac{\bar{L}}{C}} \Rightarrow \bar{\rho} \leq \frac{\bar{D}C + C\tau - F\bar{L} + F\frac{\bar{L}}{C}}{F\bar{L}} \Rightarrow \bar{\rho} \leq \frac{C^2\bar{D}}{F^2\bar{L}}$. Since $\bar{\rho} = \frac{\sum_{i=1}^F \rho_i}{F}$, then $\rho \leq \frac{C^2\bar{D}}{F\bar{L}} - \sum_{i=1}^{F-1} \rho_i$ is obtained.

Q.E.D.

Lemma 1 proves that a channel can admit a new flow f while transmitting $(F-1)$ existing flows if the average transmission rate of f meets the *scheduling condition* in (3). We now consider how to schedule the F multimedia flows in order to guarantee the performance of each flow.

Theorem 1. *For F multimedia flows transmitting via an output channel I, if the scheduling condition meets, the transmission performance of each multimedia flow is guaranteed if and only if the F flows are scheduled in such a manner: flow i $(i \in [1, F])$ takes its turn to transmit a time period $\tau_i = \frac{T(\sigma_i+\rho_i T)}{C(T+\bar{D})}$ which starts at $kT + \sum_{l=1}^{i-1} \tau_l$ $(k \in \{0\} \cup N)$, where $T = \frac{C\bar{D}-F\bar{\sigma}}{F\bar{\rho}-C}$ is called a scheduling period.*

Proof. $(\Rightarrow)$ Fig. 2 illustrates an example of flow scheduling in order to transmit three multimedia flows. There are three pairs of plots with corresponding colors in the diagram. For each pair of plots having an identical color, the straight line represents a flow's input function, and the zigzag line demonstrates a flow's output function: channel I alternates the transmission of flows in a round-robin fashion.

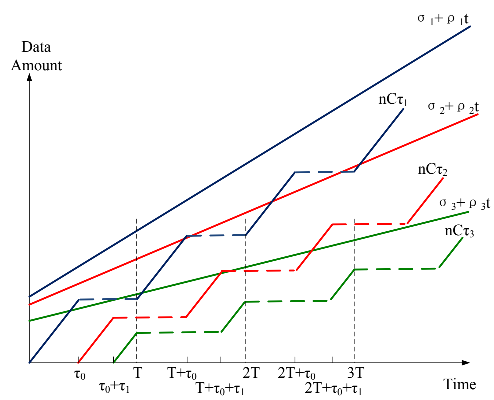


Fig. 2. An example of the flow scheduling policy.

We now generalize the flow scheduling policy for transmitting F flows. Suppose within a time period T, flow i $(i \in [1, F])$ is transmitted a period of τ_i by channel I. That is, $T = \sum_{i=1}^{F} \tau_i$ which is called as one *scheduling period* of these F flows on channel I. According to the diagram in Fig. 2, the backlog traffic of flow i (i.e., the amount of data from flow i that has not been transmitted in time) at the jth *scheduling period* is $b_j = (\sigma_j + jT\rho_j) - jC\tau_i$. In order to guarantee the full transmission of the backlog traffic in real time, we obtain that $b_j \leq C\bar{D}\frac{\tau_i}{T}$.

To find a simple expression of τ_i, it is worth to note that the backlog traffic after the first *scheduling period* (i.e., b_1) should have the largest backlog traffic because the burstiness of flow i is counted as the input of this *scheduling period*. Therefore, $b_j \leq C\bar{D}\frac{\tau_i}{T}$ holds if $b_1 \leq C\bar{D}\frac{\tau_i}{T}$. It implies that $\tau_i \geq \frac{T(\sigma_i+\rho_i T)}{C(T+\bar{D})}$. Without loss of generality, we use $\tau_i = \frac{T(\sigma_i+\rho_i T)}{\hat{C}(T+\bar{D})}$ in order to leave enough transmission time for other flows. Since there are $(i-1)$ flows transmitted before flow i, the time at which flow i should start its transmitting is then $kT + \sum_{l=1}^{i-1} \tau_l$, where $k \in \{0\} \cup N$.

For the value of T, due to $\sum_{i=1}^{F} \tau_i = T$, it is obtained that $T = \frac{\hat{C}\bar{D}-F\bar{\sigma}}{F\bar{\rho}-\hat{C}}$.

$(\Leftarrow)$ We now use the reduction to absurdity to prove the necessary condition. Suppose that flow transmissions with $\tau_i \neq \frac{T(\sigma_i+\rho_i T)}{\hat{C}(T+\bar{D})}$ can deliver acceptable multimedia quality to network users. Based on our above analysis, the total time consumed to transmit flow i within the time period $(\bar{D}+T)$ is $(\tau_i + \bar{D}\frac{\tau_i}{T})$. Hence, the time period that can be assigned for transmitting other $(F-1)$ flows is $\Delta\tau = (\bar{D}+T) - (\tau_i + \bar{D}\frac{\tau_i}{T})$.

If $\tau_i > \frac{T(\sigma_i+\rho_i T)}{C(T+\bar{D})}$, we have $\Delta\tau < [1 - \frac{\sigma_i+\rho_i T}{C(T+\bar{D})}](T+\bar{D})$ and hence $C\Delta\tau < C(T+\bar{D}) - (\sigma_i + \rho_i T)$ is obtained. It indicates that the total input data from other $(F-1)$ flows cannot be output by channel I within the delay bound $\bar{D}$.

If $\tau_i < \frac{T(\sigma_i+\rho_i T)}{C(T+\bar{D})}$, the transmission time period for flow i should be $\tau_i + \frac{\bar{D}\tau_i}{T} < \frac{T(\sigma_i+\rho_i T)}{C(T+\bar{D})} + \frac{\bar{D}}{T}\frac{T(\sigma_i+\rho_i T)}{C(T+\bar{D})} = \frac{\sigma_i+\rho_i T}{C}$ and hence $C(\tau_i + \frac{\bar{D}\tau_i}{T}) < \sigma_i + \rho_i T$ is obtained. This indicates that flow i cannot be fully output within the delay bound. Hence, the necessary condition is proved. Q.E.D.

B. Flow Scheduling for Multi-hop Wireless Transmissions

The policy in Lemma 1 and Theorem 1 are for single-hop wireless networks. It can be developed for multi-hop wireless transmissions because the flow scheduling operations are run by individual wireless nodes based on the allocated capacities of their own channels and the profiles of transmission flows. To make this improvement, the delay bound $\bar{D}$ should be guaranteed after multiple hops instead of a single hop in the above proof. To break down $\bar{D}$ between the sender and forwarders on a H-hop wireless path, for simplicity but without loss of generality, we require the sender and forwarders to share $\bar{D}$ equally. That is, the delay bound and the rate-distortion bound at each sender/forwarder are $\frac{\bar{D}}{H}$. This process helps to balance the utilization of network capacity and therefore promotes balanced traffic load in networks.

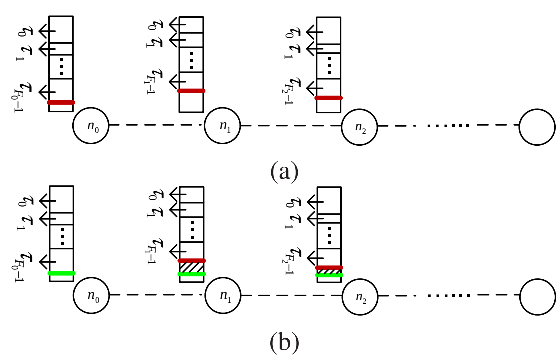


Fig. 3. An example of desynchronization of multi-hop flow scheduling.

Another issue that should be considered when developing an appropriate multi-hop flow scheduling policy is the desynchronization of time slots scheduled to transmit a flow at different hops. As illustrated in Fig. 3 (a), the red lines label the starting time for n_0, n_1, and n_2 to transmit f within a *scheduling period*. The unsynchronized starting times are mainly because nodes at different hops have different traffic load, i.e., $F_j \neq F_{j'}$ $(j \in [1, H], j' \in [1, j) \bigcup (j, H])$. This causes that *scheduling periods* are different at different hops, i.e., $T_j \neq T_{j'}$. As a result, in Fig. 3 (a), when n_1 is ready for transmitting f, n_0 has not forwarded f to n_1 yet. Therefore, f may not be transmitted with guaranteed performance by n_1 because a part of n_1's time slots scheduled for f is wasted in waiting for receiving f. To solve this problem, we use the latest starting time to synchronize the transmission schedule of f among different hops. The latest starting time T_{max} is expressed by

$$T_{max} = \max\{T_i, i \in [1, H]\},$$

where T_i is the *scheduling period* at the ith hop before admitting flow f. Then, based on the similar reasoning of Lemma 1 and Theorem 1, we achieve the following Theorem 2.

Theorem 2. *A new flow f can be admitted to transmit on a H-hop path if its average transmission rate is $\rho \leq \min\{\frac{C_i^2 \bar{D}}{HF_i\bar{L}_i} - \sum_{j=1}^{F_i-1} \rho_{i,j}\}, i \in [1, H]$, where C_i is the allocated capacity of the forwarder at the ith hop, $(F_i - 1)$ is the number of existing flows at the ith hop, $\rho_{i,j}$ is the average transmission rate of the jth flow at the ith hop, and $\bar{L}_i$ is the average packet size of F_i flows (including f) at the ith hop. The expression of ρ is also called as the multi-hop scheduling condition.*

To guarantee the performance of each individual flow, the F_i flows at the ith hop should be transmitted in turn. While all other (F_i-1) flows follow their previously scheduled time slots, starting at the time kT_{max} $(k \in N)$, the new flow f takes its turn to transmit a period of $\tau_{i,F_i} = \frac{HT_i^{'}(\sigma+\rho T_i^{'})}{C(HT_i^{'}+\bar{D})}$ at the ith hop, where σ and ρ are the average burstiness and the average transmission rate of f and $T_i^{'}$ is the scheduling period of node at the ith hop (calculated based on Theorem 1) after admitting f.

Proof. To prove the multi-hop scheduling condition, based on Lemma 1, the average transmission rate of f at the ith hop should guarantee $\rho \leq \frac{C_i^2 \bar{D}}{HF_i\bar{L}_i} - \sum_{j=1}^{F_i-1} \rho_{i,j}$, where $\rho_{i,j}$ is the average transmission rate of the jth flow at the ith hop and C_i is the allocated capacity of node at the ith hop. Therefore, considering the performance of flows passing through the H-hop wireless path, the average transmission rate of f should be

$$\rho \leq \min\{\frac{C_i^2 \bar{D}}{HF_i\bar{L}_i} - \sum_{j=1}^{F_i-1} \rho_{i,j}\}, i \in [1, H].$$

To schedule f that meets the *multi-hop scheduling condition* with guaranteed delay performance, based on the same reasoning in Theorem 1, we obtain that the length of the time slots assigned for f at the ith hop is $\tau_{i,F_i} = \frac{HT_i^{'}(\sigma+\rho T_i^{'})}{C(HT_i^{'}+\bar{D})}$, where $T_i^{'}$ is the *scheduling period* of node at the ith hop (calculated based on Theorem 1) after admitting f, σ and ρ are the average burstiness and the average transmission rate of f. In order to synchronize the transmission of f at different hops, as we have discussed, these time slots should start from kT_{max} $(k \in N)$ at different hops. Q.E.D.

The proposed policy in Theorem 2 generates unused time slots. As illustrated in Fig. 3 (b), the red lines label the finishing times of current *scheduling periods* at different hops. The green lines label the time that f actually starts its transmission in order to the synchronize f's transmission at different hops. The shadow areas therefore represent available unused capacity. To make use of the spare time spans, we propose to transmit additional data within these time spans that can result in higher quality than the basic-layer quality required by network users. The quality of this extra data (for higher multimedia quality) should be based on the transmission capacity that the spare time spans can provide.

C. Discussion

Quality of experience (QoE) becomes an important objective of broadband wireless communication networks. In [16], QoE is defined as the measure of how well a system or an application meets a user's expectations. The flow scheduling policy manages concurrent multimedia flows to meet the delay bound and throughput requirement. The delay metric is employed to guarantee the real-time performance required by communication users, while the throughput metric is used for the high multimedia quality (e.g., definition) that users can receive. These performance metrics are end-to-end performance metrics with which a sender/forwarder can schedule concurrent multimedia transmissions based on the application types and the performance expectations of individual users. Hence, the policy has the potential to support QoE services to provide transmissions that meet the quality levels required by different users.

On the other hand, because QoE is also viewed as a measure of overall performance from a user's perspective, the employment of the flow scheduling policy to support QoE services needs take more consideration of user experience factors. The factors may be different from different multimedia applications. For multimedia streaming applications that we study in this paper, these factors (apart from real time and high definition) may include continuous presentation, pause/resume delays, fast forward/rewind delays, financial budgets, etc. The flow scheduling policy is potential for being used to guarantee QoE by combining the research on the relationship between QoE and QoS[6] [17]. For example, buffers may be used for providing support for seeking functions (e.g., forward/rewind) and smoothing jitters (i.e., continuous presentation). Buffer sizes can be calculated based on user expected performance, which helps to calculate buffer delays. To guarantee the required jitter or forward/rewind performance, the flow scheduling policy can updates the end-to-end delay bounds in Theorems 1 and 2 to be $(\bar{D} - d_s)$, where d_s is the buffer delay required by a user. On the other hand, the flow scheduling policy encourages different multimedia flows to share the same wireless channels which essentially helps to reduce users' financial cost. The policy can also flexibly admit less flows if users prefer to pay more for better quality communications. Hence, we believe, with the further study of the relationship between QoS and QoE, the flow scheduling policy is available for future concurrent QoE-aware multimedia applications.

IV. Channel Aggregation Policy

The flow scheduling policy admits *full* multimedia flows to the best of a channel's capacity. The channel aggregation policy is designed to make full use of channels' residual capacity. According to (3), when $\rho < \frac{C^2\bar{D}}{F\bar{L}} - \sum_{i=1}^{F-1} \rho_i$, the channel has the spare capacity $\Delta C = \frac{C^2\bar{D}}{F\bar{L}} - \sum_{i=1}^{F} \rho_i$. ΔC is simple wasted if it is not large enough for the channel to admit a full new flow. We carry out the following studies to make use of ΔC.

A. Channel Aggregation for Single-hop Wireless Transmissions

The channel aggregation policy is applicable for a wireless node with multiple channels. The idea is to accumulate the residual capacity of channels of a wireless node for useful flow transmission. To start the discussion, we suppose a wireless node is equipped with K $(K > 1)$ channels and channel j $(j \in [1, K])$ is transmitting $(F_j - 1)$ flows. Lemma 2 analyzes the *aggregation condition* to admit a new flow by the channel aggregation policy.

[6]The goal of this relationship research is that one could estimate the QoE for a user, given a set of QoS measurements, and likewise, one could calculate the required network performance given a target QoE.

Lemma 2. *A flow f can be transmitted, with guaranteed performance, by aggregating multiple channels' residual capacity if the average transmission rate of f meets*

$$\rho \leq \frac{\sum_{j=1}^{K} C_j(\bar{D}-T_j)}{T_{max}},$$

where $T_j = \frac{C_j\bar{D}-F_j\bar{\sigma_j}}{F_j\bar{\rho_j}-C_j}$ is the scheduling period of (F_j-1) flows at channel j, $T_{max} = \max\{T_j, j \in [1,K]\}$, $\bar{D}$ is the delay bound required by network users, and C_j is the allocated capacity of channel j.

Proof. With the flow scheduling policy, flows are transmitted in the round-robin fashion proposed in Theorem 1. For a wireless node with K channels, the *scheduling period* (T_j) of (F_j-1) flows on channel j $(j \in [1,K])$ should meet $T_j \leq \bar{D}$ for the real-time performance. When $T_j \leq \bar{D}$, the time span $(\bar{D}-T_j)$ can be utilized by channel j to transmit the amount of additional data $C_j(\bar{D}-T_j)$, where C_j is the allocated capacity of channel j, and $T_j = \frac{C_j\bar{D}-F_j\bar{\sigma_j}}{F_j\bar{\rho_j}-C_j}$ is the *scheduling period* of (F_j-1) flows on channel j. Then, the total amount of data that can be admitted by the K aggregated channels is $\sum_{j=1}^{K} C_j(\bar{D}-T_j)$. For a new flow f with the burstinesss σ and the average transmission rate ρ, before the channel with the longest *scheduling period* is available, its total amount of input data is $\sigma + \rho(T_{max})$, where $T_{max} = \{T_j, j \in [1,K]\}$.

To utilize the residual capacities of all K channels of this wireless node for performance guaranteed flow transmission, the total input from a new flow f should be able to transmit out by the aggregated channels in real time. Namely, $\sigma + \rho T_{max} \leq \sum_{j=1}^{K} C_j(\bar{D}-T_j)$. Since $\sigma > L(1-\frac{\rho}{\bar{C}})$ [9], where $\bar{C} = \frac{\sum_{j=1}^{K} C_j}{K}$ and L is the average packet size of f, it can be inferred that

$$\rho \leq \frac{\sum_{j=1}^{K} C_j(\bar{D}-T_j) - L}{T_{max} - \frac{L}{\bar{C}}} \Rightarrow \rho \leq \frac{\sum_{j=1}^{K} C_j(\bar{D}-T_j)}{T_{max}}. \quad (4)$$

Q.E.D.

Lemma 2 analyzes the *aggregation condition* under which new flows can be admitted by accumulating multiple channels' capacities. When this condition is satisfied, flow f is transmitted via multiple channels by the method described in Theorem 3.

Theorem 3. *Flow f can be transmitted with acceptable performance if and only if*

1) *f is split into K' $(K' \leq K)$ subflows and each subflow has the size $S_j = C_j(\bar{D}-T_j)$ $(j \in [1,K'])$, and*
2) *the transmission of the jth subflow starts at kT_j $(k \in N)$ via channel j which lasts a time period of $(\bar{D}-T_j)$,*

where T_j is the scheduling periods before channel j admits any subflow of f.

Proof. To use K' $(K' \leq K)$ channels for transmitting f together, the amount of data assigned for a channel should be based on the channel's residual capacity. Hence, for the subflow sent to channel j $(j \in [1,K'])$, its size should not exceed the residual capacity of channel j. We therefore obtain that the size of the jth subflow is

$$S_j = C_j(\bar{D}-T_j). \quad (5)$$

To schedule the transmission of these K subflows, channel j starts transmitting the jth subflow when it completes scheduling the existing flows. Namely, the jth subflow is transmitted by channel j within a time period $(\bar{D}-T_j)$ which starts at kT_j, where $k \in N$, and T_j is the *scheduling periods* before channel j admits any subflow of f. Q.E.D.

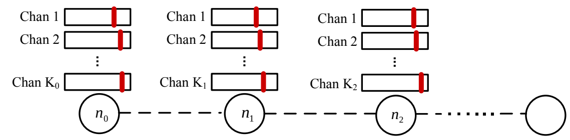


Fig. 4. An example of desynchronization in multi-hop channel aggregation policy.

B. Channel Aggregation for Multi-hop Wireless Transmissions

To improve the channel aggregation policy for flow splitting transmission via a multi-hop path, similar to the multi-hop flow scheduling policy, the sender and forwarders share $\bar{D}$ equally. Namely, the delay bound at each sender/forwarder are $\frac{\bar{D}}{H}$ respectively. Also similar to scheduling flows in multi-hop networks, due to different hops have different traffic load, the time slots arranged for splitting transmission based on Theorem 3 may not be synchronized at different hops. In Fig. 4, the time scheduled for transmitting the first subflow of f via *chan 1* at n_1 is earlier than the time scheduled for the same subflow at n_0. The desynchronization causes that n_1 cannot output all split subflows with guaranteed performance. To avoid performance degradation caused by time inconsistency, a simple and effective way is to arrange the subflows, arriving later than their scheduled time slots in current transmission periods, to be transmitted in the same time slots of the next transmission periods. This obviously increases the delays of splitting transmission. Therefore, the multi-hop channel aggregation policy should take these delays into account in order to achieve the performance satisfying network users. That is, the delay bound at each hop should be updated as $(\frac{\bar{D}}{H}-\ddot{T}_{max})$, where $\ddot{T}_{max} = \max\{T_{i,j}, i \in [1,H], j \in [1,K_i]\}$, K_i is the number of channels at the ith hop, and $T_{i,j}$ is the *scheduling period* of the jth channel at the ith hop before admitting f. Based on these insights, we achieve Theorem 4 by the similar reasoning of Lemma 2 and Theorem 3.

Theorem 4. *A new flow f can be admitted by a H-hop wireless path through using the channel aggregation policy when its average transmission rate meets $\rho \leq \frac{\sum_{j=1}^{K_i}[(\frac{\bar{D}}{H}-\ddot{T}_{max})-T_{i,j}]-L}{\ddot{T}_{max}}$, where $T_{i,j} = \frac{C_{i,j}\bar{D}-HF_{i,j}\bar{\sigma_{i,j}}}{F_{i,j}\bar{\rho_{i,j}}-C_{i,j}}$ is the scheduling period of channel j at the ith hop before admitting f, $\ddot{T}_{max} = \max\{T_{i,j}, i \in [1,H], j \in [1,K_i]\}$, K_i is the number of channels at the ith hop, $F_{i,j}$ is the number of flows transmitted by channel j at the ith hop, $\bar{\sigma_{i,j}}$ and $\bar{\rho_{i,j}}$ are the average burstiness and the average transmission rate of the $F_{i,j}$ flows.*

The performance of splitting transmission is guaranteed if and only if the jth subflow at the ith hop has the size of $C_{i,j}[(\frac{\bar{D}}{H}-\ddot{T}_{max})-T_{i,j}]$ and is transmitted by channel j at the ith hop at $kT_{i,j}$ with a period of $[(\frac{\bar{D}}{H}-\ddot{T}_{max})-T_{i,j}]$, where $k \in N$.

Proof. In order to guarantee the transmission performance after admitting a new flow f, the expression $\sigma + \ddot{T}_{max}\rho \leq \sum_{j=1}^{K_i}[(\frac{\bar{D}}{H}-\ddot{T}_{max})-T_{i,j}]$ should meet, where σ and ρ are the burstiness and the average transmission rate of f.

Since $\sigma > L(1-\frac{K_i\rho}{\sum_{j=1}^{K_i} C_{i,j}})$, the above expression infers that

$$\rho \leq \sum_{j=1}^{K_i}\{[(\frac{\bar{D}}{H}-\ddot{T}_{max})-T_{i,j}]-L\ddot{T}_{max}\}.$$

Hence, the multi-hop *channel aggregation* condition is proved. By using the similar reasoning of Theorem 3, the flow splitting

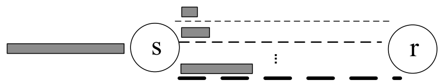


Fig. 5. An example of splitting flows to transmit by the channel aggregation policy. A thicker dotted line represents a channel with higher capacity.

and subflow transmission policy via a H-hop path can be proved. Q.E.D.

C. Discussion

Like the flow scheduling policy, the channel aggregation policy can also be employed for supporting QoE-aware multimedia services, due to the same reasons we discussed previously.

We now introduce the practical implementation of the channel aggregation policy by referring to some work in our previous study [18]. In general, we propose to use the minimum number of channels for the splitting transmission. As illustrated in Fig. 5, if the sender s has enough accumulated residual capacity for transmitting f (i.e., meets Lemma 2), s sorts its output channels in decreasing order of channels' residual capacity and then continuously selects a channel from the head of this list until the total accumulated capacity is enough for f. After these operations, s splits the data input within a period T_{max} into subflows (based on Theorem 3) and transmits these subflows via multiple channels in parallel. Instead of randomly picking channels, the proposed approach reduces overheads (generated for identifying subflows) and CPU process delays (caused by flow splitting and packet encapsulation/decapsulation) through using the minimum number of channels.

V. Efficient Multi-Flow Multicasting Transmission (EMMT)

This section proposes an *efficient multi-flow multicast transmission* (EMMT) algorithm that uses the two studied policies to multicast concurrent multimedia communications in wireless networks.

A. Multicast Criteria

1) User Experience: In multimedia multicasting, there is usually a group of receivers who may require different performance from the communications. We employ *user experience* as a criterion to design our multicast algorithm to support QoE-aware services. With our motivation in this paper which is to save network resources for useful data transmission, this criterion is designed as follows.

Multicast transmissions only need to guarantee the performance required by individual receivers. However, for a sender/forwarder whose child nodes (i.e., all downstream nodes) have different-level performance requirements, the transmission of this sender/forwarder should guarantee the highest-level performance.

2) Path Selection: Referring to our previous work in [11,14], the path selection criterion include the following metrics.

a. *Hop distance*: A path with shorter hop distance is preferred which benefits multicast by providing shorter delays and less transmission contention.

b. *Nodes with rich connectivity*: A node that has more neighbors as multicast forwarders/receivers is preferred to be a forwarding node. This metric helps to control the number of forwarders and therefore avoiding interference/conflict caused by parallel transmissions in multicasting.

c. *Reliability*: A reliable path has the priority to be a multicast path because wireless links are lost frequently which affects the continuity of multimedia presentation.

We employ the *path weight* to combine the above metrics. The *path weight* of path i that connects a sender and a receiver is

$$\omega_i = \frac{1}{h_i} \times \sum_{j=1}^{h_i} \frac{D_{i,j}}{N_{i,j}} \times \prod_{j=1}^{h_i} (1 - l_{i,j}), \tag{6}$$

where h_i is the number of hops on this path, $D_{i,j}$ and $N_{i,j}$ are the total number of child nodes and neighboring nodes at the jth hop on this path, $l_{i,j}$ is the loss rate at the jth hop on this path. In this expression, $\frac{D_{i,j}}{N_{i,j}}$ is the metric for evaluating whether a node coves more child nodes or not (i.e., the nodes with rich connectivity), and $\prod_{j=1}^{h_i}(1 - l_{i,j})$ is the metric for evaluating path reliability. Apparently, a path with larger *path weight* has the priority to become a multicast path.

B. Efficient Multi-flow Multicast Transmission

With the above criteria, we design the EMMT algorithm that employs the flow scheduling policy and the channel aggregation policy for performance guaranteed multi-flow multicasting. Since the implementation of both policies require wireless nodes to know about flow profiles, the algorithm requires a flow sender to broadcasts a light-weight PROFILE packet in the multicasting network. A PROFILE packet includes the fields listed in TABLE I. Note that the value of MESSAGE_ID is increased by 1 whenever the sender issues a PROFILE packet for a new flow.

TABLE I
Fields of PROFILE Packets

Field	Functions
MESSAGE_ID	Distinguish between different PROFILE packets sent by the same senders
FLOW_PROFILE	Record a flow's profile such as the average transmission rate, the burstiness, etc.

Once a receiver receives a PROFILE message, it replies a REPORT packet (by broadcasting) which includes the fields listed in TABLE II. Note that QUALITY_LEVEL is used to represent the *user experience* criterion, which allows the EMMT algorithm to deal with individual users' performance requirements. PATH_INF and LINK_LOSS are introduced for achieving *path weights*. With PATH_INF, the sender knows the hop distance (h) of a

TABLE II
Fields of REPORT Packets

Field	Functions
NODE_ID	Identify the receiver who issues this REPORT
QUALITY_LEVEL	Inform the sender of the quality level required by this receivers
PATH_INF	Record nodes on a path that this REPORT travels
LINK_LOSS	Record the loss rates of channels of nodes on a path that this REPORT travels
CHANNEL_PROFILE	Record the state of each node on a path that this REPORT travels

path and the number of neighbors who are also child nodes ($\frac{D}{N}$) on this path; with LINK_LOSS, the sender achieves the reliability of a path. CHANNEL_PROFILE is set for using the two policies to admit concurrent multimedia flows. In order to save control overheads, we require each wireless node to calculate its availability to admit a new flow after receiving PROFILE by the two policies. Hence, the information recorded in CHANNEL_PROFILE includes the longest *scheduling period* and the bound of average transmission rate (achieved by Theorem 4) of each node on a path that the REPORT travels. This process also reduces the calculation burden of the flow senders. We now present the detailed operations of the EMMT algorithm.

Algorithm 1 Efficient Multi-flow Multicast Transmission

Input: A new multimedia flow f input by a wireless node s.
Output: Multicast f to meet the expected performance of individual users.

1. s broadcasts a PROFILE packet;
2. After receiving PROFILE, each node checks whether it can admit f with and without the two policies by Theorems 2 and 4;
3. Each receiver replies a REPORT packet by broadcasting;
4. After receiving a REPORT packet, each forwarder fills the fields of PATH_INF, LINK_LOSS, and CHANNEL_PROFILE with the information stated in TABLE II;
5. After receiving the REPORT packets of all receivers, s selects all of the paths that can deliver f to receivers with their required performance;
6. s filters the selected paths to form the EMMT tree by using the *path weights* defined in (7);
7. f is multicasted to receivers via the EMMT tree in as follows: s or a forwarder implements the transmission by scheduling /splitting f (based on the two policies) using the highest performance requirements of all the downstream receivers as bounds.

We use an example in Fig. 6 to illustrate how the EMMT multicast architecture is set up after the sender and receivers have exchanged their PROFILE and REPORT packets. Based on (7), the EMMT algorithm selects paths with higher reliability, shorter hop distance, and via the most popular forwarders to connect a sender and a receiver. Based on the criterion of *user experience*, the path that can deliver the performance expected by a receiver is preferred. As an example, suppose $Q_4 > Q_1 > Q_3 > Q_2$, where Q_i is the quality level required by r_i, the black dotted lines should be chosen to be multicast paths based on both criteria. In order to deliver the flow to four receivers ($r_1 \sim r_4$) to meet their individual quality requirements while saving network resources, the flow with the quality level that will be not less than Q_4 after two hops is multicasted by s. That is, the flow scheduling or flow splitting at s are based on the performance bounds that can guarantee Q_4 after 2 hops. When the flow comes to s', due to the highest quality level of its child nodes is Q_1, s' schedules/splits the flow by using Q_1 as the performance bound in the two policies. Hence, unlike the transmission $s'' \Rightarrow r_4$, the multicasting of s' saves the network resources equal to the capacity for the quality $(Q_4 - Q_1)$. Similarly, at r_1, the multicast updates the quality level to Q_3 because $Q_3 > Q_2$.

In the EMMT algorithm, both PROFILE and REPORT packets are broadcasted in the network. If the average out degree of nodes in the network is η ($\eta > 1$), the broadcast of PROFILE/REPORT makes each node to forward the same PROFILE/REPORT packets η times. That is, the overheads issued to the network are $n\eta$ times of a PROFILE/REPORT packet size, where n is the number of nodes in the network. In order to control the overheads (for saving network resources), in our algorithm, the PROFILE/REPORT packets of the same multimedia flows (which can be identified by the same MESSAGE_IDs) are only transmitted once by the same wireless node. Hence, the overheads generated by PROFILE/REPORT packets are reduced to at most n times of the PROFILE/REPORT packet size.

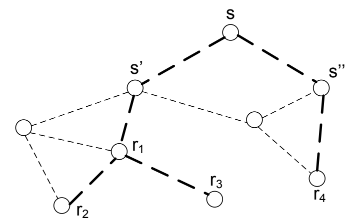


Fig. 6. An example of multicasting flows with the EMMT algorithm.

VI. Simulation Evaluations

In this section, we conduct an extensive simulation-based evaluation. There are 3 groups of simulations implemented to study the flow scheduling policy (FC), the channel aggregation policy (CA), and the EMMT algorithm separately by using the discrete event network simulator NS2.33. The evaluation of FC and CA is conducted in both single-hop and multi-hop wireless networks that use AODV as the routing protocol. The evaluation of EMMT is carried out in wireless multicast networks with single channel and multiple channels respectively, and compared with three other multicast schemes LCRT [11], CDT [10], and NFIC-MF [8] which we will introduce later.

The simulations mainly observe the average delays and the average data loss rates of multimedia transmissions. Average delays are defined as

$$AD = \frac{\sum_{i=1}^{n} d_i}{n},$$

where n is the number of receivers, and d_i is the average delay of all packets received by the ith receiver. Delays exceeding the bound cause lag which adversely affects the ability of users to communicate in real time. ADs demonstrate how well a network transmission scheme can guarantee real-time communications to different receivers. Average data loss rates are defined as

$$AL = \frac{\sum_{i=1}^{n} l_i}{n},$$

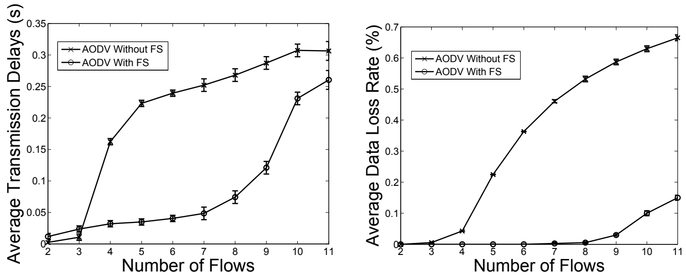
where l_i is the average data loss rate expressed by the ith receiver. Data loss rates affect the quality (e.g., definition, continuity) of multimedia streams received by users. ALs demonstrate how well a network transmission scheme can deliver high quality multimedia flows to users. The smaller AL is, the better presentation quality that users receive.

A. Evaluation of the Flow Scheduling Policy

We first observe the performance of flow scheduling policy. We implement the simulations in a single-hop wireless network which simply includes a sender and a receiver, and a 4-hop wireless network which includes a pair of sender/receiver and 3 forwarding nodes that connect into a line topology. The parameters that we use for the simulations are listed in TABLE III. As shown in the table, the simulated video flows are generated based on the MPEG-4 file *StarWarsIV.dat*. We create these flows with different transmission rates which randomly vary between 500Kbps and 1Mbps in the single-hop simulations. We also suppose that the receiver requires an average data loss rate $\leq 5\%$ for each flow in order to achieve the expected quality and continuity of multimedia flow presentations. Meanwhile, there are 2 interfering nodes who introduce 256Kbps and 500Kbps interfering traffic to the single-hop network. We report the average delay performance and the average data loss rates in Fig. 6 (a) and (b) respectively. Each value on the curves are the mean value of 20 simulation runs.

TABLE III
PARAMETERS USED IN FLOW SCHEDULING SIMULATIONS

	Single-hop simulation	4-hop simulation
NS-2 version	2.33	2.33
Radio propagation model	Nakagami	Nakagami
MAC protocol	802.11	802.11
Number of output channels at the sender or forwarders	1	3
Bandwidth of output channels at the sender or forwarders	5.5Mbps	5.5Mbps
Number of nodes sharing/interfering output channels at each hop	2	4
NS-2 video trace file	StarWarsIV.dat	StarWarsIV.dat
Delay bound	200ms	200ms



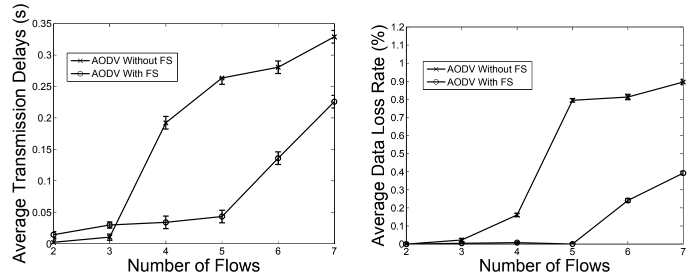
(a) (b)

Fig. 7. The average delays (a) and the average data loss rates (b) achieved by the single-hop simulations when the number of flows increases.

From the curves plotted in Fig. 7 (a), we knows that, for AODV without using the flow scheduling policy, when the 5th flow comes, the sender's output channel becomes saturated because it cannot deliver real-time flows to the receiver to meet the receiver's requirement. However, when scheduling the flows to transmit in turn, the network admits 5 more video flows with accepted delay performance. The curves in Fig. 7 (a) also show that AODV without flow scheduling generates shorter delays than AODV with flow scheduling when there are only 2 $\sim$ 3 input flows. It indicates that flow scheduling may generate longer delays when the network traffic load is light because the network has the capacity to deal with the amount of input data simultaneously (i.e., the network is under saturated) and the round-robin scheduling of flows actually causes delays in such a situation. Although the longer delays in this situation, AODV with flow scheduling still delivers flows meeting the receiver's delay requirement (200ms).

For the average data loss rates, Fig. 7 (b) shows that, for AODV without flow scheduling, the receiver is satisfied with the video quality when the sender only inputs the first 4 flows. The flow scheduling policy however enables AODV to transmit 9 flows with accepted average loss rates. The performance improvement of AODV with flow scheduling can be explained by the similar reasons for its achieved improvement in average delays.

For the simulations of evaluating the flow scheduling policy in the 4-hop wireless networks, the simulation parameters listed in TABLE III are used. Also, we generate interfering flows to nodes at each hop of the network. The transmission rates of interfering flows vary between 256Kbps and 500Kbps. Video flows transmitted from the sender to the receiver are the same ones used in the single-hop simulations. Based on the mean values of 20 runs, we plot the average delay performance in Fig. 8 (a) and the average data loss rates in Fig. 8 (b). Considering both delay bound (200ms) and acceptable data loss rates ($\leq 5\%$), AODV with flow scheduling enables the sender to admit 6 video flows which allows the opportunity to transmit 3 additional flows as compared to AODV with the flow scheduling policy.

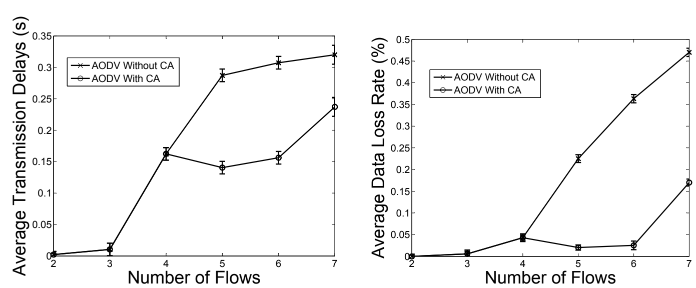


(a) (b)

Fig. 8. The average delays (a) and the average data loss rates (b) achieved in the 4-hop simulations when the number of video flow increases.

B. Evaluation of the Channel Aggregation Policy

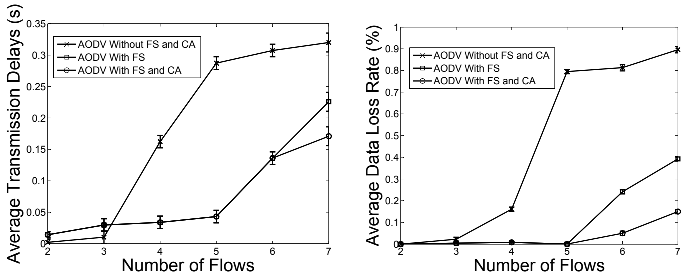
In the second group of simulations, we evaluate the performance of the channel aggregation policy. Likewise, we study this policy in a single-hop wireless network and a 4-hop wireless network. In the simulations with the single-hop wireless network, the sender is equipped with 3 radio interfaces/channels and each channel has the bandwidth of 2Mbps. All other simulation settings are the same as the settings used for the flow scheduling simulations in the single-hop networks. Based on the average results of 20 runs in the single-hop simulations, we plot the average delays in Fig. 9 (a) and the average data loss rates in Fig. 9 (b).



(a) (b)

Fig. 9. The average delays (a) and the average data loss rates (b) achieved by the single-hop simulations when the number of video flows increases.

In the simulations, the channel aggregation policy is employed when the channel capacity becomes saturated. The average delays and the average data loss rates achieved from the simulations are plotted in Fig. 9 (a) and (b) respectively. Each value on the curves is the mean value of 20 runs. The results show that the channel aggregation policy additionally enables the network to admit 2 more video flows with acceptable delay and data loss rate performance. This proves the effectiveness of using the residual capacity to improve flow transmission opportunity.



(a) (b)

Fig. 10. The average delays (a) and the average data loss rates (b) achieved in the 4-hop simulations when the number of video flows increases.

In the simulations conducted in the 4-hop networks, the channel aggregation policy is employed when the flow scheduling policy cannot admit more video flows with acceptable performance. Based on the mean values of 20 runs, the achieved average delays and average data loss rates are plotted in Fig. 10 (a) and (b) respectively. By utilizing the residual capacity of the three channels, the channel aggregation policy manages to transmit 1 extra video flow continuously and in real time, as compared to AODV without flow scheduling.

C. Evaluation of the EMMT Algorithm

We evaluate the EMMT algorithm in this section. For the purpose of comparative evaluation, we selected the following three related multicast schemes from the literature.

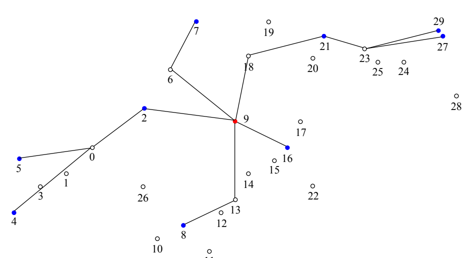
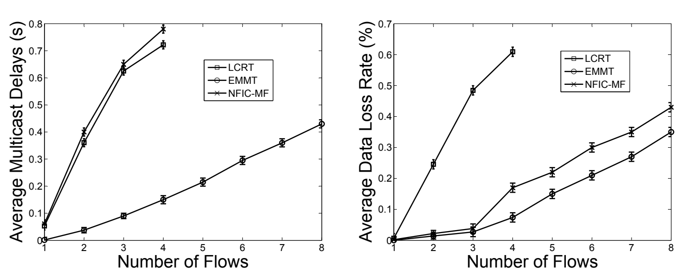
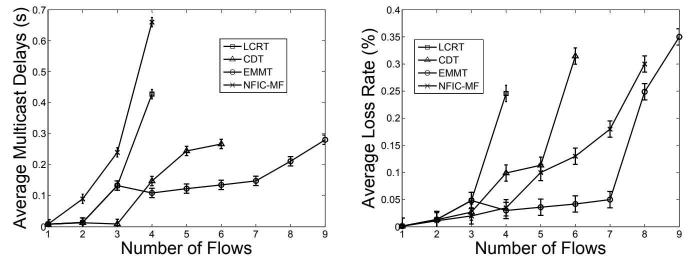


Fig. 11. The simulated network topology for flow multicasting.



(a) (b)

Fig. 12. The average multicast delays (a) and the average data loss rates (b) of EMMT and LCRT in the single-channel network when the number of multicasting flows increases.



(a) (b)

Fig. 13. The average transmission delays (a) and the average packet loss rates (b) of the three schemes in the multi-channel network when the number of flows increases.

1) The Link-Controlled Routing Tree (LCRT). This scheme [11] constructs a distribution tree by using the least number of forwarders that can provide reliable multicasting.
2) The Channel Diversity Transmission (CDT). This scheme [10] constructs a distribution tree by using channel diversity to address interference problem between multiple flows.
3) The Network-Flow Interaction Chart with Multiple Flows (NFIC-MF). This scheme [8] constructs a multi-flow transmission architecture on a temporal basis for controlling the interactions between flows.

The simulations are carried out in a 30-node wireless network (the topology is shown in Fig. 11) for multicasting video flows to a group of receivers. In this multicast, node 9 (i.e., the red node) is the multicast sender, and nodes 2, 4, 5, 7, 8, 16, 21, 27, and 29 (i.e., the blue nodes) are the multicast receivers. Nodes 1, 10, 14, 20, 26, and 28 introduce interference traffic to the multicast transmission. The transmission rates of interference traffic are in the range of [256Kbps, 1Mbps]. The bandwidth of channels in the simulations is 11Mbps. Other simulation settings are the same as the ones used in previous simulations (i.e., in TABLE III).

We first study the multicast schemes when each wireless node in the network has only one channel. With this setting, we only compare the performance of EMMT, NFIC-MF, and LCRT because CDT is a scheme based on the channel diversity. Fig. 12 presents the comparison on the metrics of average multicast delays and average data loss rates respectively. Each value on the curves is the mean result of 20 runs. NFIC-MF causes the longest delays as this transmission scheme focuses on avoiding packet loss or collapse by selecting available low-interference paths, which may cause longer transmission delays. LCRT constructs a multicast tree by using the minimum number of forwarders that can cover all receivers. This scheme therefore uses paths with shorter hop distances but may cause interference/conflicts between the parallel transmissions of multicasting. This explains the reason why LCRT may have shorter delays than NFIC-MF but the largest average data loss rates among the three schemes in the simulations. EMMT, on the other hand, admits 3 video flows (2 more flows as compared to NFIC-MF and LCRT) with acceptable performance. This is mainly because the EMMT algorithm takes the hop distances of paths into account and uses the flow scheduling policy transmitting flows in turn to reduce interference/conflict between peer forwarders in multicasting.

We then equip two radio interfaces/channels at each node to evaluate the three schemes. For the simulation of CDT, we assigns two different channels at two adjacent nodes in order to avoid interference/conflict between these nodes. The results in Fig. 13

prove that EMMT, adaptively employing the flow scheduling policy and the channel aggregation policy, improves the network ability to accept more video flows in real time and with accepted video quality than NFIC-MF, LCRT, and CDT do. For CDT, given the overall assessment of average multicast delays and average data loss rates, it admits one more flows than NFIC-MF and LCRT due to the utilization of channel diversity. However, there are only two different channels in the simulations which limits CDT to overtake the performance of EMMT. In general, EMMT transmits 4 more flows than CDT.

We also observe the overheads generated by NFIC-MF, LCRT, CDT, and EMMT (in TABLE IV) when they construct their architectures for multicasting a multimedia flow. The overheads

TABLE IV
OVERHEADS GENERATED BY DIFFERENT SCHEMES

	Overheads
NFIC-MF	4.32K bits
LCRT	1.82K bits
CDT	2.61K bits
EMMT	3.78K bits

generated by NFIC-MF are mainly for searching low-inference paths in different time slots. The overheads generated by CDT are mainly for scheduling and assigning different channels to adjacent nodes. The overheads generated by EMMT are mainly for exchanging the information of flow profile and node availability. Although the larger overheads in EMMT multicasting as compared to those in LCRT and CDT multicasting, EMMT achieves extra transmission opportunities (around 50% more) for concurrent multimedia flows.

VII. CONCLUSION

In this paper, we studied the issue of improving a wireless network ability to admit more numbers of performance guaranteed concurrent multimedia flows. We are among the first to tackle this issue by investigating the transmission opportunity provided by the performance gap. The flow scheduling policy and the channel aggregation policy were theoretically analyzed and presented as a way to strategically utilize network resources for concurrent multimedia flow transmission. A set of theoretical results regarding the scheduling condition, the aggregation condition, and the appropriate methods to schedule and split flows were contributed to increase the number of concurrent multimedia flows that can be admitted by a wireless channel. We then introduced how to apply these theoretical results to practical multimedia multicasting by proposing a novel efficient multi-flow multicast transmission algorithm.

We used ns2 simulations to evaluate the two flow management policies and the EMMT algorithm. The simulation results demonstrated that the EMMT algorithm achieved at least 50% improvement in increasing the number of admitted flows as compared to some related work. In addition, we discussed the potential to use the two policies and the algorithm to support QoE-aware multimedia services. With the further development of QoE schemes on relating user experience to QoS, our study will be useful for QoE applications. Moreover, the practical implementation of the two proposed policies can be easily integrated into other existing wireless routing protocols without further hardware or network architecture deployment. In our next-step work, we are interested in developing the two policies and the algorithm for multi-source multimedia applications.

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