Multipath Routing over Wireless Mesh Networks For Multiple Description Video Transmission

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***Abstract*—In the past few years, wireless mesh networks (WMNs) have drawn significant attention from academia and industry as a fast, easy, and inexpensive solution for broadband wireless access. In WMNs, it is important to support video communications in an efficient way. To address this issue, this paper studies the multipath routing for multiple description (MD) video delivery over IEEE 802.11 based WMN. Specifically, we first design a framework to transmit MD video over WMNs through multiple paths; we then investigate the technical challenges encountered. In our proposed framework, multipath routing relies on the maximally disjoint paths to achieve good traffic engineering performance. However, video applications usually have strict delay requirements, which make it difficult to find multiple qualified paths with the least joints. To overcome this problem, we develop an enhanced version of Guaranteed-Rate (GR) packet scheduling algorithm, namely virtual reserved rate GR (VRR-GR), to shorten the packet delay of video communications in multiservice network environment. Simulation study shows that our proposed approach can reduce the latency of video delivery and achieve desirable traffic engineering performance in multipath routing environment.**

***Index Terms*—Video communications, wireless mesh network, multipath routing, multiple description, traffic engineering.**

I. INTRODUCTION

WIRELESS mesh networking has recently emerged as a promising technology for the next-generation wireless networks [1]. Nevertheless, there are still many challenges in the design of wireless mesh networks (WMNs). One of the most important issues is how to efficiently support the video communications, which are expected to be the killer application for future wireless networks.

Video applications in WMN are usually Internet oriented and thus the traffic is either from end user to Internet gateway (IGW) or vice versa [2]. In most WMNs deployed today, the routing protocol focuses on finding a single best possible route from the source to the destination. Consequently, certain links could be heavily loaded while many others are significantly underutilized. Such a phenomenon is a breach of traffic engineering principle and could deteriorate the overall performance of the network.

To solve this problem, we propose to employ multipath routing in WMN, so that traffic can be uniformly distributed across the network. Multipath routing was firstly proposed by N. F. Maxemchuk in his seminal work on dispersity routing [3], [4], for the purpose of equalizing the load on the links in the ARPA-net and surviving link failures. Since then, multipath routing has been extensively studied in various wireline networks to achieve increased aggregate capacity, better load balancing, and path redundancy for failure recovery. Most recently, multiple path routing techniques are being considered for mobile ad hoc network (MANETs) to survive path changes [5], since the broadcast nature of radio transmissions makes it possible to obtain disjoint paths without physically installing separate wires from a source node. Unlike MANETs, WMN in general has the support of infrastructure, i.e., the backbone built amongst wireless mesh routers. Rather than surviving the path change, we utilize multipath routing to counter the uneven distribution of traffic in WMN.

In this paper, we investigate the delivery of multiple description (MD) videos over WMN. MD video is an important coding technique of error resilience and control for multimedia applications and has been recognized as an ideal candidate for video streaming in multi-hop wireless networks [6], [7]. With MD coding, multiple equivalent substreams (or descriptions) are generated from a video source for transmission. At the receiver, any received subset of these descriptions can be combined to reconstruct the original video and the quality of the reconstructed video is commensurate with the number of received descriptions. This video coding technique is drastically different from traditional layered video coding, where video reconstruction hinges upon successful delivery of the base layer.

After the video is MD coded, the multipath routing scheme constructs multiple paths from the source to the destination and video traffic is then delivered with each path carrying one substream. In the early-stage study on multipath routing, more attention was paid to the problem of finding fully disjoint paths. For instance, Sidhu et. al. [8] stated that multiple disjoint paths could increase the effective bandwidth between node pairs, reduce congestion in a network, and reduce the probability of dropped packets. Since fully disjoint paths are

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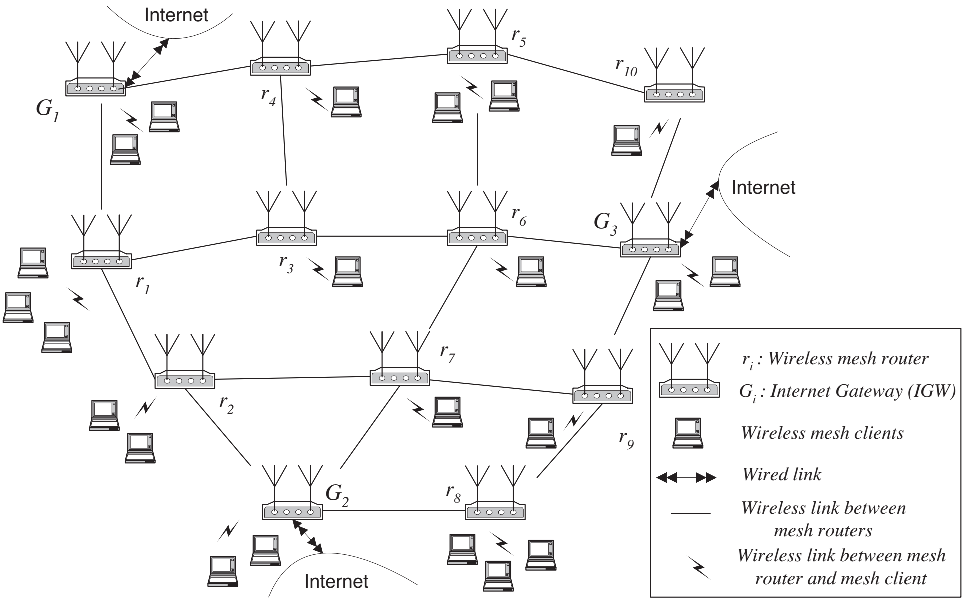

Fig. 1. An example of wireless mesh network.

not always available in the network, some researchers then turned to study maximally disjoint paths as an approximation to fully disjoint paths [9].

In our application, it is important to find the multiple paths with minimum joints to achieve traffic engineering in WMNs. On one hand, mesh topologies of WMNs provide a good basis for fully disjoint paths. On the other hand, video applications have strict delay requirements, which may make it difficult to setup multiple qualified paths with the least joints. It is clear that multipath routing will have a better traffic engineering performance if a WMN can shorten the delay of video transmission. The reason is lower video transmission delay means more qualified candidate paths.

We investigate the multiservice environment in this paper, where video traffic can coexist with other types of traffic in WMN. To reduce the delay of video traffic, our work starts with the investigation on IEEE 802.11 MAC layer, from which we further develop an enhanced version of Guaranteed-Rate (GR) packet scheduling algorithm, namely virtual reserved rate GR (VRR-GR), to give video traffic high preference in multiservice environment. The fundamental difference between VRR-GR and conventional GR is that the former employs virtual reserved rate instead of real reserved rate to calculate the Guaranteed-Rate clock value.

The rest of the paper is organized as follows. We first introduce the video communications over WMN in Section II. We then propose our approach of multipath routing for multiple description video transmission in Section III. In Section IV, we discuss the IEEE 802.11 MAC layer and develop a new approach of packet scheduling to reduce the delay of video traffic in WMN. In Section V, we present the numerical results to demonstrate the advantage of our proposed approach. In Section VI, we conclude this work.

II. VIDEO COMMUNICATIONS OVER WMN

As shown in Fig. 1, a WMN consists of two types of nodes: mesh routers and mesh clients. The mesh routers form an infrastructure of mesh backbone for mesh clients. In general, mesh routers have minimal mobility and operate just like a network of fixed routers, except being connected by wireless links through wireless technologies such as IEEE 802.11. We can observe from Fig. 1 that, a WMN can access the Internet through a gateway mesh router or IGW, which is connected to the IP core network with physical wires.

In a WMN, every mesh router is equipped with a traffic aggregation device (similar to an 802.11 access point) that interacts with individual mesh clients. The mesh router relays aggregated data traffic of mesh clients to and from the IP core network. Typically, a mesh router has multiple wireless interfaces to communicate with other mesh routers, and each wireless interface works corresponding to one wireless channel. These wireless channels have different characteristics, because wireless interfaces are running on different frequencies and built on either the same or different wireless access technologies, e.g., IEEE 802.11a/b/g/n. Unlike many other wireless networks, such as MANETs and wireless sensor networks (WSN), WMN in general has the support of infrastructure. Therefore, it is reasonable to assume that the wireless link between two mesh routers has fixed bandwidth capacity.

There is a rapid growth of video applications in WMN recently, including video on demand (VoD), video conferencing, video surveillance, online games, etc. These applications are usually in real time and have stringent delay requirement. Apart from its advantages, WMN still has to face many challenges to support video communications. For example, video applications in WMN are mostly Internet oriented and thus the traffic is either from the end user to the Internet gateway or reversely [2]. As a result, traffic congestion will happen to some hot links, with the probability to further degrade the performance of the whole network.

To solve this problem, we propose to employ multipath routing in WMN. The goal of our scheme is to distribute video traffic uniformly across the network, as well as survive the video transmission on some error-prone wireless links.

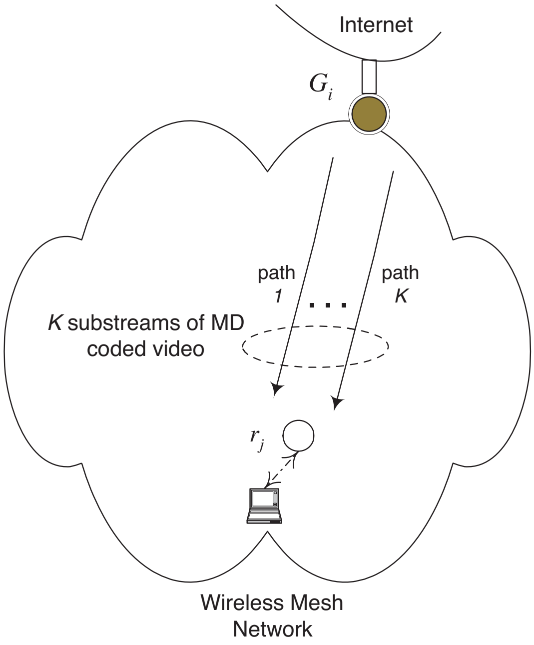

Fig. 2. Multipath routing framework for MD coded video communications over WMN.

III. MULTIPATH ROUTING FOR MD CODED VIDEO COMMUNICATIONS OVER WMN

A. General Framework

Fig. 2 demonstrates our proposed multipath routing framework for MD coded video communications over WMN. In the framework, we assume that the source node in the Internet splits the video into K substreams using MD coding. The video traffic of K substreams first reaches the IGW at the border of WMN cloud. At IGW, a multipath routing layer constructs K paths between itself and destination mesh router, each with a set of quality-of-service (QoS) parameters in terms of bandwidth, delay, and so on. Then the video traffic is delivered over K paths in the way that each path carries one substream. At the destination mesh router, K substreams are merged into one and transmitted to the mesh client. Above design relies on two key issues, i.e., MD coding and multipath routing.

B. Multiple Description Coding

To employ multipath routing in WMN, one must carefully design the video coder to generate substreams so that the loss in one substream does not adversely affect the decoding of other substreams. One well known way of generating multiple streams is to use layered coding (LC), which generates a base layer (BL) and one or more enhancement layers (ELs) [10]. The base layer includes the crucial part of the video frames and guarantees a basic display quality. Each enhancement layer improves the video quality. But without the BL, video frames cannot be reconstructed sufficiently. As a result, one major obstacle of LC is that the base layer must be delivered almost error free to guarantee a basic level of quality.

To overcome the shortcoming of LC, multiple description coding (MDC) generates multiple equally important streams, each giving a low but acceptable quality [11]–[16]. A high-quality reconstruction is decodable from all bit streams together, while a lower, but still acceptable quality reconstruction is achievable if only one stream is received. The correlation among the substreams introduced at the encoder makes it possible to partially recover lost information of one substream, using information carried in other correctly received substreams. However, such a correlation limits the achievable coding efficiency, as compared with a conventional coder designed to maximize it. With MDC, the transport layer design can be simpler than that with layered coding. Because all the descriptions are equally important, the transport layer does not need to protect one stream more than another. Also, because each description alone can provide a low but acceptable quality, no retransmission is required, making MDC more suitable for applications with stringent delay requirements.

C. Multipath Routing

In the literature, multipath routing has been widely investigated for the performance improvement in both wired and wireless networks. Early research effort on multipath routing focused on two issues in wired network: 1) how to select a maximally disjoint set of routes for a source-destination pair [8]; 2) how to allocate traffic to multiple end-to-end routes with different granularities [17]. Later on, the study of multipath routing has been extended to MANETs. The efforts in this area aim to provide wireless networks with strong error resilience, large aggregate bandwidth, and good load balancing [5], [18]–[21].

Multipath routing can achieve the best performance, if fully disjoint paths are found. Given an undirected graph $G = (V, E)$ and a source-destination pair (s, d), we define K paths from s to d as fully node/link disjoint path if there exist no shared nodes/links among these paths. According to the definition, the disjointness of paths may refer to nodes or to links. In this paper we mainly investigate the latter case, since it is the most commonly studied version in networking field.

Fully disjoint paths are not always available in WMN. If so, an alternative solution is to use partially disjoint paths, especially the maximally link-disjoint paths. A pair of paths from a source to a destination are defined to be maximally link-disjoint if the number of links shared by both paths is minimum. Considering fully disjoint paths have zero joint, we conclude that multipath routing prefers the paths with minimum joints in general.

When constructing multiple paths for a video application, each selected path must meet the corresponding QoS requirements independently. Bandwidth and delay are usually considered as two major metrics in QoS routing protocols. Bandwidth is a concave metric and generally not a severe problem in WMN due to the following reasons: 1) WMN has the support of infrastructure and the capacity of a single link could be improved by advanced technologies such as directional antenna; 2) MD coding splits the high-rate video traffic into low-rate multiple substreams, each of which only has a low bandwidth requirement along its path. Delay, however, is an additive metric and thus poses a significant challenge on multi-hop based network like WMN. It is clear that the overall delay of multipath routing solution is determined by the sub-path of worst performance. In this respect, the delay problem becomes a bottleneck to achieve multiple paths with least joints, though the mesh topology of a WMN provides a good basis for implementing multipath routing approach. To overcome this difficulty, we investigate next on how to reduce the video delivery delay in WMNs.

IV. REDUCING THE DELAY OF VIDEO TRAFFIC IN WMN

The packet delay in WMN is mainly caused by MAC layer latency and packet scheduling latency. In the following, we first introduce the enhanced distributed channel access (EDCA), which is standardized in IEEE 802.11e [22] to support multimedia services with tight QoS requirements. Based on EDCA, we then propose a new packet scheduling algorithm to further reduce the delay of video transmission in multiservice environment.

A. IEEE 802.11 MAC Layer

Currently, distributed coordination function (DCF) [23] is the most popular IEEE 802.11 MAC protocol in use. DCF is based on the scheme of carrier sense multiple access with collision avoidance (CSMA/CA), which does not differentiate traffic types. As a result, a station might have to wait for an arbitrarily long time to send a packet, and multimedia services may suffer intolerable delay and jitter. To solve this problem, IEEE 802.11e proposes EDCA as an enhanced version of DCF.

EDCA supports the QoS by introducing four access categories (ACs). Each packet arrives at the MAC layer with a priority from higher layer, and is mapped to an AC according to the priority. AC3, AC2, AC1, and AC0 are for voice, video, best effort data, and background traffic, respectively. To differentiate the traffic types, EDCA grants AC i ($i = 0, ..., 3$) a set of specific parameters, including minimum contention window ($CWmin[i]$), maximum contention window ($CWmax[i]$), and arbitration inter-frame space ($AIFS[i]$). With above parameters, the support of QoS can be achieved by differentiating the probability of channel access among different ACs [24]–[26].

EDCA has shortened the MAC layer latency of video transmission, but there still remains a problem to reduce the packet scheduling latency. To overcome this difficulty, in the following we first review the conventional Guaranteed-Rate (GR) packet scheduling, we then propose a new approach of virtual reserved rate GR (VRR-GR) to give video applications preference during packet scheduling process in multiservice environment.

B. Basics of GR Packet Scheduling Algorithms

To meet the QoS requirements of multimedia services, a number of packet scheduling algorithms have been proposed in the literature, including Virtual Clock, Packet-by-Packet Generalized Processor Sharing (PGPS), Self Clocked Fair Queuing (SCFQ), and so on [27]–[31]. In [32] and [33], Goyal et. al. defined the class of Guaranteed-Rate (GR) scheduling algorithms as the ones that allocate a given reserved rate to a flow of packets. GR scheduling algorithms guarantee a delay bound to a packet based on its expected arrival time. It has been proven that Virtual Clock, PGPS, and SCFQ all belong to GR.

Assuming that packet is the unit of data transmission at the network level, we refer to the sequence of packets transmitted by a source as a flow. Each packet of a flow is then served by a sequence of network nodes (switching elements, i.e., mesh routers in WMN) along the path from the source to the destination.

Consider a general case where flow f is switched by network node i. To provide guaranteed performance, network node i allocates flow f a reserved rate r_f (in bits/second) as requested by the source of flow f. Network node i is called a GR network node if it complies with the service discipline of work-conserving and non-preemptive and uses the Guaranteed-Rate clock value of a packet as scheduling priority. In other words, when GR network node switches a new packet, the packet in queue with the smallest Guaranteed-Rate clock value is selected for service.

Let p_f^j and l_f^j represent the jth packet of flow f and its length, respectively. Let $A^i(p_f^j)$ represent the arrival time of packet p_f^j at GR network node i. Then, Guaranteed-Rate clock value for packet p_f^j at network node i, denoted by $GRC^i(p_f^j, r_f)$, is given by

$$GRC^i(p_f^j, r_f) = \max\{A^i(p_f^j), GRC^i(p_f^{j-1}, r_f)\} + l_f^j / r_f, \tag{1}$$

where $GRC^i(p_f^0, r_f) = 0$ and $j \geq 1$.

With the definition of Guaranteed-Rate clock value, a scheduling algorithm at network node i belongs to class GR for flow f, if it guarantees that packet p_f^j will be transmitted by $GRC^i(p_f^j, r_f) + \beta^i$. Here, β^i is a constant which depends on the scheduling algorithm and the network node. For example, the virtual clock algorithm belongs to GR with $\beta^i = l_{max}^i / C^i$, where C^i and l_{max}^i represent the capacity of network node i and the maximum length of packet served by network node i, respectively.

As shown in [32], [33], strict delay bounding is a salient feature of GR packet scheduling algorithms. Let K be the total number of network nodes along the path of a flow, i be the ith network node on the path, network node 0 be the source, and network node $K+1$ be the destination. Since the packet arrives at the first network node at time $A^1(p_f^j)$ and network node K guarantees that packet p_f^j will be transmitted by $GRC^K(p_f^j, r_f) + \beta^K$, the end-to-end delay of p_f^j, denoted by d_f^j, is bounded by

$$d_f^j \leq GRC^K(p_f^j, r_f) + \beta^K + \tau^{K,K+1} - A^1(p_f^j), \tag{2}$$

where $\tau^{K,K+1}$ is the propagation delay between network node K and the destination.

From Eq. (1) we observe that $GRC^K(p_f^j, r_f)$ depends on $A^K(p_f^j)$, which in turn depends on $GRC^{K-1}(p_f^j, r_f)$ by $A^K(p_f^j) = GRC^{K-1}(p_f^j, r_f) + \beta^{K-1} + \tau^{K-1,K}$. Applying this argument recursively, $GRC^K(p_f^j, r_f)$ can be replaced with $GRC^1(p_f^j, r_f)$ in Eq. (2) by

$$d_f^j \leq GRC^1(p_f^j, r_f) - A^1(p_f^j) + (K-1) \max_{n\in[1\cdots j]} \frac{l_f^n}{r_f} + \sum_{n=1}^{K}(\beta^n + \tau^{n,n+1}). \tag{3}$$

Since $GRC^1(p_f^j, r_f)$ is completely determined by the traffic arrival characteristics of the source and the rate associated with the flow, the end-to-end delay can be determined if source specification is known.

For instance, leaky bucket is a source traffic specification that bounds the maximum deviation from the average rate [34]–[37]. Let $AP_f(t_1, t_2)$ be a function that denotes the bits of flow f that arrive in the interval $[t_1, t_2]$. A flow f conforms to leaky bucket with burst size σ_f and average rate r_f if

$$AP_f(t_1, t_2) \leq \sigma_f + r_f(t_2 - t_1). \tag{4}$$

If flow f conforms to a leaky bucket with parameters σ_f, r_f and the scheduling algorithm at each of the network node on the path of a flow belongs to GR, then the end-lo-end delay of packet p_f^j is bounded by

$$d_f^j \leq \frac{\sigma_f + (K-1) \max\limits_{n\in[1\cdots j]} l_f^n}{r_f} + \sum_{n=1}^{K}(\beta^n + \tau^{n,n+1}). \tag{5}$$

As a statistical relaxation of leaky bucket, exponentially bounded burstiness (EBB) process characterization has been proposed in [38]. A flow conforms to EBB if the deviation probability of a source from the average rate decreases exponentially. Let Λ_f be the prefactor and γ_f be the decay rate of the exponential decay function, then a flow f is an EBB process with parameters $(r_f, \Lambda_f, \gamma_f)$, if

$$Pr\{AP_f^i(t_1, t_2) \geq r_f(t_2 - t_1 + x)\} \leq \Lambda_f \exp(-\gamma_f x), \quad x \geq 0, \quad t_1 \geq t_2. \tag{6}$$

If flow f conforms to EBB with parameters $(r_f, \Lambda_f, \gamma_f)$ and the scheduling algorithm at each of the network nodes on the path of a flow belong to GR, then the end-to-end delay of packet p_f^j is bounded by

$$Pr\{d_f^j \geq x + \frac{(K-1) \max\limits_{n\in[1\cdots j]} l_f^n}{r_f} + \sum_{n=1}^{K}(\beta^n + \tau^{n,n+1})\} \leq \Lambda_f \exp(-\gamma_f x r_f), \quad x \geq 0. \tag{7}$$

In addition to delay, jitter is another important issue to multimedia services [39]. Jitter can be derived from delay by

$$J_f^j = |d_f^{j+1} - d_f^j|. \tag{8}$$

Since $d_f^{j+1}, d_f^j \geq 0$, we have $J_f^j < \max(d_f^{j+1}, d_f^j)$. The value of J_f^j depends on j and thus varies from time to time. Let $d_f = \max_{j\geq 1}(d_f^j)$, we then define J_f^j as a random variable ranged in $[0, d_f]$ with probability density function (pdf) $p(J_f^j)$. As a result, the average jitter of flow f is given by

$$E[J_f^j] = \int_0^{d_f} J_f^j p(J_f^j) dJ_f^j. \tag{9}$$

C. VRR-GR Packet Scheduling Algorithm for Multiservice Networks

1) Architecture Design: Multiservice networks are replacing older generation of single service networks. Instead of using three different networks for data, voice, and video, a multiservice network is an all-in-one platform for broadband, phone, and TV services. Our study focuses on a typical multiservice environment, i.e., video communications over WMN with other possible services. Particularly, we consider that the services may have various QoS requirements, and thus the WMN has to support four scheduling categories corresponding to the ACs defined in EDCA: SC3 for voice, SC2 for video, SC1 for best effort data, and SC0 for background traffic.

Traditional GR scheduling algorithms address mainly the reserved rate service. To accommodate the diversity of multiservice environment, we emphasize the differentiating capability during packet scheduling process in this paper. Fig. 3 demonstrates the architecture of extended GR packet scheduling architecture. It shows that four categories of services, are scheduled together in the four priority queues, where the Guaranteed-Rate clock value of a packet is utilized as scheduling priority.

Let C^i be the total capacity of the network node i, the capacity allocated to different service levels can then be represented by $C^i = C_3^i + C_2^i + C_1^i + C_0^i$, where C_3^i, C_2^i, C_1^i, and C_0^i are the sub-capacity allocated to SC3, SC2, SC1, and SC0, respectively. For SC3 and SC2, each connection is viewed as a flow and guaranteed with a fixed bandwidth (the reserved rate as required by the flow). For SC1 and SC0, the fixed bandwidth is allocated to all the connections as a whole; thus, each connection can only share a part of it, which may vary from time to time.

2) Virtual Reserved Rate GR Scheduling Algorithm: We develop a virtual reserved rate GR (VRR-GR) scheduling algorithm to determine the Guaranteed-Rate clock value of each flow in the priority queue of Fig. 3. The major difference between VRR-GR and conventional GR is that the former uses virtual reserved rate instead of real reserved rate to calculate the Guaranteed-Rate clock value. Here, the virtual reserved rate comes from the concept of virtual sub-capacity, which is defined as

$$VC_j^i = C_j^i + O_j^i, \quad j = 0, 1, 2, 3 \tag{10}$$

where VC_j^i, C_j^i, and O_j^i represent the virtual sub-capacity, real sub-capacity, and virtual offset-capacity of SCj on network node i, respectively. Our VRR-GR algorithm aims to integrate multiple service levels into one packet scheduling framework. In this framework, VRR-GR has the responsibility to prioritize video service by deriving the Guaranteed-Rate clock value from virtual reserved rate rather than real reserved rate. Bandwidth and delay bound are two important issues when designing VRR-GR algorithm. Real sub-capacity decides the bandwidth that a flow finally obtains, whereas virtual sub-capacity decides the induced delay during packet scheduling. Using conventional GR as a reference, virtual offset-capacity explains the additional reserved rate that VRR-GR grants to a flow during packet scheduling. It is worth noting that virtual offset-capacity is introduced to influence the delay bound but not the bandwidth of a flow.

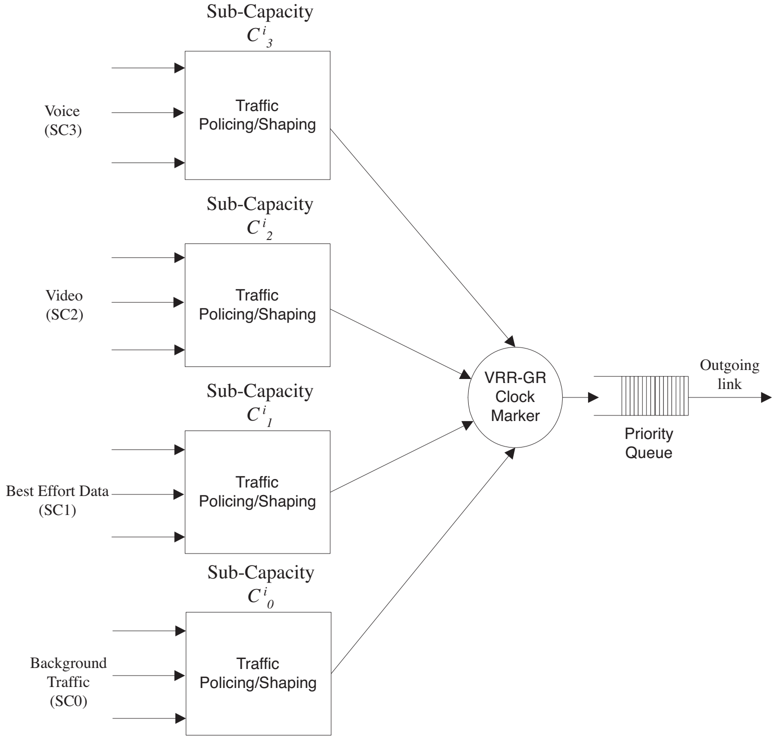

Fig. 3. The architecture of Extended GR Packet Scheduling

In VRR-GR, virtual sub-capacity and virtual offset-capacity must conform to the following constraint

$$\sum_{j=0}^{3} VC_j^i = \sum_{j=0}^{3} C_j^i = C^i \quad or \quad \sum_{j=0}^{3} O_j^i = 0. \tag{11}$$

Virtual offset-capacities, i.e., $O_3^i, O_2^i, O_1^i, O_0^i$, serve as priority indicators in VRR-GR algorithm. Greater virtual offset-capacity means more scheduling priority. $O_3^i, O_2^i, O_1^i, O_0^i$ may take positive or negative values. Positive/negative value results in lower/higher delay bound than conventional GR can achieve. In practice, we usually configure the parameters as $O_3^i > 0, O_2^i > 0, O_1^i < 0, O_0^i = -C_0^i$, so that video service can borrow some packet scheduling resource from best effort data and background traffic.

Assuming the reserved rate of a SC3 or SC2 flow f is r_f, then the virtual reserved rate of this flow is $r_f^v = r_f VC_3^i/C_3^i$ for SC3, or $r_f^v = r_f VC_2^i/C_2^i$ for SC2. As for SC1 or SC0, we consider all the connections in that service category as one flow. That is, the virtual reserved rate is $r_{SC1}^v = VC_1^i$ for SC1, or $r_{SC0}^v = VC_0^i$ for SC0. In this paper, we set $VC_0^i = 0$, which means that the SC0 packets will be scheduled only when the network node is idle. Moreover, inside the SC0 packets, network node employs the strategy of first in first out (FIFO). VRR-GR packet scheduling does have a negative impact on best effort and background traffic, since more resource is allocated to video/audio applications. However, above phenomenon shall not be viewed as a performance compromise in WMNs, since best effort and background traffic have no strict requirements on the delay bound.

As shown in Fig. 3, we assume that the packets are shaped and policed before entering the WMN, so that the following admission control rule is guaranteed:

$$\sum_{f\in a_j(t)} r_f \le C_j^i, \quad j = 0, 1, 2, 3 \tag{12}$$

where $a_j(t)$ denotes the set of SCj flows that are active at time t. It is clear that, if above condition holds, VRR-GR scheduling algorithm can provide SC1 and SC0 with their real sub-capacities, i.e., C_1^i and C_0^i, respectively.

3) Delay and Jitter Gain of VRR-GR Scheduling Algorithm: It is a salient feature that VRR-GR scheduling algorithm can provide video service with lower delay bound than conventional GR. Using Eq. (2) the delay bound of VRR-GR scheduling algorithm is given by

$$\begin{aligned} d_f^{'j} \le\; & GRC^1(p_f^j, r_f^v) - A^1(p_f^j) + (K-1) \max_{n\in[1\ldots j]} \frac{l_f^n}{r_f^v} \\ & + \sum_{n=1}^{K} (\beta^n + \tau^{n,n+1}). \end{aligned} \tag{13}$$

Accordingly, we define the delay gain of VRR-GR scheduling algorithm on video service as

$$\begin{aligned} DG_f^j = d_f^j - d_f^{'j} = & GRC^1(p_f^j, r_f) - GRC^1(p_f^j, r_f^v) \\ & + (K-1) \max_{n\in[1\ldots j]} l_f^n \left(\frac{1}{r_f} - \frac{1}{r_f^v}\right). \end{aligned} \tag{14}$$

In addition to delay, VRR-GR can suppress jitter as well. Let $d_f^{'} = \max_{j\ge1}(d_f^{'j})$, then the jitter of VRR-GR scheme, represented by $J_f^{'j}$, can be defined as a random variable ranged in $[0, d_f^{'}]$ with probability density function $p_V(J_f^{'j})$. We further assume that the probability density functions of conventional GR and VRR-GR are subject to the property of

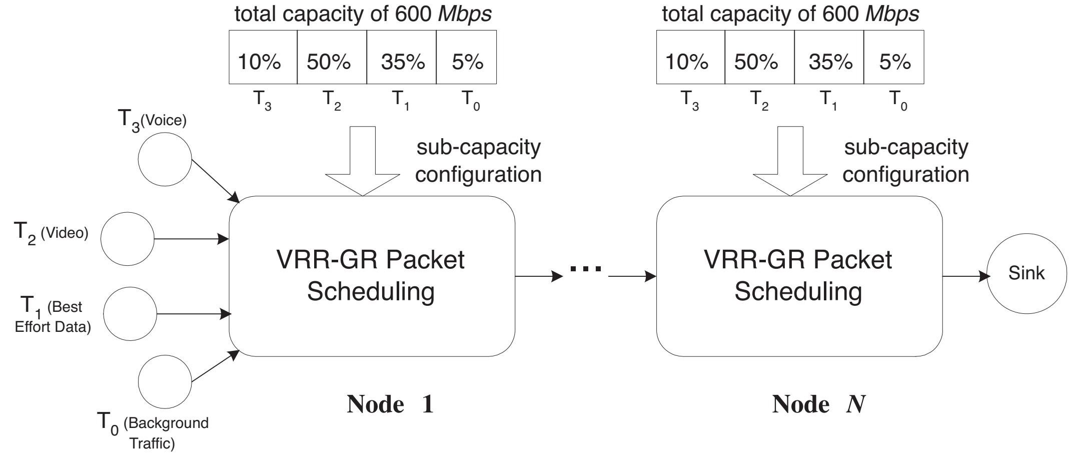

Fig. 4. Simulation model of VRR-GR packet scheduling algorithm

scaling similarity, i.e., $p_V(J_f^{'j}) = \frac{d_f}{d_f^{'}} p(\frac{d_f}{d_f^{'}} J_f^{'j})$. It then results in $E[J_f^{'j}] = \frac{d_f^{'}}{d_f} E[J_f^j]$, which implies a jitter suppression from VRR-GR as $d_f^{'} < d_f$.

V. PERFORMANCE EVALUATION

In this section, we present numerical results to demonstrate the advantages of our proposed VRR-GR scheduling algorithm in multipath routing framework for MD coded video communications. Particularly, we first study the performance of VRR-GR scheduling algorithm to find the appropriate parameter configuration for multiservice WMN. We then investigate the gain that VRR-GR scheduling algorithm can contribute to multipath routing approach, in terms of finding fully disjoint paths or maximally disjoint paths for video streams. As discussed in previous sections, GR packet scheduling could be practically implemented in the form of Virtual Clock, PGPS, or SCFQ. In our simulation study, we employ Virtual Clock to obtain the numerical results of conventional GR and VRR-GR.

A. Performance of VRR-GR Scheduling Algorithm

We have implemented a simulation using OPNET Modeler to evaluate the performance of VRR-GR through a line of N nodes (mesh routers in WMN). Fig. 4 illustrates that our simulation model consists of three classes of components, including traffic sources (T_3, T_2, T_1, T_0), VRR-GR nodes, and a sink. The traffic sources generate packets according to special profiles, with (T_3, T_2, T_1, T_0) representing the source of voice, video, best effort data, and background traffic, respectively. The N VRR-GR nodes are programmed to be identical and to schedule the packets using VRR-GR scheduling algorithm according to the configuration. The sink node is placed at the end of the line to recycle the packets.

Fig. 5 demonstrates the average packet delay of (T_3, T_2, T_1, T_0) while configuring voice and video as leaky bucket traffic sources. In this scenario, we characterize the simulation model in Fig. 4 with the following features.

1) VRR-GR nodes: The simulation involves $N = 5$ nodes, where each of them has a total capacity of 600 $Mbps$, with 10% allocated to SC3, 50% allocated to SC2, 35% allocated to SC1, and 5% allocated to SC0. Moreover, we vary O_1^i, the virtual offset-capacity of SC1, from 0 $Mbps$ to -210 $Mbps$, while keeping $O_2^i/O_3^i = C_2^i/C_3^i = 5/1$ and $O_0^i = -30$ $Mbps$. Due to the constraint $O_3^i + O_2^i + O_1^i + O_0^i = 0$, we know O_2^i is varying from $30 * 5/6 = 25$ $Mbps$ to $240 * 5/6 = 200$ $Mbps$ and O_3^i is varying from $30/6 = 5$ $Mbps$ to $240/6 = 40$ $Mbps$ accordingly.

2) Traffic source: T_2 and T_3 are configured to be leaky bucket traffic sources as shown in Tables I and II. T_1 consists of 35% of 600 $Mbps$ traffic, which includes ftp, email, and http with packet size randomly ranged in (0.5 $Kbytes$, 1.2 $Kbytes$). T_0 consists of 5% of 600 $Mbps$ traffic, which includes probing, backscatter, and misaddressed traffic with packet size randomly ranged in (0.5 $Kbytes$, 1.0 $Kbytes$).

3) Others: To simplify the simulation, we assume that the propagation delay of each link is 0 second.

Fig. 5 shows that video and audio services (T_2 and T_3) have lower packet delay as O_1^i decreases. This phenomenon implies VRR-GR has better performance than conventional GR, noting that VRR-GR degenerates to conventional GR when $O_2^i = 0$ $Mbps$ and $O_3^i = 0$ $Mbps$. Moreover, as O_1^i plunges from 0 $Mbps$ to -210 $Mbps$ and O_0^i is fixed to be -30 $Mbps$, the packet delay of best effort data (T_1) increases, whereas the packet delay of background traffic (T_0) keeps almost the same. In practice, we usually set $O_1^i = -80\% C_1^i$, so that best effort data still have $20\% C_1^i$ virtual sub-capacity.

Besides packet delay, jitter is another important QoS parameter. Fig. 6 illustrates the average jitter of (T_3, T_2, T_1, T_0) with the same simulation configuration that generates Fig. 5. It reveals that the jitter of T_2 is suppressed considerably as O_1^i plunges. We then conclude that VRR-GR can provide video services with better jitter performance than conventional GR.

Similar to Fig. 5 and Fig. 6, Fig. 7 and Fig. 8 demonstrate the simulation results of delay and jitter using EBB traffic source. This simulation uses the parameter configuration the same as in Fig. 5 and Fig. 6, except that leaky bucket traffic sources are changed to EBB traffic sources with $(\Lambda_f = 0.92, \gamma_f = 1.76)$. Fig. 7 and Fig. 8 show that EBB traffic source incurs larger delay and jitter than leaky bucket traffic source, since the former is not regulated as strictly as the later. However, in case of EBB, it still holds that VRR-GR has better performance than conventional GR in terms of delay and jitter.

TABLE I

TRAFFIC LOAD CONFIGURATION OF T_2 ($leaky\ bucket\ traffic\ that\ occupies\ 50\%\ of\ 600Mbps$)

	average rate r_f (per connection)	burst size σ_f (per connection)	packet size (per connection)	total bandwidth (all connections)
Traffic Class 1	200 Kbps	150 Kbits	0.8 Kbytes	25% of T_2 traffic
Traffic Class 2	400 Kbps	300 Kbits	1.0 Kbytes	35% of T_2 traffic
Traffic Class 3	1.5 Mbps	1 Mbits	1.2 Kbytes	25% of T_2 traffic
Traffic Class 4	2.5 Mbps	1.5 Mbits	1.5 Kbytes	15% of T_2 traffic

TABLE II

TRAFFIC LOAD CONFIGURATION OF T_3 ($leaky\ bucket\ traffic\ that\ occupies\ 10\%\ of\ 600Mbps$)

	average rate r_f (per connection)	burst size σ_f (per connection)	packet size (per connection)	total bandwidth (all connections)
Traffic Class 1	8 kbps	5 kbits	70 bytes	30% of traffic T_3
Traffic Class 2	16 Kbps	10 Kbits	90 bytes	10% of traffic T_3
Traffic Class 3	32 Kbps	20 Kbits	130 bytes	25% of traffic T_3
Traffic Class 4	64 Kbps	50 Kbits	210bytes	35% of traffic T_3

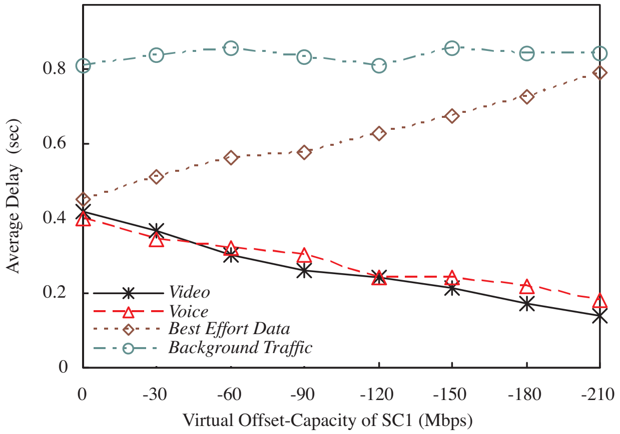

Fig. 5. Average packet delay through a line of VRR-GR nodes with leaky bucket traffic sources

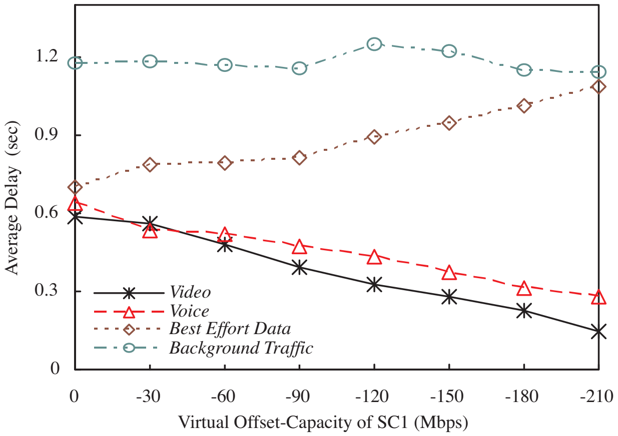

Fig. 7. Average packet delay through a line of VRR-GR nodes with EBB traffic sources

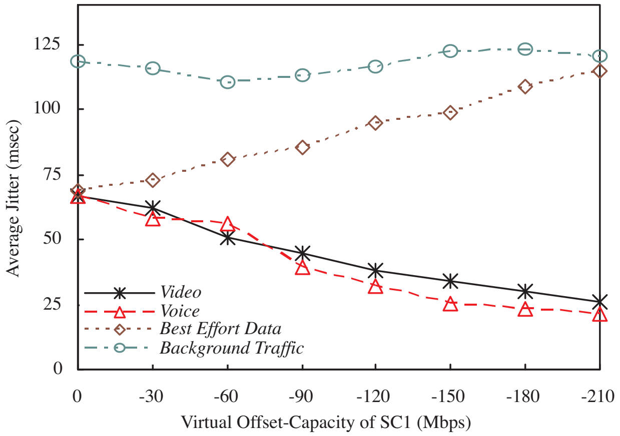

Fig. 6. Average packet jitter through a line of VRR-GR nodes with leaky bucket traffic sources

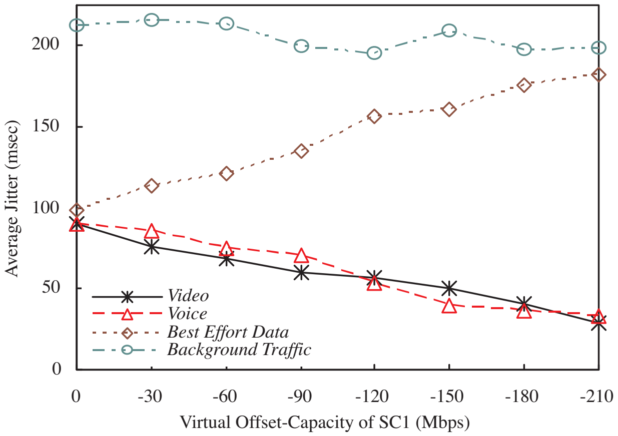

Fig. 8. Average packet jitter through a line of VRR-GR nodes with EBB traffic sources

In summary, Fig. 5, Fig. 6, Fig. 7 and Fig. 8 show that in a network of five nodes ($O_2^i/O_3^i = C_2^i/C_3^i$, $O_1^i = -80\%C_1^i$, $O_0^i = -C_0^i$) is a suitable parameter configuration to achieve low delay and jitter for video services while satisfying non-video services with their requirements as well. This conclusion from five node network is also true for the network of other sizes, although the numerical results are not shown here due to the limit of space.

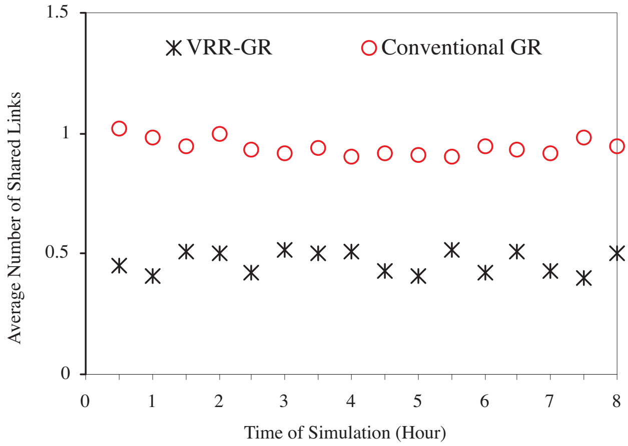

Fig. 9. The average number of shared links for all multipath routes during a simulation of eight hours.

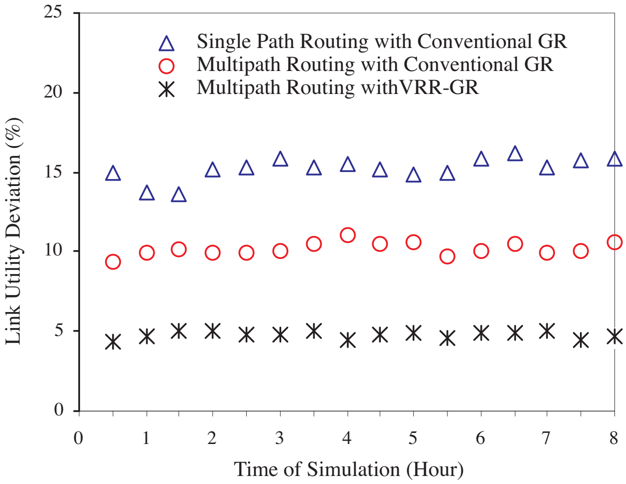

Fig. 10. Link utility deviation in WMN during a simulation of eight hours.

B. Performance of VRR-GR in multipath routing framework for MD coded video communications

This subsection presents numerical results to demonstrate the advantage of VRR-GR in multipath routing framework for MD coded video communications. Simulations are programmed using OPNET Modeler to emulate the IEEE 802.11 EDCA based WMN in Fig. 1. Specifically, Table III assigns each link of the WMN a weight, denoted by w_b^l, representing the bandwidth capacity of link l. Upon each link, we employ the complete sharing (CS) as the admission control policy on bandwidth resource according to Eq. (12).

In the simulation, we set the load of WMN as $3.0\ Gbps$ traffic, where voice, video, best effort data, background traffic take 10%, 50%, 35%, and 5% respectively and are configured as in the previous simulation scenario of EBB traffic over five VRR-GR nodes. For each connection request, the source/destination pair is randomly chosen from $\{G_i, 1 \leq i \leq 3\}$ and $\{r_i, 1 \leq i \leq 10\}$. Video and voice connections are given an arbitrary delay bound selected from 150 to 400 milliseconds. We apply different routing algorithms to different traffic load, i.e., multipath routing (two paths) for video services and shortest path (least hops) routing for others. Moreover, for VRR-GR scheduling, we run simulation with the configuration ($C_3^i = 10\%C^i, C_2^i = 50\%C^i, C_1^i = 35\%C^i, C_0^i = 5\%C^i$) and ($O_2^i/O_3^i = C_2^i/C_3^i$, $O_1^i = -80\%C_1^i$, $O_0^i = -C_0^i$); for conventional GR scheduling, we switch all traffic classes with the same priority.

We first compare conventional GR and VRR-GR packet scheduling algorithms in Fig. 9 regarding the number of shared links for multipath routed video traffic. It is observed that VRR-GR has much less average shared links than conventional GR during a simulation of eight hours. The reason is VRR-GR provides video traffic with good delay performance, which can significantly free the delay bound requirement when conducting multipath routing to find the maximally link-disjoint paths.

We next illustrate in Fig. 10 the load balancing performance of our multipath routing framework with conventional GR and VRR-GR packet scheduling algorithms, while using single path routing (shortest path first (least hops) for all SCs) as the reference. Fig. 10 evaluates the load balancing performance in WMN using the concept of link utility deviation, which is defined as

$$U_d = \sqrt{\frac{1}{N_l}\sum_{link\ l\ in\ WMN}(U_l - U_{avg})^2}. \qquad (15)$$

In Eq. (15), N_l stands for the number of links in WMN, U_l stands for the utility of link l, and U_{avg} stands for the average utility of all links. The above definition indicates that U_d is a concept involving all traffic classes, i.e., video and others.

Fig. 10 shows that multipath routing with conventional GR is setback by delay constraint, and achieves only limited load balancing gain over single path routing approach. VRR-GR, on the other hand, can significantly counteract the negative effect of delay constraint, and thus help multipath routing approach to obtain satisfying load balancing performance in WMN.

VI. CONCLUSIONS

This paper investigates the multipath delivery of MD coded videos over IEEE 802.11 based WMN. We begin our study with the framework design from traffic engineering perspective. We further identify that the strict delay requirement of video applications is the major obstacle hindering the performance of our proposed scheme. To solve this problem, we develop an algorithm of VRR-GR to give video traffic high preference during packet scheduling process. Simulation study shows that our proposed approach can shorten the packet delay of video communications and achieve satisfying load balancing performance in WMN.

APPENDIX
LIST OF NOTATIONS

AC i : access category i in IEEE 802.11e EDCA.
r_f : the reserved rate of flow f.
p_f^j : the jth packet of flow f.
l_f^j : the length of the jth packet of flow f.
$A^i(p_f^j)$: the arrival time of packet p_f^j at network node i.
$GRC^i(p_f^j, r_f)$: Guaranteed-Rate clock value for packet p_f^j at

TABLE III
THE WEIGHT CONFIGURATION: $w_b^l(Mbps)$

Links:	(G_1, r_1)	(G_1, r_4)	(G_2, r_2)	(G_2, r_7)	(G_2, r_8)	(G_3, r_6)	(G_3, r_9)
Weights:	400	500	400	600	400	660	400
Links:	(G_3, r_{10})	(r_1, r_2)	(r_1, r_3)	(r_2, r_7)	(r_3, r_4)	(r_3, r_6)	(r_4, r_5)
Weights:	500	400	600	450	300	700	400
Links:	(r_5, r_6)	(r_5, r_{10})	(r_6, r_7)	(r_7, r_9)	(r_8, r_9)		
Weights:	450	400	600	450	400		

network node i.
β^i : a constant depending on the scheduling algorithm at network node i.
C^i : the total capacity of network node i.
l^i_{max} : the maximum length of packet served by network node i.
d^j_f : the end-to-end delay of p^j_f.
$\tau^{K,K+1}$: the propagation delay between network node K and network node $K+1$.
$AP_f(t_1, t_2)$: the bits of flow f that arrive in the interval $[t_1, t_2]$.
σ_f : the burst size of leaky bucket traffic f.
Λ_f : the prefactor of EBB traffic f.
γ_f : the decay rate of exponential decay function for EBB traffic f.
J^j_f : the jth jitter of flow f defined by $|d^{j+1}_f - d^j_f|$.
SC i : the ith scheduling category in VRR-GR packet scheduling algorithm corresponding to AC i in IEEE 802.11e EDCA.
C^i_j : the sub-capacity allocated to SC j at VRR-GR network node i.
VC^i_j : the virtual sub-capacity allocated to SC j at VRR-GR network node i.
O^i_j : the virtual offset-capacity allocated to SC j at VRR-GR network node i.
r^v_f : the virtual reserved rate of flow f.
$a_j(t)$: the set of SCj flows that are active at time t.
DG^j_f : the delay gain of VRR-GR scheduling algorithm on video service.
U_d : the link utility deviation in a WMN.
N_l : the number of links in a WMN.
U_l : the utility of link l.
U_{avg} : the average utility of all links in a WMN.

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