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Resource Management and QoS Provisioning for IPTV over mmWave-based WPANs with Directional Antenna

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Abstract There is an increasing interest in 60 GHz millimeter-wave (mmWave) communication technologies for multi-Gigabit wireless personal area networks (WPAN), aiming to support broadband multimedia applications. Internet Protocol TV (IPTV) is an emerging killer application which requires high data rate and stringent quality of services (QoS) in terms of delay and packet loss. In this paper, we propose a method to efficiently support high definition video flows in a mmWave-based WPAN with QoS guarantee, considering the characteristics of both the IPTV traffic and the mmWave communication technology. We first quantify the effective bandwidth of IPTV video sources using a simple, two-level Markov traffic model. Considering the overheads of the protocol stack in mmWave WPANs, we then quantify the minimum channel time needed for each IPTV flow. Since mmWave-based WPANs will deploy directional antennas to not only extend the transmission range, but also reduce the interference level to neighboring flows, we further propose an admission control scheme and scheduling algorithm to improve the network resource utilization by taking advantage of concurrent transmissions. Extensive simulations with NS-2 using real video traces have validated our analysis and demonstrated the efficiency and effectiveness of the proposed schemes, which will be an enabling technology for future mmWave-based WPANs supporting IPTV services.

Keywords IPTV · QoS · admission control · concurrent transmission · mmWave WPAN

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1 Introduction

The 7 GHz unlicensed spectrum around 60 GHz opens up numerous opportunities for future high data rate (multi-Gigabit), short-distance, wireless applications. Because of the strong interests of using the 60 GHz communication for future last-meter wireless access, the IEEE 802.15.3 Task Group 3c (TG3c) was formed in March 2005 to develop a mmWave-based alternative physical layer (PHY) for the existing 802.15.3 high rate wireless personal area network (WPAN) Standard, operating in the unlicensed 57-63 GHz frequency band [1]. With the recent advances in silicon circuits, the mmWave prototype chipsets are within reach [2].

On the other hand, broadband multimedia applications, such as Internet Protocol Television (IPTV) and mobile TV, are emerging killer applications that will be transfered over the ubiquitous IP networks anywhere, anytime. High-definition (HD) video streams have very high data rate and stringent Quality of Services (QoS) requirements, which may not be possible to deliver in traditional, low data rate wireless systems. Therefore, the mmWave-based WPAN is considered an ideal candidate for last-meter IPTV distribution, because of its high capacity, high spatial reuse capability, and low interference with other wireless systems and electronic devices.

The ever-increasing popularity of wireless multimedia services leads to the ever-growing density of wireless devices [3]. For instance, in broadband hotspots such as airport and soccer stadium, users can use their personal wireless devices to view (on demand) a variety of video media and exchange rich multimedia information with their neighbors. Thus, researchers are striving to push up the limit of the number of users/applications that can share the available wireless resources. In other words, given the mmWave communication channels, it is of critical importance to design network protocols to efficiently utilize the resources and ensure the user-perceived QoS of multimedia applications.

Video streaming over variable bit rate (VBR) wireless channels has been extensively studied in the literature. In [4–6], different adaptation schemes were proposed to provide QoS guarantee for a video flow, by managing the playout buffer and adjusting the playback rate at the end user, or adapting the transmission rate according to the channel conditions. Call admission control based QoS provisioning for video traffic in a network environment, e.g., ATM and cellular networks, was investigated in [7, 8], using either an effective bandwidth approach or a measurement-based approach. For resource management in mmWave networks, an exclusive region (ER) based resource allocation scheme was proposed in [9] to exploit the spatial multiplexing gain of mmWave WPANs supporting persistent traffic flows. The optimal ER size towards the maximum capacity of mmWave UWB networks with omni-directional and directional antennas was analytically obtained in [10]. In [11], a multihop MAC architecture for in-room mmWave WPAN with directional antennas was proposed to provide robust network connectivity in scenarios where single hop communication suffers significant outages due to obstacles. To the best of our knowledge, efficient resource management and QoS provisioning for high quality TV/video applications over mmWave-based WPANs is still an open area, which is the major motivation of this work.

The main contributions of this paper are as follows. First, we investigate the traffic characteristics and QoS requirements of IPTV, and the special features of mmWave-based WPAN. Then, we quantify the minimum channel time that should be allocated to each IPTV flow to ensure its QoS. Second, a novel admission control and scheduling algorithm allowing concurrent transmissions is proposed, which exploit the spatial multiplexing gain of mmWave-based WPANs using directional antennas. In addition, extensive simulations with NS-2 simulators using real HD video traces have demonstrated the effectiveness and efficiency of the proposed admission control and scheduling scheme.

The remainder of the paper is organized as follows. In Section 2, we present an overview of the network architecture and unique features of mmWave-based WPANs; we further investigate the IPTV traffic characteristics and describe a two-level traffic model for IPTV. In Section 3, we first analyze the effective bandwidth for an IPTV flow, i.e., the minimum bandwidth that can guarantee its QoS requirements. Then, we calculate the minimum channel time that should be allocated to each flow, considering its effective bandwidth and system overheads. Next, we propose a scheduling algorithm that allows concurrent transmissions efficiently. Simulation results with NS-2 are presented in Section 4, followed by concluding remarks and future work in Section 5.

2 System model

2.1 mmWave-based WPANs

The IEEE 802.15.3 standard defines the basic structure and MAC protocol of high date rate WPANs [12]. In this subsection, we first present an overview of the standard and then discuss the special features of mmWave-based WPANs.

A hybrid MAC protocol is adopted in the IEEE 802.15.3 WPAN standard: it uses random access periods for network initiation/association and resource requests, etc., and contention-free periods for scheduled data transmission. Several wireless devices can autonomously form a piconet in which one of them is selected as the piconet coordinator (PNC). The PNC can collect global information about the piconet and allocate wireless resources and schedule transmissions for all devices in the piconet according to their requirements. At the scheduled channel time, devices can communicate in a peer-to-peer fashion. Such a semi-ad hoc setting can provide better QoS than a pure ad hoc network.

Timing in the 802.15.3 WPAN is based on a superframe structure, which is illustrated in Fig. 1. IEEE 802.15.3 defines two methods for communicating data between devices [12]: a) sending asynchronous data or communicating commands in the contention access period (CAP), if present; b) allocating channel time for isochronous and asynchronous streams in the channel time allocations period (CTAP). Although both commands and asynchronous data can be transmitted in the CAP, it is recommended that only commands be transmitted to minimize the length of contention period. This is desirable for reducing the protocol overheads and potential collisions. In addition, for multimedia applications like IPTV, devices may need channel times on a regular basis, and they send channel time requests during the CAP to reserve isochronous channel time in the CTAP. Based on the successfully received requests from all devices, the PNC will schedule and allocate channel time in the CTAP to all devices in a Time Division Multiple Access (TDMA) manner.

For mmWave-based WPANs, we should further consider their special features and design resource management schemes and network protocols accordingly. Since oxygen absorption peaks at around 60 GHz, mmWave signals attenuate much faster in the air than lower frequency signals [13]. Thus, directional antenna and associated techniques are not only useful, but essential for mmWave to extend transmission range and improve transmission quality. On the other hand, because of the small wavelength of mmWave communication systems, it is feasible to implement multiple antenna elements in a single device to achieve high directivity gain. As shown in Fig. 2, compared to an omni-directional antenna which distributes signal energy equally in all directions (in dotted lines), a directional antenna can achieve a higher transmission gain over a longer distance (in solid lines) by

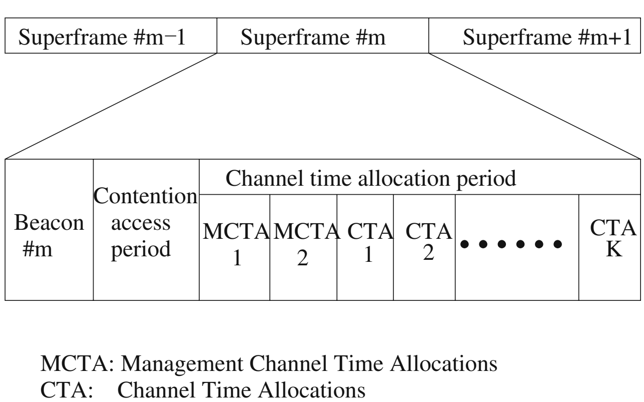

Fig. 1 IEEE 802.15.3 superframe structure

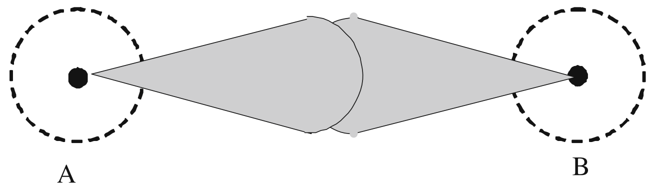

Fig. 2 Transmission and reception patterns of directional antenna

radiating the signal energy only in the desired direction [14]. With directional antennas, we can further exploit spatial multiplexing gain and allow the peer-to-peer transmissions to occur concurrently. Therefore, in our proposed scheduling algorithm, we will consider the usage of directional antenna to schedule appropriate concurrent transmissions that can significantly improve the resource utilization efficiency and guarantee the QoS requirements of IPTV.

2.2 IPTV traffic model

IPTV traffic with HD content is typically encoded by MPEG-4 (H.264), which has a high compression ratio, resulting in highly variable data rates (VBR) for the compressed videos. A continuous video stream is sampled to generate a sequence of frames as input to the encoder. The main task of a video coder is to remove the spatial and temporal redundancy within each and consecutive frames to save bandwidth. After encoding, frames are emitted periodically, comprising Group of Pictures (GoP). Each GoP contains an I frame and a number of P and B frames. For example, with an MPEG codec, the generic GoP is $I\ B_1\ B_2\ P_1\ B_3\ B_4\ P_2\ B_5\ B_6\ P_3\ B_7\ B_8$. The first frame in each GoP is an I frame, which is intra-coded without reference to any other frames. The subsequent P frames are both intra-coded and inter-coded with respect to the previous P or I frame. The remaining B frames are also intra-coded and inter-coded, and they use both the previous and following P or I frames as references. Since only I frames are encoded without exploiting temporal redundancy, typically an I frame has the largest frame size in each GoP.

A good IPTV traffic model should be able to consider the temporal and spatial correlations of video, and the inter-GoP and intra-GoP correlations. A simple, two-level Markov model was proposed in [15] for IPTV traffic, which can be used to derive the effective bandwidth of an IPTV flow and can be easily incorporated to any network simulators. Here, we briefly describe the steps to obtain the traffic model which contains a GoP-level Markov chain and a frame-level Markov chain.

In the GoP level, video data rate depends on the texture and motion complexity of the video content, or its spatial and temporal domain correlations. In the spatial and temporal domains, we categorize the video into a number of levels, S and T, respectively. Thus, we can use $S \times T$ states to represent the correlations in both domains. Since the duration of a GoP is less than half a second, the spatial and temporal correlations of the video source in each GoP are assumed to be at the same level, or in the same state. Experimental results show that choosing $S = 3$ and $T = 3$ makes a good trade-off between model accuracy and complexity. The three correlation levels in each domain are denoted as low (L), medium (M) and high (H) states, as shown in Fig. 3a. In the spatial domain, since the I frames are independently intra-coded, the I frame size is used to determine the texture complexity of the entire GoP. In the temporal domain, the ratio of the size of the first P frame, P_1, to the I frame size in the same GoP is used to indicate the temporal correlation. This is because P_1 and I have similar texture complexity, and the ratio of their frame sizes reflects the motion vector from the I frame to the following P_1 frame.

Since the GoP-level Markov chain cannot capture the burstiness of the traffic arrival rate with the GoP, a frame-level Markov chain is further established to capture the intra-GoP correlation. The time step for the frame-level Markov chain equals the duration of a video frame. Each state in the GoP-level model corresponds to a 12-step Markov chain at the frame-level, as shown in Fig. 3b. Each state in the frame-level model corresponds to a different frame type with a different traffic arrival rate. The state transition probabilities within the GoP are deterministic. The size of an I frame is determined by the spatial domain correlation only. Therefore, the I frame sizes in states XL, XM, and XH are the same, where $X = L, M, H$. A simple method to determine the frame size of each I frame state is to average the size of all I frames belonging to that state. Similarly, we can determine the P_1 frame size for each state using averages. For the remaining B and P frames, since the correlation coefficients of frames in the same GoP are very high, the remaining P/B frame sizes in each state are generated based on the P_1 frame size using the linear equations

$$\overline{F_T^K} = \alpha_T^K \overline{P_1^K}, \quad \text{for } T \in \{B_1, B_2, ..., B_8, P_2, P_3\}, \tag{1}$$

where the α_T^K's are constant coefficients determined by the average frame size of T.

The discrete-time Markov model can be translated to a continuous-time one, and the transition rate matrix M of the continuous time Markov chain is given by

$$\mathbf{M} = f(\mathbf{P} - \mathbf{I}) \tag{2}$$

where f is the frame frequency of the video source, $\mathbf{P}$ is the state transition probability matrix obtained from the video trace file, and $\mathbf{I}$ is the identity matrix with the same rank as $\mathbf{P}$.

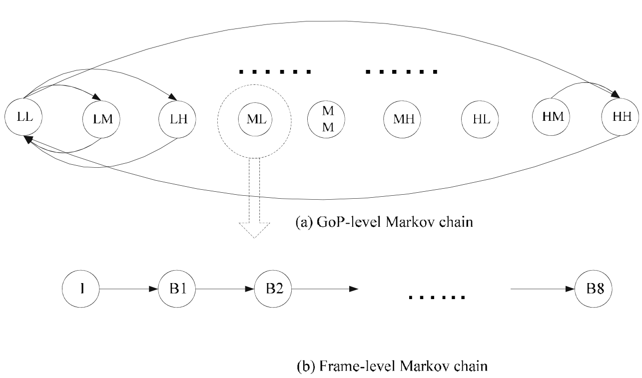

Fig. 3 Two-level Markov model for IPTV traffic

3 Resource management

3.1 Stochastic QoS and effective bandwidth for IPTV flows

Generally, the main QoS metrics of IPTV applications include throughput, delay, jitter, and packet loss. First, different from traditional low rate multimedia services, IPTV service can be very bandwidth-intensive. An IPTV flow typically requires from several Mbps (for standard definition video) to tens or hundreds of Mbps (for high definition video). Thus, appropriate bandwidth allocation is of critical importance for supporting IPTV over WPANs. Second, IPTV traffic have strict delay constraints. Because IPTV packets suffering excessive delay are useless for presentation and have to be discarded by the receiver, we should determine the maximum queueing delay in the bottleneck link, and calculate the maximum buffer size that can bound the queueing delay. For example, with the available bandwidth of 20 Mbps and the delay bound of 60 ms for the last-meter transmission over the WPAN, the maximum buffer size is obtained as $B = 60 * 20/8 = 150$ packets with a packet payload of 1000 bytes. The delay jitters resulting from the last hop transmissions can be efficiently reduced by employing a playout buffer at the receiver, which is not the main focus of this work. Third, with highly efficient video coding, high packet loss rate will severely degrade the user-perceived video quality. The industry standard requires that the packet loss rate

(PLR) of IPTV applications should be below 10^{-6} [16]. Therefore, we need to quantify the buffer overflow probability to limit the PLR.

To support IPTV applications, we need to jointly consider efficient resource management and delay performance provisioning. Specifically, we use the effective bandwidth to provide stochastical delay guarantees and achieve statistic multiplexing gain for VBR video flows. The effective bandwidth of an IPTV flow is the minimum amount of bandwidth required to satisfy its QoS, which depends on the characteristics of the source traffic in conjunction with the QoS constraints [17, 18]. Given the two-level Markov model for IPTV traffic introduced in Section 2.2, we can quantify the required effective bandwidth with the achieved the PLR performance using the fluid-flow approach [19].

Let $F_i(x)$ denote the probability that the queue length is less than x, given that the video source is in state i ($i \in \{1, 2, \cdots N\}$). $\mathbf{F}(x)$ is the row vector $[F_0(x)\ F_1(x)\ ...\ F_N(x)]$. As shown in [19], the equilibrium queue length distribution at the bottleneck link is subject to

$$\frac{d\mathbf{F}(x)}{dx}\mathbf{D} = \mathbf{F}(x)\mathbf{M} \tag{3}$$

where $\mathbf{M}$ is the rate transition matrix from Eq. 2 and $\mathbf{D}$ is an $N \times N$ diagonal matrix

$$\mathbf{D} = \mathrm{diag}\{\mathrm{D_1 - C, D_2 - C, ..., D_N - C}\}.$$

C is the allocated bandwidth and D_i is the source rate generated in state i, which equals the frame size of the state over the frame duration. For the two-level Markov model in Fig. 3, the frames in a generic GoP have nine states "LL, LM, LH, ML, MM, MH, HL, HM, HH", and thus the total number of states is $N = 12 * 9 = 108$. Given the QoS constraints in terms of delay bound and PLR, we can choose an appropriate buffer size to bound the queueing delay and limit the maximum number of connections to ensure the PLR performance. The cumulative distribution function (CDF) of the queue length is given by,

$$F(x) = 1 + \sum_{i:Re[z_i<0]} a_i \sum_{j=1}^{M} \phi_{ij} \exp(z_i x), \tag{4}$$

where z_i and $\vec{\Phi}_i$ are respectively the eigenvalue and eigenvector of $\mathbf{MD^{-1}}$, and a_i's are the coefficients that can be derived according to the conditions that the queue is empty for the overload states $J_0 \leq j \leq N$,

$$F_j(0) = 0 = \pi_j + \sum_{i=J_0}^{N} a_i \phi_{ij}.$$

π is the vector representing the state probability of the Markov chain and satisfies $\pi\mathbf{M} = 0$. Thus, the probability of packet loss due to buffer overflow is

$$\begin{aligned} G(x) &= 1 - F(x) \\ &= -\sum_{i:Re[z_i<0]} a_i \sum_{j=1}^{M} \phi_{ij} \exp(z_i x), \end{aligned} \tag{5}$$

Since the PLR is a non-increasing function of C, given the buffer size and the traffic model, we can use a simple binary search algorithm to determine the minimum C (i.e., the effective bandwidth of the flow) that satisfies the delay and PLR requirements.

3.2 Resource reservation in mmWave WPANs

Given the effective bandwidth calculated in the above subsection, we need to quantify the number of time slots reserved for each flow, considering the achievable raw data rate of the communication channel and the overheads from the protocol stack. The result should also reveal the impact of system parameters on resource utilization, which will be useful guidelines for network planner and service providers.

First, with large bandwidth, the PHY layer of mmWave-based WPANs will use different spreading factors and forward error correcting codes to adapt the raw data rate according to the received signal-to-noise ratio (SNR). For instance, in [20], using the single carrier mode, the mandatory data rate can be adapted in the range of 50 Mbps to 1.5 Gbps. According to the Shannon capacity of additive white Gaussian noise (AWGN) channel, we obtain the achievable raw data rate R

$$R = \eta W \log_2(SNR + 1), \tag{6}$$

where W is the signal bandwidth, and η (< 1) indicates the efficiency of the transceiver design.

The average received SNR can be estimated using the path-loss model:

$$SNR = \frac{P_t G_t G_r \cdot d^{-\alpha}}{P_n}, \tag{7}$$

where P_t is the transmission power, G_t and G_r are the transmitter and receiver antenna gains, respectively, d is the transmission distance between the sender and the receiver, α is the path loss exponent, and P_n is the noise power. We use the flat-top model for directional antennas, and assume the antenna gain is a constant within the radiation angle and zero outside [21]. Therefore, given the radiation angle θ, the antenna gain within the beam is $2\pi/\theta$, and that outside the beam is zero.

Next, we consider the PHY and MAC overheads in a mmWave WPAN to determine the number of time slots reserved for an IPTV flow, based on its effective bandwidth. As shown in Fig. 4, for each link-layer frame, the overhead includes preamble transmission time, frame headers (which include PHY header, MAC header, HCS, parity symbols, and optional frame header), the short inter-frame space (SIFS), and acknowledgment (ACK). Since frame header is typically transmitted using the basic data rate R_0 (e.g., 28 Mbps) which is much lower than the raw data rate (up to a few Gbps), it is desirable to aggregate a number of data packets into a single frame to reduce the overhead and improve the transmission efficiency.

Let n be the maximum number of packets (subframes) that can be aggregated in a frame and P be the size of each subframe. If the superframe duration is S_T, the minimum number of time slots (s) reserved for each flow with effective bandwidth C should be

$$s = \left\lceil \frac{C \cdot S_T}{nP}\left(\frac{nP}{R} + \frac{\text{frame header}}{R_0} + \text{SIFS} + \text{ACK}\right)\right\rceil \text{ (slots)}. \quad (8)$$

Note that the required number of slots may not be multiples of a slot time to satisfy the effective bandwidth plus the protocol overheads of video transmissions, and we take the upper bound of the required slots, which allocates more than required minimum bandwidth that can further reduces the PLR of video flows.

3.3 Admission control and concurrent scheduling algorithm

Given the required channel time for each flow, the PNC can use admission control and appropriate scheduling schemes to ensure the QoS of all admitted flows. Considering the use of directional antenna in mmWave WPANs, we can further improve the resource utilization by allowing conflict-free video flows to transmit concurrently. Two flows can transmit in

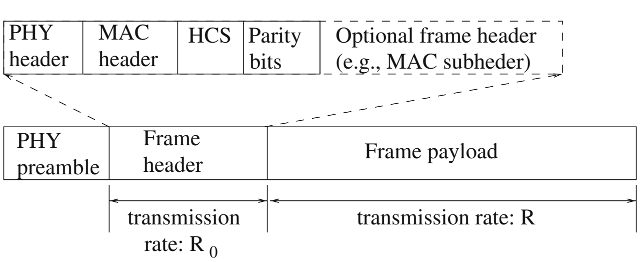

Fig. 4 Frame structure

Algorithm 1 Admission control & concurrent scheduling

BEGIN:
1: A new flow f_i requests s_i slots
2: **for all** Non-empty group (G_j != *Null*) **do**
3: **if** f_i does not collide with any flow in G_j **then**
4: **if** f_i requires extra slots ($s_i - s_{G_j} > 0$) **then**
5: **if** The remaining slots $S > s_i - s_{G_j}$ **then**
6: Schedule f_i in G_j;
7: Update the available slots $S = S - (s_i - s_{G_j})$;
8: Update the reserved slots for group G_j ($s_{G_j} = s_i$);
9: go to END;
10: **else**
11: go to line 18;
12: **end if**
13: **else**
14: Schedule f_i in G_j;
15: go to END;
16: **end if**
17: **end if**
18: Next group;
19: **end for**
20: **if** The remaining slots $S > s_i$ **then**
21: Schedule f_i in a new group G_k;
22: Update the reserved slots for group G_k ($s_{G_k} = s_i$);
23: Update the available slots $S = S - s_i$;
24: go to END;
25: **else**
26: Reject f_i due to no sufficient network resource;
27: go to END;
28: **end if**
END;

the same time slot if and only if they do not conflict with each other, i.e., the data and ACK frames of one flow do not interfere with those of the other flow. Because the transmission range can be effectively extended by the directional antenna, we assume two flows are conflict-free if the radiation angles of any two transceivers belonging to two flows do not point at each other. The pseudo-random code for the admission control and concurrent scheduling algorithm is given in Algorithm 1.

Initially, the total available slots per superframe is S. When the first flow f_0 requests s_0 slots, the scheduler (PNC) checks if there are enough available network resources to support this flow and schedules it in the first group G_0 if $S > s_0$. The scheduler allocates s_0 slots to flows in G_0, $s_{G_0} = s_0$, and updates the available number of slots by $S \leftarrow S - s_0$. Without loss of generality, for any new flow, f_i, requesting s_i slots, the scheduler checks if f_i satisfies the concurrent transmission condition with the existing flows. If f_i can concurrently transmit with all flows in G_j, the scheduler checks whether the allocated slots for G_j is sufficient for f_i, $s_{G_j} \geq s_i$. If yes, f_i can be scheduled in G_j. Otherwise, the scheduler needs to check if the available number of slots is larger than the extra required slots of f_i, and schedules f_i in G_j if and only if $S > s_i - s_{G_j}$. If the condition is satisfied, the scheduler updates the allocated slots for all flows in G_j, $s_{G_j} = s_i$, and the available slots $S \leftarrow S - (s_i - s_{G_j})$. If f_i cannot concurrently transmit with some flows in the existing groups, the scheduler will schedule f_i in a new group if the network resources is sufficient, and updates the group and available slots information accordingly. f_i will be rejected if f_i can not be accommodated in any existing groups and the available number of slots is not sufficient to support f_i.

In the proposed scheduling scheme, at least s_i slots are allocated to satisfy the required bandwidth of flow i and guarantee its stochastic delay performance. We simply group conflict-free video flows for concurrent transmissions which may result in more than required bandwidth allocation to some video flows. We can further improve the resource utilization by allocating exact required bandwidth to each flow at the expense of scheduling complexity. With very high data rate (e.g., a few Gbps) mmWave communication technologies, data transmissions are usually on the order of μs, and a complex scheduler requiring heavy computation is less desired. By exploiting the spatial reuse opportunities of mmWave channels, the resource utilization can be significantly improved with the simple and efficient concurrent scheduling scheme.

4 Simulation results

We use Network Simulator (NS2-2.32) and a real video source, "From Mars to China" in HDTV format (1920 x 1080i) [22] to validate the analysis. The frame level and GoP level statistics of the video trace are listed in Table 1. The simulated network is setup in a 10 m × 10 m room. The senders and receivers of video flows are randomly distributed in the area. The simulation parameters are listed in Table 2. The signal bandwidth is 500 MHz. The senders use the maximum transmission power $P_t = 0.1$ mW and the background noise power is -117 dBm/MHz. mmWave signals attenuate fast in the air due to oxygen absorption and atmospheric attenuation and the path loss is measured as 71.5 dB at the reference distance 1.5 m with path loss exponent $\alpha = 2$ [23]. The radiation angle of transceiver is set to $\theta = 90$ degrees unless otherwise specified, and the corresponding antenna gains are $G_t = G_r = 360/90 = 4$. To further protect the concurrent transmissions of video flows and allow a certain level of angle estimation errors, we use a 15-degree angle margin at each side. The parameter η is set as 0.293 so that the achievable data rate at the reference distance is 1 Gbps. The duration of a superframe is 20 ms, and the duration of a time slot is 17.3 μs. There are totally 1,000 time slots per superframe available for CTA, and the remaining 2.7 ms is used for beacon period and contention access period. Each simulation lasts 1200 s. To eliminate the warming-up effects, the results of the initial 20s are not counted. We repeat the simulation 100 times with different random seeds to calculate the average value.

We first study the relationship of the reserved bandwidth and the buffer size. For delay sensitive multimedia flows, the maximum buffer size is determined according to the maximum tolerable queueing delay. The packet loss rate, which is obtained as the ratio of the maximum number of dropped packets over the total number of transmitted packets, is tabulated in Table 3. For a higher reserved bandwidth, a larger buffer size can be applied for a given delay bounds, e.g., 60 ms, and a lower packet loss rate is achieved. Both the simulation and analytical results show that, for a 30 Mbps reserved bandwidth, the PLR is on the order of 10^{-6}. The simulation results demonstrate that the analysis using the

Table 1 Statistics of video trace

	Frame level	GoP level
Number	51715	4309
Mean size (bytes)	2e+4	2.4e+5
Variance of size	5.8e+10	1.2e+12
Mean bit rate	4.8e+6	7.8e+5
Peak bit rate	7.8e+7	3.5e+6
Peak/mean ratio	16.2	4.5

Table 2 Simulation parameters

Signal bandwidth (W)	500 MHz
Transmission power (P_t)	0.1 mW
Background noise (N)	-117 dBm/MHz
Path loss exponent (α)	2
Reference distance (d_{ref})	1.5 m
Path loss at d_{ref} (PL_0)	71.5 dB
Superframe length (S_T)	20 ms
A slot time	17.3 µs
Transceiver efficiency (η)	0.293
Radiation angle (θ)	90 degrees
Slot number (S)	1000

Table 3 Packet loss rate (PLR)

BW (Mbps)	Buffer (Packets)	Packet loss rate Analysis	Simulation, $n = 5$	Simulation, $n = 10$	Simulation, $n = 20$
20	150	7.87E−3	3.21E−3	3.02E−3	2.8E−3
25	187.5	6.35E−4	4.12E−4	3.66E−4	3.34E−4
30	225	1.19E−6	1.46E−6	1.53E−6	1.50E−6
35	262.5	5.79E−126	0	0	0
≥40	300	0	0	0	0

two-level Markov model is reasonable. We also observe that the average PLR performance does not change much with the aggregation parameter n. This is because our proposed concurrent scheduling scheme takes the transmission overheads into consideration and ensures the achieved throughput and the PLR performance of all admitted video flows.

We further study the PLR performance of each flow in Fig. 5. It can be seen that the PLRs of most video flows are on the order of 10^{-3}, 10^{-4} and 10^{-6} for 20 Mbps, 25 Mbps, 30 Mbps reserved bandwidth, respectively. Since many flows are concurrently transmitted, some concurrent flows may take advantages of the extra slots requested by other flows in the group and achieve better delay and throughput performance. With the proposed admission control and concurrent scheduling scheme, the delay and PLR performance of any single flow can always be guaranteed.

We then examine the number of admitted video flows under different scenarios. The relationship between the average number of admitted video flows and the reserved bandwidth is shown in Fig. 6. The maximum number of aggregated subframes is 10, i.e., up to 10 subframes can be aggregated in one single transmission. When there are 40 video flows requesting bandwidth, only around 17 video flows can be accommodated with a 20 Mbps reserved bandwidth using serial one-by-one transmission. The number of admitted video flows further decreases when a higher bandwidth is reserved to achieve better PLR performance. It is observed that only 10 flows can be supported with a 35 Mbps reserved bandwidth. Using the proposed concurrent scheduling scheme, all 40 flows can be accommodated in the network with a 20 Mbps reserved bandwidth, and 33 flows for 35 Mbps reservation with a PLR below 10^{-6}. With radiation angle $\theta = 90$ degrees, the probability that two flows cannot transmit concurrently is as low as $1/16$. Thus, a new flow is very likely to be scheduled for concurrent transmission with existing flows. When more video flows join the network, the number of admitted flows increases accordingly; but the increase rate slows down as the size of the network density increases, as shown in Fig. 7. With effective admission control, the packet loss performance can always be guaranteed and the network throughput is proportional to the number of scheduled video flows.

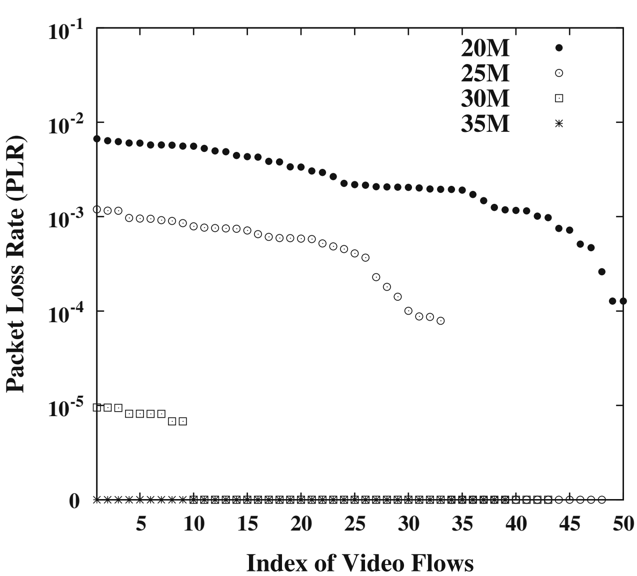

Fig. 5 Packet loss rate (50 flows, $n = 10$)

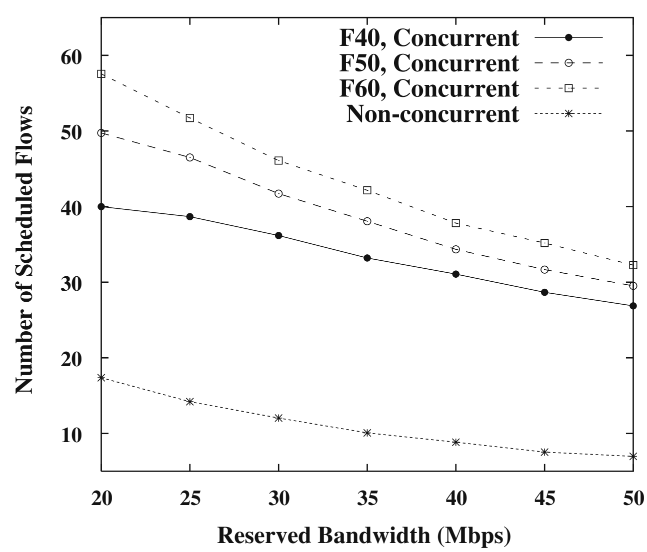

Fig. 6 Number of admitted flows vs per-flow reserved bandwidth

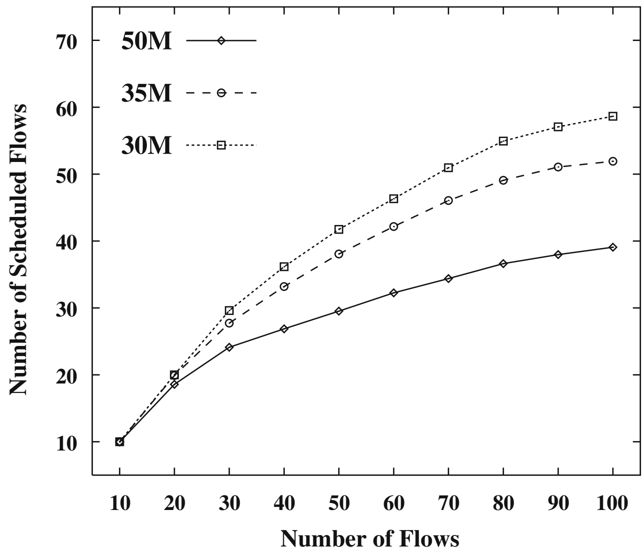

Fig. 7 Number of admitted flows vs network density

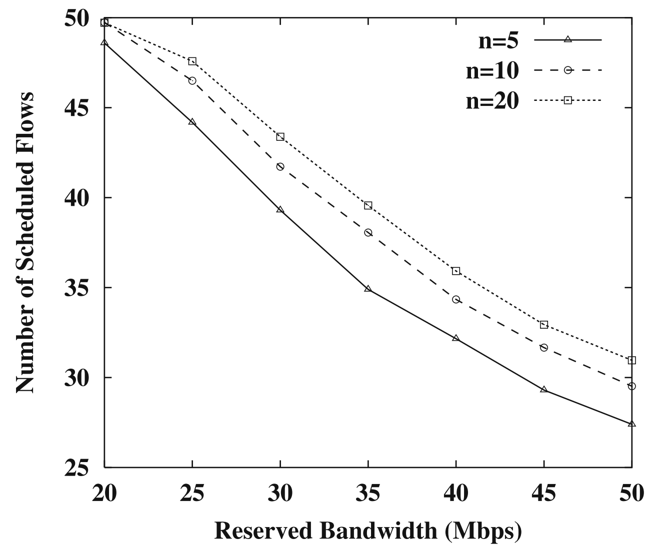

Fig. 9 Number of admitted flows under various aggregation parameters

By exploiting the high spatial multiplexing gain of mmWave WPANs, the proposed admission control and concurrent scheduling scheme can significantly improve the network throughput.

The number of admitted video flows under different radiation angles is shown in Fig. 8. With a smaller radiation angle, more flows can be scheduled for concurrent transmissions due to the reduced probability that two flows conflict with each other. In addition, a higher directivity gain can be achieved with a smaller radiation angle and thus a fewer number of slots is required. It can be seen in Fig. 8 that a smaller θ results in a larger number of admitted flows, especially for a higher reserved bandwidth.

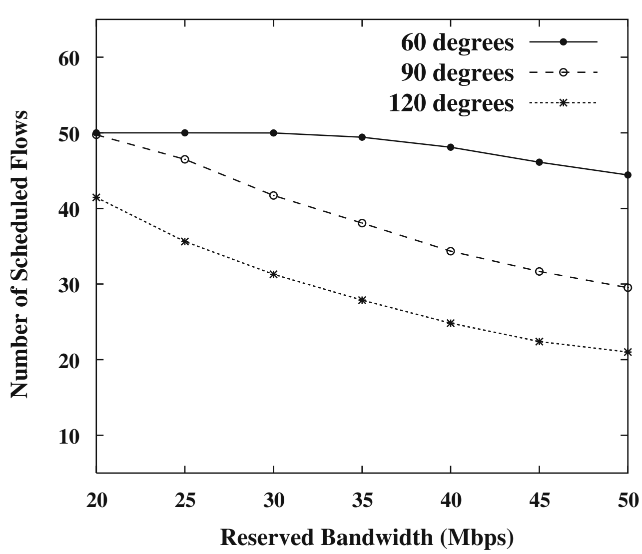

Fig. 8 Number of admitted flows under various radiation angles (50 flows)

We further investigate the impacts of parameter aggregation, n, on the network performance, and the results are shown in Fig. 9. Frame aggregation is generally used to improve the transmission efficiency and achieve high throughput. It can be seen that the maximum number of video flows that can be accommodated in the network increases from 34 for $n = 5$ to 40 for $n = 20$, with a 35 Mbps reserved bandwidth per flow. Less transmission overhead is involved when more subframes are aggregated in a single transmission. For bursty HD video flows, some video frames may exceed 300, 000 bytes, which may be segmented into more than 300 UDP packets of 1000 bytes each. In this case, aggregating multiple packets (subframes) in one frame transmission can significantly improve the transmission efficiency and the number of admitted video flows.

5 Conclusions

In this paper, we have quantified the effective bandwidth of IPTV video sources using a simple two-level Markov traffic model. The minimum channel time allocated for each IPTV flow has been derived, considering the overheads of the protocol stack in mmWave WPANs. We have further proposed an admission control scheme and scheduling algorithm which take advantage of concurrent transmissions in mmWave-based WPANs with directional antenna. Extensive simulations with NS-2 using real video traces have validated our analysis and demonstrate the efficiency and effectiveness of the proposed schemes. Future research is needed to analytically study the performance of the proposed schemes in the presence of fast fading and shadowing in the mmWave channel.

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