ENERGY-EFFICIENT COOPERATIVE MEDIA STREAMING OVER WLANS

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ABSTRACT

In wireless local area networks (WLANs), Energy-Efficient Media streaming, in particular, is a promising technology for providing services such as news clips, live sports.Toavoid service interruption when the node switching to low-power sleep mode to save energy,proper scheduling and data management strategies must be employed. However, if sleep time is not properly scheduled, significant delays can occur, which is undesirable in delay-constrained applications. In this paper, we propose an energy-efficient cooperativemedia streaming where access point transcode the transport stream to minimize the energy usage of mobile node. We first consider the multimedia transport stream transcoding and buffering to be made by the streaming access point.Second we consider the basic steps that the mobile node takes to decide when and how long it can sleep. For the multiuser scenario, to mitigate channel contention in packet downloading after sleeping, the sleep schedules requested by the nodes are coordinated by the access point (AP) to avoid overlapping active epochs. Cooperated media streaming solves the issues of transcoding delay in mobile node by giving transcoding and buffering responsibility to access points.

Keyword: Multimedia *Streaming, Online Transcoding, Energy efficiency, Resource reservation.*

I INTRODUCTION

In wireless local area networks (WLANs) based on the IEEE 802.11 standard have been applied in delay sensitive applications such as real-time multimedia [1]. Real-time transport of live video is the predominant part of real-time multimedia. In the streaming mode, the video content need not be downloaded in full, but is being played out while parts of the content are being received and decoded. Due to its real-time nature, video streaming typically has band-width, delay and loss requirements. Presently smartphones are normally equipped with WLAN interfaces that can support real-time multimedia data offloading [2]. WLANs also facilitate online calls through the voice over Internet protocol (VoIP) where audio and video data must be played out continuously [3], [4]. If the data does not arrive in time, the playout process will pause, which is annoying to human ears and eyes. Each packet needs to arrive at its destination before a prescribed deadline; otherwise, it will be dropped.

Recent advances in interactive region-of-interest (IRoI) video streaming [5] [6] offer a user's to independently choose an arbitrary region-of-interest (RoI) from a high-resolution video. Since only the user-selected RoIs are transmitted, IRoI video streaming system can provide high-quality RoIs playback and eliminate the need to transmit the full-spatial resolution video on a small screen display over a wireless network. Regardless of the availability of inexpensive high-definition (HD) video recording technology, constraints including small display sizes, limited bandwidth and low computational power are often the hurdles for delivering HD video content to mobile devices over a wireless network. One straight-forward approach to overcome these challenges is to either transcode a high-definition video into multiple file representations at lower spatial and quality resolution or the streaming service can employ scalable video coding (SVC), such as H.264/AVC-SVC extension [7], and create embedded bit streams for efficient rate and device capability adaptation. However, both of these approaches often lower the video quality of the entire scene.

Another critical challenge in wireless networks is saving energy [8]–[12]. The IEEE 802.11 standard defines a few power saving techniques for WLANs, including power-saving mode (PSM) [13] and automatic power-saving delivery (APSD) [14], which reduce energy consumption by switching the node from idle sensing mode to sleep mode. However, an energy-efficient design should also meet the network quality-of-service (QoS) requirements such as end-to-end delay [15]. If sleep time is not properly scheduled, significant delays can occur, which undesirable in IRoI video is streaming. Therefore, these power-saving techniques need to be enhanced to better accommodate sensitive delay constraints [8].

The GreenCall [16] algorithm takes advantage of this fact and puts the node into sleep mode according to the amount of spare time before the play out deadline. While sleeping, the downlink packets to the nodes are buffered at the access point (AP).When a node wakes up; it then retrieves the buffered packets from the AP and plays them out. The Sleep Scheduling [17] algorithm solve the issue of prolong waiting for channel availability by ensure that at any time there can be at most one node accessing the channel to retrieve buffered packets. To this end, each node can make a reservation with the AP in advance by sending a sleep request to the AP indicating that it would like to sleep now and will later occupy the channel after waking up. The AP checks whether there is any conflict with the schedules of other nodes and then decides to approve or decline the current sleep request. Once a request is approved, the node enters sleep mode; otherwise, it will stay active and send another request later. With this coordination mechanism, the active periods of different nodes are properly shifted to avoid collision, and a node can immediately start data retrieving after waking up.In streaming video applications running on WLAN devices must be simple. In particular, low decoding complexity is desirable. To address this issue, Lin et al. [18] employed a least-cost scheme to reduce decoding complexity.

In this paper, we develop energy-efficient cooperative IRoI media streaming technique for video streaming applications over WLANs where our goal is to minimize the transmission delay, power usage and decoding complexity of end device. In Fig. 1, raw video and audio data are pre- compressed by video compression and audio compression algorithms and then stored in transmit buffer. Upon the client's request, a streaming server retrieves compressed video/audio data from transmit buffer and then the application-layer QoS control module adapts the video/audio bit-streams according to the network status and QoS requirements. After the adaptation, the transport protocols packetize the compressed bit-streams and send the video/audio packets to the Internet. For packets that are successfully delivered to the access point, they first pass through the transport layers and then are processed by the application layer and buffered at the MPEG2 In buffer.

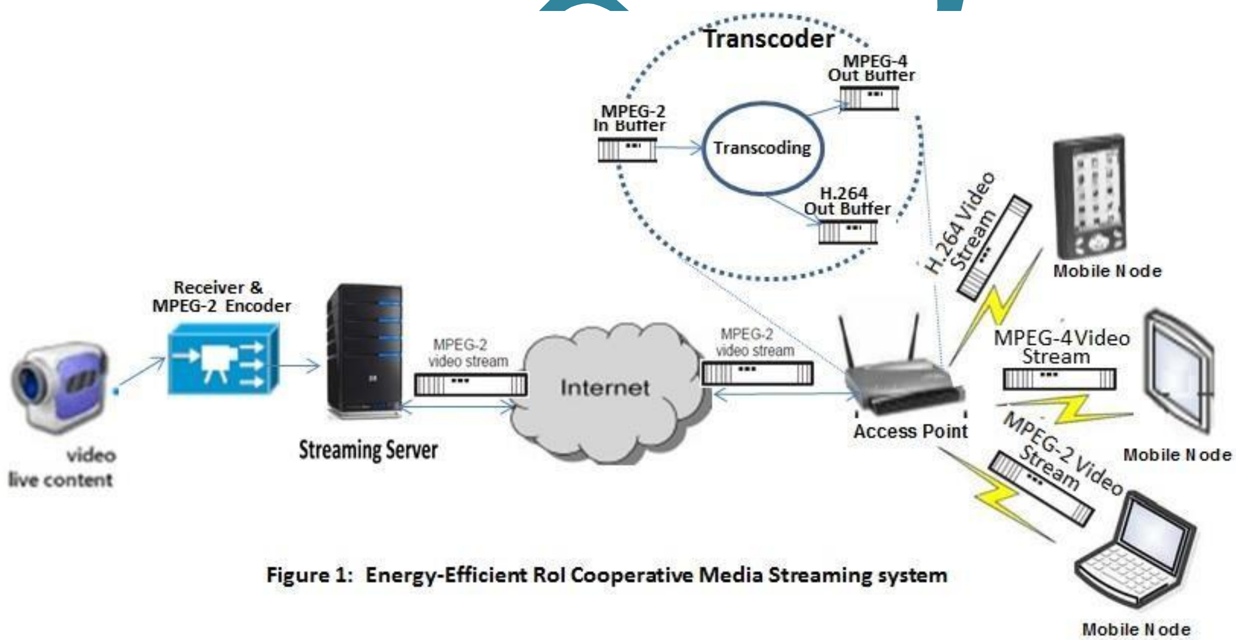

Figure 1: Energy-Efficient RoI Cooperative Media Streaming system

Access point will do the handshaking with mobile node to get the QoS requirement such as either H.264 or MPEG-4 streams interested by the mobile node. The buffered transport stream packet in access point will transcode a high-definition video into multiple file representations at lower spatial, quality resolution and create embedded bit streams for efficient rate and device capability adaptation. Each node download its region of interest file from access point then placed in its respective decoding buffer. Analyses results demonstrate that our techniques can significantly decrease streaming disruptions, reduce bandwidth consumption, increase energy efficiency than that of the existing Sleep Scheduling method.

The remainder of this paper is organized as follows.Section II reviews related work. Section III presents the problemstatements. Section IV develops the energy-efficient cooperative media streaming algorithms. Analysis results are presented in section VI and Section VII gives the concluding remarks.

II RELATED WORK

Real-time transport of live video is the predominant part of real-time multimedia. In the streaming mode, the video content need not be downloaded in full, but is being played out while parts of the content are being received and decoded. Due to its real-time nature, video streaming typically has band-width, delay and loss requirements.

In video streaming raw video and audio data are pre-compressed by video compression andaudio compression algorithms to achieve efficiency and then stored in transmit buffer. Since scalable video is capable of gracefully coping with the bandwidth fluctuations in the Internet [19], we are primarily concerned with scalable video coding techniques. Upon the client's request,a streaming server retrieves compressed video/audio datafrom transmit buffer and then the application-layer QoScontroltechniques adapts the video/audio bit-streams according to the network status and QoS requirements [20]-[23]. The application-layer techniques include congestion control and error control. Their respective functions are as follows. Congestion control is employed to prevent packet loss and reducedelay. Error control, on the other hand, is to improve video presentation quality in the presence of packet loss. Afterthe adaptation, the transport protocols packetize the compressed bit-streams and send the video/audio packets tothe Internet. Packets may be dropped or experience excessive delay inside the Internet due to congestion. To improve the quality of video/audio transmission, continuousmedia distribution services (e.g., caching) are deployed inthe Internet.For packets that are successfully delivered to the receiver, they first pass through the transport layers and then are processed by the application layer before being decoded at the video/audio decoder. To achieve synchronization between video and audio presentations, media synchronization mechanisms arerequired.

The region that are of interest to the viewers may also become too small to see when the original content is intended for large displays. Recent advances in interactive region-of-interest (IRoI) video streaming [5] [6] offer a user’s to independently choose an arbitrary region-of-interest (RoI) from a high-resolution video. Since only the user-selected RoIs are transmitted, IRoI video streaming system can provide high-quality RoIs playback and eliminate the need to transmit the full-spatial resolution video on a small screen display over a wireless network. Despite the availability of inexpensive high-definition (HD) video recording technology, constraints including small display sizes, limited bandwidth and low computational power are often the hurdles for delivering HD video content to mobile devices over a wireless network. One straight-forward approach to overcome these challenges is to transcode a high-definition video into multiple file representations at lower spatial and quality resolution. Alternatively, the streaming service can employ scalable video coding (SVC), such as H.264/AVC-SVC extension [7], and create embedded bit streams for efficient rate and device capability adaptation. However, both of these approaches often lower the video quality of the entire scene.

The carrier sense multiple access with collision avoidance (CSMA/CA)-based IEEE 802.11 medium access control (MAC) protocol requires the nodes to keep sensing the channel or staying in idle mode when they are neither transmitting nor receiving, which consumes a significant portion of the energy resources of mobile nodes. Therefore, a promising strategy for power saving is to switch nodes when doing idle listening to sleep mode by turning off the wireless interfaces, thus saving a considerable amount of energy. With the IEEE 802.11 PSM technique, a node periodically enters sleep mode and wakes up to retrieve buffered downlink packets from or report uplink packets to the associated AP [8], [17]. Downlink packets suffer from both the buffering delay at AP and the delay due to channel contention of the destination nodes after waking up. The GreenCall algorithm with the information of single-packet transmission deadlines, where they use dynamic sleep period to accommodate delay constraints [16]. The length of sleep period is selected to make sure packets can arrive before playout deadline. Since the downlink rate from the AP to the mobile node is much larger than the packet arrival rate, the node only needs to stay awake for a short time period to retrieve downlink packets and use the spare time for sleep.

The GreenCall algorithm works well for single user case, but when there are multiple nodes, the active periods of different nodes may overlap and the subsequent contention will lead to additional delay, which causes the packets tomiss their deadlines. The Sleep Scheduling [17] algorithm solve the issue of prolong waiting for channel availability by ensure that at any time there can be at most one node accessing the channel to retrieve buffered packets. To this end, each node can make a reservation with the AP in advance by sending a sleep request to the AP indicating that it would like to sleep now and will later occupy the channel after waking up. The AP checks whether there is any conflict with the schedules of other nodes and then decides to approve or decline the current sleep request. Once a request is approved, the node enters sleep mode; otherwise, it will stay active and send another request later. With this coordination mechanism, the active periods of different nodes are properly shifted to avoid collision, and a node can immediately start data retrieving after waking up.

The Joint Scalable Video Model (JSVM) reference software is a part of ISO/IEC 14496-10 standard [24] -[26]. In this standard, the video compression is performed by generating a unique hierarchical bit-stream structured in several levels or layers of information, consisting of a base layer and several enhancement layers. The base layer provides basic quality. The enhancement layers provide improved quality at increased computational cost. The region that are of interest to the viewers may also become too small to see when the original content is intended for large displays. Therefore, video retargeting techniques such as [27] [28] are proposed to automatically adapt to target displayby preserving the important region of the scene.

IETR has developed the Open SVC Decoder [25], a C language Scalable baseline profile decoder supporting all tools to deal with spatial, temporal and quality scalabilities. It is based on a fully compliant H.264 baseline decoder with most of the tools of the main profile. In the Scalable baseline profile the base layer has to be conformant with AVC baseline profile. In this profile, contrary to quality and temporal scalability which are supported without any restriction, the spatial scalable coding is restricted to 1.5 and 2 resolutions ratios between two successive spatial layers

III. PROBLEM STATEMENT

Here, we present the energy-efficient and cooperative region of interest video streaming problem over WLANs. WLAN can be used to bridge mobile VoIP nodes to the Internet, which is also advantageous over cellular Internet method in terms of low energy cost and extended coverage in indoor environments [16]. In the WLAN, the uplink and downlink communications for sending and receiving VoIP packets of the VoIP nodes are coordinated by an AP, based on the IEEE 802.11 protocol. Specifically, the CSMA/CA MAC protocol is applied to coordinate transmissions and avoid packet collisions. Upon receiving apacket, the AP will download it to the corresponding receivernode. Once the packet reaches the node, it will be stored in aplayout buffer for a while to compress jitter in playback. Forsmooth playback, the final playout time of a packet should notexceed a specific deadline; otherwise, the packet is dropped. Packet drop rate is an important measurement of the VoIP performance and is mainly caused by large delay introduced during transmission throughthe Internet.

Since energy is an important concern particularly for powerconstrainedmobile nodes, VoIP nodes with WLAN interfaceswant to work with low power while matching the playout deadlinerequirements. The contention-based CSMA/CA protocolincurs a considerable amount of energy waste for a node inidle listening. The PSM provided in the IEEE 802.11 protocolallowing a node's radio to switch from active to sleep modecan save a significant amount of idle listening energy. WithPSM [13], whenever a node wants to sleep, it should inform andget acknowledged by the AP, such that, during the node's sleepperiod, its downlink packets will be buffered at the AP forfuture transmission. The AP periodically broadcasts beaconscontaining traffic indication maps (TIMs) to indicate the bufferstate for each node. On the other hand, the node frequentlywakes up to check the beacons and prepares to downloadpackets from the AP if the corresponding TIM indicates to doso; otherwise, if there is no packet buffered in the AP, the nodewill go back to sleep again.

Despite the availability of inexpensive high-definition (HD) video recording technology, constraints including small display sizes, limited bandwidth and low computational power are often the hurdles for delivering HD video content to mobile devices over a wireless network. One straight-forward approach to overcome these challenges is to transcode a high-definition video into multiple file representations at lower spatial and quality resolution. Alternatively, the streaming service canemploy scalable video coding (SVC), such as H.264/AVC-SVC extension [21], and create embedded bit streams for efficient rate and device capability adaptation. However, both of these approaches often lower the video quality of the entire scene. The region that are of interest to the viewers may also become too small to see when the original content is intended for large displays.

PSM is suitable for VoIP communications in two aspects.Each packet has a certain level of delay tolerance; mobile nodes may have large portions of time without transmitting or receiving packets. Therefore, by turning the mobile nodes radio from the idlelistening state to the sleep state.First, Sleepperiods of the mobile nodes in WLAN must be appropriately scheduled, if its sleep period is excessively long, thedownlink packets buffered in the AP during the node's sleepperiod may suffer from long delays and may fail to reach thenode in time. Second, Multiple nodes in theWLAN must appropriately coordinated, if not appropriately coordinated then the active periods ofnodes may introduce downloading conflict to each other. Forexample, when a node wakes up it may have to wait for a whilesince the APis busy handling another node's downloading. Finally, in AP bufferedthe RoI video packet by recoding. For example, mobile node can download the user selected high- quality RoIplayback instead of downloading full-spatial resolution video. In the next section, webdesign energy-efficient cooperative RoI media streaming algorithms for VoIPsystems.

IV. ENERGY-EFFICIENTCOOPERATIVE MEDIA STREAMING

In this algorithms, Event manager of each node determine its desired transcoding type when the user activate the media then send the transcoding requestmessages to the AP spontaneously. Only when the transcoding request isaccepted by transcoding manager of AP will be transcoded. If the transcoding request is declined by transcoding manager of AP will ask the node to redetermine the transcoding type. (See Fig. 2).The cooperative transcoding algorithm is extended to multiuser caseswhere there is multiple mobile devices can associate with AP at the same time in WLAN.

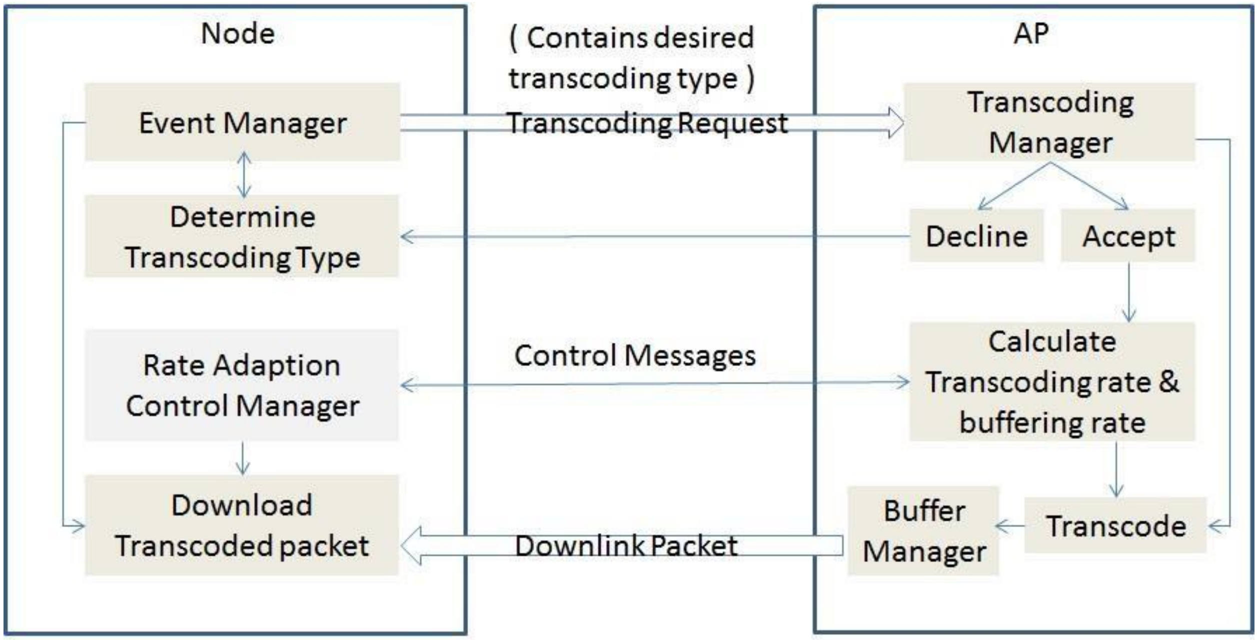

Fig 2. Block Diagram of System

A. Single-User Scenario

Generally, a node sends the request to AP with channel number and compression type. If the request is accepted by AP then a transcoding manager creates the thread for serving that node. Therefore, thread will transcode the multimedia packetsat AP then put it in a downlink buffer for download by the node. Taking advantage of this point,the node can switch to sleep mode for some time and wake upbefore the deadline to retrieve the packet from the AP and playit out.

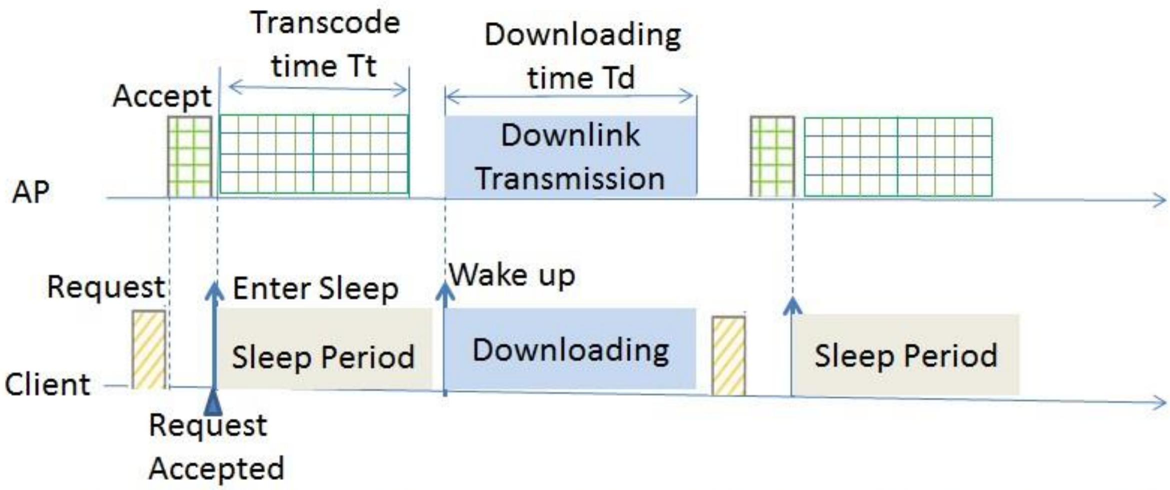

Fig. 3. Scheduling of Transcode and Downlink in the single-user scenario.

*Algorithm:*Node has to sleep for the transcoding period *Tt*when the request is accepted by AP. In PSM, during the sleep period, theAP should transcode M number of packet and put it in a downlink buffer. As shown in Fig. 3, suppose thatthe node hasreceived *M* downlink buffered packets continuouslywhen it is in active mode. Since sleep will introduceadditional delay to the packets, we first need to figure out howmuch extra delay one packet can tolerate without missing itsdeadline. Upon reception at the node, each packet has delaytolerance, denoted as *Dn*, which is the tolerable amount of timebefore being played out excluding the constant time *T*dec spentin packet decoding at the node. Let *Dn*,min be the minimum delay tolerance of those*M* packets. If the value of *Dn*,min is large enough, the nodecan consider switching to sleep mode. Specifically, if *Dn*,min>*Ts*,min, the node will decide to sleep. *Ts*,min, the minimum sleep period, is used to prevent each node from frequently switching between active and sleep modes, which can cause significant amounts of energyconsumption to the node.In our algorithm, *Ts*,min is set as 500 ms, i.e.,a moderate value that each node has a good chance to sleepupon receiving a fresh packet while preventing frequent modeswitching.

Once sleep switching is decided, the node determines a sleep period *Ts*and sends the sleep ending time (i.e., the next wake-up time) via a *Request* message to the AP. The latter then checks the future channel condition and confirms the request by replying with aAcceptsignal. However, the AP can decline the request by replying nothing. The node switches to sleep immediately after receiving accept and wakes up at the determined time. Otherwise, if no accept is received after a period of *T*wait, it remains in active mode and generates a new request later. All the downlink packets that transcoded during the sleep period will be buffered atthe AP and will be transmitted to the node after the wake-up time. In addition, the sleep time is chosen to guarantee that packets arrive before playout deadlines. In this way, idle listening time is greatly reduced and energy efficiency can be improved without sacrificing network performance.

B. Multiuser Scenario

In the multiuser scenario, the challenge is that when a noderequest for its desired transcoding type,the APmay not be accept or the transcoder may be busy due to otheruser's transcoding. Additional delay will be introduced whenwaiting for allocating the resource, which may result in violationof deadlines and packet drop. In addition, if the resource isoccupied by another node, which is also running this algorithm,the delay will be much longer since the transcoded packets are bufferedcontinuously. To solve this issue, transcoder in AP will create the new thread and buffer pool for each node to transcode.Transcoding manager maintain each thread's information and shake hands with each nodes to calculate the transcoding rate and buffering rate. Transcode threads get the free buffer from buffer pool then filled with transcoded packet and kept it for downloading.On downloading at any time there is at most one node usingthe channel to retrieve buffered packets so that the consequentdata downloading will not collide with other transmissions,In our algorithm rate adaption control manager shake hands with AP to adjust the speed of downlink packet to avoid the packet drop. Since our downlink packets are decoder specific we can achieve the energy efficiencyon node.

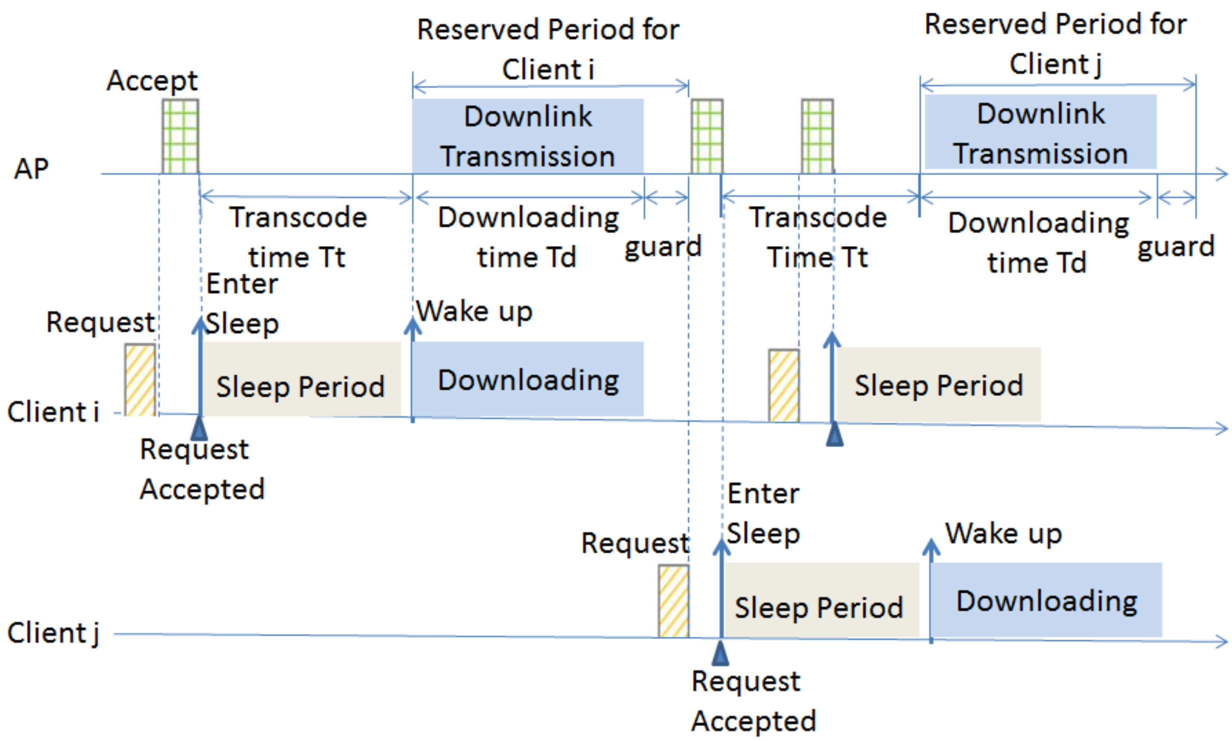

Fig. 4. Scheduling of Transcode and Downlink in the multi-user scenario.

Our algorithm in the multiuser scenario is illustrated inFig. 4. For ease of exposition, we consider two nodes, namely,node *i*and *j*. A node decides whether and how long tosleep based on the same strategy previously described in thesingle-user scenario. Once it has decided to sleep, the node willsend the request message encapsulating the wake-up time tothe AP using the contention-based IEEE 802.11 CSMA/CA protocol to avoid collisions with other nodes' transmissions.Upon receivingthe request, the AP estimates the downloadingperiod $\hat{T}_d$ and attempts to reserve a period of$\hat{T}_d + T_{d,\mathrm{guard}}$ for the requesting node at its proposed wake-uptime. $T_{d,\mathrm{guard}}$represents a *reservation guard time* immediatelyafter $\hat{T}_d$. Since there could be bursts of transcoded packets bufferedat the AP when the node is sleeping or busy downloadingbuffered packets, which cannot be predicted by the AP inestimating $\hat{T}_d$, the guard time can make room to transmit thosepackets.

Under a constant or small-fluctuation arrival rate orwhen the downloading rate is high enough to accommodate anunpredicted burst, $T_{d,\mathrm{guard}}$can be set to a relatively small value.Otherwise, it should be increased accordingly. In our simulations,since the arrival rate is constant and the downloadingrate is relatively high, a short $T_{d,\mathrm{guard}}$, i.e., 5 ms, is selected.If the calculated reservation period does not overlap with anyother served periods, the AP will mark the reservation for thisnode and confirm the request by replying the node with a acceptmessage. Otherwise, the request is denied and AP will not replyto the node. In this case, the node will retry sending the requestafter waiting for a period of time. During the reserved period,the AP transmits buffered downlink packets to the designatednode without collisions and buffers the downlink packets of theother nodes. Note that the reservation guard time is only usedby the AP in the decision process and the actual downloadingperiod may be shorter than the reservation. After the nodehas retrieved all buffered packets from AP in the downloadingperiod, both AP and the node will return to normal activemode immediately (terminating the reservation) such that anydownlink packet can be transmitted, and new request/acceptmessages can be also exchanged. As previously mentioned,after the downloadingperiod, the node must stay in active modefor at least $T_{\text{awake,min}}$to receive non buffered packets.

In the case that the sleep request is lost and not received byAP, the node will not receive the accept message after T_{wait} andit can start a new request. Therefore, the packet loss of sleeprequest will not affect the normal working of this algorithm. Ifthe accept message is lost, the AP makes a reservation, whichis in fact invalid. The node will treat its request as rejectedand send a new request. Then, on the AP side, it will receivea request from a node that is thought to be sleeping. Therefore,the AP can be aware of the loss of accept message then cancel the invalid reservation and recalculate the schedule according tothe new request. From the analysis, it can be seen that the proposedalgorithm is robust to the loss of request/accept packets.

In the proposed algorithm, although the AP participates inthe coordination, the sleep time of each node is determined bythe node in a distributed manner. Regarding fairness, it canbe ensured in view of the following aspects. First, since thelength of reservation time is decided by the AP, it can declinea request if the corresponding $T_d$is too large, i.e., to preventsome node occupying channel for too long, thus ensuring theaccess probabilities of all the nodes. Second, each time a nodesuccessfully makes a reservation, it will then go to sleep mode,and during this time, the other nodes can apply for sleep andreservation. Meanwhile, during the consequent awake time of anode after its reservation period, other nodes can shake handswith the AP for sleeping and reserving a designated period inthe future. In this way, the AP can be prevented from allocatingreservation periods for some node consecutively.In the multiuser scenario, the algorithm can alsomitigate the channel contention since most of the packets aretransmitted in reserved periods.

CONCLUSION

We present a new application of video streaming system for mobile devices over a wireless network. Our system overcomes the display size, bandwidth and computational power constraints to deliver high-resolution video for interactive display on mobile devices. When high bandwidth is available, one may argue that the system should stream the high-resolution content; however, playing such video is not feasible on a mobile device with limited computation power and display size without using video streaming. we present an energy-efficient sleep scheduling technique for video streaming applications over WLAN. Each node calculates its sleep period under packet delay constraint and requests that the AP reserve a future period for exclusive transmission. The AP examines thescheduling of all the nodes and makes sure that there are no overlapping reservation periods. In this way, the active periods of different nodes are staggered to mitigate contention and prevent further packet losses. Some of future work that we would like to explore include tile prefetching selection byjointly optimizing playback quality and latency, prediction look-ahead estimation by considering both network bandwidth and prediction accuracy, and improved crowd-driven prediction by clusteringviewing patterns according to users' profiles and interests.

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(IJIEST) 2015, Vol. No. 1, Jan-Dec **e-ISSN: 2454-9584; p-ISSN: 2454-8111**

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(IJIEST) 2015, Vol. No. 1, Jan-Dec **e-ISSN: 2454-9584; p-ISSN: 2454-8111**

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