Evaluating Video Streaming over UWB Wireless Networks

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ABSTRACT

Video streaming is going to drive the Internet to a new height. However, how to distribute the high-quality video streams that are already delivered to the doorsteps of ordinary customers to almost all rooms in their dwellings still remains a technique challenge. There are various proposals for wired, “no-new-wires” or wireless approaches, but their performance in real environment is yet to be examined. In this paper, we evaluate the performance of high-quality video streaming over short-range wireless networks, specifically Ultra-Wide Band (UWB), for a two-tiered broadband home network. Our analysis and experiment results reveal the unique features of UWB reservation schemes and their intrinsic throughput-latency tradeoffs, which indicates the efficacy of UWB for high-speed, in-room wireless access with both quality and mobility provisioning.

Categories and Subject Descriptors
C.2.1 [COMMUNICATION NETWORKS]: Network Architecture and Design—Wireless communication

General Terms
Experimentation, Measurement, Performance

Keywords
Video streaming, Ultra-Wide Band (UWB)

1. INTRODUCTION

With the advent of whole-house entertainment applications such as Internet Protocol Television (IPTV) and Digital Video Recorder (DVR), the need for video distribution among almost all rooms in a household environment is obvious. Nowadays, service providers have the capability of delivering tens to hundreds of megabits per second (Mbps) to the doorsteps of IPTV subscribers, but how to distribute the video, voice and data traffic within the premise of ordinary customers becomes a challenge, mainly due to the lack of broadband home networks economically available with Quality of Service (QoS) provisioning [1].

Ethernet is often suggested by service providers, but for the vast majority of existing houses, Category 5 (Cat 5) or better Ethernet cables with Structured Wiring is not available. Retrofit or rewiring turns out to be very expensive, and running cables along corners or outside houses is also very awkward. Both service providers and customers are looking for alternatives. Several industry groups prompt the so-called “no-new-wires” technologies to transport Ethernet frames over existing household cable, phone and power wires, but their availability and achievable performance are still quite uncertain, and these wires may not be conveniently connected to video devices in all rooms [2].

Whenever possible, consumers still prefer wireless solutions, evidenced by the proliferation of IEEE 802.11 Wireless Local Area networks (WLAN), due to their availability, affordability and flexibility [3]. Even with the data rate increase in IEEE 802.11g/n and the QoS improvement in IEEE 802.11e, delivering high-quality video over WLAN with QoS guarantee still remains an open problem, especially in a household environment full of obstacles and interferers. Existing research reveals that the conventional infrastructure-based, single-hop wireless Access Point (AP) structure may not be sufficient to cover the entire house with satisfactory performance around all corners [4].

In this paper, we follow a two-tiered approach to broadband home networks [5], with a particular focus on in-room access facilitated by the Ultra-Wide Band (UWB) technologies [6]. With a cross-room home network backbone, the in-room access networks can better utilize the high-speed, short-range wireless technologies for mobility and spatial reuse. Using a large portion of radio spectrum, UWB can achieve higher data rate (up to 480 Mbps in the current, off-the-shelf products) in a short range (around 10 m) with very low power emission (-41.3 dBm/MHz). In addition, the media access control (MAC) in UWB is designed to transport both asynchronous and isochronous traffic suitable for data and voice/video applications, respectively.

In order to evaluate the feasibility and performance of high-quality video streaming over UWB networks, we have constructed a small UWB testbed using commercially available products. By carefully choosing the experiment scenarios to represent a typical household environment, we can identify the intrinsic tradeoffs in UWB physical and MAC layers with regard to transmission rate (TxRate), retry limit, reservation percentage and pattern. For a given channel condition, a suitable TxRate and retry limit have to be...
chosen for high throughput and reliability. To reduce the turnaround overhead for higher throughput, clustered reservation is preferred; on the other hand, to reduce the service interval for lower latency, a scattered reservation is better. Thus, it is necessary to take TxRate, retry limit and reservation jointly into account to meet the stringent requirement on bandwidth, reliability and latency for high-quality video streaming. To the best of our knowledge, this is the first work reported in open literature on the performance of video streaming over UWB networks with an experiment-based, application-oriented approach.

The reminder of this paper is structured as follows. We briefly examine the technologies available for high-quality video streaming and review the related work on UWB in Section 2. We outline the testbed configuration and evaluation methodology in Section 3, with performance analysis in Section 4 and followed by throughput, latency and video performance results in Section 5. We further discuss the intrinsic tradeoffs in physical and MAC layers for high-quality video streaming in Section 6. Section 7 concludes the paper with the direction of future work.

2. BACKGROUND AND RELATED WORK

2.1 IEEE 802.11 WLAN

Currently, IEEE 802.11 [3] is the dominant technology for WLAN due to its low cost, easy deployment and high flexibility. Most portable devices now have WLAN interfaces embedded to support one or more modes of IEEE 802.11a/b/g. In theory, IEEE 802.11b can support raw data rate up to 11 Mbps and 802.11a/g up to 54 Mbps, which appears to suffice for high-quality video streaming. However, due to the high overhead in IEEE 802.11 physical and MAC layers, less than 50% of the raw data rate is available to the application layer. Newer technologies, such as IEEE 802.11n, are emerging, but they are still at a very early stage and not widely available yet, so here we just discuss IEEE 802.11a/b/g.

In a household environment, cross-room wireless signals are attenuated and reflected by floors, walls, doors and moving objects such as human beings and pets, which reduces the received signal strength. Also, IEEE 802.11 devices are working in the same 2.4 and 5 GHz unlicensed frequency bands as other home appliances such as cordless phones and microwave ovens, which introduces interference and further reduces the received signal-to-noise ratio (SNR). Given the limited number of channels available, it is not unusual to see IEEE 802.11 devices working in a low SNR condition with limited capacity. [4] points out that the average throughput of IEEE 802.11g devices in a household environment is about 10 Mbps due to high attenuation, interference and shadowing, which makes it less ideal for high-quality video streaming, especially with multiple video streams.

IEEE 802.11 MAC, even in 802.11n, is mainly based on Carrier Sense Multiple Access with Collision Avoidance, and each device has equal probability to access the channel. Due to channel contention, devices are constrained by the one with the lowest date rate, and some devices have to wait a relatively long time to access the channel. Obviously, contention-based MAC is not suitable for real-time applications such as IPTV that have stringent delay and jitter requirement. Contention-based MAC also reduces achievable throughput due to channel idle and collision times. IEEE 802.11e Enhanced Distributed Channel Access (EDCA) [3] is proposed to prioritize channel access and targets at multimedia applications. However, EDCA is a statistical priority scheme and cannot guarantee the performance for high priority traffic and may starve the low priority one.

2.2 Ultra-Wide Band Technologies

UWB is a radio technology that can be loosely defined as any wireless transmission schemes with bandwidth more than 25% of its center frequency, or more than 500 MHz [6]. There are two camps of UWB, DS-UWB and MBOA-UWB. DS-UWB, referred to as Direct Sequence UWB, is based on Direct Sequence Spread Spectrum (DSSS) technology. MBOA-UWB, which eventually became WiMedia UWB, is based on the combination of Time-Frequency Coding (TFC) and Orthogonal Frequency-Division Multiplexing (OFDM) technology. Here we focus on WiMedia UWB as it now becomes commercially available off the shelf.

UWB has a few unique features to make it a better candidate for high-quality video streaming: ultra wide band, high data rate, and low power emission. WiMedia UWB uses a 528 MHz band with TFC hopping in the 3.1 to 10.6 GHz frequency range, which enables many more channels than IEEE 802.11 to accommodate more device groups. With such a wide band, WiMedia UWB can support raw data rate up to 480 Mbps and potentially to 1,000 Mbps, even with lower SNR. By transmitting at a very low power level, UWB has very little interference to other devices, and is less susceptible to interference from other devices as well. Low power consumption also means better energy conservation for battery-powered portable consumer electronics.

In addition to the physical layer features mentioned above, UWB MAC has its own features to further benefit high-quality video streaming. WiMedia [7] has two types of MAC schemes: Distributed Reservation Protocol (DRP) and Prioritized Contention Access (PCA). Time duration in WiMedia MAC is equally divided into superframes of 65 ms each, and each superframe has 256 Medium Access Slots (MAS). The first 32 slots can be used for Beacon Period (BP), during which each UWB device can broadcast and inform others about the slots in the following Data Period (DP) it reserves for exclusive access. Consequently, in the reserved slots, the "owner" device transmits at the beginning of the slot without contending for the channel, and others have to wait for their reserved slots. DRP provides guaranteed access and performance. For unreserved DP slots within the same superframe, each device contends for the channel with PCA. PCA is similar to EDCA with prioritized channel access.

2.3 Related Work

There are a few efforts reported in the literature on UWB prototyping and experimentation, but they mainly focus on the physical layer with proprietary software for demo purpose. For example, [8] presented a wireless display system to transport raw, uncompressed analog video signals through UWB from a smartphone to a data projector. [9] implemented a UWB-based wireless communication system using multi-FPGA hardware and discrete RF design to achieve a maximum data rate of 110 Mbps. Our work experiments with an off-the-shelf UWB system and investigates the performance of high-quality video streaming in a more realistic, home-like environment for IPTV services.

On the other hand, there are several published papers on WiMedia MAC [10–14] with analysis and simulation ap-
3. EVALUATION METHODOLOGY

In this section, we first give a brief outline of our testbed configuration, and then illustrate the performance metrics of our interest and the approaches to obtaining them.

3.1 Testbed Configuration

As shown in Fig. 1, our testbed has two UWB nodes referred to as uw1 and uw2 with Tzero ZeroWire Mini PCI 700 Revision B card and dual antenna [15]. Tzero firmware (tz7110), host driver (Version 3.3.10), configuration (Version 1) and control software are used on these two nodes. Tzero configuration file allows us to manually select TxRate, retry limit and receiver diversity (RxDiversity), and set reservation percentage and pattern. By experimenting with the reservation map, we have confirmed that we can reserve with arbitrary percentage and pattern. Also, each node has a Gigabit Ethernet link for control purpose, which allows us to remotely access them without affecting with ongoing tests.

These two UWB nodes are about 10 meters away in a line-of-sight setting. In order to emulate a household environment full of obstacles and interference, we used two tea cans to cover uw2’s antenna for a resultant Received Signal Strength Indicator (RSSI) around -73 dBm, which appears to be background noise for non-UWB devices. We select channel 14 (TFC 6), which is a fixed frequency interleaving at 3.96 GHz, TxRate among 53.3, 80, 106.7, 160, 200 and 480 Mbps, retry limit from 0 to 7, and automatic RxDiversity. Unless otherwise stated, we use the so-called latency schedule with close to 50% slots reserved for uw1 and uw2, respectively, and we set reservation patterns with different scatter levels through the configuration file.

However, there are some limitations in our experimentation. First, not all WiMedia data rates, even the mandatory one at 300 Mbps, are supported by the Tzero cards in our testbed. Since the supported data rates use both QPSK and DCM modulation, we believe our testbed is representative for other missing data rates with the same modulation scheme. Second, our cards have a limited support on DRP and no support on PCA: if a slot is not reserved by either uw1 or uw2, the slot is not attempted with PCA, even when both uw1 and uw2 have data to send; if a slot is reserved by both uw1 and uw2, they will attempt to send packets in the same slot, which results in collision. Since we focus on DRP-supported video streaming, we can arrange the reservation map on uw1 and uw2 properly to avoid under or over utilization. Third, in addition to the first 32 slots reserved for BP, the last 16 slots, if reserved, will break the connection between uw1 and uw2. We believe this is an implementation issue of Tzero cards, so we can reserve at most 208 slots for uw1 and uw2, or 104 slots each with the 50% reservation.

3.2 Network Characterization

In order to evaluate the DRP capacity at a given TxRate, retry limit, reservation percentage and pattern, we need to first find out how many packets can be sent out and received in one superframe by uw1 and uw2, respectively. We use throughput to denote the capacity achieved by the sender and limited by the reservation, and we use goodput to denote the capacity achieved by the receiver, taking into account TxRate and retry limit. By increasing the offered load, we can obtain the saturated throughput, which gives an upper bound of the performance achievable for video streaming.

We used D-ITG for network performance analysis [16]. D-ITG has a sender-receiver logger structure to provide both sender and receiver traces at packet level. Given the high data rate of UWB, it requires very high precision of time synchronization if we want to obtain one-way delay. Instead, we instruct the upper-layer acknowledgment of data packets from uw1 to uw2 to return back to uw1 through the Gigabit Ethernet control link and obtain the round-trip time. In fact, we have modified D-ITG to take additional arguments to transport all signaling and upper-layer acknowledgment packets through the control link, so these packets will not affect the data packets sent through the UWB link.

3.3 Video Evaluation

In our experiments, we used a two-minute high-definition video camera demo video clip as an example. The video has a resolution of 1920*1080 and refresh rate of 24 frames per second. We applied the MPEG-4 AVC reference encoder on the raw video. MPEG-4 AVC, also known as H.264, is the newest video coding standard and has been widely adopted for high-definition TV (HDTV) services. Figure 2 shows the frame size of our encoded sample video. From the figure, we can tell the average frame size is about 21.152 kilobytes, or the average data rate around 4.186 Mbps. The figure
also shows the high peak-to-average ratio due to the high efficiency of MPEG-4 AVC. We used multiple video streams to fully utilize the UWB link, e.g., four streams to represent quad-HDTV scenarios for cinema-like experience.

Video frames are further segmented in packets. MPEG-4 AVC employs a Group-of-Pictures (GoP) structure, and some frames (e.g., P or B-frames) are predicted based on others (I or P-frames). In this case, traditional network performance metrics such as packet loss and delay are not sufficient, since losing an I-frame will affect all frames in a GoP. To obtain application-level performance metrics, we used EvalVid [17] to capture the packet trace at both the video streaming server (uw1) and client (uw2). By comparing the sequence number and timestamp of each frame at both sides, we can calculate frame loss, delay and jitter. In addition, we can reconstruct the video stream with the received packets, and calculate Peak-Signal-to-Noise-Ratio (PSNR), which is regarded as an objective metric for video quality evaluation and also a good indicator for subjective ones for perceptual video quality evaluation.

4. PERFORMANCE ANALYSIS

In this section, we analyze the performance achievable by UWB at a given TxRate, by taking into account the protocol overhead in physical, MAC, logical link control (LLC) and upper layers. The analysis will be validated by the performance results given in the next section.

Video streams are often transported in Realtime Transport Protocol (RTP), which is consequently encapsulated in UDP, IP and LLC protocol with 8, 20 and 16-byte header, respectively; or for an overall overhead of 44 bytes above the MAC layer. In WiMedia UWB, an OFDM symbol lasts $T_{SYM} = 0.3125 \times \mu s$ and can carry a different number of information bits depending on modulation and coding schemes, which jointly determine the physical layer data rate. In the Physical Layer Convergence Protocol (PLCP), a standard or burst PLCP preamble of $N_{sync} = 30$ or 18 symbols is prefixed for packet synchronization and channel estimation.

In a DRP reservation with no acknowledgment, PLCP packets are separated by a Minimum Inter-Frame Space (MIFS) of $pMIFS = 1.875 \mu s$ if the burst mode is used or otherwise a Short Inter-Frame Space (SIFS) of $pSIFS = 10 \mu s$. The last PLCP packet in a DRP reservation should have a minimum guard time of $mGuardTime = 12 \mu s$ before the end of the reserved slot, in addition to a SIFS regardless whether the burst mode is used. Therefore, for a DRP reservation covering $n$ consecutive MAS slots of $mMasLength = 256 \mu s$ each, the maximum number of PLCP packets can go through in the reservation is

$$N(n) = \left\lfloor \frac{mMasLength \times n - mGuardTime}{T + pSIFS} \right\rfloor$$

or

$$N(n) = \left\lfloor \frac{mMasLength \times n - mGuardTime - \Delta}{T + pMIFS} \right\rfloor$$

with the burst mode, where $\Delta = pSIFS - pMIFS$.

5. PERFORMANCE RESULTS

In this section, we first present the network and video performance affected by TxRate and retry limit, and then we look further into the throughput-latency tradeoff affected by reservation percentage and pattern.

5.1 TxRate and Retry Limit

5.1.1 Packet Loss

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$$T = \left\lfloor N_{sync} + N_{hdr} + 6 \times \left\lceil \frac{L + 44 + 4}{N_{IBP}} \right\rceil \times T_{SYM} \right\rfloor$$

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or

$$N(n) = \left\lfloor \frac{mMasLength \times n - mGuardTime - \Delta}{T + pMIFS} \right\rfloor$$

with the burst mode, where $\Delta = pSIFS - pMIFS$.
Link-layer retransmission is an often-used technique to combat channel error. In our testbed, we can set a retry limit to determine how many local retransmissions are allowed before dropping a packet. In Fig. 3, we show the Packet Loss Ratio (PLR) with different TxRate and retry limit, when the reservation is at 50% with map \{FF00\}*, which means only the first eight slots in every 16 slots are reserved for \textit{uw1}. PLR increases with the increased TxRate at a given SNR. When the TxRate is above 200 Mbps, the RSSI (-73 dBm in our testbed, which is very close to the Minimum Receiver Sensitivity at 200 Mbps) cannot sustain any packet transmission, and PLR is 100%. On the other hand, PLR decreases with the increased retry limit due to local retransmission. When the TxRate is below 160 Mbps, the PLR is almost 0 with a non-zero retry limit. However, to facilitate local retransmission, the transmitter has to wait for the link-layer acknowledgment back from the receiver over the air and subject to channel errors, and retransmits when a timeout event occurs. As we shall see next, this waiting period also affects achievable performance.

5.1.2 Receiver’s Goodput

In Fig. 4, we show the achieved goodput for different TxRate and retry limit. When the TxRate is increased, packet transmission time is reduced, but the PLR is increased as shown in Fig. 3. Therefore, when TxRate is slightly increased, we see a considerable increase in achievable goodput. However, when TxRate is above a threshold (106.7 Mbps in our testbed), the increase in PLR is more significant, which greatly reduces the achievable goodput. When the TxRate is above 200 Mbps, the PLR is 100%, and the achieved goodput is 0. Figure 4 also shows that the increased retry limit actually reduces achievable goodput, which seems counter-intuitive. This is due to the link-layer acknowledgment required for local retransmission, since the transmitter has to wait for the acknowledgment to come. For high-speed links such as UWB, such a waiting will keep the channel idle for a while in the reserved slots, which reduces channel utilization and eventually goodput. Therefore, block acknowledgment is necessary to improve both link utilization and reliability with retransmission. Unfortunately, block acknowledgment is not available through the existing configuration options for our Tzero cards.

<table>
<thead>
<tr>
<th>Table 1: Maximum goodput (retry limit=0)</th>
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<tr>
<td>TxRate (Mbps)</td>
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<tr>
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</tr>
<tr>
<td>53.3</td>
</tr>
<tr>
<td>80</td>
</tr>
<tr>
<td>106.7</td>
</tr>
<tr>
<td>160</td>
</tr>
<tr>
<td>200</td>
</tr>
<tr>
<td>480</td>
</tr>
</tbody>
</table>

The experimentation results in Fig. 4 have been validated by the goodput analysis results listed in Table 1, following (1)–(3). For a UDP packet with a 1024-byte payload, the total transmission time at 53.3 Mbps is 174.375 $\mu$s. For a DRP reservation without acknowledgment (i.e., retry limit = 0), the number of UDP packets can be transmitted within 8 consecutive MAS slots is

$$\left\lfloor \frac{256 \times 8 - 20.125}{174.375 + 1.875} \right\rfloor = 11,$$

or 143 packets for the entire superframe with \{FF00\}* reservation. Thus, the calculated goodput is 17.875 Mbps, while the measured one is 17.78 Mbps, as shown in Table 1. In this table, the increased difference between the calculated and measured goodput when the TxRate increases is due to the increased PLR shown in Fig. 3.

5.1.3 Video Quality

After obtaining the saturated goodput, we deliver video streams from \textit{uw1} to \textit{uw2}, and calculate frame loss, delay and jitter. With an increased TxRate, frame loss ratio (FLR) increases as well, due to the higher PLR at a given SNR. By reconstructing the received frames and comparing with the original ones, we can obtain the average PSNR. As shown in Fig. 5, the average PSNR decreases with the increased TxRate, due to a higher FLR. The higher the PSNR value, the better the reconstructed video quality, and a video stream of PSNR below 20 dB is considered not acceptable. As we can tell, local retransmission greatly improves link reliability and hence PSNR. In fact, with a retry limit of 7, the PSNR can achieve almost 50 dB when TxRate is below 200 Mbps between \textit{uw1} and \textit{uw2}, which is the upper bound for a
lossy compression scheme such as MPEG-4 AVC. Given the almost-noise-like RSSI at uw2, this demonstrates the strong capability of UWB supporting high-quality video streaming in a household environment.

5.2 Reservation Percentage and Pattern

DRP is a unique feature in WiMedia UWB MAC and designed to support isochronous voice/video traffic. By reserving a certain number of slots for exclusive access in a superframe, a node is guaranteed to have a certain portion of link airtime, or the equivalent service rate. On the other hand, the gaps between these reserved slots determine the service interval. The service interval and queuing delay, which is determined by the service rate and the peak-and-average data rate, will further determine the access latency. In order to support high-quality video streaming, it is desired to achieve both high throughput and low latency at the same time with high channel utilization.

5.2.1 Sender’s Throughput

In order to show the effect of reservation patterns, we reserve 50% available slots with different cluster levels. For example, we can reserve every other 2 slots where 1 \( \leq i \leq 6 \) for a total of 104 slots, except for the last few dozens when \( i \geq 5 \) as shown in Table 2. \( i \) is referred to as Reservation Pattern Index (RPI), and the higher the RPI is, the more clustered the reservation becomes. The RPI of our default reservation pattern \( \{CCCC\}^* \) is 3. Figure 6 shows the achievable throughput with different retry limit and reservation pattern at 53.3 Mbps. As we can tell, when the reservations become clustered, due to the reduced turnaround overhead and guard time, the achievable throughput is increased. Again, due to the time for acknowledgment, the achievable throughput is reduced with a higher retry limit, corresponding to the reduced goodput in Fig. 4.

The experimentation results in Fig. 6 have been also validated by the throughput analysis results in Table 3 as well, by following (1)–(3). At 53.3 Mbps and for RPI=1, every two consecutive MAS slots can accommodate

\[
\left\lfloor \frac{256 \cdot 2 - 20.125}{174.375 + 1.875} \right\rfloor = 2
\]

UDP packets with a 1024-byte payload, i.e., 104 packets in a superframe with \( \{CCCC\}^* \) reservation. Thus, the calculated throughput is 13 Mbps, while the measured one is 12.85 Mbps. Other reservation patterns show similar trends, but the measured throughput for RPI=2 is considerably lower than the calculated one, which needs further investigation.

5.2.2 Packet Delay

In Fig. 7, we show the packet delay affected by service interval for RPI 1 and 5, i.e., every other 2 and 32 slots are reserved, respectively. Note that the transmission and receiving time (Tx and Rx, respectively) curves are shifted horizontally to align with the start of a superframe at packet sequence number 104. The vertical time difference between the Rx and Tx curves indicates the packet delay in round-trip time, which can approximate the one-way access latency due to the low, stable delay on Gigabit Ethernet. From the figure, we can tell packets are served regularly with small service interval when RPI=1. Due to the offered load (25 Mbps) is higher than this pattern can sustain (around 13 Mbps), we see an increased packet delay due to the increased queuing delay. When RPI=5, there are more packets served in the same time interval, indicating higher throughput; however, it suffers more obvious service interval that is longer than the case with RPI=1.

5.2.3 Video Jitter

To further identify the effect of different reservation patterns on service rate and interval and consequently on access latency, in Fig. 8, we show the maximum accumulated jitter (MAJ) for the sample video. Frame jitter is defined as the time difference to deliver two consecutive frames, and is

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Table 2: Reservation Patterns

<table>
<thead>
<tr>
<th>Index</th>
<th>Reservation Pattern for the Entire Superframe</th>
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<tbody>
<tr>
<td>1</td>
<td>0000 0000 CCCC CCCC CCCC CCCC CCCC CCCC CCCC CCCC CCCC CCCC 0000</td>
</tr>
<tr>
<td>2</td>
<td>0000 0000 F0F0 F0F0 F0F0 F0F0 F0F0 F0F0 F0F0 F0F0 F0F0 F0F0 0000</td>
</tr>
<tr>
<td>3</td>
<td>0000 0000 FF00 FF00 FF00 FF00 FF00 FF00 FF00 FF00 FF00 FF00 FF00 0000</td>
</tr>
<tr>
<td>4</td>
<td>0000 0000 FFFF 0000 FFFF 0000 FFFF 0000 FFFF 0000 FFFF 0000 FFFF 0000</td>
</tr>
<tr>
<td>5</td>
<td>0000 0000 FFFF FFFF 0000 0000 FFFF FFFF 0000 0000 FFFF FFFF 0000 0000</td>
</tr>
<tr>
<td>6</td>
<td>0000 0000 FFFF FFFF FFFF 0000 0000 FFFF FFFF 0000 0000 FFFF FFFF 0000</td>
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<table>
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<tr>
<th>Table 3: Maximum throughput (TxRate=53 Mbps)</th>
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<tr>
<td>Index</td>
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</tr>
<tr>
<td>1</td>
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caused by the variation in service rate and interval. Again, the reservation is at 50%. As we can tell, when the reservations become clustered, due to the increased service interval, the MAJ is increased. The MAJ will determine the minimum initial buffering required to smooth the video playback. With an increased MAJ and given the high data rate supported by UWB, a much larger buffer is required to absorb the jitter and smooth the video, with a longer initial buffering time, which degrades video quality.

6. FURTHER DISCUSSION

So far, we have exhibited the efficacy of UWB for high-quality video streaming in a household environment. Due to the ultra-wide band, high data rate and low power emission, existing UWB devices can transport quad-HDTV or multiple HDTV video streams with satisfactory PSNR performance, even in our testbed where the SNR is very low. This substantiates our approach of using high-speed, short-range wireless technologies such as UWB and Millimeter Wave (mmWave) [18] as in-room access for two-tiered broadband home networks [5].

In addition, we have validated the performance tradeoff affected by TxRate and retry limit. At a given SNR, the choice of TxRate and retry limit has to be balanced properly with the achieved throughput and reliability. For high-quality video streaming, it is important to achieve both high throughput and high reliability, due to the high data rate video traffic for HDTV and the high-efficiency video coding schemes that make it much more sensitive to packet loss. A higher TxRate implies higher packet loss ratio at the same received SNR, and a higher retry limit can reduce packet loss but may reduce goodput.

Distributed reservation is a unique feature in WiMedia UWB. For a single reservation request, there are many possible reservation patterns. As shown in Section 5, with a clustered reservation, there are fewer turnarounds in a superframe, which leads to less guard time and higher channel utilization. On the other hand, for the same number of reserved slots, a clustered reservation will increase service interval, which potentially increases access latency. Therefore, in order to increase throughput and reduce latency, we have to strike a balance between scattered and clustered reservations, particularly for video traffic.

Although reservation can guarantee service rate and interval, when there are multiple reservation requests, how to arrange these reservations properly to meet their bandwidth and latency requirement individually is a nontrivial problem. It is possible there are enough slots available for a new request in terms of bandwidth requirement, but the location of these slots may lead to a large service interval and eventually violate the latency requirement. This situation can be alleviated by rearranging some existing reservations within their bandwidth and latency requirement. However, for WiMedia UWB, all these reservations have to follow the MAS allocation guideline specified in the WiMedia Wireless Link Protocol (WLP) due to certain physical layer constraints and fairness concerns [19].

7. CONCLUSIONS

We followed an experiment-based, application-oriented approach in this paper to the performance evaluation of high-quality video streaming over short-range wireless networks,
specifically UWB. By leveraging the physical and MAC layer features, we have demonstrated the efficacy of using UWB as the high-speed, short-range in-room access for two-tiered broadband home networks to support IPTV and DVR services. We also identified the intrinsic tradeoffs in UWB physical and MAC layers and their effect on network and application performance.

Ongoing and future work focuses on the performance of mixed traffic over UWB networks, as well as the MAS allocation schemes to accommodate multiple requests with heterogeneous bandwidth and latency requirement.

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8. REFERENCES