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Cross-Layer WLAN Measurement and Link Analysis for Low Latency Error Resilient Wireless Video Transmission

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Abstract – This paper introduces a cross-layer measurement and link analysis strategy for video transport over IEEE 802.11. Field trial measurement data is presented for streamed H.264 video over ad-hoc 802.11g links. The data is used to analyze the interactions between the physical/network/transport and application layers. The development of cross layer optimized low latency error resilient video transmission schemes is discussed.

I. INTRODUCTION

The use of WLAN technologies [1-3] for data communication is well established in the consumer electronics industry; however video streaming over Wi-Fi has yet to reach the same levels of maturity as its data counterpart. Unlike internet-based video streaming, unicast video streaming over Wireless LAN (WLAN) can take advantage of MAC level (and possibly transport level) retransmission to improve the QoS observed at the higher layers [4]. This is feasible due to the close vicinity of the client/server, which provides a predictable route for the IP packets. Through the use of suitable buffers, issues such as packet loss (for UDP/IP-based streaming) and delay variance/jitter (TCP/IP-based streaming) can be mitigated. However, if the performance of the physical layer is poor, excessive Packet Error Rate (PER) or delay may be observed at the application layer. Today, suitable higher layer performance is commonly achieved at the expense of server responsiveness in multimedia applications (eg. navigation control in Set Top Box (STB) and DVD applications). When interactivity and timeliness are crucial, low latency communications are necessary without the reliance on excessive packet retransmission. In such situations, robust wireless communication and error resilient video codecs [5] become vital. To better understand interactions between the physical/network/transport and application layers, a cross-layer measurement method is introduced and a number of measurements presented using 802.11g cards for an outdoor measurement route.

II. MEASUREMENT SYSTEM

The measurement system is a typical client-server based setting as shown in Fig. 1. Both platforms are implemented with laptops running Windows XP. Ad-hoc 802.11b/g channels are established between the client and server using Broadcom-based adaptors. TCP/IP and UDP/IP

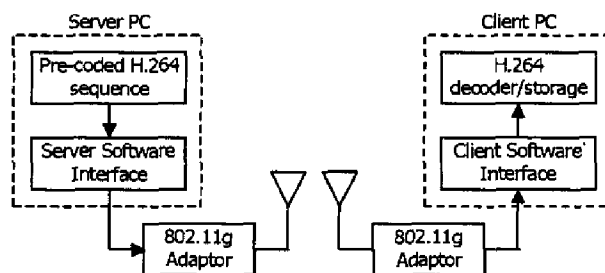


Fig 1. Overall measurement system.

communication software applications are developed via socket programming, while low-level card control is achieved through NDIS function calls.

Pre-coded H.264 sequences are used for the TCP/UDP payload. The payload packets are encapsulated using a proprietary set of headers to facilitate cross-platform measurement. The information in the header includes:

- PSIZE – packet size for TCP.
- PCNT – a running packet number.
- POFFSET – offset of packet since session starts.
- LNKSPD – mode-dependent link speed Mbps.
- SENT_TIME – time when packet is sent from server.

At the client the following measurements are made upon reception of each packet:

- RSSI – received signal strength indicator.
- RCV_TIME – time when client receives the packet.

PSIZE is used to demarcate the stream-oriented TCP packets for packet-based application data, such as H.264 streams; UDP packet sizes are intrinsic information and hence this parameter can be omitted from the header. PCNT is a running number attached to each packet sent from the server; by detecting missing packet numbers, the client is able to detect and count missing packets at the application layer and hence deduce the packet error rate (PER). RSSI is used to determine the received channel strength while LNKSPD determines the modulation/coding mode chosen at the MAC layer. Along with PER, the combination of RSSI and LNKSPD give a strong insight into the operation of the underlying link adaptation scheme. The use of POFFSET enables the client to reconstruct the pattern of lost packets. Using packet loss patterns, offline simulations can be performed on different sequences to determine the performance of error concealment and decoding schemes.

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Timing information from SENT_TIME and RCV_TIME allows packet delay to be calculated. Another piece of vital information that can be obtained from the timing data is the buffer occupancy at the client, from which a more effective buffering strategy can be derived.

To determine the Quality of Service (QoS) at the video application layer, each received packet is logged. For each packet, we record whether it was successfully delivered and if so, the delay associated with its transmission. Application level parameters (e.g. compressed video packet sizes, decoder video concealment strategies etc) are considered together with network parameters (e.g. packetisation strategies and jitter buffer size). Measurements are taken across all communication layers, including signal strength, 802.11 operating mode, delay jitter, packet error rate and the peak signal-to-noise-ratio of the reconstructed video frames and refresh rates were analysed.

III. SAMPLE RESULTS

Sample plots of the measurement results are shown in Fig. 2 (UDP/IP) and Fig. 3 (TCP/IP). The target bit rate is set to 2 Mbps. TCP is implemented with a buffer of 125ms (250 kbits). The packet size is 1200 bytes.

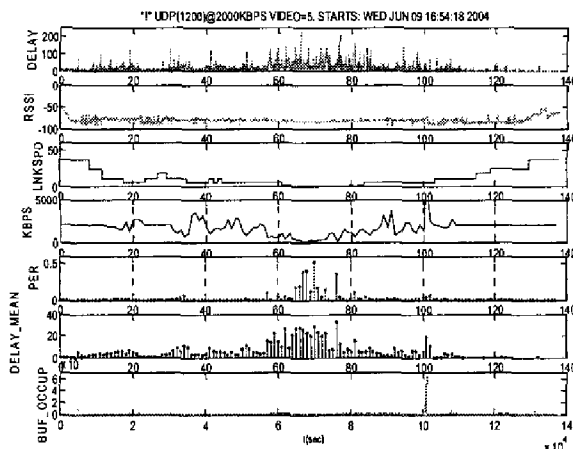


Fig 2. Transmission over a UDP link in an outdoor route environment.

For both cases, as the mobile moves further from the server, the RSSI decreases from -50dBm to -90dBm. The link speed can be seen to reduce as the mobile client moves away from the server, and then increases as the terminal returns to the server. This particular IEEE 802.11b/g card uses a total of nine link speeds: 1, 2, 5.5, 11, 18, 24, 36, 48 and 54 Mbps, with modes 6, 9 and 12 Mbps remaining unused.

The PER at the application layer is calculated by checking the received packet counter at the application layer. After the MAC layer performs a maximum number of retries, a packet is dropped. If UDP/IP is used, no retransmission occurs and the packet will be lost at the application layer. The resulting lost packets are registered as a packet error rate (PER, Fig. 2). High PER is seen at times when both the RSSI and the link-

speed are low.

Using SENT_TIME and RCV_TIME, buffer occupancy can be defined as the difference between the total number of bytes sent within a second and the total number of bytes received during the same second. With TCP, the jitter buffer at the receiver must be large enough to smooth excessive jitter. For low latency applications, such as real-time video transmission, UDP is faster and timelier than ARQ-based TCP transmissions. UDP is also a good choice if robust error resilience and error concealment techniques are applied at the video application layer in order to cope with the high PER, especially during bad channel conditions.

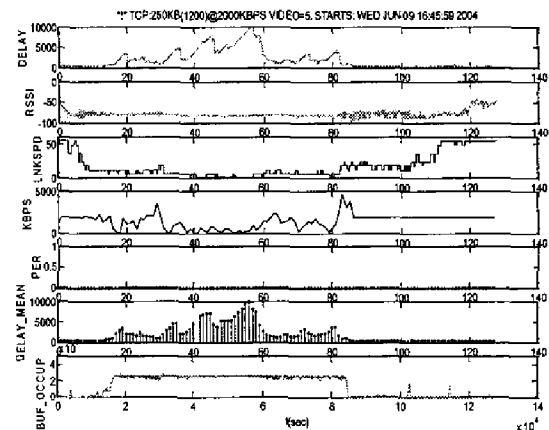


Fig 3. Transmission over a TCP link in an outdoor route environment.

IV. CONCLUSIONS AND FUTURE WORK

This paper summarized a cross-layer measurement system for characterizing error and delay parameters over an 802.11g channel. The paper explored the transmission of video packets through TCP/IP and UDP/IP transport/network layers. The use of a customized packet header allowed various link adaptation, packetisation and buffering techniques to be analyzed. Work is currently underway to develop an error-resilient H.264 decoder and a cross-layer link-adaptive H.264 encoder. This solution aims to optimize the PSNR of the reconstructed images under a wide range of PER conditions.

REFERENCES

- [1] A. Doufexi, S. Armour, P. Karlsson, M. Butler, A. Nix and D. Bull, "A Comparison of the Hiperlan/2 and IEEE 802.11a Wireless LAN Standards," IEEE Comm. Magazine, May 2002.
- [2] R. Van Nee, G. Awater, M. Morikura, H. Takanashi, M. Webster and K. Halford, "New High-Rate Wireless LAN Standards," IEEE Comm. Magazine, Dec. 1999.
- [3] Pierre Ferre, Angela Doufexi, Andrew Nix and David Bull, "Throughput Analysis of the IEEE 802.11 and IEEE 802.11e MAC," IEEE Wireless Communications and Networking Conference, Mar. 2004.
- [4] D. Qiao and S. Choi, "Goodput Analysis and Link Adaptation for IEEE 802.11a Wireless LANs," IEEE Transactions on Mobile Computing, Vol. 1, Oct. 2002.
- [5] T. Stockhammer, M. Hannuskela and T. Wiegand, "H.264/AVC in Wireless Environments," IEEE Transactions on Circuits and Systems for Video Technology, Vol. 13, July 2003.