Ferre, P. L., Doufexi, A., Chung How, J. T. H., Nix, A. R., & Bull, D. (2003). Link adaptation for video transmission over COFDM based WLANs.

Peer reviewed version

Link Adaptation for Video Transmission over COFDM based WLANs

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Abstract

Wireless Local Area Networks (WLANs) such as IEEE 802.11a/g and Hiperlan/2 operate in numerous transmission modes, each providing different throughputs and reliability levels. Link Adaptation algorithms proposed in the literature ([1], [2], and [3]) maximise data throughput based on channel conditions and assume that only error-free packets may be passed on to the higher layer. For multimedia applications such as video transmission, completely error-free communication is not essential, especially if robust video compression techniques are applied. In such scenarios, better decoded video quality might be obtained with a video stream transmitted at a higher bit-rate but with some degree of transmission error, rather than an error free video stream at a lower bit-rate. In this paper, we present a link adaptation scheme that improves the Quality of Service (QoS) for video transmission in terms of the overall received video quality, rather than by maximising error-free throughput. Simulation results using the H.263+ video compression standard over IEEE 802.11a are presented.

1. Introduction

Recently there has been considerable interest in home entertainment, video on demand and other home multimedia communication products. New local wireless networks such as IEEE 802.11a/g [4] and Hiperlan/2 [5] are already deployed and are currently available as a possible solution for these applications. These WLANs provide data rates up to 54 Mbit/s depending on the mode used. The link adaptation mechanism enables the system to adapt the transmission mode to a quality metric. The ability to change modes is used to determine the reliability of the system. This provides the radio with the capability to adapt to a better configuration to improve the QoS of transmission. Common link adaptation algorithms have mainly focused on maximising the error free throughput at the higher application layer. In this paper, we suggest a link adaptation algorithm that chooses the operating mode in order to maximise the QoS for video transmission over these networks, i.e. video quality. This work acknowledges that error resilient video codecs are capable of tolerating small levels of bit error in the video bit stream.

The paper is organised as follows. First a brief description of the IEEE and ETSI OFDM based LANs is given in section 2. Section 3 presents the common link adaptation algorithm based on throughput. Section 4 details video coding and packetisation basics for the simulation. In section 5, a description of our system is given. Section 6 details our algorithm using a video quality metric to change the operating mode. It also contains a comparison between the two link adaptation schemes. Finally, section 7 concludes this paper.

2. IEEE and ETSI LANs

Recent wireless LAN standards at 5GHz have similar physical layers (PHY) based on Orthogonal Frequency Division Multiplexing (OFDM) ([5], [4]) including an interleaver and a convolutional encoder. OFDM is used to combat frequency selective fading and to randomise burst errors. OFDM is implemented by means of an inverse FFT. 48 data symbols and 4 pilots are transmitted in parallel to form one OFDM symbol. In order to prevent Inter-Symbol Interference, a guard interval is inserted by means of cyclical extension [6].

IEEE 802.11a/g provides 8 operating modes (7 for Hiperlan/2) with different reliability levels and different bit rates at the physical layer (up to 54Mbit/s) depending on the modulation and coding rate, (see Table 1). However, the actual data throughput also depends on the MAC protocol employed and the link adaptation algorithm. Both standards are packet-based systems but differ in their MAC structure [2]. Hiperlan/2 uses a synchronous access with Time Division Duplex/Time Division Multiple Access

Proceedings Symposium IEEE Benelux Chapter on Communication and Vehicular Technology, 2003, Eindhoven

(TDD/TDMA) and a 2ms MAC frame along with a fixed packet size of 54 bytes [7]. IEEE 802.11a/g uses an asynchronous access technique: Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) with a variable length frame format (up to 4096 bytes). Selective Repeat ARQ is used for retransmission of erroneous packets for Hiperlan/2. IEEE 802.11 implements a Stop and Wait ARQ.

Table 1: Mode Dependent Parameters for WLANs at 5GHz and 2.4Hz

Mode	Modulation	Coding rate	Nominal bit rate [Mbit/s]
1	BPSK	1/2	6
2	BPSK	3/4	9
3	QPSK	1/2	12
4	QPSK	3/4	18
5	16QAM (H/2 only)	9/16	27
5	16QAM (IEEE only)	1/2	24
6	16QAM	3/4	36
7	64QAM	3/4	54
8	64QAM (IEEE only)	2/3	48

3. H.263+ Video Coding and Packetisation

In this paper, video sequences have been encoded using ITU-T H.263+ video coding standard [8] and then transmitted over IEEE 802.11a. H.263+ divides each video frame into non-overlapping macroblocks, where a macroblock (MB) consists of 16 by 16 luminance pixels. At Common Intermediate Format (CIF) and Quarter CIF (QCIF) resolutions (352 by 288 and 176 by 144 pixels respectively), each video frame consists of 396 MBs (18 rows of 22 MBs) and 99 MBs (9 rows of 11MBs) respectively. Figure 1 shows the MB structure at QCIF resolution on the *foreman* sequence.

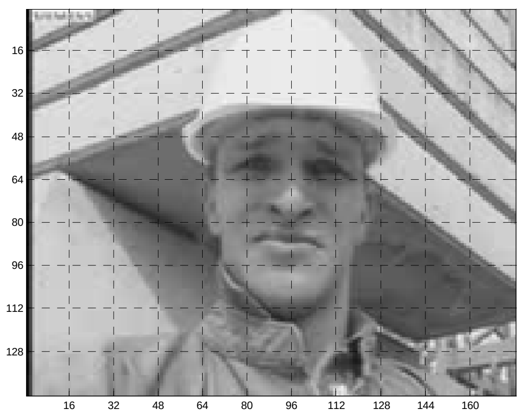

Figure 1: Macroblock structure at QCIF resolution

The slice structure mode of H.263+ (Annex K of [8]) enables a variable number of MBs to be grouped and coded together into a slice. Each slice is independently decoded. A slice has a variable length and is limited by its maximum size. The number of MBs in a slice is constrained to a maximum of 22 and 11 at CIF and QCIF resolution respectively. This ensures that at a given resolution, a slice will never contain more than one row of the image. When a MB is corrupted during transmission, it is concealed using the previous reference frame and the motion vector of the MB spatially above. The quality of the video sequence can be assessed via the *Peak-to-Peak Signal to Noise Ratio* (*PSNR*), based on the *Mean Square Error* between the received and the original sequence. The *PSNR* of one video frame is given by:

$$PSNR = 10\log\left(\frac{255^2}{MSE}\right)$$

where *MSE* is the *Mean Square Error* given by:

$$MSE = \frac{1}{N.M}\sum_{i=1}^{N}\sum_{j=1}^{M}\left[x_{original}(i,j) - x_{received}(i,j)\right]^2$$

N and M represent the width and height (in pixels) of each video frame. $x_{original}$ and $x_{received}$ represent the pixels of the original uncoded frame and the received pixels respectively. *PSNR*s of all the frames of one video sequence are then averaged to form the *PSNR* of the video sequence. The higher the *PSNR* (in dB), the better the video quality.

4. Description of the System

In the system under study, video sequences at CIF resolution have been encoded according to the H.263+ standard. Variable length slices at the output of the video encoder are first coded using an outer concatenated shortened RS code and an outer interleaver to form one IEEE 802.11a packet of variable length. Simulations have been performed with two standard video sequences: *foreman* presenting medium motion and *akiyo* presenting low motion. Each PHY operating mode carries one video stream encoded at a different bit rates: from 700kbs with mode 1 to 6350kbs with mode 7 for the *foreman* sequence and from 256kbs to 2305kbs for the *akiyo* sequence (Table 2 and Table 3).

These bit rates are obtained by varying the quantiser parameters at the encoding process. They are chosen so that each transmission will have the same usage of the radio channel, and so that ratios between encoded bit rates respect ratios between nominal bit rates of the physical layer. They are extendable to higher values but still give a good idea of the general case. For each video stream, the maximum slice size is set so that encoded videos have almost the same number of slices per frame (at around 20/21 slices per frame). As *akiyo* sequence presents slow motion features, it presents better compression performances than *foreman* sequence described as medium motion sequence. *Akiyo* presents better video quality (*PSNR*) for lower encoded bit rates than *foreman*. The maximum slice sizes are smaller with *akiyo* sequence since it contains slow motion. All the MBs in a slice do not need to be encoded, and the length of a slice is therefore smaller.

Table 2: Video Bit Rates for Transmission over IEEE 802.11a – *foreman* Sequence

Mode	Nominal bit rate [Mbit/s]	Maximum Slice Size (bytes)	Slices per Frame	PSNR (dB)	Video bit rate Mbit/s
1	6	230	21.08	37.07	0.700
2	9	345	21.67	38.71	1.050
3	12	460	20.50	39.95	1.410
4	18	690	20.52	41.52	2.120
5	24	920	21.97	42.77	2.830
6	36	1380	21.92	44.85	4.235
7	54	2100	19.28	48.78	6.350
8	48	1840	19.74	46.5	5.650

Table 3: Video Bit Rates for Transmission over IEEE 802.11a – *akiyo* Sequence

Mode	Nominal bit rate [Mbit/s]	Maximum Slice Size (bytes)	Slices per Frame	PSNR (dB)	Video bit rate Mbit/s
1	6	100	20.03	40.58	256
2	9	150	20.11	42.50	380
3	12	200	19.83	43.76	512
4	18	300	20.44	46.01	770
5	24	400	21.50	46.83	1024
6	36	600	20.01	47.91	1540
7	54	900	18.18	49.12	2305
8	48	800	18.47	47.72	2050

5. Link Adaptation based on Throughput

Link adaptation algorithms that maximise error-free data throughput have been proposed recently ([1], [2] and [3]). The throughput depends on the physical layer and therefore on the operating mode. The link quality measurement used is packet error rate (*PER*). A Cyclic Redundancy Check (*CRC*) is used for error detection. If a packet is detected to be erroneous by the CRC code (in both IEEE 802.11a and Hiperlan/2) or if the ACK time out of the IEEE 802.11 MAC has expired, the terminal will retransmit the packet [2]. As a first approximation, the link throughput with retransmission is given by: *Throughput = R*(1 – PER)*, where *R* and *PER* are the nominal bit rate and packet error rate for a specific mode [6] respectively. The link adaptation scheme based on throughput would choose the mode with the highest throughput for a given *carrier to noise ratio* (C/N). Figure 2 and Figure 3 show a possible link adaptation scheme based on throughput for IEEE 802.11a under ETSI standard Channel A conditions for the *foreman* and *akyio* video sequences respectively.

The transmitter would choose the operating mode according to the graph after determining the corresponding C/N of the receiver. We can see that modes 2 and 4 will never be chosen since it does not offer the highest throughputs for any C/N for channel model A. This can be explained by the fact that mode 2 has a less reliable channel and a smaller nominal bit rate than mode 3. We should note that *akiyo* sequence offers very little motion, so slices have smaller sizes (see Table 2 and Table 3). *Akiyo* sequence therefore shows better PER performance as small packets are less likely to be corrupted when transmitted through the WLAN. Throughput for the *akiyo* sequence is therefore slightly better for a given mode and a given C/N.

Link adaptation based on throughput assumes that the received data must be error-free. This is not necessary for video transmission where the quality of the video depends on a trade-off between bit-rate and *BER*/*PER*. Link adaptation based on throughput does not consider the video quality.

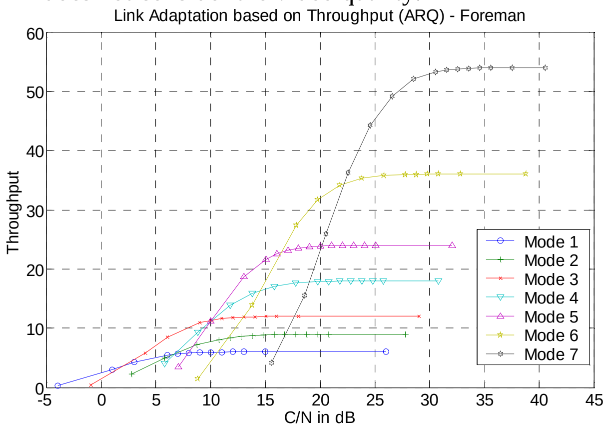

Figure 2: Link Adaptation based on Throughput for *foreman* sequence

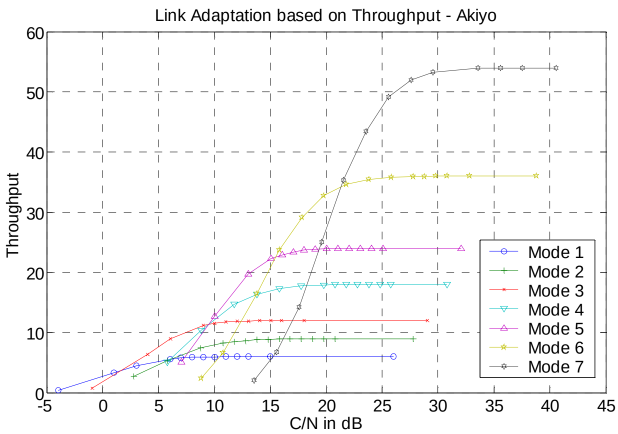

Figure 3: Link Adaptation based on Throughput for *akiyo* sequence

6. Link Adaptation based on Video Quality (*PSNR*)

The link adaptation proposed in this paper is based on video quality rather than link throughput. The 8 modes of IEEE 802.11a/g provide different nominal bit rates from 6 to 54Mbit/s (see Table 1 and Table 2). For example, mode 6 with a nominal bit rate of 36Mbit/s can carry an application with a higher bit rate than mode 1 with a nominal bit rate of 6Mbit/s. This means that, for the same time usage of the channel, mode 6 can carry video with higher quality than mode 1, at the expense of a less reliable channel. The resulting video quality is a function of both the video transmission bit rate and the reliability of the channel. Since videos can tolerate transmission errors if robust compression and good concealment techniques are applied, a less reliable mode with higher video bit rate could achieve better video performance than a good modulation/coding PHY scheme with a low video bit rate as shown in Figure 4 and Figure 5.

Figure 4: Frame 30, Foreman at 700kbit/s Mode 1, C/N = 25dB BER=0

Figure 5: Frame 30, Foreman at 4235kbs Mode 6, C/N = 25dB, BER=10^{-3}

It can be seen that Mode 1 is carrying *foreman* sequence at 700kbit/s with a Bit Error Rate (BER) of 0, i.e. with no error. The *PSNR* is then 37.07dB. On the other hand, Mode 6 is carrying the same sequence encoded at 4235kbit/s (i.e. with an original quality of 44.85dB) with a BER of 10^{-3}. Figure 5 shows better visual resolution compared to Figure 4. Impairments due to errors are almost insignificant to be visually noticed.

In Figure 2 and Figure 3, link adaptation based on throughput required the use of retransmission, so *ARQ* has been used for the simulations based on *PSNR*. In order to perform link adaptation based on video quality, the transmitter must have some knowledge of the received video quality. This can be achieved with the exchange of quality metrics via the feedback channel (available in the IEEE 802.11 MAC) or with the use of pre-defined maps for different video content types obtained from statistical measurements. This is not in the scope of this paper and we assume that the transmitter is able to choose the operating mode for the current C/N configuration.

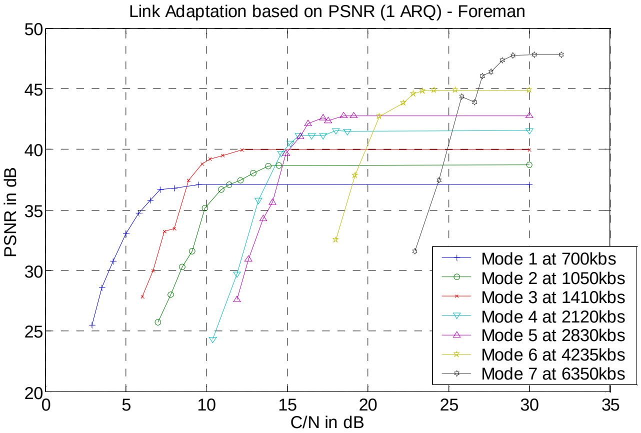

Figure 6: Link Adaptation Based on PSNR for *Foreman* sequence

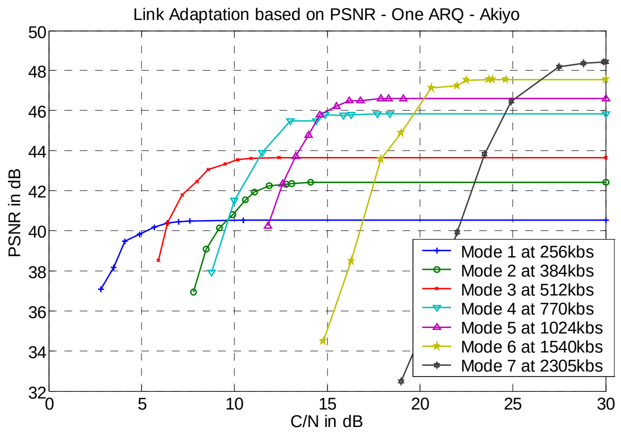

Figure 7: Link Adaptation Based on PSNR for *Akiyo* sequence

The link adaptation algorithm based on video quality chooses the operating mode that provides the highest *PSNR* for a given C/N. Figure 6 and Figure 7 show a possible adaptation based on *PSNR*, for the *foreman* and *akiyo* sequences respectively.

The schemes optimising throughput and optimising *PSNR* present different switching points. For the *foreman* sequence and the throughput based algorithm, changing from mode 1 to mode 3 occurs at a *C/N* of approximately 2 dB according to Figure 2. Following Figure 6, switching from mode 1 to mode 3 at 2dB will result in a considerable drop in *PSNR*. It is then preferable to stay with mode 1 until the degree of errors in mode 3 allows a switching based on *PSNR* (at around a *C/N* of 9dB). Figure 6 shows that better *PSNR* is obtained with mode 3 than with mode 1 from a C/N of 9dB to 15dB, where mode 3 still carries errors and where mode 1 is almost error-free. We recall that

simulations based on PSNR imply the use of one retransmission.
By comparing Figure 6 and Figure 7, we can see that switching points are slightly different for the two video sequences. Mode 4 will never be used for the transmission of *foreman*, where it is used with *akiyo*. Changing from mode 1 to mode 3 for *akiyo* occurs at around 6dB compared with approximately 9dB for *foreman*. In a general way, we can see that *akiyo* C/N changing points are lower than for *foreman*. This is mainly due to smaller packet lengths that provide better PER performances and therefore better video quality. The other reason is due to the fact that *akiyo* is a slow motion sequence, therefore it is more robust to errors.

7. Conclusions

In this paper, we present a new link adaptation strategy based on maximisation of video quality (*PSNR)* designed for enhanced video transmission over WLANs. Whereas common link adaptation algorithms maximise the error free data throughput at the higher layer, our algorithm maximises the video quality (in terms of *PSNR*). The two algorithms show different switching points. The comparison between the different sequences shows that the switching points are also dependent on the content of the sequence (slow/medium motion).
If the bit stream carries data that require error-free transmission, such as file transfer or side information in a multimedia system, then the throughput should be optimised and the link adaptation algorithm based on error-free throughput should be adopted. On the other hand, if the bit stream carries a video sequence, then video quality should be optimised and the link adaptation based on the *PSNR* presented in this paper should be adopted.

8. References

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