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Cross-Layer Application of Video Streaming for WiMAX: Adaptive Protection with Rateless Channel Coding

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1. Introduction

Video streaming is an important application of broadband wireless access networks such as IEEE 802.16d,e (fixed and mobile WiMAX) (IEEE 802.16e-2005, 2005; Andrews et al., 2007; Nuaymi, 2007), as it essentially justifies the increased bandwidth compared to 3G systems, which bandwidth capacity will be further expanded in part 'm' of the standard (Ahmandi, 2011, written by Intel's chief technology officer). Broadband wireless access continues to be rolled out in many parts of the world that do not benefit from existing wired infrastructures or cellular networks. In particular, it allows rapid deployment of multimedia services in areas in the world unlikely to benefit from extensions to both 3G such as High Speed Downlink Packet Access (HSDPA) and UMTS such as Long-Term Evolution (Ekstrom et al., 2006). WiMAX is also cost effective in rural and suburban areas in some developed countries (Cicconetti et al., 2008). It is also designed to provide effective transmission at a cell's edge (Kumar, 2008), by allocation to a mobile user of sub-channels with separated frequencies to reduce co-channel interference. Time Division Duplex (TDD) through effective scheduling of time slots increases spectral efficiency, while the small frame size of 5 ms can reduce latency for applications such as video conferencing. The transition to the higher data rates of IEEE 802.16m indicates the competiveness of WiMAX.

Mobile WiMAX was introduced in 2007, as part *e* of the IEEE 802.16 standard, to strengthen the fixed WiMAX part d standard of 2004. Mobile WiMAX, IEEE 802.16e, specifies the lower two layers of the protocol stack. Like many recent wireless systems, part *d* utilized Orthogonal Frequency Division Multiplexing (OFDM) as a way of increasing symbol length to guard against multi-path interference. The sub-carriers inherent in OFDM were adapted for multi-user usage by means of Orthogonal Frequency Division Multiple Access (OFDMA), allowing subsets of the lower data-rate sub-carriers to be grouped for individual users. Sub-channel spectral allocation can range from 1.25 MHz to 20 MHz. Adaptive antenna systems and Multiple Input Multiple Output (MIMO) antennas can improve coverage and reduce the number of base stations. Basic Multicast and Broadcast Services (MBS) are supported by mobile WiMAX. IEEE 802.16m (Ahmandi, 2011) is expected to increase data rates to 100 Mbps mobile and 1 Gbps fixed delivery. However, 802.16m is not backwards compatible with 802.16e, though it does support joint operation with it.

One of the drivers of WiMAX's development is its suitability (because of centralized scheduling using TDD) for video streaming. Video streaming, as a part of Internet Protocol TV (IPTV) (DeGrande et al., 2008), can support time-shifted TV, start-again live TV, and video-on-demand. As an example, the UK's BBC iPlayer supports the former two of these unicast services, though using a form of block-based streaming in which differences in bandwidth capacity at the access network are accommodated by changes in spatial resolution. As the iPlayer's TV display is through a browser plug-in an alternative name for this service is Internet TV. Internet TV differs from what might be termed true IPTV as it uses 'best-effort' IP routing. The iPlayer is probably the best approximation to the type of video streaming considered in this Chapter. However, this Chapter does not utilize the chunk-based pseudo streaming of the BBC iPlayer but a packet-based streaming directly from the output of the codec or from pre-encoded stored video. It also does not use the Transmission Control Protocol (TCP) that underlies the Hyper Text Transport Protocol (HTTP) as this can lead to unacceptable delays across wireless networks, as TCP reacts to adverse channel conditions as if they were traffic congestion. IPTV as a service to set-top boxes or desk-top PCs generally includes TV channel multiplexing within a coded stream encapsulated in (say) MPEG-2 Transport System (TS) application-layer packets as well as an Electronic Program Guide (EPG) service. When transferred to a mobile system, this type of IPTV may well require the video service office (VSO) (DeGrande et al., 2008), as the last step in a content delivery network (CDN) overlay to respond to channel selection by the user rather than deliver all channels to the user (as occurs in fiber-to-the-home services). Such CDNs also have the important function of caching content nearer to users. It should be remarked that the BBC, provider of the iPlayer, acts as a public service and, hence, does not require a formal business model, whereas other IPTV services generally have a traditional business plan and may employ encryption and digital rights management .

It has become increasingly clear that Next Generation Networks (NGNs) will not be based on wireline devices as previously envisaged but on mobile devices. However, the volatile nature of the wireless channel (Goldsmith, 2005), due to the joint effect of fading, shadowing, interference and noise, means that an adaptive approach to video streaming is required. To achieve this exchange of information across the protocol layers is necessary, so that the application-layer can share knowledge of the channel state with lower protocol layers. Though a cross-layer application in general has its detractions, such as the difficulty of evolving the application in the future, because of the delay constraints of video streaming and multimedia applications in general, its use is justified.

This Chapter provides a case study, in which information from the PHYsical layer is used to protect video streaming over a mobile WiMAX link to a mobile subscriber station (MS). Protection is through an adaptive forward error correction (FEC) scheme in which channel conditions as reported by channel estimation at the PHY layer serve to adjust the level of application-layer FEC. This flexibility is achieved by use of rateless channel coding (MacKay, 2005), in the sense that the ratio of FEC to data is adjusted according to the information received from the PHY layer. The scheme also works in cooperation with PHY-layer FEC, which serves to filter out packet data in error, so that only correctly received data within a packet are passed up the layers to the video-streaming application. The 802.16e standard provides Turbo coding and hybrid Automatic Repeat request (ARQ) at the PHY layer with scalable transmission bursts depending on radio frequency conditions. However, application-layer forward error correction (Stockhammer et al., 2007) is still recommended for IPTV during severe error conditions.

Rateless channel coding allows the code rate to be adaptively changed according to channel conditions, avoiding the thresholding effect associated with fixed-rate codes such as Reed-Solomon. However, the linear decode complexity of one variant of rateless coding, Raptor coding (Shokorallahi, 2006), has made it attractive for its efficiency alone. For broadcast systems such as 3GPP's Multimedia Broadcast Multicast System (MBMS) (Afzal, 2006) , as channel conditions may vary for each receiver, the possibility of adapting the rate is not exploited, even with a rateless code. However, for unicast video-on-demand and time-shifted TV streaming it is possible to adaptively vary the rate according to measured channel conditions at the sender. These services are a commercially-attractive facility offered by IPTV as they add value to a basic broadcast service.

In addition to analysis of the cross-layer protection scheme, the Chapter demonstrates how source-coded error resilience can be applied by means of data-partitioning of the compressed video bitstream. This in turn encourages the use of duplicate data, as a measure against packet erasure. Packet erasure can still occur despite adaptive FEC provision for data within WiMAX packets, i.e. Medium Access Control (MAC) protocol data units (MPDUs). Assessment of the results of the adaptive protection scheme is presented in terms of packet drops, data corruption and repair, end-to-end delay introduced, and the dependency of objective video quality upon content type.

The remainder of this Chapter is organized as follows. Section 2 sets the context for the case study with discussion of WiMAX cross-layer design, IPTV for WiMAX, together with source and channel coding issues. Section 3 presents the simulation model for the case study with some sample evaluation results. Finally, Section 4 makes some concluding remarks.

2. Context of the case study

This Section now describes research into cross-level design for mobile WiMAX in respect to video streaming.

2.1 WiMAX cross-layer design

The number of cross-layer designs for wireless network video-streaming applications has considerably increased (Schaar & Shankar, 2005) with as much as 65% of applications in mobile ad hoc networks adopting such designs. This should not be a surprise, as source coding and streaming techniques in the application layer cannot be executed in isolation from the lower layers, which coordinate error protection, packet scheduling, packet dropping when buffers overflow, routing (in ad hoc and mesh networks), and resource management.

In WiMAX multicast mode, scheduling decisions for the real-time Polling Service (rtPS) queue, one of the WiMAX quality of service queues (Andrews et al., 2007), in particular are suspended. This can cause excessive delay to multimedia applications. To avoid this, in Chang & Chou (2007) knowledge of the application types and their delay constraints is conveyed to the datalink layer, where the scheduling mode is decided upon. The network layer can also benefit from communication with the datalink layer in order to synchronize

WiMAX and IP handoff management (Chen & Hsieh, 2007) and in that way reduce the number of control messages. For further general examples of cross-layer design in WiMAX, the reader should consult Kuhran et al. (2007).

Video applications using PHY layer information were targeted in Juan et al. (2009) and She et al. (2009). In Juan et al. (2009), layers of a scalable video stream were mapped onto different 802.16e connections. The base station (BS) periodically reports average available bandwidth to a collocated video server, which then dynamically allocates video packets to the connections. The base layer occupies one connection while the remaining enhancement layer(s) packets occupy the second connection. If base layer packets (and certain key pictures) are lost, then the BS only retransmits these if available bandwidth permits. In She et al. (2009), cross-layer design was applied to WiMAX IPTV multicast to guard against channel diversity between different receivers. The solution again utilized scalable video layers but, instead of a mapping onto different connections, superposition coding is employed. In such coding, more important data are typically modulated at Binary Phase Shift Keying (BPSK) whereas enhancement layers are transmitted at higher order modulation such as 16QAM (16-point Quadrature Amplitude Modulation). A cross-layer unit performs the superposition at the BS, whereas, at the subscriber stations, layers are selected according to channel conditions. Both these schemes fall into the class of wireless medium-aware video streaming. However, neither of these papers explained how signaling between lower and higher level protocols can take place.

In Neves et al. (2009) it was pointed out that IEEE 802.21 Media Independent Handover (MIH) services (IEEE 802.21, 2008) already provides a framework for cross-layer signaling that could be enhanced for more general purposes. In fact, another WiMAX specific set of standardized communication primitives is IEEE 802.16g. However, it could be that legacy WiMAX systems will need to be provided with a different interface. In 802.21, a layer 2.5 is inserted between the level 2 link layer and the level 3 network layer. Upper-layer services, known as MIH users or MIHU communicate through this middleware to the lower layer protocols. One of the middleware services, the Media Independent Event Service (MIES) is responsible for reporting events such as dynamic changes in link conditions, link status and quality, which appears suitable or at least near to the requirements of the adaptive scheme reported in this Chapter.

There are penalties in applying a cross-layer scheme (Kawadia & Kumar, 2003), namely it may result in a monolithic application that is hard to modify or evolve. However, for wireless communication (Srivastava & Motani, 2005) an adaptive scheme that leverages information across the layers can cope with the volatile state of the channel due to fading and shadowing and the constrained available bandwidth of the channel. It is not necessary to abandon layering altogether in a 'layerless' design but simply to communicate between the layers. Video applications break protocol boundaries with limited objectives in mind, though improvements in performance remain the goal. Performance may be defined variously in terms of reduction of delay, reduction of errors, throughput efficiency, and, in wireless networks, reduction of energy consumption. This list by no means exhausts the possible trade-offs that can be engineered through cross-layer exchange of information.

2.2 IPTV video streaming

The ability to provide TV over wireless (and digital subscriber line) access networks has undoubtedly been encouraged by the increased compression achievable with an

H.264/Advanced Video Coding (AVC) codec (Wiegand et al., 2003), for example reducing from at least 1.5 Mbps for MPEG-2 video to less than 500 kbps for equivalent quality TV using H.264/AVC compression. The density of subscribers is linked to the number of sub-channels allocated per user, which is a minimum of one per link direction. In a 5 MHz system, the maximum is 17 uplink and 15 downlink sub-channels. For a 10 MHz system (FFT size 1024) 35 downlink and 30 uplink sub-channels are available. For a mobile WiMAX (IEEE 802.16e) 10 MHz system, capacity studies (So-In et al., 2010) suggest between 14 and 20 mobile TV users per cell in a 'lossy' channel depending on factors such as whether simple or enhanced scheduling and whether a single antennas or 2×2 MIMO antennas are activated. However, given the predicted increase in data rates arising from IEEE 802.16m, the number of uni-cast video users (Oyeman et al., 2010) with 4×2 Multi User (MU)-MIMO antennas, will be 44 at 384 kbps and 22 at 768 kbps in an urban environment. For a similar configuration but using IEEE 802.16m 20 MHz (FFT size 2048) rather than IEEE 802.16m 80 MHz channels (4 FFT of size 2048 each) , the authors of Oyeman et al. (2010) reported the number of uni-cast video users to be 11 and 6 depending on data-rates. However, it should be born in mind that the capacity of a WiMAX cell can be scaled up by means of sectored antennas, whereas the above capacities for IEEE 802.16m are for a single sector. A typical arrangement (Jain et al., 2008) is to have three sectors per cell. It should be remarked that in Oyeman et al. (2010), the subscriber density of LTE-Advanced is assessed as very similar to that of IEEE 802.16m.

In Degrande et al. (2008), ways to improve IPTV quality were discussed with the assumption that intelligent content management would bring popular video content nearer to the end viewer. The typical IPTV architecture considered, Fig. 1a, assumes a super head-end (SHE) distributor of content across a core network to regional video hub offices (VHOs). VHOs are connected to video serving offices (VSOs) over a regional metro network. It is a VSO that interacts with users over an access network. While Degrande et al. (2008) have managed networks using IP framing but *not* 'best-effort' routing in mind, CDNs such as iBeam and Limelight originated for the unmanaged Internet. Microsoft TV IPTV Edition is probably the best known of the managed network proprietary solutions and this too can utilize WiMAX delivery (Kumar, 2008).

An overview of how an IPTV system with WiMAX fixed or mobile delivery is presented in Uilecan et al. (2007). The system takes advantage of WiMAX's point-to-multipoint (PMP) mode for the broadcast of TV channels. MPEG2-TS packets containing multiplexed TV channels are encapsulated in RTP/UDP/IP packets. Header suppression and compression techniques reduce the overhead. In Issa et al. (2010), IPTV streaming was evaluated on a WiMAX testbed for downlink delivery of TV channels and uplink delivery of either TV news reports or video surveillance; refer to Figure 1b. Broadly for streaming media WiMAX's application class 3 supports medium bandwidth between 0.5 and 2 Mbps and jitter less than 100 ms. In fact, the ITU-T's recommendations for IPTV (not mobile TV) are even more stringent with jitter less than 40 ms and packet loss rates less than 5%. Video conferencing (not covered in this Chapter) will require jitter less than 50 ms but probably much lower bandwidths and end-to-end latency less than 160 ms.

In a native Real-Time Protocol (RTP) solution for IPTV distribution, the Real-Time Protocol Streaming Protocol (RTSP) is available for TV channel selection and can support pseudo video cassette recorder functions such as PAUSE and REWIND. The Real-Time Control

Protocol (RTCP) is suitable for feedback that may be used to reduce the streaming rate for live video, or by stream switching or a bitrate transcoder if pre-encoded video is being streamed.

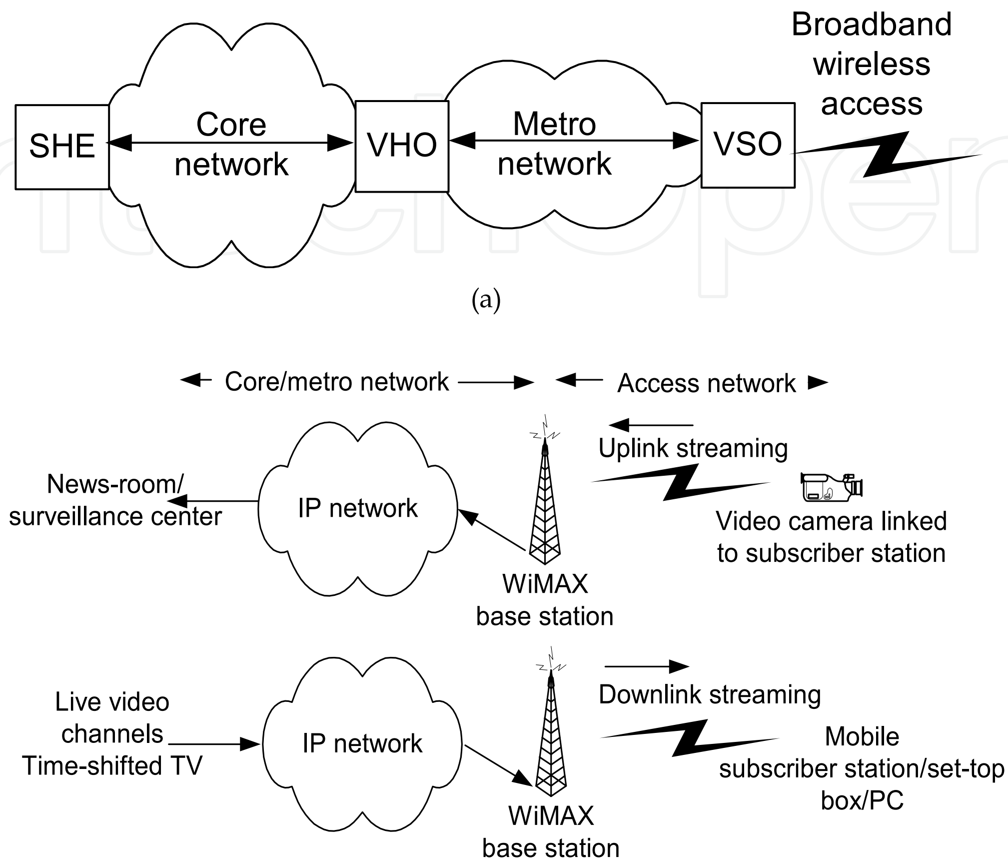


Fig. 1. (a) Schematic IPTV distribution network (b) Downlink and uplink streaming scenarios.

Originally, it was assumed (Kumar, 2008) that the IP networks involved would form "walled gardens", which would be managed by telecommunications companies ('telcos') and which might exclude competitors in the speech communication market such as Skype voice-over-IP and include traditional forms of mobile broadcast. Originally also it was thought that WiMAX's extended coverage would function as a backhaul service to IEEE 802.11 networks, which are limited in range by their access control mechanism, whereas WiMAX has been developed as a replacement for many smaller but isolated IEEE 802.11 hotspots. The IP Multimedia Subsystem (IMS) then allows roaming across networks with a common framing standard, outside the 'walled garden'. In the IMS view, WiMAX is an underlying network just as LTE would be. WiMAX's real-time Polling Service (rtPS) is the scheduling service class suited to IPTV video streaming.

2.3 Source coding for video streaming

Source coding issues are now briefly discussed. As mentioned in Section 1, data-partitioning was enabled for error resilience purposes. In an H.264/AVC codec (Wenger, 2003), when data-partitioning is enabled (Stockhammer & Bystrom, 2007), inter-coded slices are normally divided into three separate partitions according to decoding priority. These data are packed into different Network Abstraction Layer units (NALU's). Each NALU is encapsulated into an IP/RTP/UDP packet for possible IMS transport. Each partition is located in either of type-2 to type-4 NAL units. A NAL unit of type 2, also known as partition-A, comprises the most important information of the compressed video bit stream of P- and B-pictures, including the MB addresses, MVs, and essential headers. If any MBs in these pictures are intra-coded, their frequency transform coefficients are packed into the type-3 NAL unit, also known as partition B. Type 4 NAL, also known as partition-C, carries the transform coefficients of the motion-compensated inter-picture coded macroblocks. When motion-copy error concealment is enabled at a decoder, then receipt of a partition-A carrying packet is sufficient to enable a partial reconstruction of the frame. When the quantization parameter (QP) is appropriately set, the smaller size of partition-A results in smaller packet length and, hence, a reduced risk of error.

In adverse channel conditions, duplicate partition-A packets are transmitted. On the other hand, the duplicate partition-A stream should be turned off during favorable channel conditions. In an H.264/AVC codec, it is instead possible to send redundant pictures slices (Radulovic et al., 2007), which employ a coarser quantization than the main stream, but this can lead to encoder-decoder drift. Besides, for data-partitioning, replacing one partition with a redundant slice with a different QP to the other partitions would not permit reconstruction in an H.264/AVC codec.

In order to decode partition-B and -C, the decoder must know the location from which each MB was predicted, which implies that partitions B and C cannot be reconstructed if partition-A is lost. Though partition-A is independent of partitions B and C, Constrained Intra Prediction (CIP) should be set in the codec configuration (Dhondt et al., 2007) to make partition-B independent of partition-C. By setting this option, partition-B MBs are no longer predicted from neighboring inter-coded MBs. This is because the prediction residuals from neighboring inter-coded MBs reside in partition-C and cannot be accessed by the decoder if a partition-C packet is lost. There is a by-product of increasing overhead from extra packet headers in a reduction in compression efficiency but the overall decrease in packet size may be justified in error- prone environments.

2.4 Rateless channel coding for video streaming

Rateless or Fountain coding (MacKay, 2005), of which Raptor coding (Shokorallahi, 2006) is a subset, is ideally suited to a binary erasure channel in which either the error-correcting code works or the channel decoder fails and reports that it has failed. In erasure coding, all is not lost as flawed data symbols may be reconstructed from a set of successfully received symbols (if sufficient of these symbols are successfully received). A fixed-rate (n, k) Reed-Solomon (RS) erasure code over an alphabet of size $q = 2L$ has the property that if any k out of the n symbols transmitted are received successfully then the original k symbols can be decoded. However, in practice not only must n, k, and q be small but also the computational complexity of the decoder is of order $n(n - k) \log_2 n$. The erasure rate must also be estimated in advance.

The class of Fountain codes allows a continual stream of additional symbols to be generated in the event that the original symbols could not be decoded. It is the ability to easily generate new symbols that makes Fountain codes rateless. Decoding will succeed with small probability of failure if any of k (1 + ε) symbols are successfully received. In its simplest form, the symbols are combined in an exclusive OR (XOR) operation, according to the order specified by a random, low density generator matrix and, in this case, the probability of decoder failure is $\partial = 2^{-k\varepsilon}$, which, for large k, approaches the Shannon limit. The random sequence must be known to the receiver but this is easily achieved, through knowledge of the sequence seed.

Luby transform (LT) codes (Luby, 2002) reduce the complexity of decoding a simple Fountain code (which is of order k^3) by means of an iterative decoding procedure. The 'belief propagation' decoding relies on the column entries of the generator matrix being selected from a robust Soliton distribution. In the LT generator matrix case, the expected number of degree one combinations (no XORing of symbols) is $S = c \ln(k/\partial)\sqrt{k}$, for small constant c. Setting $\varepsilon = 2 \ln(S/\partial)$ S ensures that, by sending k(1 + ε) symbols, these symbols are decoded with probability $(1 - \partial)$ and decoding complexity of order k ln k.

The essential differences between Fountain erasure codes and RS erasure codes are that: Fountain codes in general (not Raptor codes) are not systematic; and that, even if there were no channel errors, there is a small probability that the decoding will fail. In compensation, they are completely flexible, have linear decode computational complexity, and generally their overhead is considerably reduced compared to fixed erasure codes. Apart from the startling reduction in computational complexity, a Raptor code (Shokorallahi, 2006) has the maximum distance separable property. That is, the source packets can be reconstructed with high probability from any set of k or just slightly more than k received symbols. A further advantage of Raptor coding is that it does not share the high error floors on a binary erasure channel (Palanki & Yedidai, 2004) of prior rateless codes. However, it is probably the combination of closeness to the ergodic capacity and the low rate of decoder error (Castura & Mao, 2006) that most determines the advantage of Raptor codes over other forms of rateless channel coding.

3. Case study

A video application can adopt at least three methods of protection for fragile video streams. The first method is application-layer channel coding. However, application coding is only effective to the extent that a packet actually reaches a wireless device and is not lost beforehand. Packets can be lost in a variety of ways: because of buffer overflow; or because the signal-level drops below the receiver's threshold; or because the physical-layer forward error correction is unable to reconstruct enough of the packet to be able to pass data up to the application layer. Therefore, the second method of protection is duplication of all or part of the original bitstream. The duplicated packets are sent alongside the original video stream. A third method is to anticipate errors at the source-coding stage through error resilience, with a good number of such techniques presented in Stockhammer & Zia (2007). Error resilience can act as an aid to reconstruction through error concealment. The scheme described in this Chapter's case study utilizes all three methods of protection. Simulations show that in particularly harsh channel conditions the scheme is able to protect the video stream against data loss and subsequently achieve reasonable video quality at the mobile device. Without the protection scheme the video quality would be poor.

In the protection scheme, application-layer channel coding takes advantage of rateless channel coding (MacKay, 2005) to dynamically adapt to channel conditions. Extra redundant data are 'piggybacked' onto a new packet so as to aid the reconstruction of a previous packet. To achieve adaptation (and also to turn off duplicate slices during favorable conditions) channel estimation is necessary. As an example, the IEEE 802.16e standard (IEEE 802.16e-2005, 2005) specifies that a mobile station or device should provide channel measurements, which can either be received signal strength indicators or may be carrier-to-noise-and-interference ratio measurements made over modulated carrier preambles. Therefore, to aid in this process the method assumes one of these methods is implemented.

Error resilience is provided by data partitioning (Stockhammer & Bystrom, 2007). Data-partitioning rearranges the video bitstream according to the reconstruction priority of the compressed data. There is less overhead than other forms of error resilience such as the popular Flexible Macroblock Ordering (Lambert et al., 2005). Consequently, data-partitioning can operate during favorable channel conditions, as well as unfavorable channel conditions. On the other hand, the duplicate stream protection mentioned previously should be turned off during favorable channel conditions, as its transmission involves a significant overhead. 'Redundant' data at coarser quantization levels can be sent instead of duplicated data but redundancy results in encoder-decoder drift, unless a memory-intensive, multiple-reference scheme (Zhu et al., 2006) is employed.

3.1 Implementing the protection scheme

In the adaptive channel coding scheme, the probability of channel byte loss through fast fading (BL) serves to predict the amount of redundant data to be added to the payload. In an implementation, BL, is found through measurement of channel conditions. If the original packet length is L, then the redundant data is given simply by

$$\begin{aligned} R &= L\times BL+(L\times BL^2)+(L\times BL^3)\ldots \\ &= L/(1-BL) - L, \end{aligned} \tag{1}$$

which adds successively smaller additions of redundant data, based on taking the previous amount of redundant data multiplied by BL.

Rateless code decoding in traditional form operates by a belief-propagation algorithm (MacKay, 2005) which is reliant upon the identification of clean symbols. This latter function is performed by PHY-layer forward error correction, which passes up correctly received blocks of data (checked through a cyclic redundancy check) but suppresses erroneous data. For example, in IEEE 802.16e (Andrews et al., 2007), a binary, non-recursive, convolutional encoder with a constraint length of 7 and a native rate of 1/2 operates at the PHY layer.

If a packet cannot be decoded, despite the provision of redundant data, extra redundant data are added or 'piggybacked' onto the next packet. In Figure 2, packet X is corrupted to such an extent that it cannot be immediately decoded. Therefore, in packet X+1 some extra redundant data are included up to the level that decode failure is no longer certain.

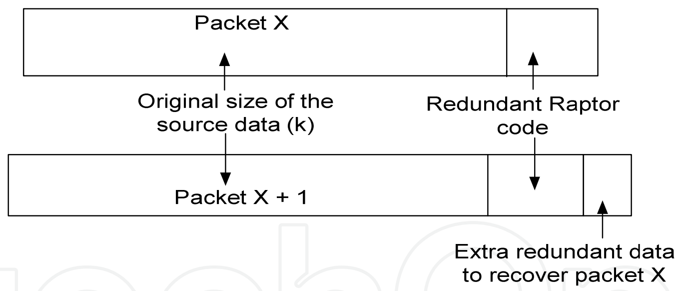


Fig. 2. Division of payload data in a packet (MPDU) between source data, original redundant data and piggybacked data for a previous erroneous packet.

3.2 Modeling the WiMAX environment

To evaluate the scheme, transmission over WiMAX was carefully modeled. The PHY-layer settings selected for WiMAX simulation are given in Table 1. The antenna heights are typical ones taken from the standard (IEEE 802.16e-2005, 2005). The antenna was modeled for comparison purposes as a half-wavelength dipole, whereas a sectored set of antenna on a mast might be used in practice to achieve directivity and, hence, better performance. The IEEE 802.16e Time Division Duplex (TDD) frame length was set to 5 ms, as only this value is supported in the WiMAX forum simplification of the standard. The data rate results from the use of one of the mandatory coding modes (IEEE 802.16e-2005, 2005) for a TDD downlink/uplink sub-frame ratio of 3:1. The base station (BS) was assigned more bandwidth capacity than the uplink to allow the WiMAX BS to respond to multiple mobile devices.

Parameter	Value
PHY	1024 OFDMA
Frequency band	5 GHz
Bandwidth capacity	10 MHz
Duplexing mode	TDD
Frame length	5 ms
Max. packet length	1024 B
Raw data rate (downlink)	10.67 Mbps
Modulation	16-QAM 1/2
Guard band ratio	1/16
MS transmit power	245 mW
BS transmit power	20 W
Approx. range to SS	1 km
Antenna type	Omni-directional
Antenna gains	0 dBD
MS antenna height	1.2 m
BS antenna height	30 m

OFDMA = Orthogonal Frequency Division Multiple Access,
QAM = Quadrature Amplitude Modulation, TDD = Time Division Duplex

Table 1. IEEE 802.16e parameter settings

Channel model

To establish the behavior of rateless coding under WiMAX, the ns-2 simulator augmented with a module or patch [12] that has proved an effective way of modeling IEEE 802.16e's behavior. Ten runs per data point were averaged (arithmetic mean) and the simulator was first allowed to reach steady state before commencing testing.

A two-state Gilbert-Elliott model served to simulate the channel model for WiMAX. In (Wang & Chang, 1996), it was shown that this model sufficiently approximates to Rayleigh fading, as occurs in urban settings during transmission from a base station to a mobile device. Moreover, in Jiao et al. (2002) it was shown that a first-order Markov chain can also model packet-level statistics. The main intention of our use of the twofold Gilbert-Elliott model was to show the response of the protection scheme to 'bursty' errors. These errors can be particularly damaging to compressed video streams, because of the predictive nature of source coding. Therefore, the impact of 'bursty' errors (Liang et al., 2008) should be assessed in video-streaming applications.

To model the effect of slow fading at the packet-level, the PGG (probability of being in a good state) was set to 0.95 and the PBB (probability of being in a bad state) = 0.96. The model has two hidden states which were modeled by Uniform distributions with PG (probability of packet loss in a good state) = 0.02 and PB (probability of packet loss in a bad state) = 0.01. The selection of a Uniform distribution is not meant to model the underlying physical process but to reflect the error patterns experienced at the application.

Additionally, it is still possible for a packet not to be dropped in the channel but, nonetheless, to be corrupted through the effect of fast fading. This byte-level corruption was modeled by a second Gilbert-Elliott model, with the same parameters (applied at the byte level) as that of the packet-level model except that PB (probability of byte loss) was increased to 0.165.

Assuming perfect channel knowledge of the channel conditions when the original packet was transmitted establishes an upper bound beyond which the performance of the adaptive scheme cannot improve. However, we have included measurement noise into the estimate of BL to test the robustness of the scheme. Measurement noise was modelled as a zero-mean Gaussian (normal) distribution and added up to a given percentage (5% in the evaluation) to the packet loss probability estimate.

In order to introduce sources of traffic congestion, an always available FTP source was introduced with TCP transport to a second mobile station (MS). Likewise, a CBR source with packet size of 1000 B and inter-packet gap of 0.03 s was also downloaded to a third MS. WiMAX has a set of quality-of-service queues at a BS. While the CBR and FTP traffic occupy the non-rtPS (non-real-time polling service) queue, rather than the rtPS queue, they still contribute to packet drops in the rtPS queue for the video, if the packet rtPS buffer is already full or nearly full, while the nrtPS queue is being serviced. Buffer sizes were set to fifty packets, as larger buffers lead to start-up delays and act as a drain upon MS energy.

The following types of erroneous packets were considered: packet drops at the BS sender buffer and packet drops through channel conditions; together with corrupted packets that were received but affected by Gilbert-Elliott channel noise to the extent that they could not be immediately reconstructed without a retransmission of piggybacked redundant data.

Notice that if the retransmission of additional redundant data still fails to allow the original packet to be reconstructed then the packet is simply dropped.

Raptor code model

In order to model Raptor coding, we employed the following statistical model (Luby et al., 2007):

$$\begin{aligned} P_f(m,k) &= 1 && \text{if} \quad m > k \\ &= 0.85 \times 0.567^{m-k} && \text{if} \quad m \geq k \end{aligned} \tag{2}$$

where $P_f(m,k)$ is the decode failure probability of the code with k source symbols if m symbols have been successfully received (and 1 - P_f is naturally the success probability). Notice that the authors of Luby et al. (2007) remark and show that for k > 200 the model almost perfectly models the performance of the code. In the experiments reported in this Chapter, the symbol size was set to bytes within a packet. Clearly, if instead 200 packets are accumulated before the rateless decoder can be applied (or at least equation (2) is relevant) there is a penalty in start-up delay for the video stream and a cost in providing sufficient buffering at the MSs. In the simulations, the decision on whether a packet can be decoded was taken by comparing a Uniformly-distributed random variable's value with that of the probability given by (2) for k > 200. The Uniform distribution was chosen because there is no reason to suppose that a more specific distribution is more appropriate.

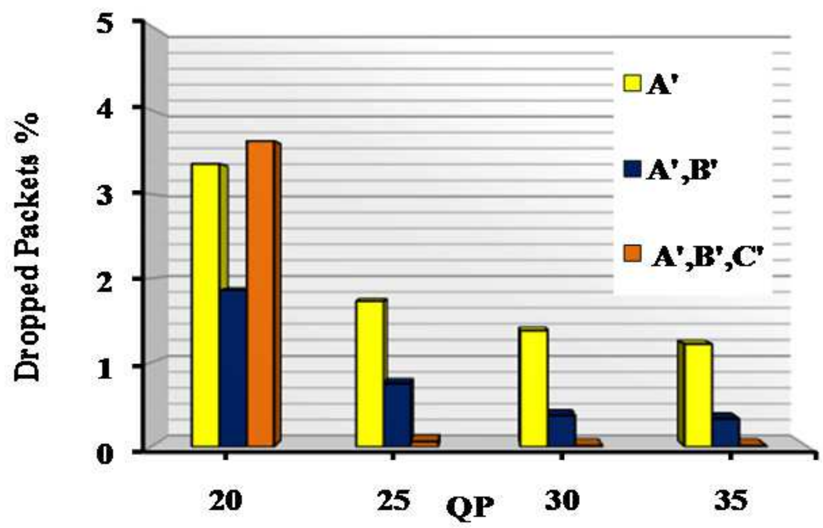
It is implied from (2) that if less than k symbols (bytes) in the payload are successfully received then a further k - m + e redundant bytes can be sent to reduce the risk of failure. In the evaluation tests, e was set to four, resulting in a risk of failure of 8.7 % in reconstructing the original packet if the additional redundant data successfully arrives. This reduced risk arises because of the exponential decay of the risk that is evident from equation (2) and that gives rise to Raptor code's low error probability floor.

Test video sequence

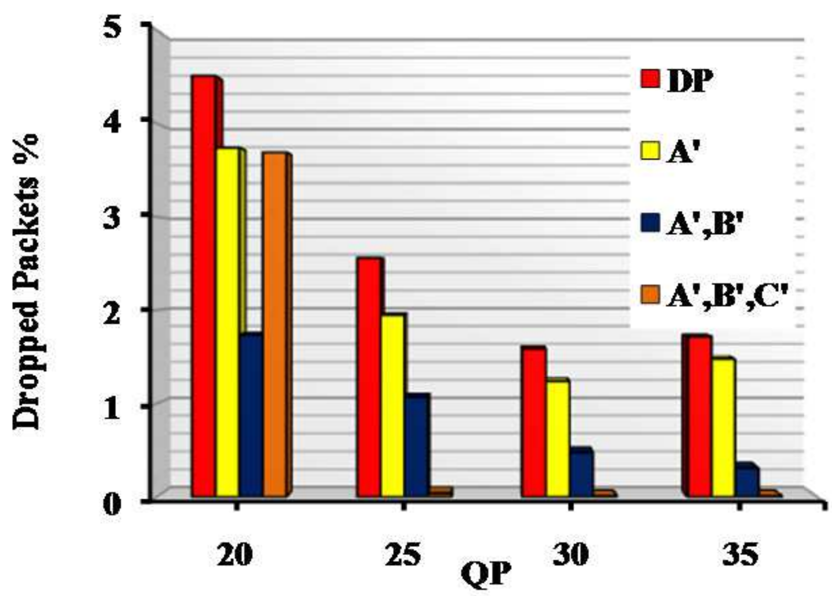
The test sequence was Paris, which is a studio scene with two upper body images of presenters and moderate motion. The background is of moderate to high spatial complexity. The sequences was variable bitrate encoded at Common Intermediate Format (CIF) (352 × 288 pixel/picture), with a Group of Pictures (GOP) structure of IPPP….. at 30 Hz, i.e. one initial Instantaneous Decoder Refresh (IDR)-picture followed by all predictive P-pictures. This structure removes the coding complexity of bi-predictive B-pictures at a cost in increased bit rate. Similarly, in H.264/AVC's Baseline profile, B-pictures are not supported to reduce complexity at the decoder of a mobile device. As a GOP structure of IPPP.... was employed, it is necessary to protect against temporal error propagation in the event of inter-coded P-picture slices being lost. To ensure higher quality video, 5% intra-coded MBs (randomly placed) (Stockhammer & Zia, 2007) were included in each frame (apart for the first IDR-picture) to act as anchor points in the event of slice loss. The JM 14.2 version of the H.264/AVC codec software was utilized, according to reported packet loss from the simulator, to assess the objective video quality (PSNR) relative to the input YUV raw video. Lost partition-C carrying packets were compensated for by error concealment at the decoder using the MVs in partition-A to predict the missing MB.

3.3 Evaluation results

Figure 3 shows the effect of the various schemes on packet drops when streaming Paris. 'Data-partition' in the Figure legend refers to sending no redundant packets. 'Duplicate X' refers to sending duplicate packets containing data-partitions of partition type(s) X, in addition to the data-partition packets. The proposed redundant schemes were also assessed for the presence of CIP or its absence. From Figure 3, the larger packet drop rates at quantization parameter (QP) = 20 will have a significant effect on the video quality. However, the packet size changes with and without CIP have little effect on the packet drop rate.



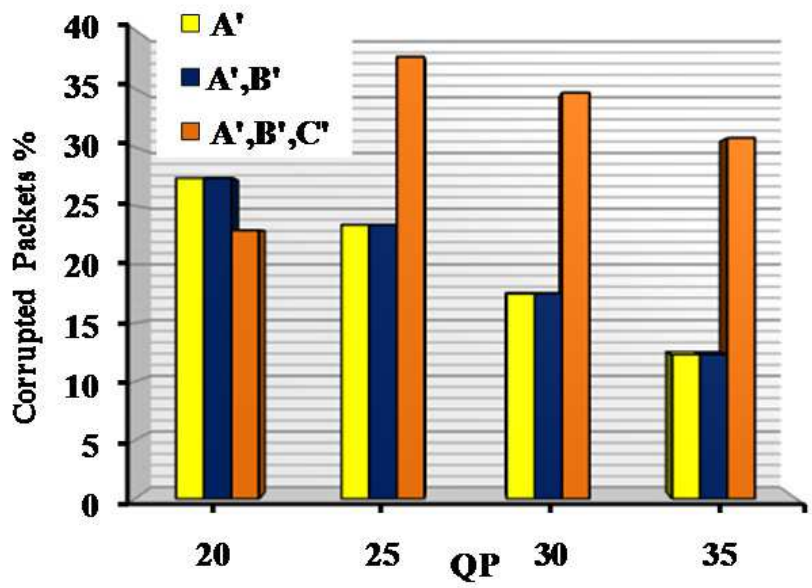
(a)



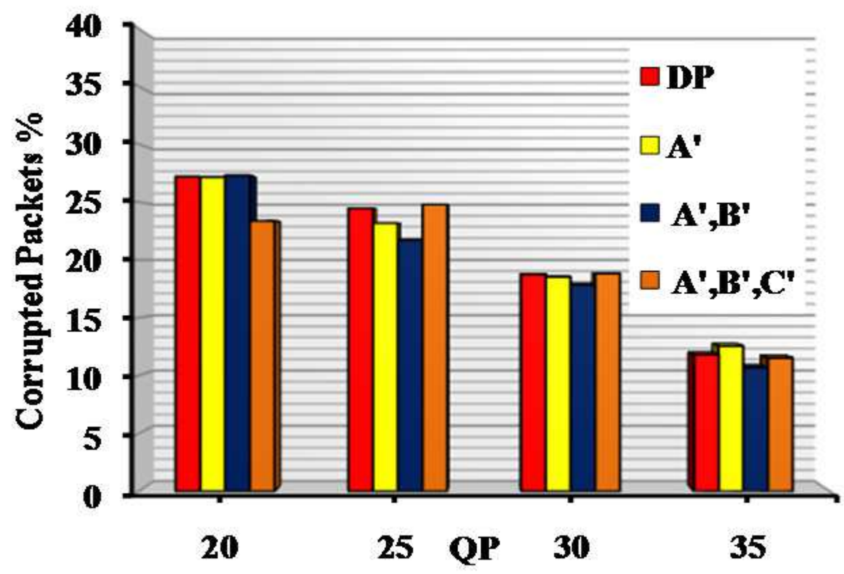
(b)

Fig. 3. *Paris* sequence protection schemes packet drops, (a) with and (b) without CIP.
A` = duplicate partition-A; A`,B` = duplicate partitions A and B; A`, B`, C` = duplicate partitions A`, B`, and C`; DP = data-partitioning without duplication.

Figure 4 shows the pattern of corrupted packet losses arising from simulated fast fading. There is actually an increase in the percentage of packets corrupted if a completely duplicate stream is sent (partitions A, B, and C), though this percentage is taken from corrupted original and redundant packets. However, the effect of the corrupted packets on video quality only occurs if a packet cannot be reconstructed after application of the adaptive retransmission scheme.



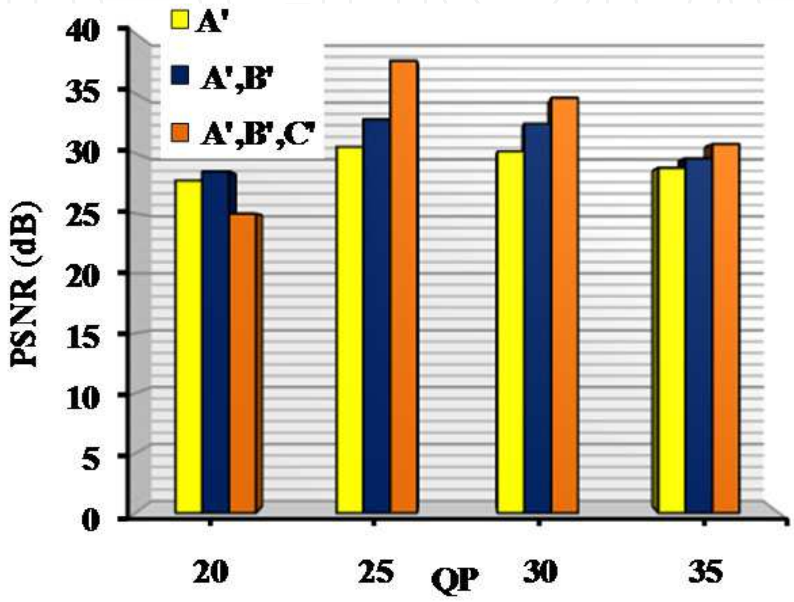
(a)



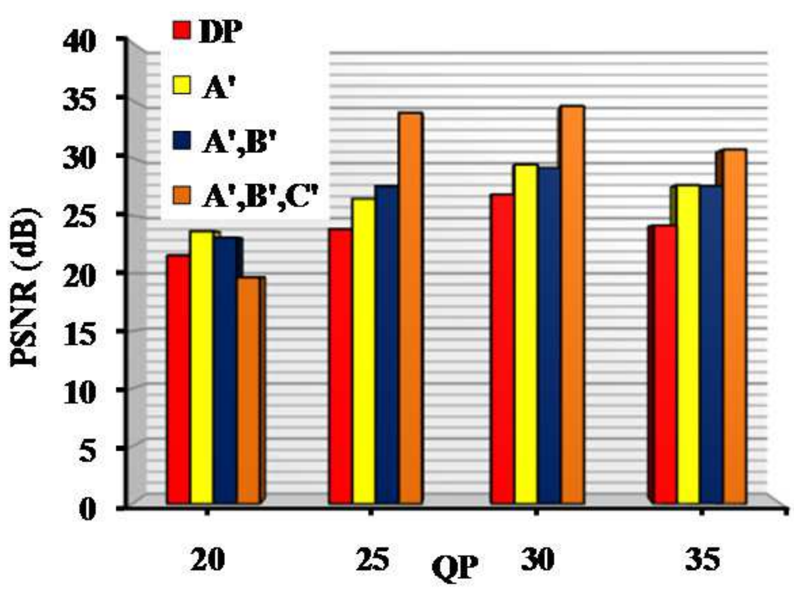
(b)

Fig. 4. *Paris* sequence protection schemes corrupted packets, (a) with and (b) without CIP. A\` = duplicate partition-A; A\`, B\` = duplicate partitions A and B; A\`, B\`, C\` = duplicate partitions A\`, B\`, and C\`; DP = data-partitioning without duplication.

Examining Figure 5 for the resulting objective video quality, one sees that data partitioning with channel coding, when used without duplication, is insufficient to bring the video quality to above 31 dB that is to a good quality. PSNRs above 25 dB, we rate as of fair quality (depending on content and coding complexity). However, it is important to note that sending duplicate partition-A packets alone (without duplicate packets from other partitions) is also insufficient to raise the video quality to a good rating (above 31 dB). Therefore, to raise the video quality to a good level (above 31 dB) requires not only the application of the adaptive rateless channel-coding scheme but also the sending of duplicate data streams with duplication of more than just partition-A packets.



(a)



(b)

Fig. 5. *Paris* sequence protection schemes video quality (PSNR), (a) with and (b) without CIP. A` = duplicate partition-A; A`, B` = duplicate partitions A and B; A`, B`, C` = duplicate partitions A`, B`, and C`; DP = data-partitioning without duplication.

The impact of corrupted packets, given the inclusion of retransmitted extra redundant data, is largely seen in additional delay. There is an approximate doubling in per- packet delay between the total end-to-end delay for corrupted packets, about 20 ms with CIP and 17 ms without, and normal packet end-to-end delay. Normal packets do not, of course, experience the additional delay of a further retransmission prior to reconstruction at the decoder. Nevertheless, the delays remain in the tens of millisecond range, except for when QP = 20, when end-to-end delay for the scheme with a complete duplicate stream exceptionally is as high as 130 ms. It must be recalled that, for the duplicate stream schemes, there is up to twice the number of packets being sent. This type of delay range is acceptable even for interactive applications, but may contribute to additional delay if it forms part of a longer network path.

4. Concluding remarks

IEEE 802.16 and more narrowly the WiMAX Forum's simplification of the standards are well suited to video streaming but some form of application layer error protection will be necessary, of the type presented in this Chapter's case study. For severe channel conditions combined with traffic congestion, not only does forward error correction seem a necessary overhead, together with source-coded error resilience, but additional duplication of some part of the encoded bit-stream may be advisable. In the case study, data partitioning had the dual role of providing a way to reduce packet sizes (MPDUs) and a way to scale layer duplication. However, alternative schemes exist such as the MPEG-Pro COP #3 (Rosenberg & Schulzrinne) IP/UDP/RTP packet interleaving scheme which includes FEC as separate packets, and it is worth considering how application layer packet interleaving could be included in the presented scheme, though at a cost in increased latency. Such schemes have the advantage that they can be applied to multicast as well as unicast delivery, as there is no requirement for repair packets. However, the feedback implosion at a remote multicast server that results from repair packet requests from multiple video receivers can be avoided in the Chapter's scheme as the single request for extra 'piggybacked' redundant data can be turned off. This will require a determination of what level of adaptive FEC is necessary to support multicast delivery without repair packets. All the same in the Internet TV version of IPTV, multicast from a remote server prior to reaching the WiMAX access network is unlikely. This is because the Internet Group Management Protocol (IGMP) should be turned on at routers to support multicast, which is difficult to ensure.

5. References

Afzal, J.; Stockhammer, T.; Gasiba, T. & Xu, W. (2006). Video Streaming Over MBMS: A System Design Approach, *Journal of Multimedia*, Vol. 1, No. 5, pp. 25-35.

Ahmandi, S. (2011). *Mobile WiMAX: A Systems Approach to Understanding IEEE 802.16m Radio Access Technology,* Academic Press, ISBN978-0-12-374964-2, Elsevier, Amsterdam.

Andrews, J. G.; Ghosh, A. & Muhamed, R. (2007). *Fundamentals of WiMAX: Understanding Broadband Wireless Networking*, Prentice Hall, ISBN 0-13-222552-2, Upper Saddle River, NJ.

Castura, J. & Mao, Y. (2006). Rateless Coding over Fading Channels, *IEEE Communications Letters*, Vol. 10, No. 1, pp. 46-48.

Chang, B.-J.; & Chou, C.-M. (2007). Cross-layer Based Delay Constraint Adaptive Polling for High Density Subscribers in IEEE 802.16 WiMAX Networks, *Wireless Personal Communications*, Vol. 46, No. 3, pp. 285-304.

Chen, Y.-W. & Hsieh, F.-Y. (2007). A Cross-layer Design for Handoff in 802.16e Network with IPv6 Mobility, *IEEE Wireless Communications and Networking Conference*, pp. 3844-3849.

Cicconnetti, C.; Lenzini, L.; Mingozzi, E.; and Eklund, C. (2006). Quality of Service Support in IEEE 802.16 Networks, *IEEE Network*, Vol. 20, No. 2, pp. 50-55.

Degrande, N.; Laevens, K. & Vleeschauwer, D. De (2008). Increasing the User Perceived Quality for IPTV Services, *IEEE Communications Magazine*, Vol. 46, No. 2, pp. 94-100.

Dhondt, Y.; Mys, S.; Vermeirsch, K. & Walle, R. Van de (2007). Constrained Inter Prediction: Removing Dependencies between Different Data Partitions, *Proceedings of Advanced Concepts for Intelligent Visual Systems*, pp. 720-731.

Ekstrom, K. et al. (2006). Technical Solutions for the 3G Long-Term Evolution, IEEE Communications Magazine, Vol. 44, No. 3, pp. 38-45.

Lambert, P.; Neve, W. De; Dhondt, Y. & Wall, R. Van de (2005). Flexible Macroblock Ordering in H.264/AVC, *Journal of Visual Communication and Image Representation*, Vol. 17, No. 2, pp. 358-375.

Goldsmith, A. (2005) *Wireless communications*, Cambridge University Press, UK.

IEEE 802.16e-2005 (2005). IEEE Standard for Local and Metropolitan Area Networks. Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems.

IEEE 802.21-2008 (2008) 802.21 WG IEEE standard for local and metropolitan area networks, media independent handover services.

Issa, O.; Li, W.; & Liu, H. (2010). Performance Evaluation of TV over Broadband Wireless Access Networks, *IEEE Transactions on Broadcasting*, Vol. 56, No. 2, pp. 201–210.

Jain, R.; So-In, C. & Al-Tamimi, A.-K (2008) System-level Modeling of IEEE 802.16e Mobile WiMAX networks, *IEEE Wireless Communications*, Vol. 15, No. 5, pp. 73–79.

Jiao, Ch.; Schwiebert, L. & Xu, B. (2002). On Modeling the Packet Error Statistics in Bursty Channels, *IEEE Conference on Local Computer Networks*, pp. 534- 541.

Juan, H.-H.; Huang, H.-C.; Huang, C. & Chiang, T. (2009). Cross-layer Mobile WiMAX MAC Designs for the H.264/SVC Scalable Video Coding, *Wireless Networks*, Vol. 16, No. 1, pp. 113-123.

Kawadia, V. & Kumar, P.R. (2003). A Cautionary Perspective on Cross layer Design, *IEEE Wireless Communications Magazine*, Vol. 12, No. 1, pp. 3-11.

Kuhran, M.S.; Gür, G.; Togcu, T. & Alagöz, F. (2010). Applications of the Cross-layer Paradigm for Improving the Performance of WiMAX. *IEEE Wireless Communications*, Vol. 17, No. 3, pp. 86-95.

Kumar, A. (2008). *Mobile Broadcasting with WiMAX: Principles, Technology, and Applications*, Elsevier, ISBN 978-0-240-81040-9, Amsterdam.

Liang, Y.J.; Apostolopoulos, J.G. & Girod, B. (2008). Analysis of Packet Loss for Compressed Video: Effect of Burst Losses and Correlation Between Error Frames, *IEEE Transactions on Circuits and Systems for Video Technology*, Vol. 18, No. 7, pp. 861-874.

Lee, J. M.; Park, H.-J.; Choi, S. G.; & Choi, J.K. (2009). Adaptive Hybrid Transmission Mechanism for On-demand Mobile IPTV over WiMAX, *IEEE Transactions on Broadcasting*, Vol. 55, No. 2, pp. 468-477.

Luby, M. (2002) LT codes, *34rd Annual IEEE Symposium on the Foundations of Computer Science*, pp. 271–280.

Luby, M.; Gasiba, T.; Stockhammer, T. & Watson, M. (2007). Reliable Multimedia Download Delivery in Cellular Broadcast Networks, *IEEE Transactions on Broadcasting*, Vol. 53, No. 1, pp. 235-246.

MacKay, D.J.C. (2005). Fountain Codes, *IEE Proceedings: Communications*, Vol. 152, No. 6, pp. 1062–1068.

Neves, P.; Sargneto, S.; Pentikousis, K. & Fontes, F. (2009) WiMAX Cross-layer System for Next Generation Heterogeneous Environments, pp. 357-376, In *WiMAX, New Developments*, Dalal, U.D.D. & Kosta, Y.P. (eds.), In-Tech, ISBN 978-953-7619-53-4, Vienna, Austria.

Nuaymi, L. (2007) *WiMAX: Technology for Broadband Wireless Access*, Wiley & Sons, ISBN 978-0-470-02808-7, Chichester, UK.

Oyman, O.; Foerster, J.; Tcha, Y.-j.; & Lee, S.-C. (2010). Towards Enhanced Mobile Video Services over WiMAX and LTE, *IEEE Communications Magazine*, Vol. 48, No. 8, pp. 68–76.

Palanki, R. & Yedidai, J. (2004). Rateless Codes on Noisy Channels, *International Symposium on Information Theory*, p. 37.

Radulovic, J.; Wang, Y.-K.; Wenger, S.; Hallapuro, A.; Hannuksela, M.H. & Frossard, P. (2007). Multiple Description H.264 Video Coding with Redundant Pictures, *International Workshop on Mobile Video*, pp. 37-42.

Rosenberg, J. & Schulzrinne, H. (1999). An RTP Payload Format for Generic Forward Error Correction, Internet Engineering Task Force, RFC 2733.

Schaar, M. van der & Shankar, N.S. (2005). Cross-layer Wireless Multimedia Transmission: Challenges, Principles, and New Paradigms, *IEEE Transactions on Wireless Communications*, Vol. 12, No. 4, pp. 50–58.

She, J.; Yu, X.; Po, P.-H.; & Yang, E.H. (2009) A Cross-layer Design Framework for Robust IPTV Services over IEEE 802.16 Network, *IEEE Journal of Selected Areas in Communications*, Vol. 29, No. 2, pp. 235-245.

Shokorallahi, A. (2006) Raptor codes, *IEEE Transactions on Information Theory*, Vol. 52, No. 6, pp. 2551-2567.

So-In, C.; Jain, R. & Tamini, A.-K. (2010). Capacity Evaluation of IEEE 802.16e WiMAX, *Journal of Computer Systems, Networks, and Communications*, Vol. 2010 [online], 12 pages.

Srivastava, V. & Motani, M. (2005). Cross-layer Design: A Survey and the Road Ahead, *IEEE Communications Magazine*, Vol. 43, No. 12, pp. 712-721.

Stockhammer, T. & Bystrom, M. (2004). H.264/AVC Data Partitioning for Mobile Video Communication, *Proceedings of IEEE International Conference on Image Processing*, pp. 545-548.

Stockhammer, T. & Zia, W. (2007). Error-resilient Coding and Decoding Strategies for Video Communication, pp. 13-58, In Schaar, M. van der & Chou, P.A. (eds.) *Multimedia in IP and Wireless Networks,* Academic Press, ISBN 978-0-12-088480, Burlington, MA.

Stockhammer, T.; Luby, M. & Watson, M. (2008). Application Layer FEC in IPTV Services, *IEEE Communications Magazine*, Vol. 46, No. 5, pp. 94-101.

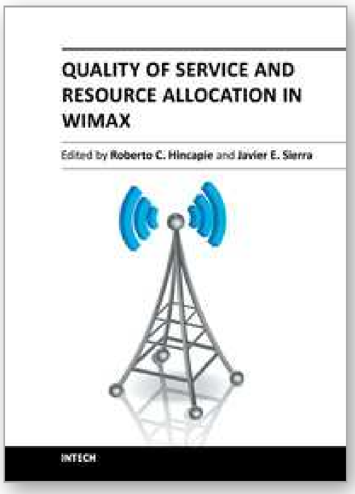
Uilecan, I.V.; Zhou, C. & Atkin, G.E. (2007). Framework for Delivering IPTV Services over WiMAX Wireless Networks, *IEEE International Conference on Electronics/Information Technology*, pp. 470–475.

Wang, H. & Chang, P. (1996). On Verifying the First-order Markovian Assumption for a Rayleigh Fading Channel Model, *IEEE Transactions on Vehicular Technology*, Vol. 45, No. 2, pp. 353–357.

Wiegand, T.; Sullivan, G.J.; Bjontegaard, G. & Luthra, A. (2003). Overview of the H.264/AVC Video Coding Standard, *IEEE Transactions on Circuits and Systems for Video Technology*, Vol. 13, No. 7, pp. 560-576.

Wenger, S. (2003). H.264/AVC over IP, *IEEE Transactions for Circuits and Systems for Video Technol*ogy, Vol. 13, No. 7, pp. 645-655.

Zhu, C.; Wang, Y.K.; Hannuksela, M. & Li, H. (2006). Error Resilient Video Coding Using Redundant Pictures, *IEEE International Conference on Image Processing*, pp. 801–804.



Quality of Service and Resource Allocation in WiMAX

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This book has been prepared to present state of the art on WiMAX Technology. It has been constructed with the support of many researchers around the world, working on resource allocation, quality of service and WiMAX applications. Such many different works on WiMAX, show the great worldwide importance of WiMAX as a wireless broadband access technology. This book is intended for readers interested in resource allocation and quality of service in wireless environments, which is known to be a complex problem. All chapters include both theoretical and technical information, which provides an in depth review of the most recent advances in the field for engineers and researchers, and other readers interested in WiMAX.

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