Broadcast/Multicast

MPEG-2 Video over Broadband Fixed Wireless Access Networks

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Abstract
With the emergence of broadband fixed wireless access networks, there is an increasing interest in providing broadband video services over outdoor wireless networks. In this article we investigate some fundamental issues related to the broadcasting or multicasting of CBR MPEG-2 videos over fixed wireless channels in B-FWANs using FEC strategies. In B-FWANs, high-frequency wireless channels are used and a direct or indirect LOS propagation path is usually required between a transmitter and a receiver. The wireless channel is modeled by a K factor Rician fading model instead of a Rayleigh fading model. The unique characteristics of the physical channel require special consideration at the system design level. In order to evaluate the overall system performance properly, a set of parameters for objective video quality assessment is introduced and used in our simulation studies, including a definition of the objective grade point value, the number of reconstructed frames, and the conventional peak signal-to-noise ratio value. The feasibility of cell interleaving is also addressed. MPEG-2 control information (i.e., the control block) plays a critical role in the decoding process and can influence the reconstructed video quality dramatically; special consideration and excess protection should be given to this information. In this article the concept of a new FEC strategy, header redundancy FEC, is introduced to address this issue. In HRFEC, selected important (high-priority) MPEG-2 control blocks (such as sequence_header, sequence_extension, and picture_header and corresponding extensions) are replicated before transmission, and duplicate copies are transmitted over the wireless link. Our results indicate that HRFEC is a simple, flexible, and effective error control strategy for broadcasting or multicasting MPEG-2 videos over error-prone and time-varying wireless channels.

During the past years, there has been an increasing demand for broadband services (video, fast Internet access, etc.) to the home or office. In addition, the 1996 Telecommunication Deregulation Act allows different service providers (telephone, television, CATV, and new entrants) to compete in each other's territory to provide broadband services to potential residential and business customers. It has propelled the need to find fast deployable and cost effective solutions to enable market penetration as soon as possible.

Current competing technologies in the wireline world are digital subscriber loop (xDSL), fiber to the curb (FTTC), cable modem over hybrid fiber cable (HFC), and, ultimately, fiber to the home (FTTH). In the wireless arena, broadband wireless access networks and satellite networks are two promising technologies as a result of efforts in the commercialization of technologies and infrastructures which were originally limited to military purposes, and progress in the semiconductor industry and digital technologies during the past decades [1]. For most of the practical and marketable broadband services, such as digital broadcast TV, tele-education, video on demand (VoD), and home shopping, mobility is not a top-priority consideration. For that reason, fixed wireless access networks (FWANs), especially broadband FWANs (B-FWANs), have attracted significant attention recently.

FWANs use wireless access loops connecting a fixed customer premises or terminal to the broadband network. They
provide an effective solution for new competitors to capture the local access market because of their fast deployment capability, cost-effective infrastructure, and low system maintenance cost. FWANs can also function as a redundant backup system for current wireline networks in case of emergency or natural disasters. Currently, there are two major B-FWAN candidates for broadband services to the home or office: local multipoint distribution/communication system (LMDS/LMCS) and multichannel multipoint distribution system (MMDS). Both LDMS and MMDS are proposed to distribute high-quality digital video and offer fast Internet access using low-power high-frequency or millimeter-wave (MW) radio signals over short to medium distances. LMDS and MMDS systems have a similar architecture, as shown in Fig. 1. The major differences between them include cell coverage size, one-way or two-way wireless services, and spectrum bands (LMDS uses the 28 GHz band, while MMDS uses 2.15–2.682 GHz). The cell radius of a typical MMDS cell is around 25–35 mi, while that of LMDS is around 1–3 mi, depending on terrain and antenna placement. LMDS can offer two-way wireless services for home users. MMDS requires terrestrial wired networks for upstream traffic (from users to the base stations, BSs) [2]. We describe LMDS in more detail in subsequent sections.

In the literature, most of the previous research on MPEG-2 video transmission over wireless channels focused on indoor wireless asynchronous transfer mode (WATM) systems [3, 4]. In this article we investigate some fundamental issues with regard to the transmission of MPEG-2 encoded video sequences over outdoor high-frequency fixed wireless channels used in B-FWANs, such as LMDS. The focus is on broadcast video services. In the remainder of this section we give a brief overview of LMDS, WATM, and MPEG-2 video. How we model the short-term characteristics of wireless channels is described in the second section. A brief summary of multipath wireless channel models is given, and the K factor Rician fading model for multipath line of sight (LOS) wireless channels of B-FWANs is described in detail. In LMDS systems, the wireless channel requires an LOS path. When combined with the multipath fading effect, the channel can be modeled as a Rician fading model, which is different from the conventionally used Rayleigh fading model. A simulation is used to study the channel performance. In the third section a description of the simulation system we used to study broadcast or multicast MPEG-2 video sequences over fixed wireless channels in B-FWANs is given. The system includes an MPEG-2 encoder/decoder, an error control encoder/decoder, a wireless channel simulator, and a system performance evaluation module. In the fourth section, simulation results of broadcast or multicast MPEG-2 video sequences over high-frequency fixed wireless channels under different channel conditions and with different error control schemes are presented. Issues regarding MPEG-2 video transmission over wireless channels are also addressed. In the fifth section, the concept of a header redundancy forward error correction (HRFEC) scheme is introduced for broadcast or multicasting MPEG-2 video over wireless channels. HRFEC allocates excess error protection to MPEG-2 control information due to the critical role MPEG-2 control information plays in the decoding process. Simulation results show that reconstructed video quality can be improved substantially by using HRFEC when compared to a scheme using pure FEC only. The sixth section summarizes our studies and concludes this article.

LMDS

The system architecture of LMDS is illustrated in Fig. 1, and the frequency spectrum allocated to LMDS is given in Table 1 [5]. LMDS uses the 28 GHz MW spectrum to transmit information to users located in service areas. Service areas are divided into cells (or sectors), very similar to cellular systems. Customers inside each cell communicate to one BS over wireless links. Several BSs are connected together to a central office or head-end via optical fibers at the OC-3 or OC-12 rate. A central office acts as the gateway to broadband backbone networks. The key features of LMDS include:

Broadband and Flexible Services in One Infrastructure — With 1–1.3 GHz spectrum bandwidth, nearly all kinds of broadband services can be supported in one single infrastructure. LMDS can also be scaled to fit the deployment requirements of small businesses, telecommuters, and residential (urban, suburban, and rural) services. Some potential services are briefly summarized in Table 2.

A True Two-Way System — Unlike MMDS, satellite networks, and other competing systems, LMDS offers true duplex services. This allows for effective two-way voice and high-speed data services to be deployed.

<table>
<thead>
<tr>
<th>License block</th>
<th>Spectrum band</th>
<th>Potential applications</th>
</tr>
</thead>
<tbody>
<tr>
<td>A (1,150Mhz)</td>
<td>(A-1) 27.5–28.35 GHz</td>
<td>License A is for asymmetric/symmetric services.</td>
</tr>
<tr>
<td></td>
<td>(A-2) 29.10–29.25 GHz</td>
<td>(A-2) is limited to BS to subscriber transmission.</td>
</tr>
<tr>
<td></td>
<td>(A-3) 31.075–31.225 GHz</td>
<td>(A-2) and (A-3) are for symmetric services.</td>
</tr>
<tr>
<td>B (150 Mhz)</td>
<td>(B-1) 31.000–31.075 GHz</td>
<td>Small business applications.</td>
</tr>
<tr>
<td></td>
<td>(B-2) 31.225–31.300 GHz</td>
<td></td>
</tr>
</tbody>
</table>

Table 1. LMDS spectrum allocation.
now and new interactive video services to be added in the future with little additional infrastructure cost. For example, with the increasing popularity of the Internet, more and more home or personal businesses require faster and broader uplink channels.

Fast Deployment and a Cost-Effective Solution — Regional telecommunications companies find it difficult to deploy cable and fiber systems in certain areas where installing underground infrastructure is undesirable due to existing buildings, impractical due to terrain, and costly because of the infrastructure system itself. LMDS provides a cost-effective low-maintenance solution that can be rapidly deployed.

Digital Technology — Although digital wireless transceivers are more expensive, broadband services using digital wireless techniques will still be used because digital signals tolerate more noise, allow use of advanced coding, modulation, and error control techniques, and are less susceptible to corruption by rain and foliage.

The Primary Disadvantage — Besides the free space attenuation and multipath effects, one primary disadvantage of the wireless channels in LMDS is directly related to the properties of millimeter waves. Because millimeter waves can not penetrate vegetation and buildings, an LOS propagation path is required between a transmitter and a receiver. When a direct LOS path is not available, high reflective mirrors or virtual BSs can be used to achieve an (indirect) LOS path. The requirement for an LOS path also limits the service coverage in one cell. In addition, the rain effect is another problem. Water absorbs MWs, so rainfall introduces fading effects on signals.

<table>
<thead>
<tr>
<th>Broadband services</th>
<th>Downlink bandwidth</th>
<th>Uplink bandwidth</th>
</tr>
</thead>
<tbody>
<tr>
<td>Broadcast, multicast:</td>
<td>1.5–6 Mb/s</td>
<td>9.6–64 kb/s</td>
</tr>
<tr>
<td>Pay-per-view Digital TV</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Interactive services: Interactive video</td>
<td>64 kb/s–6 Mb/s</td>
<td>9.6–64 kb/s</td>
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<tr>
<td>Video on demand</td>
<td></td>
<td></td>
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<tr>
<td>Interactive games</td>
<td></td>
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<tr>
<td>Home shopping</td>
<td></td>
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<tr>
<td>Telemedicine</td>
<td></td>
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<tr>
<td>Tele-education</td>
<td></td>
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<tr>
<td>Information retrieval</td>
<td></td>
<td></td>
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<tr>
<td>Internet access: Internet browsing</td>
<td>14.4 kb/s–10 Mb/s</td>
<td>14.4–128 kb/s</td>
</tr>
<tr>
<td>Desktop multimedia</td>
<td></td>
<td></td>
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<tr>
<td>Software download</td>
<td></td>
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<tr>
<td>Electronic banking</td>
<td></td>
<td></td>
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<tr>
<td>Symmetric service: Work at home</td>
<td>9.6 kb/s–2 Mb/s</td>
<td>9.6 kb/s–2 Mb/s</td>
</tr>
<tr>
<td>Videoconferences</td>
<td></td>
<td></td>
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<tr>
<td>Video telephony</td>
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<tr>
<td>Voice telephony</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Small business/home: Internet homepage</td>
<td>9.6–384 kb/s</td>
<td>64 kb/s–1.5 Mb/s</td>
</tr>
<tr>
<td>Internet server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Internet download</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Others: Fax, e-mail, file transfer</td>
<td>9.6–64 kb/s</td>
<td>9.6–64 kb/s</td>
</tr>
</tbody>
</table>

Table 2. Potential broadband services over LMDS.

Small Cell Size — Because LOS propagation paths are required from the transmitters to the receivers, LMDS uses very small cell configuration with a radius less than 5 km. Frequency reuse can be adopted to increase system capacity. The spectrum can be reused more efficiently by using sector antennas and beamforming technologies.

There has been an increasing demand for the services and functions provided by LMDS. B-FWANs, especially LMD, systems are well positioned to play a significant role in the local access market to provide broadband services to residential and business customers.

Wireless ATM and B-FWANs

ATM technology uses fixed-size (53-byte) cells to transport multimedia services over bandwidth-rich, low-bit-error-rate (BER) media. WATM was proposed to combine the advantages of both ATM and wireless technologies to achieve ubiquitous ATM connectivity. It is a promising technology that can support broadband services over wireless links, and has been the focus of the ATM Forum in recent years [3, 6, 7]. B-FWANs are proposed to provide broadband and multimedia services to customers. WATM over B-FWANs will make that possible.

However, for WATM to succeed, it must be able to seamlessly support standard ATM services with quality comparable to that on the wired network. Contrary to the fundamental assumptions of ATM, wireless links are time-varying with higher BER and limited bandwidth. Before WATM becomes a reality, many issues (e.g., error control, flow control, and QoS control) must be addressed in the design and implementation of B-FWANs. The bandwidth allocation is a crucial element of the efficient use of the radio frequency spectrum and is a key factor in the design of the WATM layer, which consists of media access control (MAC) and data link control (DLC).

MPEG Video and B-FWANs

The Motion Picture Experts Group (MPEG), a standard recommended by the International Organization for Standards/International Electrotechnical Commission/International Telecommunication Union (ISO/IEC/ITU) [9], is the most widely used broadband video compression scheme. The MPEG video standard specifies the syntax and semantics of an encoded video bitstream for decoding which includes parameter sets such as bit rates, picture sizes and pixel resolutions, and the decoding parameters to reconstruct the pictures. MPEG-1 is standardized at a maximum bit rate of 1.866 Mb/s for CD-based applications and does not support full interlaced video. MPEG-2 video is an extension of MPEG-1 and supports fully interlaced video. It can provide studio-quality pictures at bit rates between 2 and 9 Mb/s. CATV companies are using MPEG-2 to compress and decompress video for distribution and broadcasting. Direct broadcast satellite (DBS) and high-definition TV (HDTV) use MPEG-2 video for direct broadcast. Digital video disk (DVD) has also defined in its specification...
that the video decoding standard is to be MPEG. Most video-on-demand (VoD) systems proposed are planning to use the MPEG-2 standard.

The MPEG standard relies on two basic techniques: intraframe discrete cosine transform (DCT) for the reduction of the spatial redundancy, and interframe motion compensation for the reduction of temporal redundancy. Syntactically, each encoded MPEG-2 video sequence starts with a sequence header, which is optionally followed by several extensions before the appearance of the picture headers, which indicate the beginning of each coded frame. At the end of each sequence, a special sequence_end_code (0x00001087) terminates the current sequence. The whole video sequence can be divided into groups of pictures, and each group of pictures consists of one or more frames which use three different encoding types. Intra-coded frames (I-frames) are encoded independently (spatial compression only). Predictive-coded frames (P-frames) are coded using prediction obtained from the temporarily preceding I- or P-frame, or using intra-coding without prediction if the difference between the two frames is too large. Bidirectionally coded frames (B-frames) are coded using predictions from preceding and/or upcoming I- or P-frames in the video sequence. Frames are output from an MPEG-2 encoder in the order in which they are going to be processed by the decoder, each one different from the order in which they are displayed due to the temporal coding scheme. Although theoretically I-frames can be generated only when a scene change happens in the video sequence, the interval between two I-frames (N) is chosen to be around 12 to 16 in practice. Intraframe coding is performed on the block level. Each block is transformed from the spatial domain to the frequency domain by DCT. The frequency coefficients are then quantized with a quantization matrix. The quantization coefficients are organized in a zigzag order before variable-length encoding is applied. Motion compensation is applied at the macroblock level, and the compressed stream consists of motion vectors and predicted errors. Motion matching techniques are used to obtain the motion vector.

MPEG-2 video can be encoded to generate variable or constant bit rate (CBR) traffic. For some applications such as VoD, the number of transmitted bits per unit time on the channel varies depending on picture content and resource constraints. This is called variable bit rate (VBR) coding. VBR video has nearly constant quality and higher encoding gains. However, for some other applications such as broadcast services, it is necessary to transmit video information at a fixed bit rate. That can either mean buffering the VBR streams and transmitting at a constant bit rate, or CBR encoding, where the number of transmitted bits per unit time is constant on the channel. For CBR, the coded picture quality may vary depending on its content.

Broadband video services, such as digital TV and VoD, make up a major portion of services provided over B-FWANs (Table 2). To save on storage space and channel resources, video programs are usually compressed before transmission. Most likely, MPEG will also be adopted for LMDS and other B-FWAN systems.

Modeling of Wireless Channels

Modeling of wireless channels plays an important role in simulating and evaluating a wireless communication system. Generally, the process of channel modeling can be divided into two phases: long-term and short-term modeling. Long-term modeling is used if we want to catch the long-term properties of a channel, such as the transition between channel states, while short-term modeling is aimed at capturing the short-term characteristics of the wireless channels in a particular state. For example, if we assume that there are two states — a GOOD state and a BAD state for a bursty wireless channel — long-term modeling will try to model the transition between the GOOD and BAD states by using the hidden Markov model (HMM) [10-12], and short-term modeling tries to model the channel within the BAD or GOOD state by using a multipath fading model.

In this article we focus on the short-term modeling of wireless channels only insofar as we want to observe the static behavior of the channel and its impact on MPEG-2 video services. We are currently investigating the long-term behavior of the channel in conjunction with error control schemes such as automatic repeat request (ARQ). For some preliminary results the reader is referred to [9]. A brief summary of multipath wireless channel models is given first. Then the K factor Rician fading model for multipath LOS wireless channels of B-FWANs is described in detail. The channel performance via simulation is provided subsequently.

Multipath Fading Wireless Channels

For most wireless channels, there is usually more than one path from the transmitter to the receiver. This is known as the multipath effect or multipath fading, due to atmospheric reflection, multiple scattering or reflection from other objects, the presence of a great number of obstructions, variations of the medium's dielectric constant, and so on. Different paths may experience different time delay and attenuation. The attenuation and delay may be time-varying. All these effects result in fluctuations in the received signal level. There has been some literature dealing with the modeling of multipath wireless channels [14-16]. A comprehensive summary of different multipath wireless channel models under different conditions is available in [14].

The simplest model for discrete multipath channels can have the form

$$y(t) = \sum_{n} a_n(t) \times s(t - T_n(t))$$

(1)

where $s(t)$ is the input signal, and $a_n(t)$ and $T_n(t)$ are the attenuation factor and the delay for the nth path, respectively. This model is called the tapped delay line (TDL) model, since the channel is modeled as a tapped delay line with time-varying coefficients. For indoor wireless channels, TDL models with time-varying tap gains work well. $n$, $a_n(t)$, and $T_n(t)$ can be estimated from empirical measurements made at different frequencies of interest for different types of buildings and floor plans.

However, for outdoor wireless channels TDL models are not quite satisfactory. This is because a TDL model requires identification of the individual paths, which is impractical or impossible when the channel changes randomly, and the number of paths $n$ also changes as a function of time. In fact, well-designed statistical models are far more useful for outdoor wireless system simulation and performance evaluation.

Mobile radio wireless channels usually use Rayleigh fading models [17], because the amplitude of the received signal at a mobile has a Rayleigh distribution, and the phase has a uniform distribution. An implementation proposed by Jakes [15], known as Jakes' model, has been widely used to simulate Rayleigh fading channels. However, the Rayleigh fading model holds only in the case where there are a large number of indirect paths that dominate over the direct path. In some
In the literature some research has been done to study the propagation properties of MW links [18-24]. For MW wireless channels in B-FWANs, such as an LMDS system, critical propagation issues are free space attenuation, attenuation by rain, vegetation, and buildings, signal depolarization, multipath, and coherent cell interference.

An initial study of an LMDS radio channel was conducted in suburban areas in Northglenn, Colorado, and San Jose, California, by the National Telecommunications and Information Administration (NTIA) [18]. The study showed that the most serious propagation impairment for LMDS in the two surveyed suburban areas was signal attenuation. Attenuation is caused by tree canopy extending above roof height, which blocks LOS propagation paths and causes signal attenuation, depolarization, and multipath. The primary mode of signal propagation through trees is considered to be diffraction. Diffraction combined with wind will cause signal variability. Multipath effects were also found which were caused by multiple scattering events instead of multiple reflec-
tions. To study the short-term variability of such an LOS wireless channel, the K factor Rician model can be used. In addition, a random noise model can be used to describe the randomness of channel errors.

In the K factor Rician model, the received signal power consists of the waves coming directly from the transmitter along the LOS path (the direct component, $E_d$) and a number of small waves scattered from adjacent houses, leaves, or other scattering objects (the scattered component, $E_s$). Since the scattered waves have random phase, the summation of the scattered signal waves will produce a complex Gaussian process whose amplitude is Rayleigh distributed. When the constant direct wave is included, the resultant amplitude should follow the Nakagami-Rica distribution [17]. In the K factor model, a tuple is used to describe the state of the channel ($K$, SNR), in which the ratio $K = E_d/E_s$ is the K factor, and SNR is the signal-to-noise ratio at the receiver. A special case is when $K \ll 1$; the K factor fading model then becomes a Rayleigh fading model.

The BER of the K factor Rician model under different ($K$, SNR) was studied via simulations, and the results are given in Fig. 2. A WATM infrastructure was assumed, and the central spectrum (28 GHz) of LMDS was used in the simulation. The radius of the service cell was chosen to be 5 km, and the simulated wireless channel had a bit rate $R$ of 3.4 Mbits.

When $K$ is less than 0 dB, the channel performance is close to that of a Rayleigh fading channel. When $K$ is greater than 20 dB, the channel can be treated as a direct path without fading. Generally, the K factor of a typical Rician fading channel is somewhere between these two extreme cases. In addition, the number of bits and bytes in error per cell was traced and the corresponding probability density functions (pdfs) were generated, as shown in Figs. 3, 4, and 5. (Note that only the cells with errors are used to generate these figures to emphasize the distribution.) For channels with the same SNR but different $K$, when $K$ increases, the pdfs of bits or bytes in error per cell decrease in distribution range and peak value. This results in the BER of the wireless channel decreasing. In addition, the average duration of each fade also decreases and the average interval between two deep fades increases.

A System Description of the Simulation

In Fig. 6 the block diagram of the simulation system to study broadcast or multicast MPEG-2 video sequences over fixed wireless channels in B-FWANs is illustrated. The wireless channel simulator models a time-division multiple access (TDMA) LMDS broadcast channel in the 28 GHz frequency band. An MPEG-2 encoder encodes input video sequences. The output MPEG-2 bitstreams are sent to the wireless channel simulator after error control encoding. The modulation scheme used is π/4-shifted differential quadrature phase shift keying (DQPSK). At the receiver an MPEG-2 software decoder decodes the accepted bitstreams after error correction, and then system performance evaluation is conducted. A detailed description of each function block is given in the following subsections.

**An MPEG-2 Encoder and Decoder**

The MPEG software encoder used in the simulations was developed in our research group according to ITU-T Recommendation H.262 [9]. The encoder compresses a raw video sequence in red, green, blue (RGB) or International Consultative Committee for Radio (CCIR) format into MPEG bitstreams. The encoder supports certain options in the MPEG-1/MPEG-2 standard, including CBR, VBR, and SNR scalability. Statistical data, such as the number of bits per frame, can be generated, and a user-friendly X-Motif graphics user interface (GUI) is provided for encoding control [25]. The input video sequence can also be optionally saved frame by frame in YCrCb format.

The interval between consecutive I-frames,
Figure 7. A trace of the number of bits per frame for a CBR (R = 2.8 Mbit/s) encoded Indiana Jones video sequence (in display order): a) 1000 frames; b) a segment of 180 frames.

Figures 7(a) and 7(b) show the number of bits per frame for a CBR (R = 2.8 Mbit/s) encoded Indiana Jones video sequence (in display order): a) 1000 frames; b) a segment of 180 frames.

In our system a modified version of the MPEG-2 decoder originally developed by the MPEG Software Simulation Group (MSSG) [23] was used. The block functions of the MPEG-2 decoder are illustrated in Fig. 8. The decoder can optionally output a decoded video sequence in YCrCb format frame by frame, so we can compare the original and reconstructed frame sequences to assess the reconstructed video quality.

**Error Control for Broadcast/Multicast Services**

Error control is necessary to transmit information over the error-prone and time-varying wireless channel correctly. Basically, there are two distinct classes of error control schemes: forward error correction (FEC) and ARQ. FEC employs error detection/correcting codes to combat errors by adding redundancy. ARQ combines errors by retransmitting the information over and over until it is correctly received or the retransmission limit is reached (truncated ARQ). Traditionally, FEC has been used for real-time applications such as video transmission because it guarantees constant throughput and delay. However, because of the overhead of redundant codes in FEC, which exists irrespective of the channel states, it is wasteful when the channel condition is good. ARQ is usually used for non-time-critical applications due to its delay variance. Although extensive research has been conducted on error control schemes and strategies for wireless networks as summarized in [27], it is still impractical and almost impossible to design an error control scheme which works optimally for different systems. A design and implementation of an error control strategy is a trade-off between cost and quality, and has to take into consideration channel characteristics, available system resources, and QoS requirements. Since our attention is on broadcast/multicast video services in this article, we focus on FEC strategies to combat channel errors.

The Bose-Chaudhuri-Hocquenghem (BCH) code is a widely used block code. If \( p \) is a prime number and \( q \) is any power of \( p \), there exist codes with symbols from the Galois field, \( GF(q) \). These codes are called \( q \)-ary codes. For any choice of positive integer \( s \) and \( t \), there exists a \( q \)-ary BCH code of length \( n = q^s - 1 \) which is capable of correcting up to any combination of \( t \) or fewer symbol errors and requires no more than \( 2t \) parity check bits. One special subclass of \( q \)-ary BCH codes for which \( s = 1 \) is the Reed-Solomon (RS) codes. A \( t \) error-correcting RS code from \( GF(q) \) is referred to as \((n, k, t)\), where \( k \) is the information symbol length, and the minimum distance is \( d_{\text{min}} = 2t + 1 \) [28].

BCH and RS block coding schemes have many, but not all, of the desirable properties. BCH and RS codes have optimal distance properties, that is, provide optimal error correction capability given a fixed number of parity bits for specific codes. However, there are no RS codes which are capable of correcting all \( q \)-ary BCH codes for which \( s = 1 \) is the Reed-Solomon (RS) codes. A \( t \) error-correcting RS code from \( GF(q) \) is referred to as \((n, k, t)\), where \( k \) is the information symbol length, and the minimum distance is \( d_{\text{min}} = 2t + 1 \) [28].

**System Performance Evaluation**

The major part of overall system performance evaluation is video quality assessment. Reconstructed video quality can be assessed either subjectively or objectively. In subjective assessments, the perceived quality of various video clips, and subjective grade points (SGPs) are given by the viewers. For example, one set of grade points is in the range from 1 to 5 with

- 1 for very annoying
- 2 for annoying
- 3 for slightly annoying
- 4 for perceptible
- 5 for imperceptible [29]

In objective assessments, a set of parameters are calculated based on the video clips, and an objective grade point (OGP) is derived which should match that of subjective assessment closely.

Traditionally, SNR measurements have been used to estimate the relative quality of a reconstructed image compared to an original image in image processing. The basic idea is to compute a single number that reflects the quality of the reconstructed image. Reconstructed images with higher SNRs are judged better. For video sequence quality assessment, peak SNR (PSNR) has been used. Assume that an original frame \( O(i, j) \) contains \( N \times M \) pixels, and a reconstructed frame \( D(i, j) \) is obtained after decoding; then the PSNR is calculated as

\[
\text{PSNR} = 10 \log_{10} \left( \frac{255^2}{\text{RMSE}} \right)
\]

(2)

where

\[
\text{RMSE} = \frac{1}{N \times M} \sum_{i,j} [O(i,j) - D(i,j)]^2
\]

(3)

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Conventionally, the average value of the PSNR of all the frames is used for objective video quality assessment. However, SNR measures do not equate with human subjective perception accurately (i.e., higher measures do not always mean better quality). From the definition of PSNR, PSNR only captures a small portion of the video quality; the important temporal information of video sequences is omitted. Despite that, PSNR is still extensively used to estimate objective video quality because it is easy to compute. The actual value of PSNR is not meaningful, but the comparison between two values gives a measure of quality. The MPEG Committee used an informal threshold of 0.5 dB PSNR to decide whether to incorporate a coding optimization scheme because they believed an improvement of that magnitude would be visible.

Another objective video quality assessment scheme used in our simulation was proposed by the Institute for Telecommunication Sciences (ITS) for teleconference video sequences [29]. Although it was developed and tested for teleconferencing systems, the idea behind the system is quite general, so it can be modified for other types of motion video, such as MPEG. A brief summary of the video quality assessment scheme is given here.

Let be the estimated OGP, which is the result of a linear combination of n measurements $M_i$ which are generated as the result of comparing the original and decoded video sequences, and n + 1 constants $c_i$ which are determined through an optimization procedure so that s matches the OGP closely.

$$s = c_0 + \sum_{i=1}^{n} c_i M_i$$

$M_i$ is a measure of spatial distortion and is obtained from the spatial information features of the original and decoded video. The spatial distortion measured by $M_1$ includes both blurring and false edges. $M_1$ is calculated by

$$M_1 = \text{RMS}_{a} \left( 5.81 \times \frac{\text{SI}(O_a) - \text{SI}(D_a)}{\text{SI}(O_a)} \right)$$

$\text{SI}[\cdot]$ denotes the spatial information (SI) function, and $\text{RMS}_{a}$ represents the root mean square function over frames. $O_a$ and $D_a$ denote the nth frame of the original and decoded video sequence. $M_2$ and $M_3$ both measure temporal distortion between the original and decoded video sequences. Measurements of $M_2$ and $M_3$ are given by

$$M_2 = f_a(0.108 \times \text{MAX}(\text{TI}(O_a) - \text{TI}(D_a)), 0))$$

$$M_3 = \text{MAX}_{a} \left( 4.23 \times \text{LOG}_{10} \left( \frac{\text{TI}(O_a)}{\text{TI}(D_a)} \right) \right)$$

and

$$f_a(x_n) = \text{STD}_{a}(\text{CONV}(x_n, [-1, 2, -1]))$$

$\text{TI}[\cdot]$ is the temporal information function, CONV denotes the convolution operator, and $\text{STD}_{a}$ denotes a standard deviation over frames. The SI feature differentiates between pixels across space, while the TI feature differentiates between pixels across time.

$$\text{SI}(F_n) = \text{STD}_{\text{space}}(\text{Sobel}(F_n))$$

$$\text{TI}(F_n) = \text{STD}_{\text{space}}(F_n - F_{n-1})$$

The STD is the standard deviation operator over pixels in one frame, and $F_n$ is the nth frame of the video sequence. Sobel denotes the Sobel filter [30]. Since the Sobel filter is an edge enhancement filter, edge energy gained or lost in a frame will be reflected in $\text{SI}$ and $\text{TI}$, and sharpening quality. $\text{SI}$ generates higher values when $\text{SI}$ and $\text{TI}$ are not given a score of 5 for any video clip. This objective video quality assessment scheme was also used for system performance evaluation in our simulations.

In addition, another measurement which is helpful and complementary is the number of frames reconstructed by the decoder. To evaluate overall system performance, a combination of the three OGP's is used together or alternatively in our simulations. Both luminance and chrominance components of each frame are used in the calculation process. The simulation results, the difference between two measurements determines the relative system performance between them.

**Simulation Results**

**Broadcast/Multicast CBR MPEG-2 Video Transmission**

In this subsection CBR MPEG-2 video transmission over a TDMA-based K-factor-modeled high-frequency wireless channel is investigated to obtain a quantitative understanding of the fundamental relations among overall system performance, wireless channel capacity, and end-to-end video quality. A WATM infrastructure was assumed, and WATM cells which consist of header, payload, and error control fields were used to carry MPEG-2 video bitstreams. RS codes were used as FEC to combat channel errors. The received cells or reassembled blocks are dropped when the blocks are uncorrectable by the RS decoder, and 8s are stuffed in the positions before passing up to the MPEG-2 decoder. No error concealment procedure was applied at the receiver. In order to make the MPEG-2 software decoder work properly, an error recovery algorithm has to be applied after the RS decoder. Its primary function is to prevent generating critical MPEG control bits such as the Start_Code_Prefix (0x0000001) accidentally. For example, the error recovery algorithm involves parsing a portion of the bitstream with 0s until the

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next Start Code Prefix is correctly received if the beginning Start Code of this portion of the bitstream was lost. In order to minimize the effect of the position of errors on the results, the simulation was run several times with different start times.

The quality of the reconstructed videos improves when the channel BER drops as K or SNR is increased. In Fig. 9 the OGP results under different channel states with the same RS code (t = 3 per WATM payload) were compared. As we can see, when the BER of the channel is about or less than 10^-6, the channel can be treated as a lossless channel and the OGPs almost reach the value under ideal channel conditions.

In Fig. 10, the OGP results when different error correction codes (t = 1, 3, 5, 7 symbols/cell) are used is compared. The channel rate is fixed at R = 3.4 Mb/s. The corresponding encoded MPEG-2 video bit rates are 3.02, 2.81, 2.63, and 2.47 Mb/s when t = 1, 3, 5, 7, respectively. Error control is a trade-off between cost and quality. When the channel BER is high, more error protection (larger t) provides better quality services. However, when the BER is low, video quality will be degraded a little when excess error protection (also larger t) is used because larger t reduces the information rate. The comparison of the number of reconstructed frames for different simulations are also given in Fig. 11.

We would like to emphasize here one important point regarding MPEG-2 video transmission over wireless links. During the simulations, we noticed that the results could be dramatically different even when the channel states and BER were kept the same, and only the start time (t) was changed. Changing the start time makes errors happen at different positions in the bitstream. The reason is that not all the bits in an MPEG-2 bitstream have the same importance in the decoding process. In MPEG-2 bitstreams many MPEG control blocks (control information) are inserted so that the compressed video sequence can be decoded properly. If channel errors occur and corrupt that control information, the subsequent information is pretty much useless. For example, if a picture header is lost over the wireless channel, the subsequent encoded picture data would be dropped by the MPEG-2 decoder, so the quality of the reconstructed video will degrade dramatically. For illustra-

![Figure 9](image1.png) ![Figure 10](image2.png)

**Figure 9.** System performance results under different channel states (K, SNR) when the same error protection is used (t = 3).

**Figure 10.** System performance results for different error protection capabilities (t) (K = 7.5 dB).
tion one example is listed in Table 3, where NLF is the number of lost frames.

As we already know, temporal information plays an important role in the objective video quality assessment. $M_2$ measures the effect of temporally localized changes between the reconstructed and original frames on the video quality. The convolution function enhances these changes, and the STD function quantifies the spread of the changes. $M_3$ selects the video frames that have the largest added motion or changes by taking the maximal value of a function on T1. If a picture header is lost which usually causes the loss of the whole frame, $M_2$ and $M_3$ will increase and degrade the $s$. In the examples given in Table 3, the results of simulation 0 are the ideal values when the channel is errorless. The difference between simulations 1 and 1* is the start time; the channel state and error protection were the same. Only one frame was lost for simulation 1, but none for simulation 1*. As a result, the $M_2$ and $M_3$ of simulation 1 are much larger than those of 1*, and the difference between them accounts for a difference of $0.272 \times (\Delta M_2) + 0.356 \times (\Delta M_3) = 1.32$ points in $s$ approximately. These examples clearly show that loss of frames will degrade overall video quality, and special consideration has to be given to the MPEG control information to prevent losing sequences or frames because of lost header information.

An alternative approach for combating long bursts of errors is cell interleaving. This involves interleaving symbols from multiple cells before transmission onto the wireless channel. The number of cells that are interleaved is referred to as the interleaving depth, $d$. Interleaving decreases the burst of errors per cell; therefore, a cell has a larger

<table>
<thead>
<tr>
<th>Simu</th>
<th>$t_0$</th>
<th>$t$</th>
<th>$K_{(d)}$</th>
<th>SNR($\text{dB}$)</th>
<th>$s$</th>
<th>PSNR($\text{dB}$)</th>
<th>$M_1$</th>
<th>$M_2$</th>
<th>$M_3$</th>
<th>NLF</th>
</tr>
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<tbody>
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<td>0</td>
<td>0</td>
<td>4.688</td>
<td>0.072</td>
<td>0.037</td>
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<td>0</td>
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<td>0</td>
</tr>
<tr>
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<td>60.00</td>
<td>3</td>
<td>7.5</td>
<td>2.281</td>
<td>0.512</td>
<td>1.046</td>
<td>4.762</td>
<td>1</td>
<td></td>
<td>0</td>
</tr>
<tr>
<td>1*</td>
<td>40.00</td>
<td>3</td>
<td>7.5</td>
<td>3.968</td>
<td>0.137</td>
<td>0.067</td>
<td>1.815</td>
<td>0</td>
<td></td>
<td>0</td>
</tr>
</tbody>
</table>

Table 3. A closer analysis of the simulation results (CBR MPEG-2 at 2.8 Mbps).

Figure 11. The number of decoded frames: a) different $K$ ($t = 3$); b) different $t$ ($K = 7.5$ dB).

Figure 12. System performance results when cell interleaving is used ($t = 3$, $K = 7.5$ dB).
chance to be corrected by the same error correction capability $t$ after deinterleaving. Interleaving and per-cell FEC strategy becomes effective when $t \times d$ exceeds the average burst length. However, an extra delay of $2d$ cells is introduced due to the interleaving and deinterleaving processes which exists regardless of whether the errors are bursty or nonbursty, or there are no errors. Simulations were run to determine the feasibility of per-cell coding with interleaving. The proper choice of $d$ should take into consideration the FEC parameter $t$ and the channel state ($K$, SNR), which determines the average duration of error bursts. For example, in Fig. 12 the simulation results for different interleaving depths are shown when $t = 3$ and $K = 7.5$ dB. The results indicate that the improvement in performance can be observed when $d$ is greater than 8. As a side effect, cell interleaving to this depth introduces a constant excess delay of $2 \times d$ cells. For broadcast or multicast video services, a constant delay at this level is tolerable; however, it becomes an important concern for interactive MPEG-2 video applications.

Error concealment techniques have been studied and used to improve the received video quality. Most of the error concealment techniques are based on the correlation of a lost component with its spatial or temporal neighboring components. When the BER or cell loss rate (CLR) of the wireless channel is low (CLR $< 10^{-5}$), simple error concealment algorithms such as those in [31, 32], which use previous or adjacent frame or blocks to replace a lost one, can conceal the error to a certain extent. To illustrate the effects of simple error concealment, the simulation results when the previous frame is used to replace a lost one are shown in Fig. 13, where they are compared to that when no error concealment is used. As we can see, error concealment can improve the quality to a certain extent. When the channel condition is poor, special or more sophisticated algorithms should be applied.

**Header Redundancy FEC Strategy**

From studies of the previous section, it is clear that special consideration and effort are needed to make sure that
important MPEG-2 control information can be received correctly. In this section the concept of a new FEC strategy, called header redundancy FEC (HRFEC), for broadcasting or multicasting MPEG-2 video over wireless channels is introduced. A more detailed description of the implementation of the strategy can be found in [33]. Simulation results are also presented, and a performance improvement can be observed.

The Concept of HRFEC for Broadcasting/Multicasting MPEG-2 Video Services

The main idea behind HRFEC is actually a selective protection scheme. When transmitting MPEG-2 video streams over error-prone wireless channels, the MPEG-2 control information (e.g., sequence_header, sequence_extensions, GoP_headers, and picture_headers) is transmitted repeatedly or redundantly to increase the probability of correct reception. At the receiver, the redundant information will be discarded or ignored whenever a single copy of the information is received or accepted correctly.

The overhead and delay jitter introduced by the redundancy cells in the HRFEC need to be estimated so that its efficiency can be used in deciding at which level the HRFEC scheme should be applied in the hierarchical MPEG-2 control structure. Assume that a WATM infrastructure is adopted, the redundancy is RD (RD = 1 when there is no redundancy), and there are average i cells for each frame which contain the priority MPEG-2 control information, such as the GoP_header (if it exists), picture_header, and picture_coding_extension. Usually, i is small. The overhead of HRFEC is then a function of i, RD, R (the CBR MPEG-2 bit rate), fr (frame rate), and possibly sn (the number of slices per frame) depending on which level (sequence, GoP, picture or slice) the HRFEC is applied. For comparison, the overhead per picture at the picture level and slice level are estimated as

\[
\text{overhead}_{\text{picture level}} = \frac{i \times (RD - 1) \times ((48 + \text{overhead}_{\text{watermark}}) \times 8) \times fr \times sn}{R} \quad (11)
\]

For example, if we assume that watm_{overhead} (including header and ECF) is 6 bytes and i = 1, the estimated overhead is shown in Fig. 14. For most MPEG-2 video applications, HRFEC at the picture level makes sense. When R is greater than 4 Mb/s, the overhead per picture is less than 3 percent even when RD = 10. We can also observe that the overhead drops and tends to saturate as the transmission rate R increases. When HRFEC is applied at the picture level, the delay jitter introduced per picture is also a function of i, RD, and R. The estimated results when i = 1 are given in Fig. 15. When r = 4 Mb/s and RD < 10, the delay jitter is less than 1 ms, which is definitely within the tolerance of MPEG video applications.

\[
\text{overhead}_{\text{slice level}} = \frac{i \times (RD - 1) \times ((48 + \text{overhead}_{\text{watermark}}) \times 8)}{R} \quad (12)
\]

Simulation Results

The system performance of HRFEC when applied to CBR-encoded MPEG-2 video over fixed wireless channels was investigated via simulations. An MPEG-2 bitstream was transmitted, and the same FEC (RS code of i = 3) was applied. Taking the overhead picture level into consideration, the CBR MPEG-2 rates are adjusted to R = 3.00 Mb/s, R = 2.97 Mb/s, R = 2.95 Mb/s, and R = 2.89 Mb/s when using RD = 1, 3, 5, 10. The simulation results are given in Figs. 16 and 17.

In Fig. 16 we compare the number of reconstructed frames using HRFEC with different redundancy values of RD (when K = 7.5 dB). For RD = 1, HRFEC is equivalent to the conventional FEC scheme without selective redundancy. We see that for SNR > 20 dB, there is little difference among the different RD values because the channel has low BER with rare frame losses. For SNR < 20 dB, higher redundancy guarantees a larger number of reconstructed frames. For example, when SNR is 10 dB, approximately 150 frames were lost when pure FEC (RD = 1) was used; but that number reduced to 55,
8, and 0 when HRFEC with a redundancy of 3, 5, and 10 was used, respectively. When RD = 10 almost no frames were lost, even for SNR = 10 dB. As a direct consequence, the OGP and PSNR improved dramatically for SNR < 20 dB, as shown in Fig. 17.

These simulation results show that with extra protection for MPEG control information, we get marked improvement in performance, which indicates that HRFEC is an effective error control strategy for transmitting MPEG streams over wireless channels. In addition, the overhead and delay of HRFEC are comparatively small and within the tolerance of service requirements when properly applied.

**Conclusions**

With the emergence of broadband fixed wireless access networks, there is an increasing interest in providing broadband video services over wireless networks. In this article we investigate some fundamental issues regarding broadcasting or multicasting CBR MPEG-2 video sequences over fixed wireless channels in B-FWANs using FEC strategies.

B-FWANs use high-frequency — such as 2GHz, 20 GHz, and 28 GHz (LMDS) — wireless channels to provide services to customers. The simulated system is modeled as a 28 GHz MW TDMA-based broadcast or multicast downlink. A wireless ATM infrastructure was assumed with π/4-shifted DQPSK modulation. For the error control, RS codes were used. For MW transmission, a direct or indirect LOS propagation path is usually required between a transmitter and a receiver. As a result, the wireless channel is modeled as a K factor Rician fading model instead of a Rayleigh fading model. The K factor modeled channel performance was measured via simulations. The unique characteristics of the physical channel require special consideration at the system design level, including the error control strategy. In order to evaluate overall system performance properly, a set of parameters for objective video quality assessment is introduced and used in our simulations, which includes a definition of the OGP value, conventional PSNR value, and the number of reconstructed frames. First, in order to gain a quantitative understanding of the relationship among reconstructed video quality, channel state, and error protection, the performance of transmitting CBR MPEG-2 video sequences over different channel states using different RS codes is measured via simulations. The performance of a cell interleaving scheme was also studied; we show that the performance for broadcasting or multicasting video services can be improved by cell interleaving when the interleaving depth is properly chosen. However, the excessive constant delay introduced by interleaving is a major concern for interactive real-time MPEG-2 applications. After MPEG-2 decoding, error concealment procedures can be employed to improve the video quality to a certain extent.

MPEG-2 control information (i.e., the control blocks) plays a critical role in the decoding process and can influence the reconstructed video quality dramatically; therefore, special consideration and extra protection should be used. The concept of a new FEC strategy, header redundancy FEC, is introduced to address this issue. In HRFEC, selected important (high-priority) MPEG-2 control blocks (e.g., sequence_header, sequence_extensions, GoP_header, picture_header, and corresponding extensions) are replicated before transmission, and duplicate copies are transmitted over the wireless link. The simulation results showed that by using HRFEC at the picture level on an MPEG-2 video sequence, the number of reconstructed frames increased and the overall video quality improved dramatically with low overhead and delay cost. The results indicate that the concept of HRFE is a simple and effective error control strategy for broadcasting or multicasting MPEG-2 videos over error-prone and time-varying wireless channels.

**References**


Additional Reading


Biographies

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