Quality-of-service management for broadband residential video services

by R. J. Green, S. I. Woolley, N. W. Garnham and K. P. Jones

This paper introduces an approach to managing quality-of-service (QoS) for a residential video-on-demand (VoD) service employing MPEG-2 transport streams using an experimental asynchronous transfer mode (ATM)/asymmetric digital subscriber line (ADSL) access network testbed. The paper examines the complex multilayer propagation of ATM layer parameters over an underlying physical layer and their relationship to video quality. The results obtained from the testbed provide an important insight into the factors relevant to the provisioning and management of new multiservice network infrastructures. Overall, these results contribute to an understanding of the multilayer QoS relationship and provide a basis for comparison with, and development of, similar systems. The paper proposes a method to aid the comparison of results based on a multilayer QoS approach.

1 Introduction

Use of the public access telephony network for the provision of broadband services is now feasible with the availability of asymmetric digital subscriber line (ADSL) and optical-fibre technologies. Asynchronous transfer mode (ATM) network technology was designed to support voice, video and data communication. The recent demand for fast Internet access to the home has made the development and rollout of broadband access networks and video-on-demand services commercially viable.

The telecommunications industry has put considerable investment into the development of ADSL technology for the delivery of services requiring 2–8 Mbit/s to the home. ADSL modems are connected to each end of the standard twisted-pair telephone lines. Downstream speed may be as high as 8 Mbit/s, but this rate is dependent on the performance of the copper pair circuit—its wiregauge, distribution point joints, cross-talk, radio frequency interference, line attenuation and impulse noise. The noise and interference result in single-bit errors or burst errors, which corrupt the data stream. To control these effects ADSL systems make use of forward error correction and data interleaving. This protection is traded off against increased latency and delay[1].

A particular challenge is the delivery of video services, which have stringent timing and error constraints. The delivery of high data rate video (3–6 Mbit/s) at satisfactory quality levels requires an understanding of the cumulative effect of quality-of-service (QoS) propagation through multiple protocol layers. In the work reported in this paper ATM bit rates of 3–5 Mbit/s have been used to match the data rate of ADSL.

The standards for VoD (Video on Demand) and ATM connectivity are defined by a number of bodies, the most notable of which are the ATM Forum and the Digital Video Broadcasting (DVB) group[2] for digital TV services. The Motion Picture Experts Group (MPEG) video-coding algorithm (ITU-T H.262)[3] has been adopted as the industry standard for high-quality compression of video. The coexistence of these standards led to a specification for the adaptation of MPEG-2 streams across ATM[4].

This paper presents results from a real ATM/ADSL testbed for investigating the effects of transmission impairments on the quality of received video. After introducing the essential features of MPEG video streams the paper describes the testbed for analysing QoS and introduces the metrics used for characterising the content of the video test sequence. The paper then presents test results and a QoS approach for comparing similar systems.

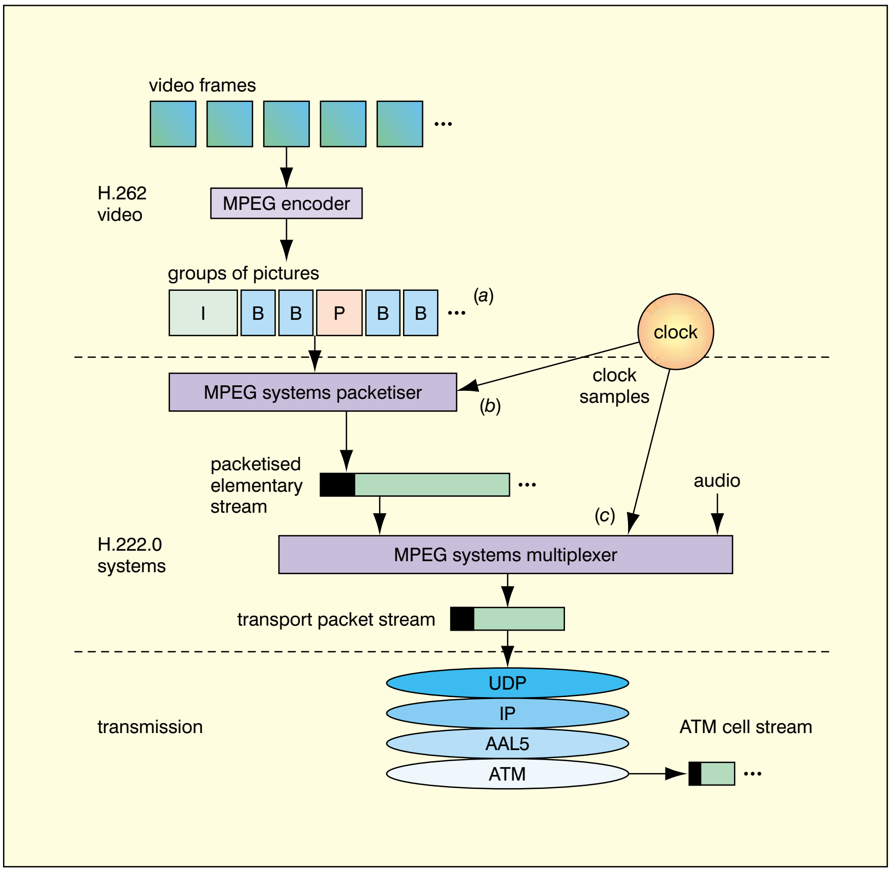


Fig. 1 MPEG transport of video over IP/ATM

contained within them and without reference to other frames. Predicted difference frames are derived from I-frames, and Bi-directional frames are estimated from past and future I- and P-frames[3].

For the purpose of successful video decoding, the I-frames should be error-free and the clocks associated with their presentation reliable. A decoder uses periodic timing references to playback a video stream containing groups of pictures. Decoding timestamps indicate when to start decoding a video frame and presentation time-stamps dictate the time to display a frame to the user ((*b*) in Fig. 1). Together, these timestamps aid long-term synchronisation and are supplied at least every 700 ms. Transport packets, carrying video and audio data, are received at irregular intervals due to the accumulation of delay variations (jitter) within the broadband network. Unique program clock references (PCRs), which occur at least every 100 ms, are used by the decoder to reconstruct the encoding clock ((*c*) in Fig. 1). This enables synchronised data delivery at a constant bit rate (CBR) to the audio and video decoding stages. To maintain jitter bounds in the CBR environment the baseline VoD model proposed by the ATM Forum delivers 2 transport packets (2×188 bytes) in each frame of ATM Adaptation Layer 5 (AAL5) data, where AAL5[7] provides the means to deliver frames between sender and receiver at a given quality of service (e.g. delay, error rate and jitter). Provided that PCR-carrying packets are delivered within acceptable delay bounds the elementary streams (for video, audio, and other data) can be regarded as decodable.

A strong dependency exists between video service applications generating traffic (their specific implementation) and the network. These dependencies need to

2 MPEG video transport streams

MPEG transport streams (TSs) provide a means of delivering video services with a degree of error-tolerance. Transport streams are a subset of the MPEG-2 Systems standard (ITU-T H.222.0)[5]. Essentially an MPEG transport packet is 188 bytes long and contains either video, audio or other data. Packets are multiplexed into a transport stream. Transport streams can carry multiple video, audio and data streams and these are indexed by periodically inserted tables. For an interactive video-on-demand service an audio and a video stream are multiplexed together to form a single program transport stream (SPTS)[4]. MPEG video streams consist of groups of pictures ((*a*) in Fig. 1) containing I-, P- and B-frames[3,6]. Intra frames are coded on the basis of spatial information

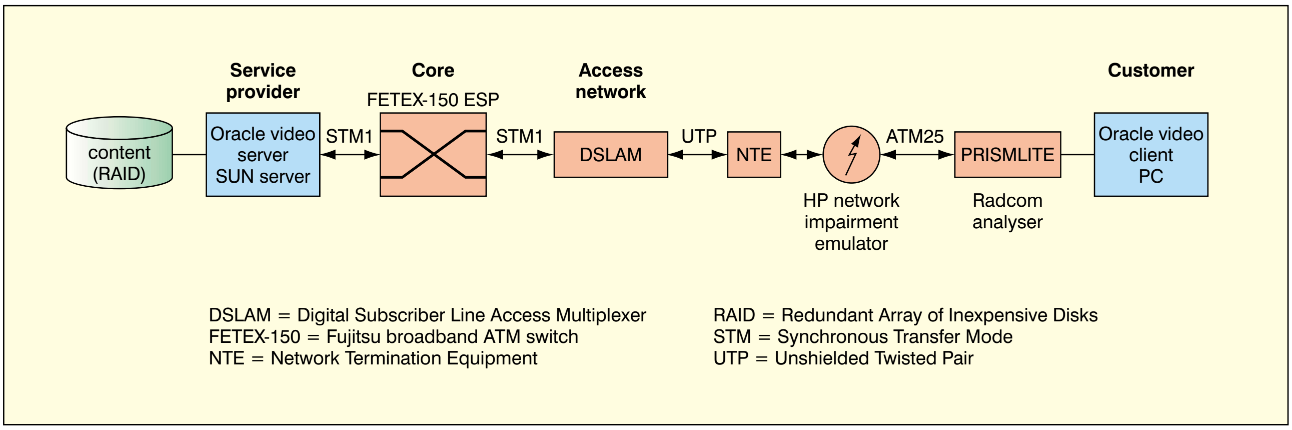


Fig. 2 Logical configuration of the testbed

be integrated in order to guarantee QoS and to maximise resource utilisation. Prior to the move towards IP (Internet Protocol), a significant research effort was made to propose schemes to pack MPEG video into ATM cells, either with or without audio. By default, AAL5 discards erroneous data. This can severely degrade video quality and it is better to pass the correct length of data to the MPEG decoder than to discard it.

Image fidelity is very important to video services. ITU-T P.910[8] defines fidelity as the transparency of the performance (system or codec) with respect to the ideal transmission system; fidelity may be defined by a human observer or by how well the output conforms to the input. This is governed by viewer expectations for a given service and requires investigation into psychovisual phenomena.

3 The experimental testbed

Experimental results were obtained from a real ATM/ADSL testbed with video content encoded with an MPEG-2 hardware encoder and a commercial video server. Fig. 2 shows the key components of the testbed, which was configured as a residential broadband access network. An ATM virtual connection was provided between customer equipment and an Oracle video server[9]. A Realmagic Netstream 2 MPEG hardware decoder was installed on the client[10]. Six MPEG-2 TS packets were used per AAL5 Frame (1·128 Kbytes). Each AAL frame is segmented into 25 cells. A Hewlett-Packard network impairment emulator module (NIM) was used to generate ATM errors[11] and a Radcom Prismlite analyser was used to capture AAL5 frames[12].

Reference	Clip name	Description
a, *b*	Fairy Advert	Screen pan and still frames
c, *d*	News	News presenter moving hands/face and background road traffic
e, *f*	CITV	Morphed computer graphics against rolling background
g, *h*	Cricket	Three scene cuts containing close-up action and fast pans

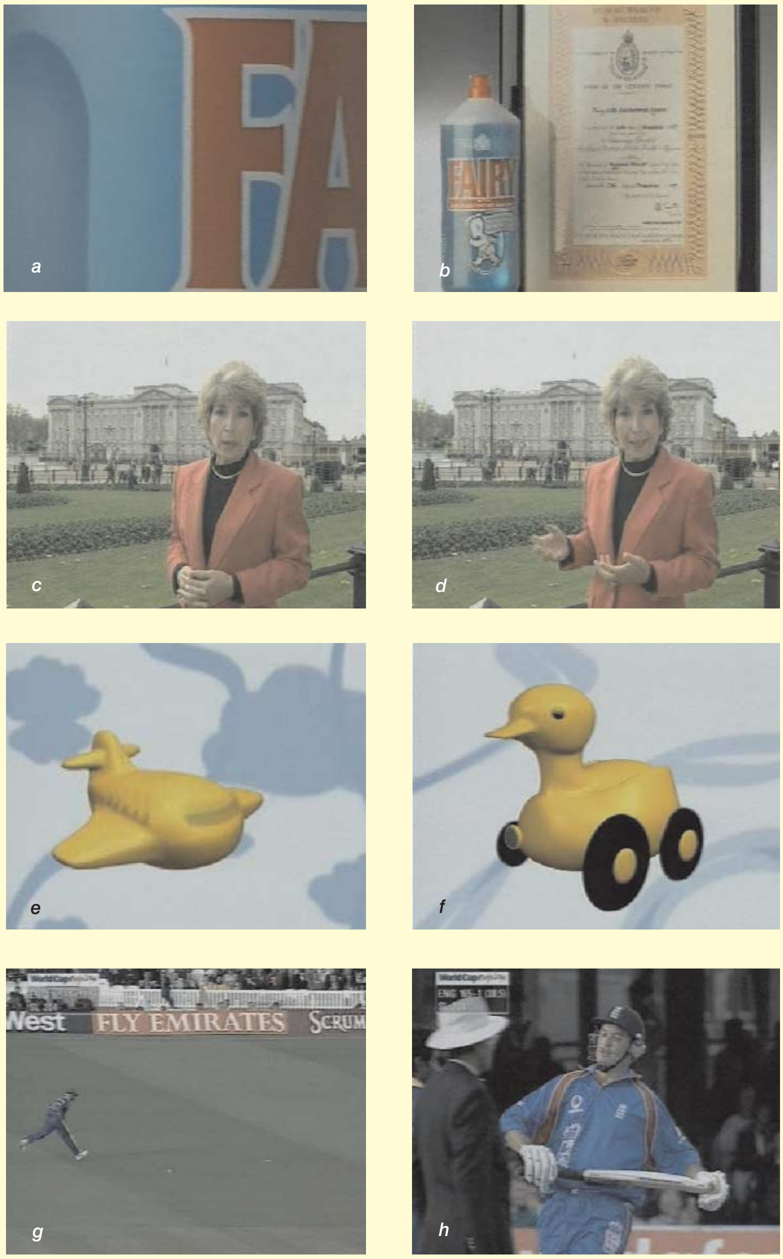


Fig. 3 Sample video frames from the test sequence showing the range of spatial and temporal content

This investigation used single cell losses distributed deterministically and exponentially. The deterministic discard case provided a control to act as a quality differentiator by providing a fixed cell loss interval (e.g. 1 in 1000 cells). Exponentially distributed losses were chosen to approximate buffer losses. Cell losses were inserted on the 25 Mbit/s ATM connection to the PC. Cell loss rates (CLRs) of 1 lost in every 1000 cells (1×10^{-3}) to 1 lost in 50 000 cells (2×10^{-5}) were chosen based on the video stream bit rates shown in Table 1. This table shows the three MPEG TS bit rates used for the test along with their respective ATM bit and cell rates.

4 Test content

This experiment used MPEG-2 video encoded at a resolution of 352 × 576 (horizontal × vertical) pixels to compress TV-quality video down to 2–4 Mbit/s. Broadcast-quality MPEG-2 (at a resolution of 704 × 576 pixels) requires an MPEG data rate of 5–10 Mbit/s.

Four test clips were selected for the test sequence based on their spatial and temporal information content. Fig. 3(*a–h*) shows representative scenes from the four 8-second clips forming the test sequence. The clips were

Table 1: ATM rates for the 2, 3 and 4 Mbit/s MPEG-2 transport streams

MPEG-2 TS, Mbit/s	ATM bit rate, Mbit/s	ATM cell rate, cells per second
2·379	2·883	6800
3·449	4·197	9800
4·516	5·469	12 900

chosen to contain a mixture of detail and slow to fast motion scenes (e.g. action shots and camera pans).

The clips were edited into a 101 second sequence in which each of the four clips was repeated three times. The video was loaded onto the video server and transferred over the testbed.

Video characterisation

The content of a video sequence may be characterised using metrics defined by an ANSI specification for the objective performance assessment of one-way video signals. ANSI T1.801.03[13] defines two equations, called Spatial Information and Temporal Information, which summarise the degree of spatially perceptible information (SI) and temporally perceivable information (TI), respectively, for each frame of a video sequence[8,13]. SI is an indicator of the amount of horizontal and vertical edge information using an edge-enhancing filter[8]. TI is an indicator of the amount of motion between adjacent video frames.

Fig. 4 plots SI and TI for part of the test sequence; the positions of the selected video frames shown in Fig. 3 are marked with a cross.

Fig. 5 illustrates the compression used to encode the test sequence into a 2 Mbit/s stream by showing the degree of quantisation[6] for each MPEG frame as a time history plot for the first 36 seconds (899 frames).

This video was loaded onto the video server and transferred over the testbed. Impaired AAL5 frames were captured using the Radcom analyser and the decoded video was captured on a Betacam video recorder. The next section details the analysis of this data.

5 Video quality assessment

Decoded video may be assessed by either subjective or objective means. Subjective assessment analyses the opinions of a carefully selected group of human viewers. Observers vote on perceived quality according to a 5-point

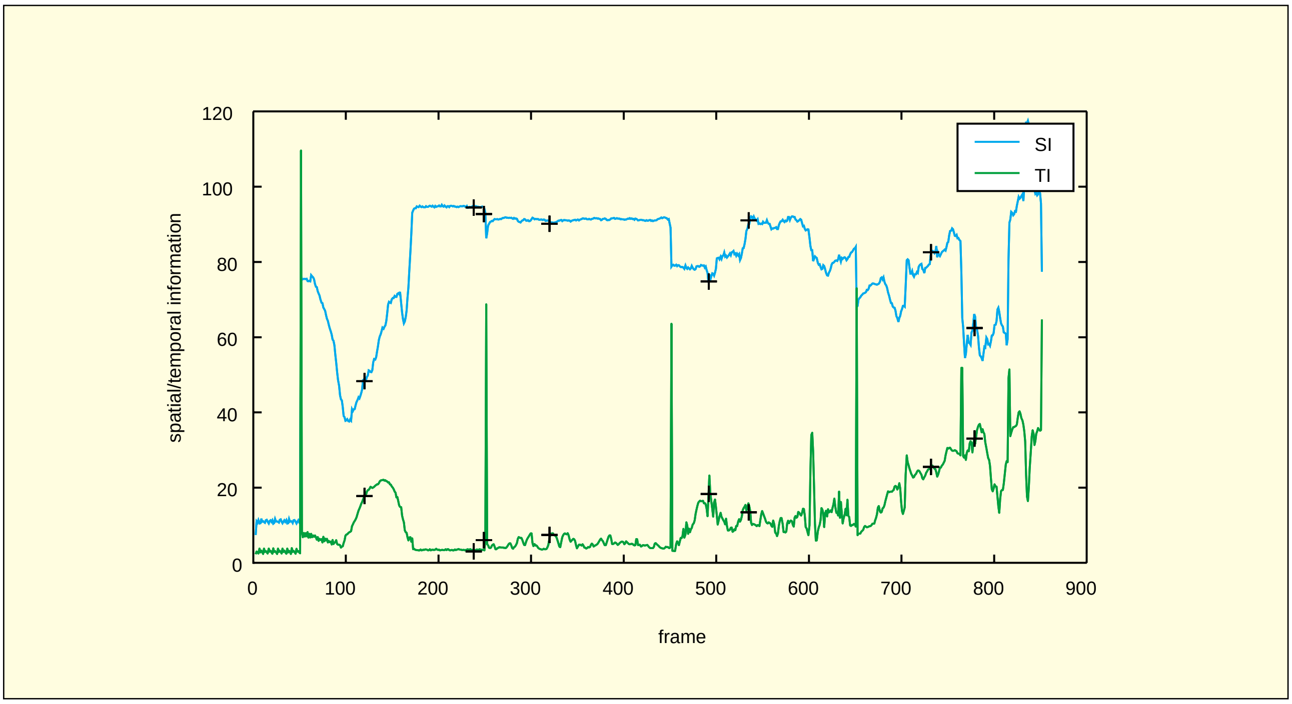


Fig. 4 Graph of spatial information and temporal information for the first 36 seconds of the test sequence

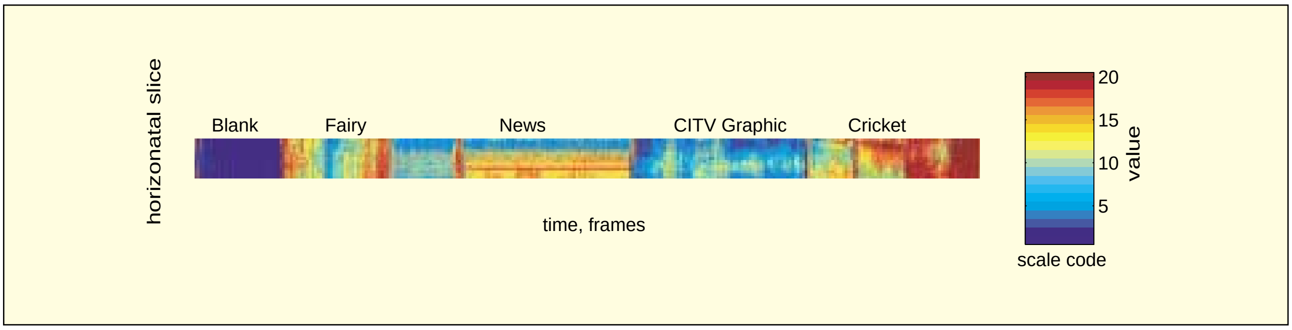


Fig. 5 Quantisation scale code: a 36 second frame time history

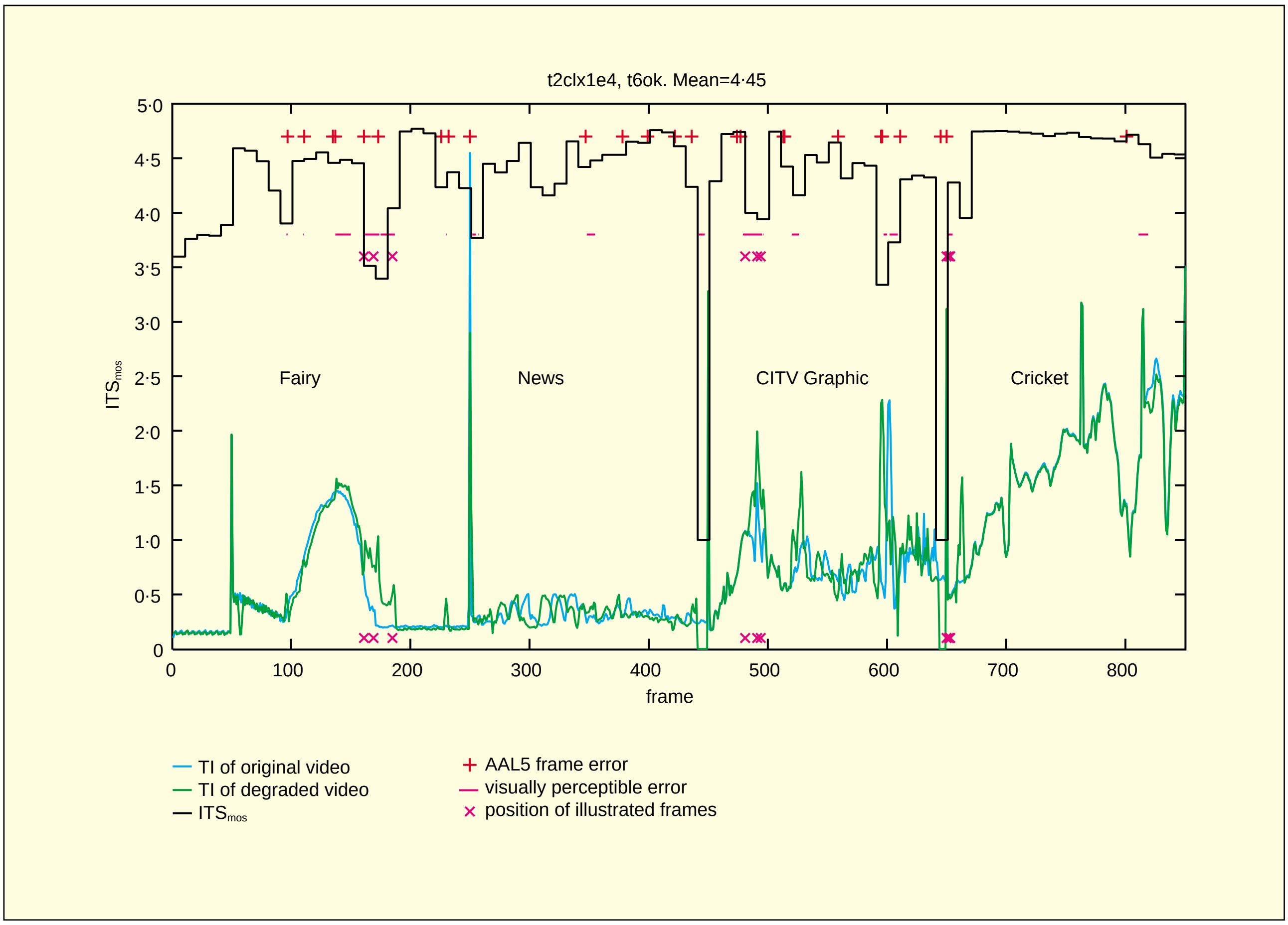


Fig. 6 ITS$_{mos}$ time history plot of the 2 Mbit/s encoding at CLR = 10^{-4} with exponential discard

scale, in which 5 is excellent and 1 is very annoying, see Table 2.

For consistent, real-time, video-quality measures, objective assessment algorithms may be used. Webster[14] presents the Institute of Telecommunications Science's (ITS's) objective assessment system, which makes meaningful evaluations of quality without viewing panels. The ITS quality rating is mapped to the standard quality scale of 1–5. As the ITS metric was designed for low bit rate applications the scores are overestimated.

Results

A visual indication of the performance of each test case is presented using the ITS video quality indicator. The ITS mean opinion score (ITS$_{mos}$) metric is used to assess differences between test cases relative to each other, having a common set of parameters and testbed. Scaled plots of the reference and degraded temporal information (TI) (against time, i.e. the frame interval of 1/25 s) provide an index into the test sequence. TI shows the degree of motion in the sequence; for example, the news sequence has low motion, whereas the cricket sequence contains fast motion, camera pans and quick scene cuts.

Categories of observable artefacts

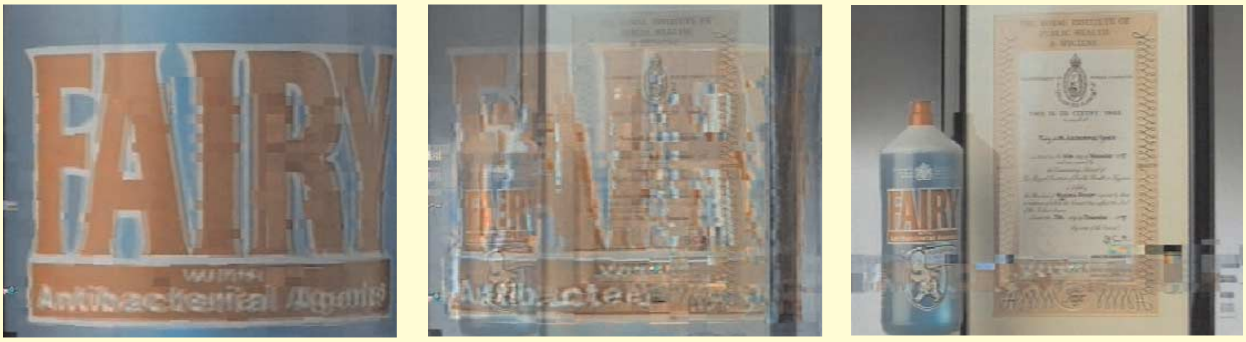
This section presents a sample of the range of observable artefacts that were in the first 650 frames of the 2 Mbit/s encoding that was subjected to exponentially distributed discards at CLR = 10^{-4}. Fig. 6 shows the first 650 frames of the test sequence. The TI of the reference sequence is shown in blue and the degraded TI is shown in green. The ITS$_{mos}$ score, which ranges between 4·77 and 1, is shown as a black staircase plot, each step representing 10 frames. The times at which AAL5-PDU discards occurred are shown by red plus signs. Each frame represents a 40 ms interval. Magenta line segments show where video errors occurred that were observable on a frame analysis of the sequence. Long line segments represent contiguous errors.

Fig. 6 shows that there were video frame losses in the second and third clip where the ITS$_{mos}$ score was 1, the lowest score on the five-point scale. In particular, in the third clip (frames 500) the loss of video frames causes the alignment of the feature vectors to be lost thereby resulting in a score of 3·25. Figs. 7, 8 and 9 show example error bursts that occurred in this sequence. The frame numbers presented correspond to the frames marked with a magenta cross (×) in Fig. 6.

All the errors were very noticeable on playback and are representative of all other test cases considered. These

Table 2: numerical grading scale for subjective video quality assessment

Score	Quality	Impairment
5	Excellent	Imperceivable
4	Good	Perceivable
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying



161 169 185

Fig. 7 Selected frames from a 1 second burst of tiling errors. Starting on frame 161 severe tiling* errors extend to frame 172. Frames 173 to 185 are affected by horizontal slice blocking errors.

481 491 494

Fig. 8 Examples from frames 478–494. Blocking and tiling* errors span 17 frames.

650 652 653

Fig. 9 Frames 650-653 contain a significant glitch. Full screen tiling is observable and a 'synchronisation' error occurs on frame 653.

artefacts characterise the observable errors seen at higher or lower cell loss rates, i.e. similar errors are seen nearly-continuously (e.g. CLR = 10^{-3}) or less frequently (CLR = 2×10^{-5}).

*Tiling artefacts are small blocks having distinct boundaries (the encoding structure). Blocking artefacts are solid colour blocks (e.g. green, yellow, black) having no resemblance to the image.[13]

Results analysis

Undegraded performance: Fig. 10 shows the difference between the 6 Mbit/s (t6ok) reference and the 3 Mbit/s encoding (t3ok). The set of ITS values for the sequence has been time collapsed into an overall mean value, which is printed above each graph. The means for the undegraded 2, 3 and 4 Mbit/s test sequences (t2–4) are all 4·7 (to one decimal place), the maximum value of the

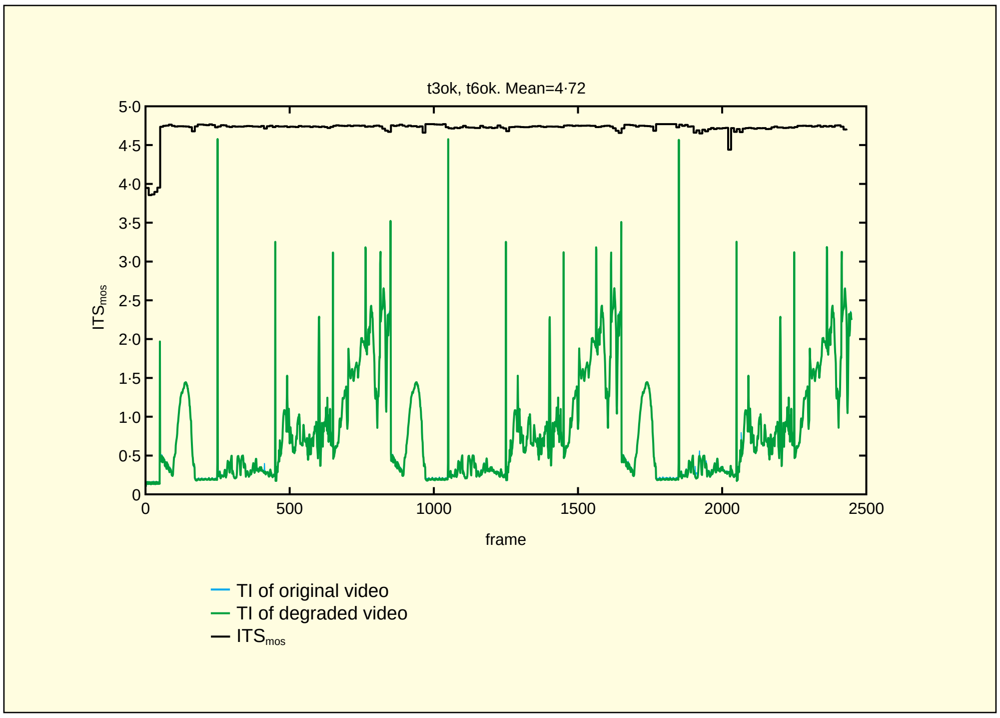


Fig. 10 ITS$_{mos}$ time history plot comparing the 3 Mbit/s undegraded encoding with a 6 Mbit/s (relatively high-quality) reference encoding

ITS$_{mos}$ metric. Fig. 10 shows that there are small deviations from 4·7 where the coding performance has dropped during changes in temporal activity.

Fig. 11 summarises the results of these measurements as comparative histograms of normalised frequency against ITS score (in 0·25 increments). Figs. 11*a* and *b* show that, compared to the 6 Mbit/s reference, 90% of the scores for each bit rate lie above 4·5.

Exponential discards: Figs. 12*a* and *b* show ITS$_{mos}$ plots for 2 and 4 Mbit/s encoding, respectively, and for CLR = 10^{-4}. Exponentially distributed frame error positions are marked by red plus signs (+). The mean ITS$_{mos}$ quality values of 4·5 and 4·0, respectively, (4·4 for 3 Mbit/s encoding—not illustrated) show that the 4 Mbit/s case (t4) was significantly affected. The sequences were resynchronised at each scene cut (as necessary) to compensate for the loss of playback frames resulting from data losses. Loss of synchronisation resulted in the degradation shown until the next scene cut. This was the motive for using the mean of ITS scores over a short number of frames.

Fig. 12 shows that the 4 Mbit/s case was significantly degraded.

Overall results

Fig. 13 shows the probability distributions of the ITS$_{mos}$ scores for the test cases. The notation used for designating test cases is of the form t⟨videostream_bitrate⟩cl⟨ **d**eterministic|e**x**ponential⟩⟨CLR⟩; for example, t4clx1e4 indicates 4 Mbit/s encoding, expo-

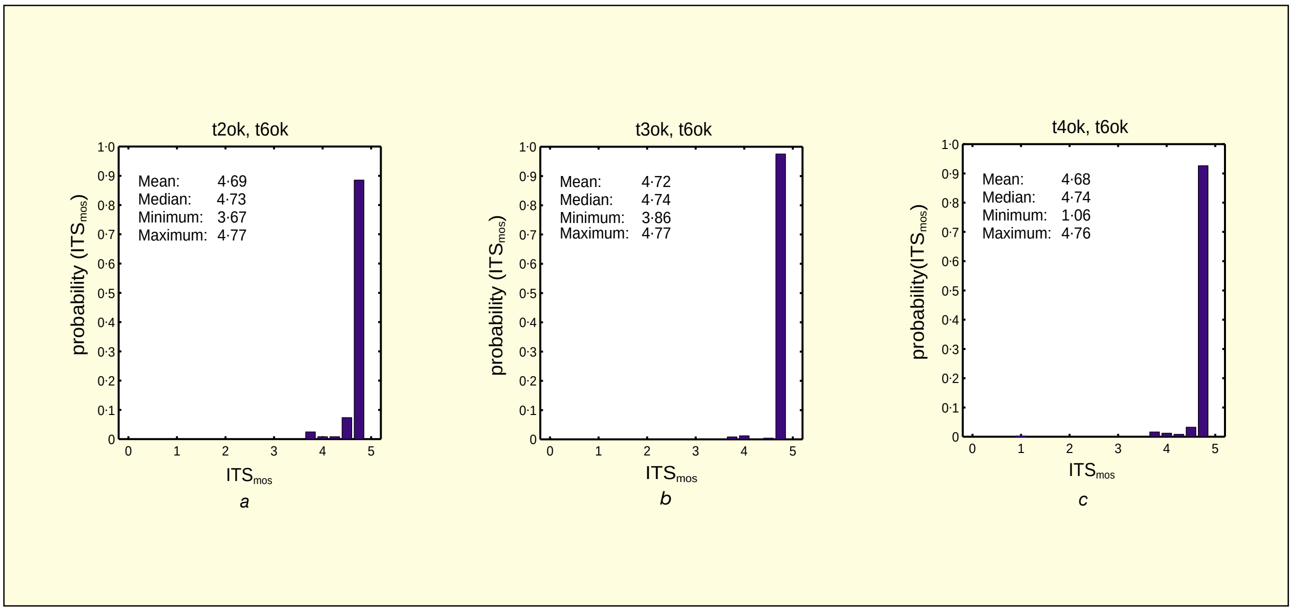


Fig. 11 Histograms showing the probability of ITS$_{mos}$ values for the (*a*) 2 Mbit/s, (*b*) 3 Mbit/s and (*c*) 4 Mbit/s undegraded test sequences compared to the 6 Mbit/s encoding. Bin width = 0·25 'impairment' units.

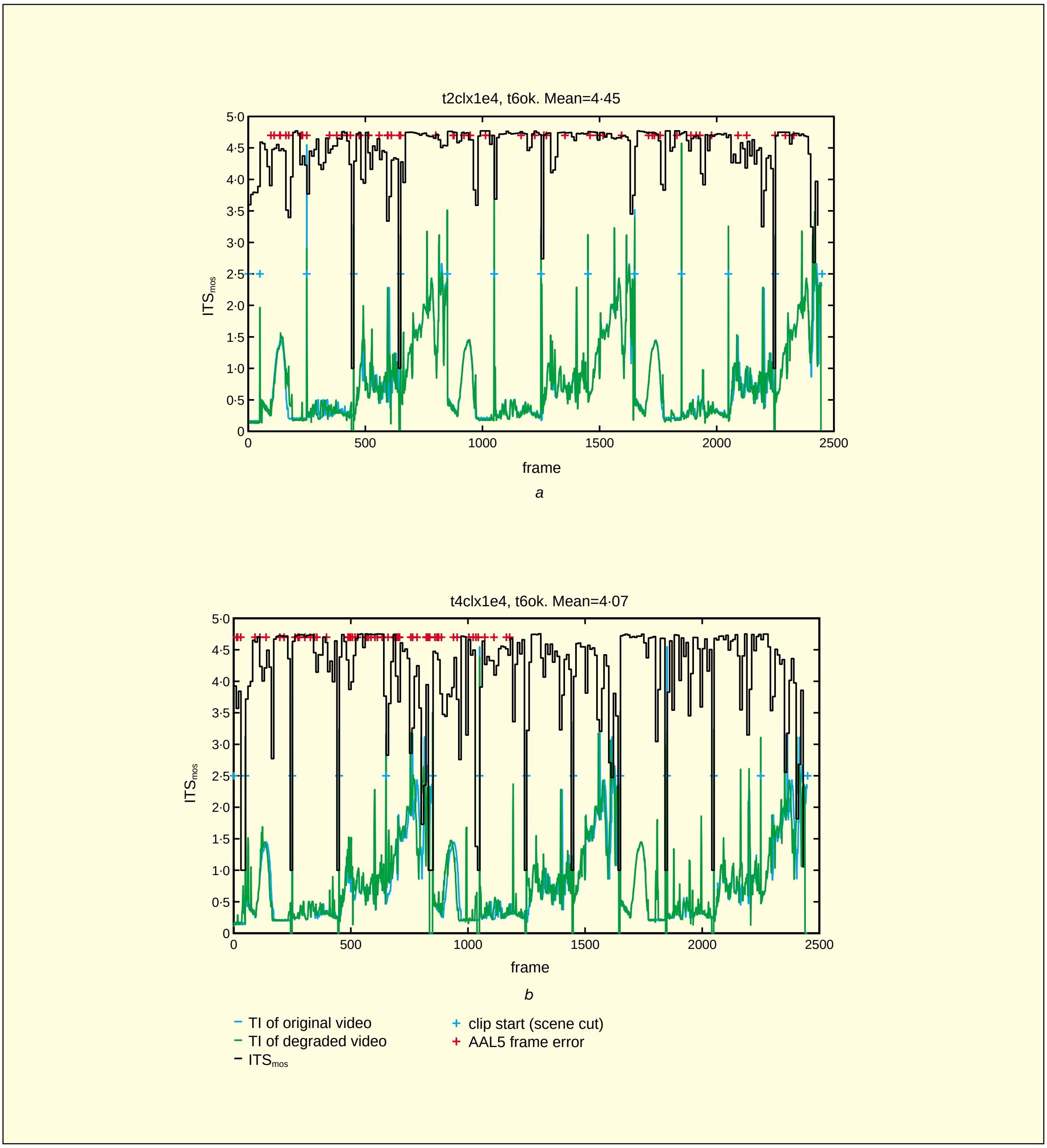


Fig. 12 ITS_{mos} time history plots for exponentially distributed discards at CLR = 10^{-4}: (a) 2 Mbit/s encoding; (b) 4 Mbit/s encoding

nentially distributed discards and a cell loss rate of 10^{-4}.

It can be seen that a CLR of 10^{-3} produced very poor results. This was due to frame-freezing during playback. Picture freezes are not acceptable for video services. The deterministic and exponential test cases at CLR = 10^{-4} also clearly show poor performance. It was noted that bursts of errors were visibly more disturbing than regular, deterministic errors. This is not evident from the histograms. The most important conclusion of this work is that a CLR of less than 2×10^{-5} (i.e. 1 in 50 000 cells) is required to avoid video frame losses for video encoded at 2 and 3 Mbit/s. At this CLR, 32 video frames (1·3 seconds) were lost from the 4 Mbit/s encoding having a cell rate of 12 900 cells per second. Table 3 summarises the results for other CLRs.

A proposed multilayer QoS approach for video services

The analysis of the results of our experiment have shown how picture quality relates to network errors, AAL5 frame inter-arrival times and MPEG transport stream packet losses. The selection of experimental parameters is non-trivial and is beyond the scope of this introduction. However, this paper proposes an approach involving a higher level of description of the key experimental equipment set-up to aid the comparison of results with, and development of, similar systems. This

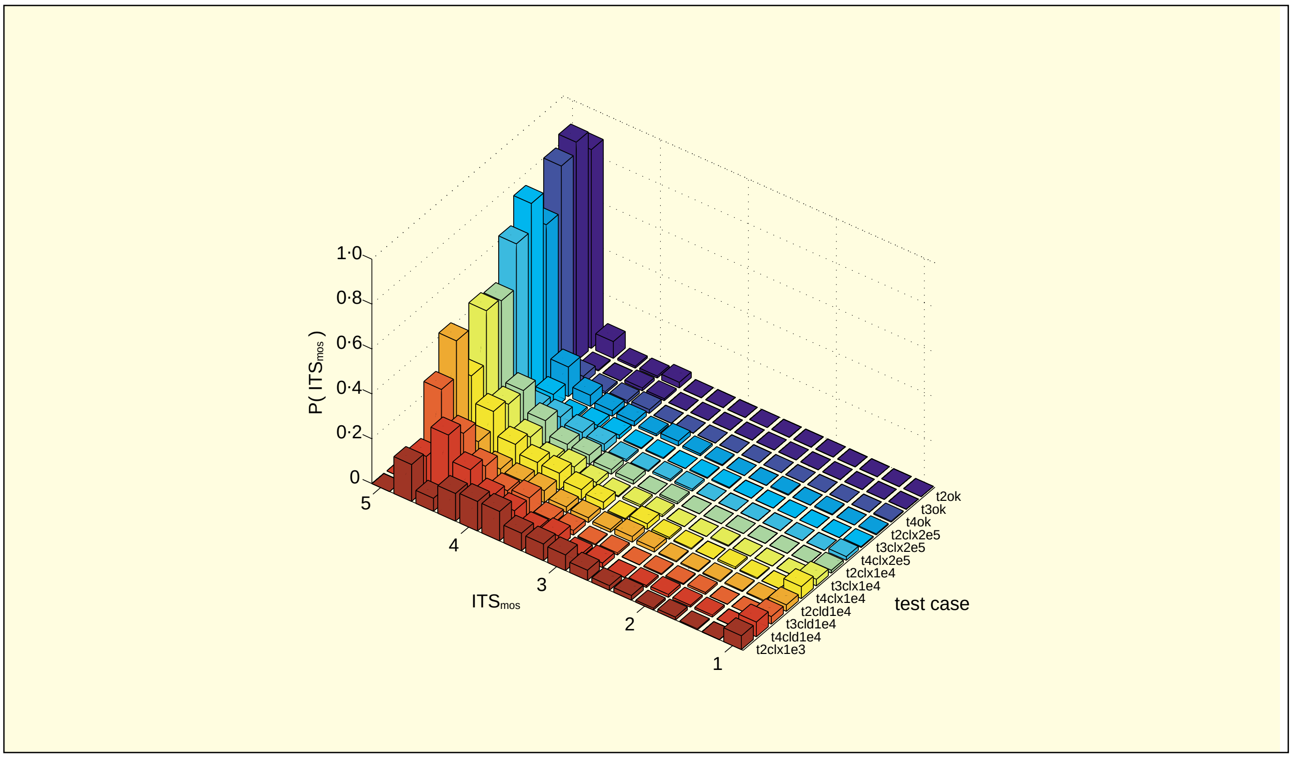


Fig. 13 Probability distributions of the ITS_{mos} scores for various cell loss and video rate test cases

approach incorporates a summary diagram and a table.

Fig. 14 shows the protocol layers used for the experiment described in this paper. Consideration of this analysis illustrates how we may approach comparing systems. The essence of this approach is a bottom-up analysis, centred around the ATM-layer. From the physical layer measurement point, MP–1, ATM layer parameters can be calculated. These can be translated from measurement point MP0 up to MP1, where the elementary streams can be assessed and checked for validity (loss, delay and synchronisation). This localises the analysis of error propagation up to MP2, which is dependent on the implementation of motion compensation and passive concealment between MP1 and MP2.

The controlled variables of the experiment presented in the preceding sections of this paper are listed in Table 4.

Fig. 14 and Table 4 indicate where layer accesses take place to collect data as a measurement point. In the experiment described the Radcom analyser captures AAL5 frames at MP0(i), and the Betacam SP video recorder captures the decoded MPEG video at MP2. Note that the Radcom analyser can decode higher protocol layers but not accurate frame inter-arrival times, as these are dependent on processing scheduling in the end system.

It is important to note that it is not always possible to access data, e.g. the decoder system clock, at the layers we should like as they may be embedded inside a commercial vendor's hardware, e.g. a set-top box or PC decoder card, and therefore inaccessible.

6 Conclusions and further work

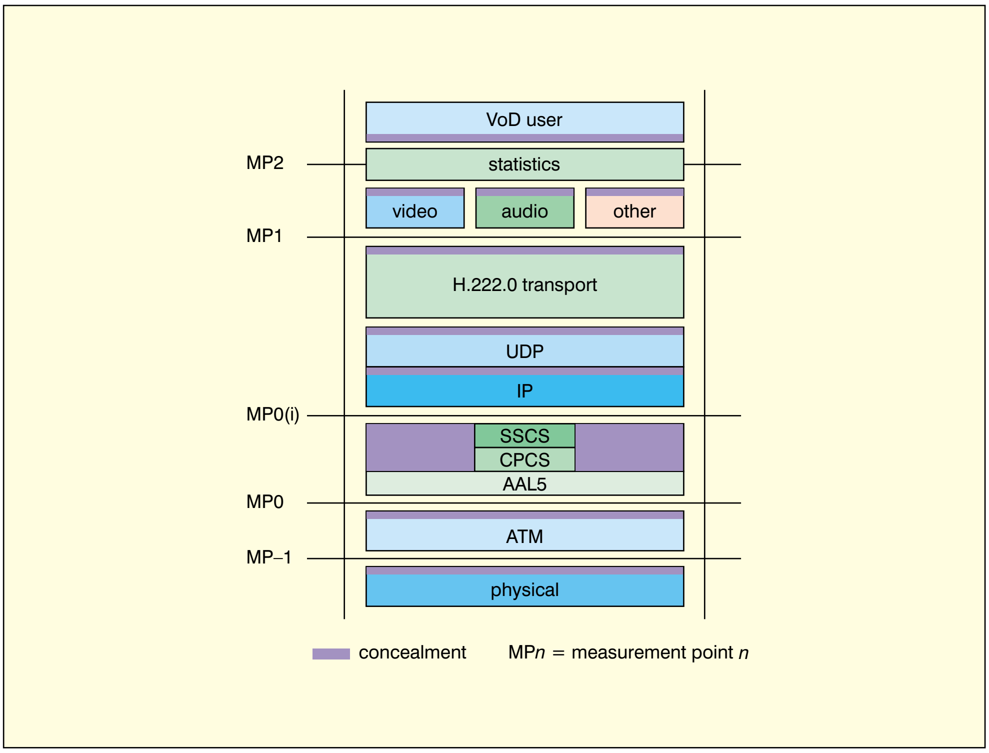
This paper has presented results from an investigation into the way the multiple protocol layers of a real access network platform built from commercial equipment impact on the QoS of high data rate video services. Ultimately, QoS is defined subjectively as the user's degree of satisfaction with the service. However, from the networking viewpoint it is the collective effect of performances that may be measured at a hierarchy of access points. Using the Fujitsu Telecommunications testbed the effects of errors have been shown to have a direct relationship to real access network deployments using ADSL/ATM /AAL5/IP/UDP/MPEG-TS/MPEG-2 video protocol layers. The results of this work convey the key issues that need to be understood and point the way for future research work. These issues include the coding of video test content, the testbed, the video server, the ATM interface parameters and the capture of results by video recording and by data capture.

The work has shown that to avoid frame losses and to provide services using ATM bit rates greater than 3 Mbit/s the cell loss rate needs to be less than 1 in 50 000 cells. Although the hardware decoder was developed for

Table 3: Categorisation of artefacts at different CLRs and video bit rates

CLR	Bit rate, Mbit/s	Observable artefacts
1×10^{-3}	2	Screen freezing, jerkiness and continuous tiling
1×10^{-4}	2, 3	Widespread tiling and jerkiness
	4	Significant jerkiness in fast-action sequences
2×10^{-5}	2, 3	Artefacts significantly fewer
		Tiling and jerkiness observable
	4	Artefacts become more frequent
$\geq 1 \times 10^{-5}$		Tiling and jerkiness still annoyingly perceptible

Fig. 14 A measurement-point model for network termination and MPEG protocol layers



ATM/ADSL networked video applications, tiling or blocking error artefacts were visible. To prevent significant video-frame losses CLRs lower than 2×10^{-5} are required. The loss of 1 Kbyte data packets is severely detrimental to playback performance. This research used half-D1, half TV resolution, MPEG video coding of test sequences in order to provide high-quality compression in the 2–5 Mbit/s range. The results presented here have shown that 4 Mbit/s video is particularly intolerant to cell losses of more than 1 in 10^5.

The objective video quality measure known as the ITS mean opinion score has been used to compare results obtained by modifying two experimental variables—the MPEG transport stream data rate and the ATM cell loss ratio—i.e. to compare the quality of video sequences that have been subjected to different levels of impairment. The approach that has been presented provides a basis for the development and comparison of future systems.

Although protocol layers are often thought of as being transparent this is not normally the case for the delivery of video services. It is hard to define appropriate QoS metrics for them in advance and it can even be difficult to apply these metrics to evaluate service quality. For

Table 4: Summary of the experimental equipment configuration, data capture access layers and experimental variables

Measurement points	Layer	Managed variables and equipment specification	Measurement points accessed	Experimental variable
MP–1	Physical	STM-1 5·5 Mbit/s raw data rate ADSL/ATM 6·2 Mbit/s raw data rate		
MP0	ATM	Equipment HP network impairment emulator Cell discard experimental parameter		CLR
MP0(i)	AAL5	Corrupt data delivery option off Fixed frame size: 1128 bytes Equipment: Radcom analyser	✓	
	IP/UDP	Nonfragmented IPv4		
	Network adaptation	None		
MP1	MPEG transport	Encoding: Optibase Moviemaker 200 Basic Server: Oracle video server Decoder: Real Magic Netstream 2 Accelerator PCR-PID on videostream PCR interval fixed at 22 ms		
	MPEG video	Half D1, PAL-I, GOP = 12,3 Bit rate 2, 3, 4 Mbit/s experimental parameter		Bit rate
MP2		Decoded video: Betacam SP video recorder	✓	

example, the ATM Severely Errored Cell Block Ratio metric (SECBR) would provide a better description of severe degradation of the physical layer than simply error-free seconds (e.g. CLR). Thus, in order to choose optimum frame sizes and provide appropriate error protection to protect video from impairments introduced in the physical layer requires an improved understanding of the physical layer parameters. This paper has shown that exponentially distributed errors can have a serious effect on video quality and it is these bursts that need to be transparent to the video-on-demand viewer.

Acknowledgments

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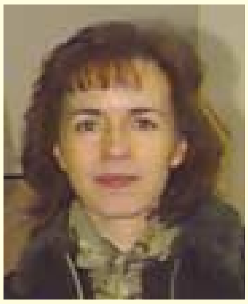

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