Packet-by-Packet Adaptive Scheme for Video Coding and Transmission Over Wireless IP Networks Using Rate-Compatible Punctured Turbo (RCPT) Codes

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***Abstract*— In this paper, we consider real-time video coding and transmission over packet-switched wireless IP networks, such as WLAN, using RCPT codes and joint source-channel coding (JSCC) with concentration on a packet-by-packet adaptive scheme. We present a systematic design methodology to enable the applicability of JSCC techniques. The performance of H.263+ video coding and transmission over wireless channel modelled as slow Rician fading channels using this approach is studied. Results indicate that a packet-by-packet adaptive RCPT-JSCC approach is of significant advantage for real-time video applications and leads to more acceptable video delivery quality over interference-limited and time-varying wireless networks.**

Keywords: Video transmission, RTP-H.263+, RCPT codes, joint source-channel coding, Rician fading.

I. Introduction

Multimedia is expected to be a major source on future wireless networks, and the design of appropriate application-specific transport protocols will play an important role in the provisioning of such services. These transport protocols must interact seamlessly with their peer counterparts on the wired backbone network as well as wired subnetworks. Regardless of the wireless transport protocol in use, the transport layer must utilize the services provided by the data link layer to provide end-to-end service-dependent QoS guarantees. The design issues associated with the transport layer become particularly important for multimedia delivery, e.g., voice, images and video. The multimedia data will typically be provided to the transport layer in compressed form, perhaps using a scalable or layered compression approach, in the form of multiple data streams each with their own QoS requirements. Furthermore, there will generally be strict requirement on the relative latency of these different streams with respect to each other. The transport layer must then provide reliable end-to-end transport of these multimedia data streams, within appropriately defined QoS guarantees, while meeting the strict relative latency requirements despite the relatively unreliable packet transport provided by the data link layer. More specifically, the transport layer must operate reliably in the face of errored or dropped packets on the wireless subnetworks, due to noise, fading, and interference effects, as well as the tandem effects of dropped packets on the wired backbone network due to congestion. This places considerable burden on the transport layer.

In order to provide reliable operation then, an appropriate multimedia transport protocol must be provided, in addition to segmentation and reassembly of compressed data streams to/from packets, some form of FEC across packets perhaps combined with some type of packet interleaving. Generally this is performed in an adaptation sublayer and must be carefully designed to avoid introducing additional latency in the form of delays and jitter, either of which could be catastrophic to a real-time multimedia service such as video. Furthermore,the design of the adaptation sublayer is further compounded by the fact that the different layers or streams of a multimedia source have different QoS requirements and thus different adaptation sublayers may be required for each source layer [1].

The effective solutions should be in an integration fashion across multiple networking layers. Existing protocols for wireline networks are very limited in their ability to accomplish this; they are generally designed and implemented to provide specified and narrowly-defined services with little ability to adapt to the highly time-varying conditions associated with wireless networks. What is required is an appropriate suite of adaptive, event-driven protocols that pass state information across layers in an effort to cope with this variability. Furthermore, these protocols must interact in a relatively seamless fashion with their counterparts on the fixed wireline backbone networks. To accomplish this will require a closer coupling between communications and networking research.

The use of middleware to manage wireless networks is of interest. Whereas the use of intelligent software in communication protocol layers has not been extensive, there is reason to believe that a greater degree of intelligence can and should be incorporated into middleware layers. Decision-making to support smart caching, mobility awareness, error control, packet-by-packet protocol selection, and other link-enhancing adaptations fits naturally into middleware layers.

In this paper we will investigate the performance of a packet-by-packet adaptive scheme for digital video transmission over wireless IP networks using RCPT codes in a JSCC manner as a direct extension of our previous research on cross-layer design. Considering the superior performance of Turbo codes, some further performance improvement may be possible compared to the use of Rate-Compatible Punctured Convolutional (RCPC) codes. Our objective is to quantitatively compare the performance between JSCC using RCPC and RCPT as well as investigating the JSCC scheme structure using RCPT over wireless channels modelled as an AWGN channel as well as more general slow-fading Rician channels which incorporate fading/multipath effects.

The remainder of this paper is organized as follows: In Section II we provide some technical preliminaries describing RTP-H.263+ packetization, slow Rician fading channel model, channel loss patterns and passive error recovery mechanisms. In Section III we describe rate-compatible punctured Turbo (RCPT) channel codes, and the proposed packet-by-packet adaptive RCPT-JSCC approach. In Section IV we provide some selected experimental results and discussion. Finally, Section V provides a summary and conclusions.

II. PRELIMINARIES

In this paper we describe and investigate a video coding and delivery system for wireless IP networks which consist of a H.263+ video codec, a RTP-H.263+ packetization scheme, RCPT channel coding and QPSK modulation. Considering the typical bandwidth limitations of wireless channels, QCIF-format (176×144) video sequences are used in this work.

Figure 1 illustrates the video coding and wireless delivery scheme proposed and investigated in this paper. A H.263+ source coder encodes the input video which is applied to a RTP-H.263+ packetizer. Before the packets are transmitted, they are protected against channel errors using FEC codes. The channel coding rates can also be selected adaptively according to the prevailing channel conditions. This channel rate matching is achieved by employing a set of RCPT or RCPC codes. Finally, the bitstreams are modulated before being transmitted over a wireless link. During transmission, the modulated bitstreams typically undergo degradation due to additive white Gaussian noise (AWGN) and/or fading. At the receiver side, the received waveforms are demodulated, channel decoded and depacketized, and then source decoded to form the reconstructed video sequence. The reconstructed sequence may differ from the original sequence due to both source coding errors and possible channel error effects.

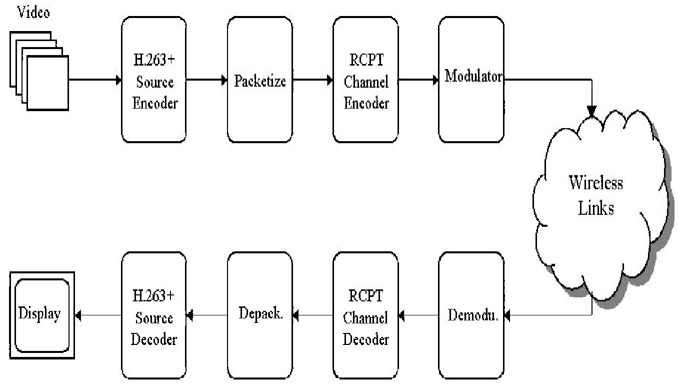

Fig. 1. Illustration of video coding and delivery system over wireless networks.

A. RTP-H.263+

In order to transmit real-time H.263+ video over wireless IP networks, the H.263+ bitstream must first be packetized. The protocol of choice for IP-based real-time packet video applications is the real-time transport protocol (RTP). A payload format for H.263+ video has been defined for use with RTP (RFC 2429) [2]. According to the RTP-H.263+ payload format specification, the H.263+ encoded bitstream is packitized and then transmitted as RTP packets.

In our experiments, the GOB mode was selected for the H.263+ coder and packetization was always performed at group of blocks (GOBs) boundaries, i.e., each RTP packet contains one or more complete GOBs. Since every packet begins with a picture or GOB start code, the leading 16 zeros are omitted in accordance with RFC 2429 [2].

B. Packet Video Over Wireless Networks

Knowledge of the radio propagation characteristics is usually a prerequisite for effective design and operation of a communication system operating that environment. The fading statistics of different radio channels and their associated communication performances has been studied considerably in the past [3]. Despite the fact that Rayleigh fading is the most popular model, Rician fading is observed in radio channels as well as in indoor cordless telecommunications (CT) [3].

A slow and flat Rician fading is assumed here, where the duration of a symbol waveform is sufficiently short so that the fading variations cause negligible loss of coherence within each received symbol. At the same time, the individual waveform is assumed to be sufficiently narrow-band (sufficiently long in duration) so that frequency selectivity is negligible in the fading of its spectral components. As a result, the receiver can be designed and analyzed on the basis of optimal processing of the received waveform, e.g., by a matched filter or other appropriate substitute in the same manner used in the nonfading case.

C. Channel-Induced Loss Models

In this work, we restrict our attention to a random loss model, i.e., the wireless channel is characterized by uncorrelated bit-errors. This is a reasonable model for a fairly benign wireless channel under the assumption of sufficient interleaving to randomize the burst errors produced in the channel decoder.

By means of FEC, some of these bit errors can be corrected. Depending on the FEC code parameters and the channel conditions, there will be residual bit errors. Generally, over existing wired IP networks, UDP is configured to discard any packet with even a single error detected in the entire packet, including the header, although UDP itself might not implement this error-detecting functionality. In the wireless video telephony system described by Cherriman and Hanzo [4], such packets are also discarded without further processing. However, we have shown in [5] the advantage of a transparent transport layer for video transmission over noisy channels such as wireless links.

In this paper, we will again assume the transport layer is transparent to the application layer, i.e., a packet with errors is not simply discarded in the transport layer. Instead, the application layer should be able to access the received data although such data may have one or more bit errors. Such a model corresponds to a transport layer scheme allowing bit errors in the payload. The channel-induced impairment to the video quality is then in the form of residual bit errors in the video stream. It is the responsibility of the application layer to deal with the possible bit errors. Specifically, here we make use of the H.263+ coding scheme where, based on syntax violations, certain error patterns may be detected by the video decoder and use of the corresponding errored data can be avoided by employing passive error-recovery techniques.

For simplicity, we make use of the error-detecting and recovery scheme suggested in Test Model 8 [6]. More detailed discussions of available error recovery techniques can be found in [7]. The major objective of the PER schemes in [6] is to detect the severe error patterns and prevent the use of such errors which may substantially degrade the video quality. The remaining undetected error patterns in the payload which are not detected by the H.263+ decoder will result in the use of incorrectly decoded image data which can cause quality degradation of the reconstructed video.

III. Packet-by-Packet Adaptive Transport Scheme

Unlike separation-based techniques, joint source-channel coding design techniques rely on the joint or cooperative optimization of communication system components. The joint source-channel coding approach allows for strategies where the choice of source coding parameters caries over time or across users in a manner that in some way depends on the channel or network characteristics. Likewise, joint source-channel coding allows for systems for in which the choice of channel code, modulation, or network parameters varies with the source characteristics. The goal for time-varying systems is to achieve graceful performance fluctuations as the system evolves and change with time.

The objective of JSCC is to jointly adjust the source and channel coding rates to optimize the overall performance, due to both source coding loss and channel error effects, subject to a constraint on the overall transmission bitrate budget. In this work, we present a packet-by-packet adaptive coding and transport scheme based on the JSCC approach to adjust the system operating mode according to the prevailing channel conditions.

In a packet-by-packet adaptive scheme, the source coding bit rate and the corresponding channel coding rate are jointly selected for a single video data packet based on the prevailing channel conditions. A higher rate channel code with less overhead introduced is invoked when the channel is favorable, in order to increase the effective source coding rates under overall bit budgets resulting in better video quality. Conversely, a lower rate or more powerful channel codes is employed in order to improve the error correcting capacity to obtain lower residual BER when the channel experience inferior quality, such as when new users join into the wireless network. The adaptation is achieved through the use of channel rate matching.

A. Channel Rate Matching

The channel coding rate matching is processed through threshold-control algorithm based on pre-established threshold tables obtained through the JSCC approach described in the following section. The corresponding thresholds are dependent upon the BER performance of corresponding channel codes and channel conditions as much as the error-resilience performance of coded video to the channel induced impairments, here in the form of bit errors and packet losses. Specifically, in this work we model the wireless channels as Rician fading channel with $\zeta^2 = 7$ dB as discussed previously. For the Rician fading channel model, we assume decoding with perfect channel state information (CSI) and perfect channel interleaving, i.e., channel errors are randomly located.

The error-resilience performance is obtained through the use of universal distortion-rate characteristics [8]. It depends on the characteristics of video sequence, such as the motion and scene variations, and post processing techniques used to mitigate the effects of errors, such as passive error recovery (PER). PER approaches offer the potential for exploiting the inherent redundancy in typical video sequences to recognize gross errors caused by channel error effects and replace them by appropriate spatiotemporal interpolations based upon reliably received information deemed uncorrupted by channel error effects [7]. Unfortunately, these schemes tend to break down in the presence of significantly high and sustained channel error effects and/or high spatiotemporal frequency components rendering the interpolations relatively useless [9]. In general, it is fair to say that compressed video data are extremely sensitive to the transmission-induced errors. As a result, it is of essential importance to keep the residual bit error rate under certain level in order to provide certain QoS specified. In turn, for given QoS, it leads to the selection of appropriate channel codes under specified channel conditions for certain channel model.

B. RCPT Codes

Turbo codes, introduced by Berrou et al. in 1993 [10], caused a great stir in the coding community and have prompted a great deal of research [11], [12], [13]. Rate-compatible punctured turbo (RCPT) codes have also been considered to achieve unequal error protection (UEP) [14], and the use of turbo codes in an ARQ protocol was first proposed in 1997 [15], in which additional constituent encoders are added to obtain rates lower than that of a conventional turbo encoder. Rowitch and Milstein [16] considered RCPT codes applied to ARQ schemes to provide higher throughput. Cherriman et al. [4] investigated the performance improvement for wireless video telephony using Turbo codes compared to the use of BCH codes.

In this work, a set of RCPT codes are generated using the algorithm described in [16] employing two memory $M = 4$ constituent encoders with generator $(1, 23/35)_{octal}$ and a puncturing period $P = 8$. Simulations were conducted to obtain the BER performance of the resulting RCPT codes over both the AWGN and Rician fading channel with $\zeta^2 = 7$ dB. In order to achieve statistical reliability, for each SNR, simulations employed up to 10^8 transmitted information bits or until 1000 errored blocks were detected, whichever occurred first. For Rician fading channels, we assume perfect channel interleaving, i.e., channel errors are randomly dispersed.

C. Joint Source-Channel Coding Methodology

The overall performance for the transmitted video will be measured as the average $PSNR$ over a sequence of N_f consecutive frames and includes channel error effects as well as source coding losses. For each combination of source coding rate, R_s, channel coding rate, R_c, and the packetization overhead, R_p, the corresponding optimal operational distortion-rate characteristics for a given overall channel signaling rate, R_{tot}, is given as,

$$PSNR^*(R_{tot}) = max\ PSNR(R_s, R_p, R_c)\ , \qquad (1)$$

where the maximization is performed over all R_s, R_c and packetization choices of interest, subject to the constraint

$$\frac{R_s + R_p}{R_c} \leq R_{tot}\ . \qquad (2)$$

In [17], [8], it was shown that much of the computational complexity involved in solving this optimal rate allocation problem may be avoided through use of universal distortion-rate characteristics, $PSNR(R_s, R_p, P_b)$, where R_s represents the source rate, R_p represents the packetization overhead, and P_b represents the corresponding bit-error probability. The universal distortion-rate characteristics can be obtained through simulations where the net effect of the transported channel, including FEC, is modeled as a BSC with crossover probability P_b [8].

Given a family of universal distortion-rate characteristics for a specified source encoder and a video sequence, together with appropriate evaluations of bit-error probability for a particular modulation/coding scheme as a function of channel parameters, the corresponding optimal distortion-rate characteristics for the video sequence can be determined as in [17], [8] through the following procedure: For a specified channel signal-to-noise ratio, E_S/N_I, we can find the associated P_b through the corresponding bit-error probability performance characteristics for a selected modulation/coding scheme as discussed earlier. Then, for each choice of source coding rate, R_s, of interest, use the resulting P_b to find the corresponding overall $PSNR$ from the universal distortion-rate characteristics. Finally, we evaluate the resulting component distortion-rate characteristics through the JSCC approach described in [17], [8]. More specifically, this entails solution of the rate allocation problem described by (1) or, equivalently, obtaining the convex hull of all operational points $PSNR(R_s, R_p, R_c)$ satisfying the constraint (2). In this work, as noted previously, the R_c are selected from a set of avaliable RCPT or RCPC codes of rates, $R_c = \frac{8}{9}, \frac{8}{10}, \cdots, \frac{8}{24}$, which are obtained by making use of a $R_c = 1/3$ mother code with memory $M = 4$ or 8 and a corresponding puncturing period $P = 8$. Furthermore, all results are obtained for the case of packetizing 1 GOB per packet, for QCIF video sequences at 7.5 fps, resulting in packetization overhead of $R_p = 21.6$ Kbps.

D. Middleware

The proposed packet-by-packet adaptive scheme is necessary for enabling real-time video service over wireless networks with certain QoS as have been demonstrated in [18] and [4]. The main advantage is that irrespective of the prevailing channel conditions, the system achieves the best possible video quality by intelligently adjusting the achievable video source coding rate and the corresponding channel coding rate in order to match the channel quality experienced. This is achieved on an IP packet basis in order to enable the compatibility with the IP networks. Our scheme is different

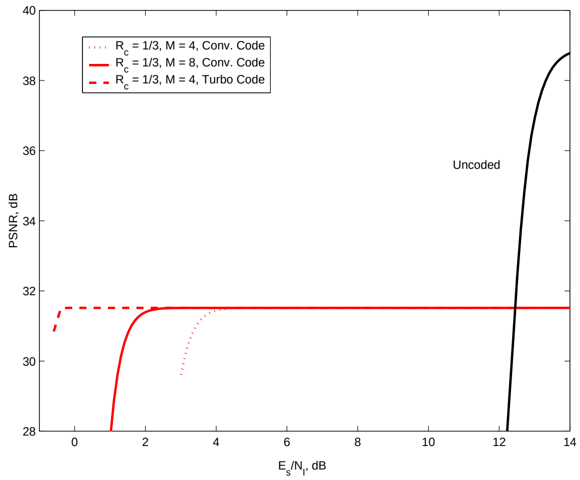

Fig. 2. Performance of RTP-H.263+ packet video transmission over an AWGN channel using fixed Turbo and convolutional codes.

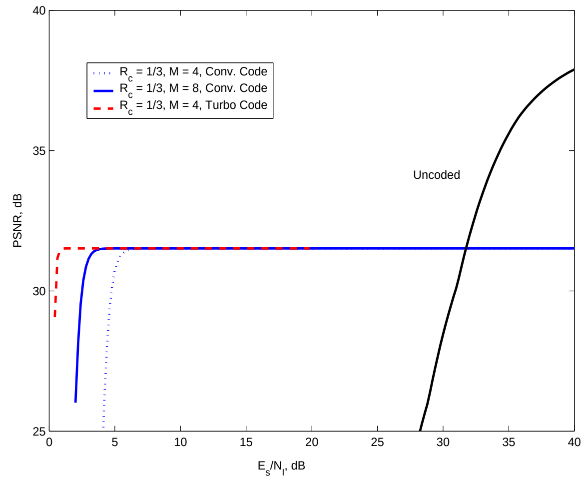

Fig. 3. Performance of RTP-H.263+ packet video transmission over a Rician channel with $\zeta^2 = 7$ dB using fixed Turbo and convolutional codes.

from the proposed adaptive modem in [4] under which a near-instantaneous system is required, considering practical time delay in acquiring channel state information, such scheme may be highly unstable and sensitive. Furthermore, too much frequent adjustment in video coding rate may resulting in highly fluctuating video qualities and make it annoying to end-users. A major objective of our proposed scheme is to enable system operation mode switch based on certain channel models when resource re-allocation happens. Among them are the join-in or leave-out the shared network of certain number of users, or channel quality change attributed to mobility or fading. Another advantage of this scheme is the fusion of information between different layers in the communication system, the physical channel state information is passed through layers to the wireless adaptation sublayer middleware, and the corresponding rate adjustment are processed. This appropriate matches the use of application-layer framing (ALF) as suggested in [19].

IV. Results and Discussion

We present some selected results using RCPT codes and a JSCC approach for a representative QCIF video-conferencing sequence, Susie at 7.5 fps. These results were obtained using a single-layer H.263+ coder in conjunction with RCPT channel codes together with QPSK modulation. The results for both AWGN and Rician fading channels are demonstrated. The maximum symbol transmission rate is set to be $r_S = 64$ Ksps, such that the overall bitrate (after channel coding) is constrained as $R_{tot} = 128$ Kbps due to the use of QPSK modulation.

Figure 2 demonstrates results for the RTP-H.263+ packet video transmission over an AWGN channel using fixed source and channel coding schemes. We provide results for this system using PER together with turbo coding with $R_c = 1/3$ and memory $M = 4$ compared to an uncoded system using only PER. For comparison, the results for the same rate convolutional codes, with memory $M = 4$ and 8, are also illustrated. It should be obvious that in the absence of channel impairments, such as with large E_S/N_I, the additional channel coding overheads force a decrease in the available source coding bitrate[1] and this results in a corresponding decrease in the video quality. This can be seen if we compare the results for the coded and uncoded systems for large E_S/N_I, the difference is about 7 dB in PSNR. However, it should be noted that passive error recovery by itself is not effective as E_S/N_I decreases below a threshold value, approximately $E_S/N_I = 13$ dB in this case. As a result, the reconstructed video quality degrades dramatically. On the other hand, the coded cases can maintain the video quality at acceptable levels for much smaller values of E_S/N_I compared to the uncoded system which can be attributed to its error-correcting capability. This is a good indication of the necessity of employing FEC coding in wireless networks. Furthermore, the system employing turbo coding outperforms those using convolutional codes with similar complexity. This is also illustrated in Figure 2 if we compare the performance between the rate 1/3 turbo code with $M = 4$ and the rate 1/3 convolutional code with $M = 8$; a performance gain of about 2 dB in E_S/N_I is obtained. Note that the block size of the employed turbo coding here is 1024 bits which is reasonably short resulting in negligible additional delay for video transmission where a reasonably large number of bits are generated and transmitted instantly. This suggest that turbo codes are a reasonable choice even for real-time video applications resulting in improved power efficiency.

[1]Recall we are holding the total transmitted bit budget at $R_{tot} = 128$ Kbps.

Next we consider a more realistic wireless channel model, the Rician fading channel. Figure 3 demonstrates results for RTP-H.263+ packet video transmission over a Rician fading channel with $\zeta^2 = 7$ dB, again using fixed source and channel coding schemes. Again, we provide results for a system using PER together with turbo coding with $R_c = 1/3$ and memory $M = 4$ compared to the uncoded system using only PER. For comparison, the results for the same rate convolutional codes, with memory length $M = 4$ and 8, are also illustrated. It is clear that to provide a fixed quality of reconstructed video, say 30 dB in PSNR, an E_S/N_I in excess of 30 dB is required for an uncoded system. For interference-limited wireless links, such large power levels will cause excessive interference to other users, resulting in reduced system capacity. To improve the system capacity, FEC must be employed as clearly demonstrated in Fig. 3. For example, for the same quality level, 30 dB in PSNR, the selected turbo coded system requires only 2 dB in E_S/N_I, a substantial coding gain of over 28 dB in E_S/N_I. This is achieved, however, at the expense of a video quality degradation of about 7 dB in PSNR for large E_S/N_I. Nevertheless, the significant performance improvement at low E_S/N_I clearly suggests the advantage of FEC coding in the design of a multiuser wireless communication system where efficient low power operation is the key to improved system capacity. Furthermore, as illustrated in Fig. 3, turbo coding still outperforms the convolutional codes with similar complexity by over 1 dB in E_S/N_I. It should be noted that the block length of turbo codes used in this paper is held to 1024 information bits, which should be considered reasonably short-length. For high bitrate video encoding and transmission systems, such as possible on wireless LANs, it can be expected that even larger coding gain is possible using turbo codes with larger block length while maintaining acceptable delay.

The above results also suggest the use of a JSCC approach to adaptively allocating resources between source and channel coding. We consider the use of RCPT codes within a JSCC approach. First, we provide results for a JSCC approach for the case of an AWGN channel. The results are illustrated in Fig. 4. Here we illustrate results for different values of channel coding rate with the source coding rate chosen to achieve the overall bitrate budget $R_{tot} = 128$ Kbps. In particular, the smaller values of R_s allow use of more powerful low-rate channel codes resulting in improved performance for small values of E_S/N_I. On the other hand, for large E_S/N_I performance improvements can be obtained using larger values of R_s together with less powerful high-rate RCPT channel codes since less error-correcting ability is required. The optimum JSCC procedure selects the convex hull of all such operating points as illustrated in Fig. 4. As can be observed, the JSCC approach clearly demonstrates significant performance improvements over either the uncoded case or the case where the channel coding rate is fixed at an arbitrarily chosen value.

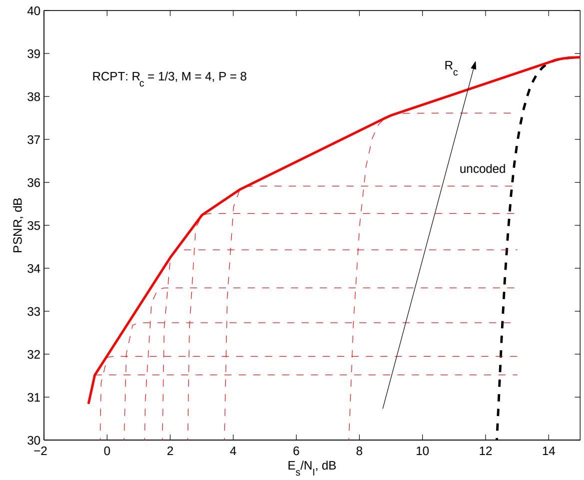

Fig. 4. Performance of RTP-H.263+ packet video transmission over an AWGN channel using JSCC approach with RCPT codes. Also shown are performance results for a set of fixed channel coding rate schemes.

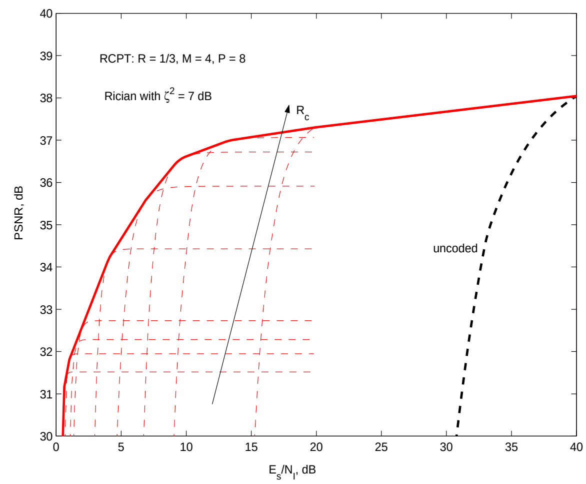

Fig. 5. Performance of RTP-H.263+ packet video transmission over a Rician channel with $\zeta^2 = 7$ dB using JSCC approach with RCPT codes. Also shown are performance results for a set of fixed channel coding rate schemes.

Now we apply the JSCC approach to Rician fading channels. In Fig. 5, we illustrate the results for the JSCC scheme together with a set of fixed coding schemes for a Rician fading channel with $\zeta^2 = 7$ dB assuming decoding with perfect CSI. It is clearly demonstrated that with fixed FEC, substantial performance improvement is obtained for small E_S/N_I with a performance penalty for large E_S/N_I in Rician fading channels due to channel coding overheads. However, the JSCC approach eliminates such penalty through the joint adjustment of source and channel coding rates, such that acceptable end-to-end quality is maintained for reasonably small values of E_S/N_I as well as improved video quality for large values of E_S/N_I. Furthermore, JSCC provides a much more graceful

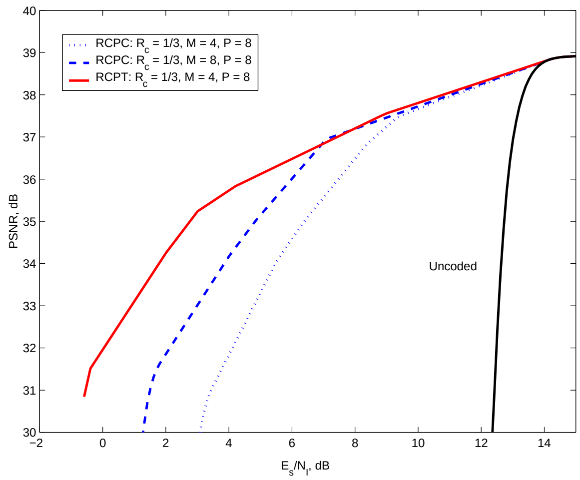

Fig. 6. Performance comparison of RTP-H.263+ packet video transmission over an AWGN channel using JSCC approach.

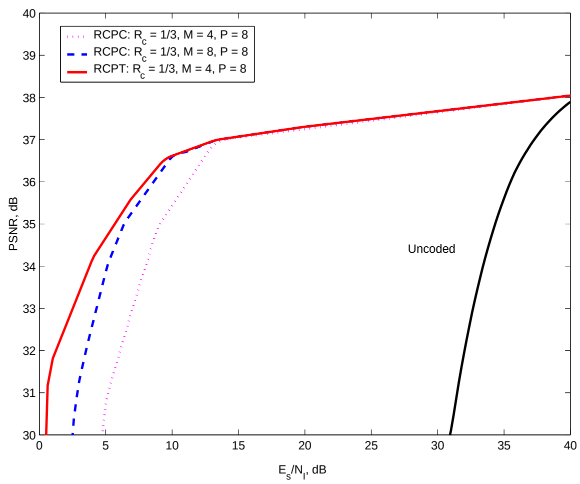

Fig. 7. Performance comparison of RTP-H.263+ packet video transmission over a Rician channel with $\zeta^2 = 7$ dB using JSCC approach.

end-to-end quality degradation compared to other fixed FEC schemes which is highly desirable for time-varying wireless communication networks.

Finally, we compare the performance of the turbo and convolutional coding schemes when used in conjunction with a JSCC approach in Fig.'s 6 and 7 for an AWGN channel and a Rician fading channel with $\zeta^2 = 7$ dB, respectively.[2] As can be seen, use of JSCC in either case can provide a more graceful pattern of quality degradation by keeping the video quality at an acceptable level for a much wider range of E_S/N_I. This is achieved by jointly selecting the channel and source coding rates based on the prevailing channel conditions, here represented by E_S/N_I. As a result, the overall system performance will be optimized for given channel conditions. The system employing RCPT-JSCC outperforms the corresponding RCPC-JSCC scheme with similar complexity due to the additional channel coding gain achieved using turbo codes. For fixed values of PSNR this amounts to about 2 dB in E_S/N_I over AWGN channels, or about 1.75 dB for the Rician fading channel with $\zeta^2 = 7$ dB, for small values of E_S/N_I.

Further objective as well as subjective results for the RCPT-JSCC system compared to uncoded systems and RCPC-JSCC systems are presented. The typical reconstructed video quality for selected channel conditions are demonstrated in Fig. 8. Figure 8 shows the 12^{th} frame of the Susie subsequence ($N = 12$) with overall rate held constant at $r_S = 128$ Ksps transmitted over a Rician fading channel with $\zeta^2 = 7$ dB for selected values of E_S/N_I. As shown in Fig. 8-(a) and (b), it is clear that extremely large E_S/N_I, certainly above 30 dB, is required for an uncoded system to achieve acceptable quality over a fading channel, resulting in extremely high interference to other users sharing the same network. Such operation is prohibitive in a multiuser wireless communication system where efficient low-power operation is the key to improved system capacity.

Instead, a JSCC system can avoid such problems and achieve graceful quality adjustment through the use of JSCC according to the prevailing channel conditions, resulting in substantially improved reconstructed video quality as demonstrated, for example, in Fig. 8-(c) and (d). Compared to Fig. 8-(a) and (b), a reconstructed video quality of $PSNR = 30.16$ dB can be obtained for the corresponding fading channel with E_S/N_I as low as 3 dB by using an RCPC-JSCC approach. Furthermore, to obtain reconstructed video with a reasonably better quality, say 34 dB, at the same E_S/N_I value of 3 dB an RCPT-JSCC system is required instead of the corresponding RCPC system. This offers the potential of significant improvements in system capacity. Considering that wireless network conditions are highly time-varying, such an adaptive feature is of significant advantage to end-user quality as well as system capacity.

V. Summary and Conclusion

We have described a packet-by-packet adaptive RCPT-JSCC approach to be used in conjunction with packetized H.263+ video over wireless IP networks. Both AWGN channel and Rician fading channel with Rayleigh fading as special case are evaluated for proposed JSCC scheme. The results clearly demonstrate that with appropriate RCPT-JSCC, packet video can be delivered over a wireless IP network with acceptable end-to-end quality while exhibiting a more graceful pattern of quality degradation compared to fixed channel coding schemes.

[2]Here we illustrate only the convex hull over all operational choices of R_s and R_c.

(a) $PSNR$ = 14.09 dB
E_S/N_I = 20 dB
Uncoded

(b) $PSNR$ = 19.21 dB
E_S/N_I = 30 dB
Uncoded

(c) $PSNR$ = 30.16 dB
E_S/N_I = 3 dB
RCPC-JSCC
$R_c = 1/3,\ M = 8,\ and\ P = 8$

(d) $PSNR$ = 34.22 dB
E_S/N_I = 3 dB
RCPT-JSCC
$R_c = 1/3,\ M = 4,\ and\ P = 8$

Fig. 8. The 12^{th} frame of Susie subsequence ($N = 12$) with overall rate held constant at $r_S = 128$ Ksps transmitted over a Rician fading channel with $\zeta^2 = 7$ dB.

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