[1] Nasruminallah; Hanzo, L., "Near-Capacity H.264 Multimedia Communications Using Iterative Joint Source-Channel Decoding," Communications Surveys & Tutorials, IEEE , vol.14, no.2, pp.538,564, Second Quarter 2012

Near-Capacity H.264 Multimedia Communications Using Iterative Joint Source-Channel Decoding

Nasruminallah and L. Hanzo

Abstract—**In this tutorial, a unified treatment of the topic of near capacity multimedia communication systems is offered, where we focus our attention not only on source and channel coding but also on their iterative decoding and transmission schemes. There is a paucity of up-to-date surveys and review articles on the unified treatment of the topic of near capcity multimedia communication systems using iterative detection aided joint source-channel decoding employing sophisticated transmission techniques - even though there is a plethora of papers on both iterative detection and video telephony. Hence this paper aims to fill the related gap in the literature.**

Index Terms—**Multimedia communications, H.264 video transmission, joint source-coding and channel coding, iterative detection, near-capacity wireless communications, EXIT charts, irregular channel codes, video standards.**

I. Advances in Multimedia Coding

ROBUST transmission of multimedia source coded streams over diverse wireless communication networks constitutes a challenging research topic [1, 2]. Recent advances in the world of telecommunication and multimedia systems resulted in the design of improved transmission techniques. However, bearing in mind the volume of information produced by high definition multimedia communication systems and the limited availability of unoccupied bandwidth at carrier frequencies, where beneficial propagation conditions prevail, the design of efficient multimedia systems at low bit-rate requires careful attention. Hence the design of improved coding techniques is important for the successful implementation of various multimedia communication systems, in order to reduce the amount of information required for flawless interactive multimedia communications [1]. An overview of advances in the field of video coding is presented in Table I.

The rest of the paper is organised as follows. A prelimenary introduction about H.264 video coding is provided in Section I-A followed by the details about our input source codec parameters in Section I-B. An overview of the iterative detection is provided in Section II. Section III portrays our employed video transmission scheme using Short Block Code (SBC) based iterative source-channel decoding. SBCs are used for achieving guaranteed convergence in soft-bit assisted iterative Joint Source-Channel Decoding (JSCD), which facilitates improved iterative Unequal Source-Symbol Probability Aided (USSPA) operations. The schematic of the proposed SBC based iterative source-channel decoding arrangement is presented in Section III-A. Section II-A provides the details about the iterative source channel decoding aided receivers, followed by the introduction about EXIT charts in Section II-B. The iterative convergence analysis using SBCs is provided in Section III-B, followed by the proposed SBCs in Section III-C. The performance of the proposed system is characterised with the aid of EXIT chart analysis in Section III-D and Section III-E.

Furthermore, the performance improvements of the proposed SBCs using RSM is described in Section IV along with its EXIT chart analysis and the overall performance results in Section IV-B. The use of RSM is to improve the convergence behaviour of the SBC coding upon incorporating additional redundancy and an improved minimum Hamming distance $d_{H,min}$. Additionally, SBC assisted UEP video using RSC codes and Sphere Packing (SP) modulated Differential Space Time Spreading (DSTS) along with the concepts of SP modulation and its performance results is presented in Section V. SP modulation is a specific scheme, which maintains the highest possible Euclidian distance of the modulated symbols, as detailed in [3]. DSTS is a low-complexity technique that does not require channel estimation, because it relies on noncoherent detection. This low-complexity detection is particularly important in the context of Multiple-Input Multiple-Output (MIMO) systems using N_T transmit and N_R receive antennas, which would require the estimation of (N_TxN_R) MIMO channels, hence substantially increasing both the cost and complexity of the receiver. Furthermore, the pilot-overhead required by the MIMO channel estimator may also be excessive. A serially concatenated three-stage scheme for near-capacity operation in terms of iteratively detected H.264 wireless video telephony is described in Section VI along with its EXIT chart analysis and system performance results. In contrast to the two-stage system constituted by a single iterative loop, the three-stage system employs two iterative loops, which exchange extrinsic information both between the inner and the intermediate decoder, as well as between the outer decoder and the intermediate decoder. The conventional two-stage turbo-detection schemes generally suffer from a Bit Error Rate (BER) floor, therefore the advantage of the three-stage design is to circumvent this deficiency by proposing an extra iteration in the turbo detection process. Finally, the paper is concluded with generic design guidelines and conclusions in Section VII and Section VIII, respectively.

A. The H.264 Video Coding Standard

Wireless systems are typically constrained owing to the availability of limited bandwidth and battery power. Therefore,

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The authors are with the School of ECS, University of Southampton, SO17 1BJ, UK (e-mail: {lh}@ecs.soton.ac.uk).

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PERFORMANCE ANALYSIS OF FORWARD ERROR CORRECTION AIDED P2P LIVE-VIDEO STREAMING

Nasru Minallah[*], Laiq Hasan[*] and Amjad Ullah Khattak[**]

ABSTRACT

Peer-2-Peer (P2P) technologies are rapidly advancing and have already proven successful in the form of various bandwidth demanding applications such as file-transfer and video streaming. The major advantage of the P2P technologies is its built-in scalability feature, relative to traditional client–server architecture. Therefore, keeping in view the large number of users involved in live video-streaming, P2P technology can be considered as a valuable tool aiming to broadcast live-video contents to a large number of users over the internet. Although, there is lot of research going on in analyzing the performance of P2P video streaming in heterogeneous networking scenarios, but to the best of our knowledge no work is done in analyzing the impact of Forward Error Correction (FEC) on the performance of P2P live-video streaming. In this paper we analyze the performance and characteristics of P2P live-video streaming on end-user quality of experience while employing FEC on the coded video packets. Additionally, in-order to carry out the performance analysis with reference to non-error protected benchmarker system, we utilized a realistic real-time French distributed Grid facility known as Grid'5000. More specifically, the performance of the system is evaluated while considering various realistic P2P video streaming constraints, such as limited playout delay, limited upload bandwidth and video rates. We concluded that the FEC will result in noticeable performance improvement in a scenario where each peer is connected with limited number of other peers that have the required contents for download. In such a situation the portion of the stream corrupted due to channel errors may not be recovered with the help of other accessible peers. However, if each peer has more connected peers with various alternative options to download the corrupted contents, then the employment of FEC may not have very noticeable advantage on the performance of the P2P video streaming system.

Key words: FEC,P2P,Live-video Streaming

INTRODUCTION

MOTIVATION AND BACKGROUND

Peer-2-Peer (P2P) video streaming is considered as an attractive candidate for the transmission of live or precoded video contents over the internet. In P2P video streaming, peers utilize their upload bandwidth by forwarding contents to their connected peers. As the cumulative upload band- width increases with increase in number of cooperating peers, therefore this application can potentially be scaled to large number of users. P2P video broadcast is considered cost effective and easy to deploy in comparison to traditional Internet Protocol (IP) based video multicast[1]. In addition to scalability the P2P paradigm brings a number of advantages, such as resilience and operability in heterogeneous communication networks. P2P video technology is currently implemented in the form various commercial applications, such as PPlive, SopCast, TVAnts and CoolStreming[1]. The P2P streaming applications are constrained by inherent delay to wait and upload a piece of content only after completely downloading that content. The impact of this inherent delay in the dissemination of contents is investigated in[2]. In[3] a methodology based on direct relation between the streaming peers upload bandwidth and its selection possibility as the destination peer is proposed. A distributed scheduling strategy which can distribute every chunk to all peers in a minimum number of steps, is presented in[4]. The different ways of frames to chunk grouping and the impact of limited playout delay on the video quality is presented in[5] in the context of P2P distribution. Furthermore, a novel P2P video streaming architecture is presented in[6]. The proposed architecture is named as TURIN stream (Totally pUsh, Robust, and efficieNt) designed to achieve low delay, robustness to peer churning, limited protocol overhead, and quality-of-service differentiation based on peers cooperation.

[*]Deptt. of Computer Systems Engg., University of Engineering & Technology, Peshawar.
[**]Deptt. of Electrical Engg. University of Engineering & Technology, Peshawar.