Perceptually Tuned Joint Source/Channel Coding for Video Streaming over DAB IP Networks.

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Abstract—The quality of service (QoS)-guarantee in real-time communication for multimedia applications is significantly important. In this paper, we propose a joint source channel coding (JSCC) scheme, which guarantees QoS requirements for video streaming over digital audio broadcast (DAB) IP tunneling networks. Joint source channel coding is an architectural framework for multimedia networks based on sub-streams or flows. Each sub-stream has a different QoS requirement, and we can control degree of satisfaction for the required QoS considering network traffic characteristics (e.g. delay, jitter, and packet loss). DAB networks based on EUREKA-147 is initially designed for digital audio broadcast, and it cannot meet the QoS requirements for the video streaming. For the reason, we should have a scheme to maximize the video quality under the poor network conditions. In this paper, firstly, we designed a classification scheme to partition video data into multiple sub-streams which have their own QoS requirements. We used MPEG-4 advanced visual coding (AVC) for our source coding, because it is already adopted in digital multimedia broadcast (K-DMB) standard in South Korea. Secondly, we designed a management (reservation and scheduling) scheme for sub-streams to support better perceptual video quality such as the bound of end-to-end jitter. We have shown that our joint source/channel coding scheme satisfies QoS requirements using real video experiment over DAB networks.

I. Introduction

Quality of Service(QoS) guarantee in real time communication for multimedia applications is significantly important, because more multimedia services, such as video on demand, video conference, and live video and audio broadcasting, require guaranteed services in the distributed heterogeneous environment. These applications require high bandwidth and real time. The real-time constraint is however not as critical as hard real-time system and is usually classified as soft real time. However, they impose other requirements which are peculiar to multimedia traffic, for example, bounded delay, bounded jitter, bounded[1] packet loss, and bounded synchronization gap. These requirements are defined as QoS of the real-time multimedia application [1], [2].

The problem is that how to support QoS in the lossy packet network. In signal processing, source coding and channel coding are used for signal transmission, where source coding removes signal redundancy as well as signal components that are subjectively unimportant, and channel coding adds controlled redundancy so that transmission impairment such as packet loss or bit errors can be reversed. Source coding is used in the service layer to increase the traffic-carrying capacity of transmission links within the bitway layer, and channel coding is used in the bitway layer to increase the traffic capacity of a network. Joint source/channel coding(JSCC) is a way to increase the traffic capacity of a network and also maximize a subjective image quality by exchanging some information between source coding and channel coding [3]-[5].

In Sec. II, we review the basic communication architecture which is used for the mobile networks model in this paper. In Sec. III, we designed JSCC, where we used the MPEG-4 advanced visual coding (AVC) as a baseline codec. We classifies packets based on motion information for grouped transport. In Sec. IV, we propose substream scheduling to maximize perceptual quality in our JSCC framework, show simulation results for our proposed scheme, and then compare the results with those of other JSCC schemes which use different scheduling schemes in Sec. V. In Sec. VI, we conclude with future research issues.

II. Review on JSCC for mobile networks Architecture.

The TCP protocol supports effective and stable network services with flow control and error control, congestion control. But, TCP protocol is more packet loss in wireless networks than in wired networks. Also, Packet-based mobile communication networks inevitably introduce three types of impairment: packet

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