Investigation of Multimedia Transmission Property-Based Wireless Multimedia Technologies

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Abstract - With the emergence of broadband wireless networks and increasing demand of multimedia information on the internet, wireless multimedia services are foreseen to become widely deployed in the next decade. However, wireless channels are unreliable and the channel bandwidth varies with time, which may cause severe degradation to the quality of the transmitted multimedia content. In addition, for image and video transmission, the heterogeneity of receivers makes it difficult to achieve efficiency and flexibility. The issue of guaranteeing the quality of multimedia is to be well solved. In this research topic, we aim to study distributed transmission and error protection schemes for multimedia transmission over wireless networks based on the multimedia transmission property. The main issues to be studied we are interested include: 1) Analyzing multimedia transmission property; 2) Building efficient analytical models for multimedia transmission property; 3) Designing distributed, adaptive and fine grained transmission and error protection schemes.

Index Terms - wireless, quality of service, scalable video, adaptive services

I. PREFACE

The importance of visual communications has increased tremendously in the last few decades. The progress in microelectronics and computer technology, together with the creation of networks operating with various channel capacities, is the basis of an infrastructure of a new era of telecommunications. New applications are preparing a revolution in the everyday life of our modern society. Emerging applications such as desktop video conferencing, mobile terminals, and Internet-based audio-visual communications have a great impact on modern professional life, education, and entertainment. Visual information is one of the richest but also most bandwidth-consuming modes of communication.

In my personal opinion, four subjects will determine the future of communications, these are: Multimedia Applications, Mobile and Wireless, IP & Networks and Photonic Networks. Therefore as a communication engineer to stay ahead with the most modern communications technologies, it is advisable to stay tuned with one of those subjects. Our project is emphasizing mainly on multimedia transmission technologies through wireless networks. Modern signal processing techniques and network adaptation techniques shall be used to design multimedia system with improved performance and flexibility. The following few pages give you an introduction to the project and will describe for you some of the major hot issues in the project. Furthermore, simulation results are presented to prove our proposed techniques. The work is sponsored by Project 60202005 Supported by National Natural Science Foundation of China.

II. INTRODUCTION TO OUR WORK

A. Definition of Multimedia

From the user's perspective, multimedia is computer information that can be represented through audio and/or video, in addition to text, image, graphics and animation.

B. Multimedia Transmission

In current and future communications networks three challenges arise when dealing with multimedia transmission, these are;

Unreliability

In addition to the noise arise in the wire-lined communication network,s wireless communication networks, such as mobile cellular networks are typically

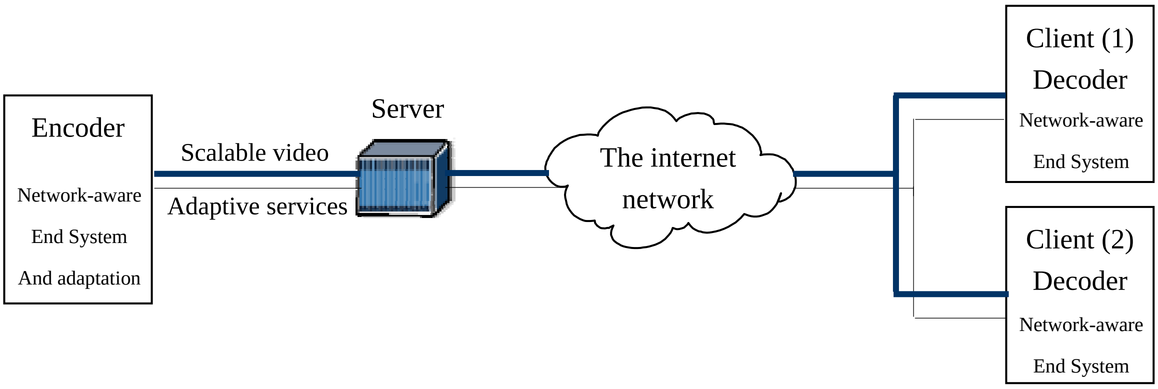

Fig. 1 System configuration for internet streaming of video
(Two client's example)

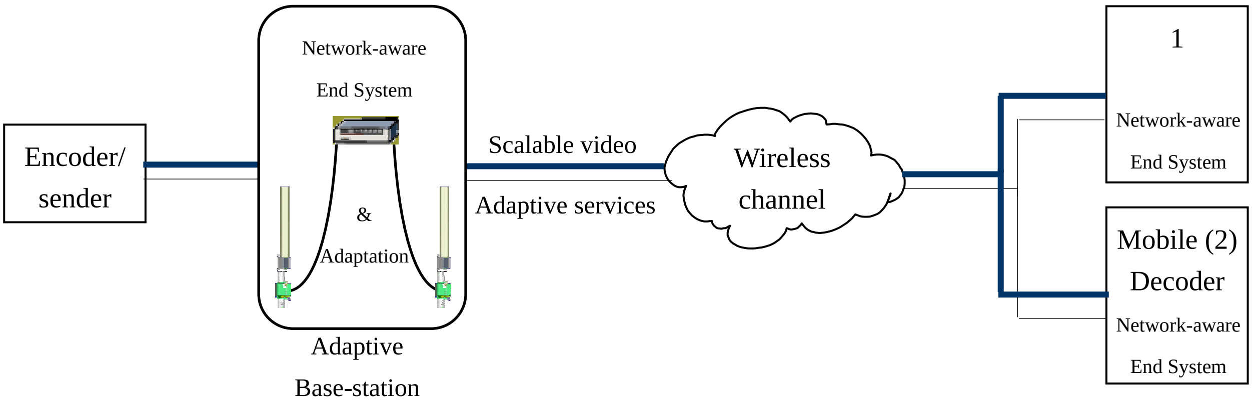

Fig. 2 System configuration for video streaming over cellular mobile network
(Two subscriber's example)

much more noisy and have both small-fade (multipath) and large-scale (shadowing) fades, making the BER very high. The resulting bit errors can have devastating effect on video quality. Therefore it is crucial to design robust transmission mechanisms for video over such networks.

Bandwidth fluctuations
When a mobile terminal moves between different networks have different configurations, for example, from WLAN to cellular network. Another example is when a mobile moves from one cell to another creating hand-off problems. Changing distance between the mobile terminal and the base-station also cause such bandwidth fluctuations and can also change the capacity of the transmission channel. In addition, the throughput of wireless transmission can also change because of multipath fading, co-channel interference, and noise disturbances. Consequently, bandwidth fluctuations pose a serious problem for real-time video transmission over wireless networks. So, remember the three parameters that are effected by bandwidth fluctuations; BER, channel capacity, and throughput.

Heterogeneity
Multicast delivery of real-time video uses point-to-multipoint transmission, where one sender and multiple receivers are involved. For applications such as video conferencing, delivery using multicast can achieve higher bandwidth efficiency while unicast delivery of such applications is inefficient. However, think once more, the efficiency of multicast is achieved at the cost of losing the service flexibility of unicast (i.e. in unicast, each receiver can individually select the QoS required). Such lack of flexibility in multicast can be problematic under heterogeneous environment, where receivers could be different in terms of latency requirements, visual quality requirements, processing capabilities, power limitations (in wired and wireless) and bandwidth limitations.

C. Wireless Multimedia Technologies
Now to address the above issues, several techniques are

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proposed, the main important are:

Scalable Video Coding

Scalable video coding is to compress a raw video sequence into multiple substreams, each of which represents different quality, image size, or frame rate. The intention of scalability is to provide interoperability between different services and to support receivers flexibility with different display capabilities. Scalability methods have recently appeared to cope with the problems of heterogeneity of receivers. Different algorithms have already been studied to perform scalability on both image and video [1, 2]. There are different modes of scalability: temporal scalability, SNR scalability, spatial scalability and object-based scalability. For more information, please refer to [3]. Video scalability sometimes referred as video layering, that's to assign different information streams to different receivers. Scalability has been standardized in MPEG-2 video compression that have two modes of scalability (SNR and spatial scalabilities). The most modern called FGS-Fine Granularity Scalability is standardized within MPEG-4 video compression standard [4]. There are many advantages of using FGS for internet video streaming, it allows separation of encoding and transmission, the server can transmit enhancement layer at any bit rate without transcoding, it enables video broadcast on the internet to reach a large audience and provide a solution to the video server overload problem.

Network-aware Adaptation of End Systems

Most current video applications are insensitive to changing network condition. In a time-varying wireless environment, however, video applications must be robust and adaptive in the presence of QoS fluctuations (unreliability and bandwidth fluctuations). To address this issue a new approach called network-aware adaptation has been proposed. As the name indicates, it contains two elements: network awareness and adaptation. Network awareness refers to having knowledge about the current status of underlying network resources (e.g. available bandwidth and bit error conditions). Adaptation, on other hand, is to adapt video streams based on network status. It has been shown that network-aware adaptation of end systems can significantly improve performance of some applications [5].

Adaptive QoS Support from Networks

Adaptive QoS support or adaptive services are a technique to adapt video streams during periods of QoS fluctuations and handoffs. Adaptive services have been demonstrated to be able to effectively mitigate fluctuations of resource availability in wireless networks. There have been many proposals on adaptive approaches and services in the literature, which include an "adaptive reserved service" framework [6], a wireless adaptive mobile information system (WAMIS) [7], an adaptive service based QoS bounds and revenue [8], an adaptive framework targeted at end-to-end QoS provisioning [9], a utility-fair adaptive service [10], a framework for soft QoS control [11], a teleservice model based on an adaptive QoS paraidigm [12], an adaptive QoS framework called AQuaFWiN [13], and an adaptive QoS management architecture [14].

III. WHAT ARE WE WORKING ON?

Fig. 1 illustrates the adaptive framework. On the sender side, raw video is compressed by a scalable video encoder. Then the compressed video is sent to the networks by a network-aware end system, which monitors the network status and adapts the video streams accordingly. On the receiver side, a network-aware end system can sense the network status and coordinate with the sender side in video transport. The received packets are decoded by a scalable video decoder. Fig. 2 proposes the same technique, but for mobile cellular network. Fig. 1 provides adaptive QoS support to the scalable video. On the receiver side, network-aware end systems coordinate with the network in video transport.

In this project many issues are being investigated with combined technologies on wireless communication, networks transmission and QoS, and scalable video coding. As a result, a distributed, flexible, scalable and adaptive wireless multimedia communications system will be designed. These issues are:

- Design an adaptive QoS framework for wireless networks, analyze priority schemes to support QoS in cellular system and wireless LAN networks, including queue management, weighted scheduling and MAC layer based priority mechanisms.
- Analyze the system capacity of cellular networks and wireless LANs supporting QoS, optimize the overall system performance on capacity and QoS, and design distributed admission control and resource management mechanisms for wireless networks.
- Design adaptive network aware end systems, analyze the multimedia transmission property, and design a distributed and adaptive multimedia transmission system.
- Investigate the technologies for programmable wireless nodes, design dynamic applications oriented models for wireless network, and design a framework for programmable wireless nodes.

IV. PROPOSED SOLUTIONS

A. Multimedia Scalability

Scalability refers to the ability to modify the resolution and/or bit rate associated with an already compressed data source in order to satisfy requirements which could not be foreseen at the time of compression [15]. However, it is necessary to maintain both high compression and scalability to meet the demands of emerging digital video applications. Much of research work has done toward achieving these goals. The successfulness of the existing image compression techniques can be useful for video compression developments. Video compression can be achieved through generalizing the image compression technique to 3-dim signals. Another way to compress video is to define a hybrid approach based on motion compensation. In [16], as an example, block-based motion-compensation followed by 3-D wavelet (packet) decomposition was used. The output coefficients of the wavelet packets are encoded using the so called tri-zerotrees TRI-ZTR data structure, which is an extension of the 2-D zero tree coding. The proposed scheme is capable of supporting multirate video scalability with fine granularity. The bit rate scalability is arbitrary assigned to the decomposition branches, after choosing a certain display spatial resolution and frame rate. Changing the bit rate assignment with constant frame rate and video resolution will considerably affect the distortion level of the compressed video. A hierarchal motion estimation procedure is proposed in [17], and used to provide video layering, since the difference between levels represents additional refinement information. And it was found that centred cubic spline filters are the optimum solution for hierarchal motion estimation constraints proposed. At higher layers (transport and data link layer in the OSI layer model) [18] introduces an adaptive QoS framework and error control procedure for network-aware end systems. Due to the heterogeneity of receivers in practical wireless and IP networks, mismatches can occur at the receivers caused by the difference between the allocated video subscription level and the dynamic requirements of the receivers. Since the number of layers in most compression standards are practically limited. A more recent research is done by J. Liu and others [19], to cup with the problem above. He optimizes the rate allocation among the layers with the objective of maximizing the expected fairness index for all the receivers. In this paper, we will use Multiplexed Wavelet Transform (MWT) for video coding that was recently proposed in [20]. A bit rate allocation procedure can easily apply without requiring strict design constraints like other methods do. In addition, the number of channels can be simply assigned according to network need, to cup with the problem of mismatches at the receivers. A reason of using MWT is that MWT provides an efficient way for multiresolution analysis of signals. When applying to videos the multiresolution representation will give scalability. In other words with MWT, a piece of video signal can be transmitted at different resolution, depending on the receiving devices as well as network environment. Among different possibilities of multiresolution analysis and synthesis, wavelet functions are the most adapted to these purposes due to their scaling and translation properties.

B. The Multiplexed Wavelet Transform

As the name indicated, MWT may be realized by multiplexing P ordinary wavelet analysis and synthesis filterbanks as shown in Fig. 3. Assume that the wavelet transforms in the MWT are not identical, that is $\{\Psi_{n,m,q}(k)\}$, are the complete and othonormal wavelet functions. In this paper, we will define the multiplexed wavelets as follows [20];

$$\zeta_{n,m,q}(k) = \sum_s \Psi_{n,m,q}(s)\delta(k - sP - q) \quad (2)$$

where n=1,2,…;q=0,1,…,P-1. The MWT of a signal $f(k)$ is defined as the set of coefficients

$$F_{n,m,q} = \sum_k f(k)\zeta_{n,m,q}(k) \quad (3)$$

and the multiplexed wavelet expansion (inverse MWT) is given by the following sum

$$f(k) = \sum_n \sum_m \sum_{q=0}^{P-1} F_{n,m,q}\Psi_{n,m,q}(k) \quad (4)$$

Notice that the multiplexed wavelets $\zeta_{n,m,q}$ are defined over three dimensions (indices). The first index corresponds to scale and the remaining two to time. The orthogonality and completeness of the multiplexed wavelet sets are inherited from the ordinary wavelet sets [6]. Notice that the multiplexed wavelet $\zeta_{n,m,q}(k)$ is obtained from wavelet $\Psi_{m,n}(k)$ by inserting P-1 zeros between successive samples and shifting by q samples. Now, in the same way one can define the multiplexed scaling sequences, $\{\phi_{0,m,q}(k)\}$ as follows;

$$\vartheta_{0,m,q}(k) = \sum_s \delta(s - q - kP)\phi_{0,m,q}(s) \quad (1)$$

The frequency spectrum of the multiplexed scaling sequence consists of multiple frequency bands that are centered on the harmonics at $\nu_j = j/P$, j=0,1,…,P/2. On the other hand, the frequency spectra of multiplexed wavelets have a multiple-band structure consisting of sidebands of the harmonics. As P grows, these bands narrow and get closer to the harmonics.

C. Multilayer Video using MWT

As mentioned above, MWT can achieve both multirate and multiresolution scalability simultaneously during the bit stream extraction process. Suppose we encode a 30 fps CIF-format video sequence of size 352 pixels*288lines, and use P=3 and 8-frames video sequence. This means that we can have three possible spatial resolutions, four different temporal resolutions and arbitrary bit rates at the decoder. In the MWT encoders shown in Fig. 3, motion compensation is applied in advance of MWT decomposition. Here, the first frame in a group of frames (GOF) is used as a reference frame, and succeeding frames in the GOF are then "mapped / registered" with respect to the reference frame by estimating a set of block motion vectors for each frame. The output is then decomposed to produce the expansion coefficients. The expansion coefficients (1) are quantized at different layers and encoded using variable length coder (VLC). The ordinary WT blocks, each implemented as an N+1 channel iterated filter bank, are represented as one input N+1 output devices. Outputs labled n=1,…,N correspond to wavelet coefficients at scale level n and output N+1 scaling coefficients. In order to compute the MWT, the input signal $f(k)$ is demultiplexed over P channels that are in the wavelet transform blocks. Different layers have different number of bits allocated to each quantizer. And at each layer, the number of stages "w" of the dyadic

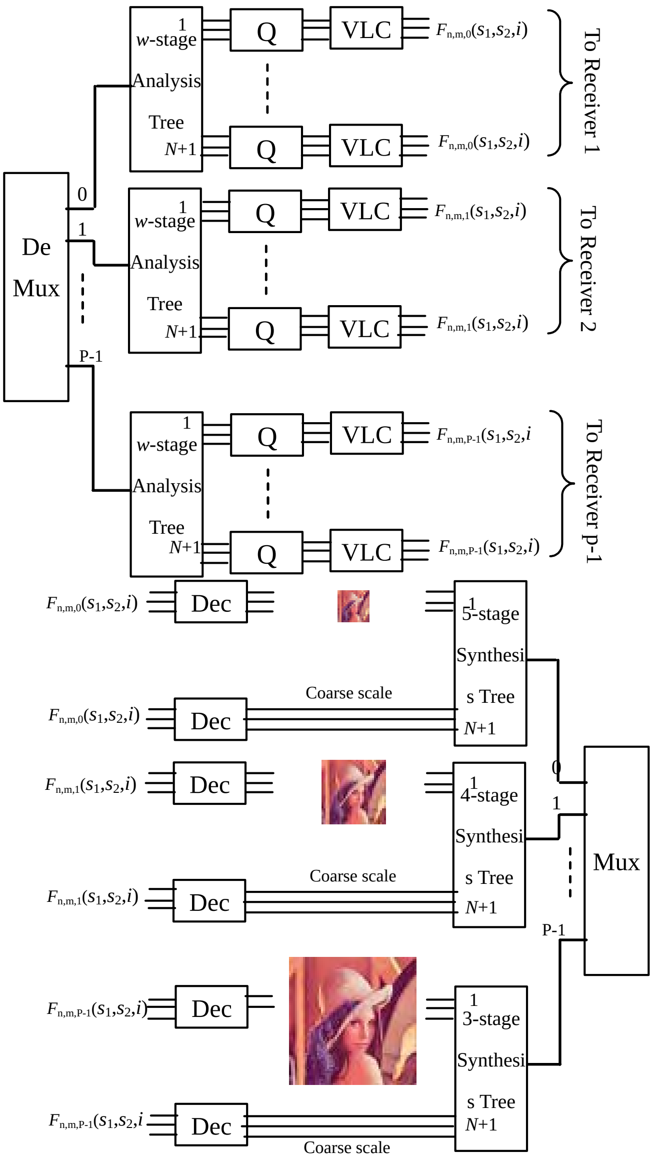

Fig. 3 Block diagram of the MWT coder

wavelet transform is adapted according to the receiver capability. The multiplexed wavelets $\{\zeta_{n,m,q}\}$ can also be designed for optimum solution for a given network. Our problem here is to expand the definition of MWT to include two and higher-dimensional signals, for example the colored RGB signal. The RGB colored image, is a three-dimensional signal, with R,G and B are two dimensional.

We shall apply a 2-dim wavelet transform at each stage individually to R,G and B. The output is also a 2-dim R,G and B signals (s_1 and s_2 represents the special location of the scaling coefficients $F_{0,m,q}(s_1,s_2,i)$ and wavelet coefficients $F_{n,m,q}(s_1,s_2,i)$), but their sizes are halved after each stage. The index "i" have the values 1,2 and 3, corresponding to R, G and B. Such that the output RGB at each stage "w" is a 3-dim signal can be used to construct the original image at different resolution. This video layering method provides many advantages; first, by using the MWT, different adaptive techniques can be designed to adapt the value of "w" and the number of resolutions "P" in response to the network requirements. Second, by choosing the suitable wavelet family, bandwidth efficiency enhancement can be achieved. These advantages are gained by using MWT come at the cost of relatively more number of operations required, and hence, more system complexity. An example is shown in Fig. 3, where three variant receivers receive different video quality. The compressed video streams can adapt to three levels of bandwidth usage. To determine the optimum bit allocation is to minimize the output error variance σ^2=E$\{e^2\}$, where $e(k)$=$fc(k)$-$f(k)$, $\{fc(k)\}$ is the decoder output samples.

D. Joint Source-Channel Coding

Delivery of real-time video typically has QoS requirements,e.g., bandwidth, delay and error requirements. First, video transmission usually has minimum bandwidth requirements (e.g., 28 kb/s) to achieve acceptable presentation quality. Second, real-time video has strict delay constraints (e.g., 1 second). This is because real-time video must be played out continuously. If the video packet does not arrive timely, the playout process will pause, which is annoying to human eyes. Third, video applications typically impose upper limits on bit error rate (BER) (e.g., 1%) since too many bit errors would seriously degrade the video presentation quality. However, unreliability and bandwidth fluctuations of wireless channels can cause severe degradation to video quality. Furthermore, for video multicast, heterogeneity of receivers makes it difficult to achieve efficiency and flexibility.

To solve above problems, different technologies have

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been investigated in source coding and channel coding areas. Since wireless channels are time-varying, it is desired that image/video coding rate can be adaptively adjusted. In source coding area, scalable image/video coding is a common approach for providing rate adaptation. An efficient scalable image and video coding methods have been discussed in the past section based on MWT.

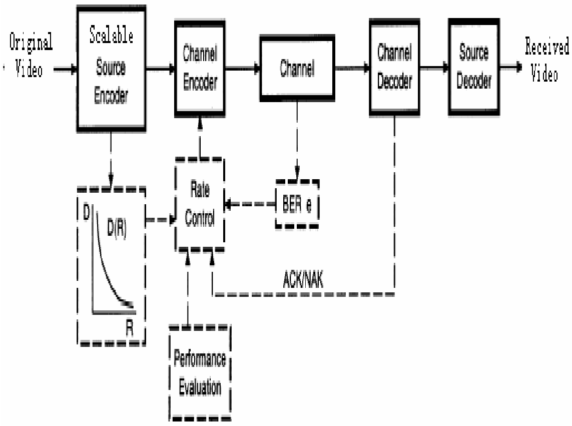

Fig. 4 System architecture of Joint Source-Channel Coding for wireless video transmission

In traditional Shannon theorem for communication one can separately design the source and channel coder to achieve the optimal performance of the over all system. This applies to source and channels that are memory less and stationary. However, in most real world applications, both the source signal and the channel environments vary rapidly, hence non-stationary. So separating design method cannot achieve optimal multimedia transmission. Joint source-channel coding technology (JSCC) is proposed to solve this problem and has been widely used for multimedia transmission quality optimization. Fig. 4 shows the system architecture of Joint Source-Channel coding for wireless video transmission.

JSCC requires too more functional entities to exchange information and cooperate with each other, this prevents JSCC to get much more progress. Currently JSCC is coarse-granularity and integrated. Coarse-granularity refers to the method of differentiating multimedia streams. The common approach is choosing different encoding methods to encoding the multimedia source according to channel conditions or applying different error protection schemas to different part of multimedia stream. However, error protection is based on subjective experience without theoretical analysis. Integrated JSCC refers to determine the joint encoding strategies according to channel conditions and the characteristics of multimedia sources. All network nodes in JSCC system should provide channel coding and channel condition information, while multimedia source should keep accurate channel information and deliver Unequal Error Protection (UEP) schemas to wireless channel. When wireless channel changes frequently, the cost for JSCC information transmission raises and the channel conditions are not real-time. Thus integrated JSCC is not very useful in time-varying wireless channel.

Our research will firstly investigate multimedia transmission property, which demonstrates the relationship between multimedia reconstructed quality over error-prone networks and channel errors (Packet Loss Rate or Bit Error Rate). Based on efficient analytical model for multimedia transmission property, we can then design distributed, adaptive and fine grained transmission and error protection schemes. Such fine-granularity and distributed JSCC technology can solve the problems of current JSCC schemas.

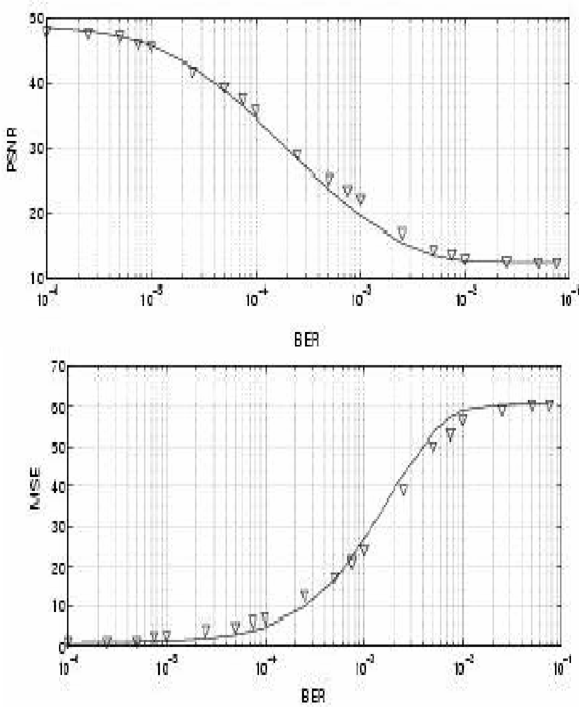

Fig.5 Image quality versus channel bit error ratio

V. PROGRESS OF OUR RESEARCH

A. JPEG2000 Encoded Image Transmission Property Model

JPEG2000 is a new scalable image standard. We analyzed the performance of error resilience tools in JPEG2000 at first, then presented an analytical model to estimate the quality of JPEG2000 encoded image transmitted over wireless channel. We validated the effectiveness of the analytical model by simulation results, which are shown in Fig. 5. Furthermore, analytical model is utilized by the base station to design efficient unequally error protection (UEP) schemas for

JPEG2000 image transmission. With the analytical

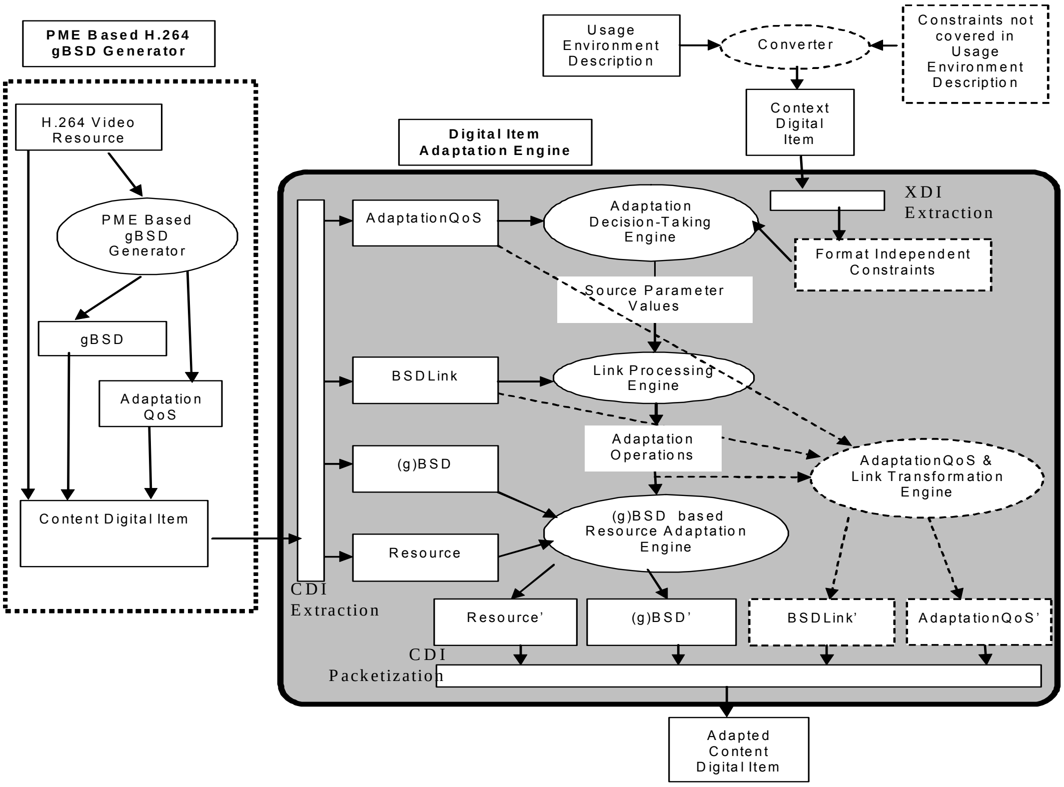

Fig.6 System Architecture of PME based H.264 video adaptation for Digital Item Adaptation

model, a utility function is defined to make a tradeoff between the image quality and the cost for transmitting the image over wireless channel, and chose which kind of UEP schemas to employ to reach the optimal image transmission quality.

B. MPEG-21 Digital Item Adaptation System for H.264/AVC Video

The new MPEG-21 standard defines a multimedia framework to enable transparent and augmented use of multimedia resources across heterogeneous networks and devices used by different communities. We incorporated the Perceived Motion Energy (PME) Model into the proposed MPEG-21 Digital Item Adaptation Framework for frame dropping in H.264/AVC encoded video adaptation. There are two advantages of this work; one is the use of PME model to reduce the viewer's perceived motion jitter due to frame dropping to a minimum. The other is the adaptation nodes can easily apply frame dropping operations without knowledge of detailed encoding syntax of H.264/AVC videos.

VI. OPEN ISSUES AND FUTURE WORK

- Transmission Property Model for H.264/AVC Video: Because the reconstructed video quality related to encoding methods, error resilience and error concealment, a general, accurate and real-time video transmission model is relative complex and still an open issue.
- Distributed unequal error protection schemes based on Multimedia Transmission Property Model: It involves using JPEG2000 image and H.264/AVC video transmission property model to design distributed unequal error protection schemas on wireless base station. Both unicast and multicast situation will be investigated.
- Wireless Video Transmission and Adaptation Prototype System: Integrated previously motioned technologies with MPEG-21 DIA Framework to achieve transparent interoperable access to distributed multimedia content. How to delivery multimedia transmission property to network nodes and how to apply JSCC schemas are still in work. Finally, a prototype system based on MPEG-21 Resource Delivery Test Bed will be expected.

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