NEAR EAST UNIVERSITY

Faculty of Engineering

Department of Electrical and Electronic Engineering

Error-Resilient H.263 Video Coding for Wideband CDMA Systems

Graduation Project
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Student: Achraf Kaddoura (20021943)

Supervisor: Mr. Izzet Agoren

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ABSTRACT

Multimedia services for the third generation communication systems have drawn much attention recently. Rich multimedia applications, such as video conferencing, telemedicine, and video-on-demand, are proposed and will be implemented in the near future owing to the capability of broadband mobile communications. However, the interfaces and noise in wireless channels become major obstacles in the implementation of bit-error-rate sensitive applications. In this project, a novel error-resilient H.263 video codec, cooperating with smart antenna techniques, is introduced which provides a robust wireless video transmission.

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INTRODUCTION

The multimedia nowadays is one of the most improving technologies that the world is focusing on. As the world is developing multimedia is also being developed and improved rapidly until we reached to what is called the multimedia of the third generation which is proposed and will be implemented in the near future. In this chapter the multimedia services for the third generation communication which have drawn much attention recently will be introduced.

To begin with, the term multimedia can be defined as the combination of different types of media into one. With the aid of digital technology, text, graphics, animation, audio and video can be combined into new interactive media forms delivered by computers and other electronic devices. The evolution of the multimedia applications can be traced through three major stages. First, even prior to the deployment of delivery-network infrastructure, stand-alone applications (e.g., video arcade games) and CD-ROM-based applications had successfully integrated multiple media, mostly in the form of games, entertainment, and educational materials.

The integration of multiple media, e.g., voice, video, image, and data, provides an effective means of communication to the users of various services. Because of the advances in computer and communication technologies, creating sophisticated multimedia user interfaces is no longer limited to special-purpose applications. The popularity of the World Wide Web, where most applications currently utilize images and data, is adequate testimony to this fact. The increasing power of electronic circuitry in workstations, personal computers, and consumer electronics, in conjunction with the decreasing cost of high-bandwidth and low-latency communication, have created a large momentum to develop sophisticated multimedia applications as well as to provide new types of services to businesses and homes. Next, high expectations of technological breakthroughs to make available lower-cost delivery bandwidth (as needed for sophisticated multimedia applications) created a tremendous excitement in the area of multimedia. The potential for the convergence of multiple services (e.g., TV, movie, and telephone) had stirred up the marketplace.

Almost every day, newspaper headlines announced new field trials and potential mergers of corporations. Nowadays the focus in multimedia is on the creation of large-scale video servers and delivery infrastructure that would be capable of delivering thousands of simultaneous high-quality video streams to homes and business. However, the economics of the marketplace has run counter to these high expectations. For example, in movie-on-demand applications, the cost of storage of a large video library and the cost of delivery bandwidth for a two-hour video per customer were found to be prohibitively expensive in comparison with traditional movie rentals. In addition, the infrastructure for delivering high-quality video was mostly unavailable, particularly to the home. This issue is referred to as the last-mile bandwidth problem. The popularity of the Web, which can use very low-bandwidth networks in the last mile, and the lack of deployment of large-scale video servers have laid the groundwork for the current stage of multimedia. Applications using high-bandwidth multimedia streams will be deployed in enterprise (high-bandwidth-network) environments and more slowly (as higher-bandwidth infrastructure is deployed) for residential consumers.

The primary focus of current research activities is on the creation of multimedia content that can be delivered inexpensively by means of the existing low-bandwidth networking infrastructure as well as on the development of platform-independent applications that can adapt to heterogeneity in environments arising from the differences in capabilities of system components and end-user devices. The new forms of content fall into two categories: low-bandwidth, real-time conferencing, and delivery of composite multimedia documents consisting of images, text, and possibly short audio and video clips. The business and research focus has shifted from just video or audio quality to information delivery. Video-conferencing applications, for example, are replacing special video-conferencing rooms (which have high operational cost) used in many business environments, and bringing the desired functionality to the desks.

As it was mentioned above these rich multimedia applications, such as video conferencing and video on demand are proposed and will be implemented in the near future owing to the capability of broadband mobile communications. However, the interference and noise in the wireless channel become major obstacles in the implementation of bit-error-rate sensitive applications. In this paper, the H.263 video codec, cooperating with smart antenna techniques, is introduced which provides a robust wireless video transmission.

The H.263 standard, published by the International Telecommunications Union (ITU), supports video compression (coding) for video-conferencing and video-telephony applications. Videoconferencing and video telephony have a wide range of applications such as: desktop and room conferencing, video over the Internet and over telephone lines, telemedicine etc. In each case video information (and perhaps audio as well) is transmitted over telecommunications links, including networks, telephone lines, ISDN and radio. Video has a high "bandwidth" (i.e. many bytes of information per second) and so these applications require video compression or video coding technology to reduce the bandwidth before transmission. A number of video coding standards exist, each of which is designed for a particular type of application: for example, JPEG for still images, MPEG2 for digital television and H.261 for ISDN video conferencing. H.263 is aimed particularly at video coding for low bit rates (typically 20-30kbps and above). The H.263 standard specifies the requirements for a video encoder and decoder. It does not describe the encoder or decoder themselves: instead, it specifies the format and content of the encoded (compressed) stream. The error resilient H.263 video codec system will be discussed in details in chapter three. An end-to-end reverse link wideband CDMA (WCDNA) system simulation is used by signal processing workstation (SPW), with the performance measured in terms of bit error rates and power of signal to noise ratio (PSNR).

To summarize, in this chapter the definition of multimedia and a brief discussion about evolution of multimedia through several generations was included. In addition, brief information about H.263 was mentioned. In the next chapters the wireless communication systems will be introduced and the error resilient H.263 and its improvements will be discussed in details. The smart antenna techniques which cooperates with error resilient H.263 to provide a robust wireless video transmission will be discussed as well.

CHAPTER ONE
BACKGROUND

1.1 Overview

On the approach of the third generation communication systems, many new wireless applications are expected. The capability of multimedia transmission is no doubt one of the most attractive. More bandwidth consuming applications, such as high quality image transmission, bi-directional video conferencing, telemedicine, and video on demand, many soon dominate the market. However many techniques used widely via the wired transmission, may not be suitable or optimal solutions for wireless communications. For, instance most existing video compression standards are developed for the nearly error-free environment. Whereas in wireless communications, encountering imperfect channels, much data error rates (compare with wired communications) are expected. The current standards for video compression, unfortunately, cannot be directly applied in such error-phone environments. In this chapter a list of improvement techniques will be discussed to recover the imperfectness and the errors of the wireless channels.

1.2 Third Generation Wireless Networks

1.2.1 Definition

Image this scenario: you check your e-mail, downloads a file for the next day's video conference while calling your friend to pick you up from airport when you back from the business trip. Sounds like typical things you would do when you are in a hotel which offer Internet access. But wait, you are not in any hotel, you are in a construction trailer in a remote region of China, hundreds of miles from the nearest village.

Welcome to the world of Third Generation (3G) Global Wireless Network. Humans have long dreamed of possessing the capability to communicate with each other anytime, anywhere. Driven by information technology, the dream is expected to become reality in a few years. 3G wireless networks is a new generation of advanced harmonized global multimedia wireless communications system which combine high speed mobile access with Internet Protocol (IP) based services.

1.2.2History

Historically, the 1st generation wireless communication system refers to the usage of the analog cordless telephone systems, which were introduced in the 1980s. They had very limited features, poor voice quality, limited radio coverage and low data transmission speed to 9.6kbs.

In the early of 1990's, the 2nd generation mobile communication systems came up with the digital technology. By digitizing signals, these 2nd generation systems supported some additional service such as three-way calling and text transmission. And the data rates increased to 14.4kbs. The leading 2nd generation technology standards are Global Systems for Mobile communications (GSM), Code Division Multiple Access (CDMA) and Time Division Multiple Access (TDMA).

Though the 3G technologies will be firstly deployed in three most industrialized regions: North America, Europe and Japan, probably this technological revolution would bring more benefits to China. Because of:

First, the existing fixed cable telecommunication network system of China is far behind the developed countries'. Currently, the telephone penetration is just 12.2% in China, compare to 96.6% of U.S.A and 98% of Japan. If China is going to follow the formal steps to build a mature wire-line network as the western countries did, it will take another 10 -- 15 years and spend astronomical amount of money to achieve the goal in this world most popular country. The 3G provide a very good chance for China to skip the cable network stage therefore save the huge amount of money for other useful projects.

Second, due to the low telephone penetration, China's Internet development is very slow. Most of the Internet users of China use dial-up system to log on Internet, the existing telephone system is narrow bandwidth and over-crowded, making China Internet users very difficult to connect with Internet. This technological limitation discouraged many potential Internet users, leading to the very low Internet users penetration in China is 0.4% of 1999 and estimated figure is no more than 2% in five years. The revolutionary 3G will free China Internet establishment from the physical constrain of fixed telephone system. China Internet users will have a competitive alternative to log on Internet. China would expect dramatically increasing in Internet usage penetration after the establishment of 3G wireless networks.

Finally, Chinese are ready to accept the 3G technologies because the 1st and 2nd wireless generation technologies have developed very well in China. Compare to the low establishing speed of fixed-line telephone system, mobile telephone usage is growing rapidly in China. In 1999, more than a million cell phones were sold each month in China. Chinese are used to the wireless communication. So there won't be any social obstacles impeding the application of 3G network. Plus, the current wireless telecommunication standard GSM will be integrated in the 3G network very well technically.

1.2.3 Standards

The main standards of 3G wireless network technology will require:

- Very high data transmission rate: 144Kbs in all fast-mobility environments (ten times faster than that of 2ndG) and 2Mbs in low-mobility and indoor environments.
- Enhanced communications: good voice quality (comparable to wire-line quality), seamless roaming across multiple networks (allowing global roaming).
- Several simultaneous services and multimedia accessibility.
- Security comparable to Integral Services Digital Network (ISDN).
- Support to both circuit-switched (1st / 2nd generation technology) and packet-based services (such as IP traffic and real time video).
- Affordable price to mass market.

1.2.4 Imperfect Channel

Third generation (3G) wireless systems are expected to provide high bit rate data services suitable for transmitting multimedia information. At the same time, they are to operate reliably in different types of environments: macro, micro, and Pico cellular; urban, suburban, and rural; indoor and outdoor. In other words, the 3G systems are expected to offer better quality and coverage, be more power and bandwidth efficient, and be deployed in diverse environments. Nevertheless, 3G systems, despite their enhanced features, are encountering imperfect channels and a high data error rates are expected. They are severely bandwidth-constrained particularly for handling video communication traffic. Whilst current methods of video compression accelerate transmission by reducing the number of bits to be transmitted over the network, they have the unfortunate trade-off of increasing signal sensitivity to transmission errors.

1.2.5 Improvement of the Channel

Many research results have shown some improvements in reducing the transmission errors by developing more adequate source and channel coding, building more robust communication protocols, and moderating the imperfect channel condition. One effective method of protecting the compressed video signal is to split the coded video signal to number of separate bit streams where each can be transmitted via a separate channel having a different degree of error protection. The bit stream splitting can be accomplished by taking into consideration the perceptual significance of coded video, where better protection is provided for the transmission of the more error-sensitive bits. This greater performance is achieved by incorporating several such strategies. An error-resilient H.263 video coding scheme is proposed for the wireless communication protocols, in addition, by taking advantage of the dedicated pilot channel in the WCDMA, a smart antenna technique is used to combat the fading channel, and also a robust decoder equipped with error recovery mechanisms is built in the receiver.

1.3 Protection Methods against Channel Errors

The transmission of data over wireless channels is challenging due to a number of factors such as high bit rates, delay, and loss sensitivity. As a result, many solutions have been proposed in different perspectives. From channel coding perspective, Forward Error Correction (FEC) techniques have been proposed to reduce delay due to retransmission at the expense of increased bit rate. In this section the FEC and ARQ techniques will be discussed.

1.3.1 Forward Error Correction (FEC)

Forward error-correction coding (also called channel coding) is a type of digital signal processing that improves data reliability by introducing a known structure into a data sequence prior to transmission or storage. This structure enables a receiving system to detect and possibly correct errors caused by corruption from the channel and the receiver. As the name implies, this coding technique enables the decoder to correct errors without requesting retransmission of the original information.

In a communication system that employs forward error-correction coding, a digital information source sends a data sequence comprising k bits of data to an encoder. The encoder inserts redundant (or *parity*) bits, thereby outputting a longer sequence of n code bits called a codeword. On the receiving end, codewords are used by a suitable decoder to extract the original data sequence.

Codes are designated with the notation (n, k) according to the number of n output code bits and k input data bits. The ratio k/n is called the rate, R, of the code and is a measure of the fraction of information contained in each code bit. For example, each code bit produced by a (6, 3) encoder contains 1/2 bit of information.

Another metric often used to characterize code bits is redundancy, expressed as $(n-k)/n$. Codes introducing large redundancy (that is, large $n-k$ or small k/n) convey relatively little information per code bit. Codes that introduce less redundancy have higher code rates (up to a maximum of 1) and convey more information per code bit. Large redundancy is advantageous because it reduces the likelihood that all of the original data will be wiped out during a single transmission.

On the down side, the addition of parity bits will generally increase the transmission bandwidth or the message delay (or both). For real-time applications, such as voice communications, the code-bit rate must be increased by a factor of $n/k = 1/R$ to avoid a reduction in data throughput. Hence, for a given modulation scheme, the transmission bandwidth increases by that same factor n/k. If, however, the communication application does not require the real-time transfer of information, then additional message delay (rather than increased bandwidth) is the usual trade-off.

The general error-performance characteristics of most digital communication systems have a waterfall-shaped appearance. System performance improves (i.e., bit-error rate decreases) as the signal-to-noise ratio increases. For example, the coded system, operating with a received signal-to-noise ratio of 8 decibels, has a smaller bit-error rate by a factor of 100 compared with the uncoded system at the same signal-to-noise ratio.

In another way, it is indicate that the coded system can achieve the same bit-error rate as the encoded system at a lower signal-to-noise ratio. This reduction in required signal-to-noise ratio, called the coding gain, is a common metric used to measure the performance of different coding schemes.

The importance of coding gain is evident when the system is viewed from the designer's perspective. For example, to obtain the same level of improved bit-error rate without the use of coding, a designer would have to achieve a larger signal-to-noise ratio (12 decibels instead of 8). To do so would require the use of larger power supplies, bigger antennas, or higher-quality components that introduce less noise. If none of these modifications can be provided, then the designer will have to tolerate some type of performance degradation—such as reduced service ranges or lower operating margins—to obtain the same improvement.

1.3.2 Automatic Repeat Request (ARQ)

The ARQ strategy for error control is based on error detection and retransmission. Consequently, ARQ system differs from FEC systems in three important respects:

1. An (n, k) block code designed for error detection generally requires fewer check bits and has a higher n/k ratio than a code designed for error correction.
2. ARQ system needs a return transmission path and additional hardware in order to implement repeat transmission of codeword with detects errors.
3. The forward transmission bit rate must make allowance for repeated word transmission.

Each codeword constructed by the encoder is stored temporarily and transmitted to the destination where the decoder looks for errors. The decoder issues a positive acknowledgment (ACK) if no errors are detected or a negative acknowledgment (ANK) if errors are detected. A negative acknowledgment causes the input controller to retransmit the appropriate word from those stored by the input buffer. A particular word may be transmitted just two ore more times, depending on the occurrence of transmission errors. The function of the output controller is to assemble the output bit stream from the codeword that have been accepted buy the decoder. When an error is detected in a word, the receiver signals back to the transmitter and the word is transmitted again.

There are three basic ARQ systems:

1. The stop-and wait ARQ system: it is the simplest to implement. In this system the transmitter sends a codeword to the receiver during a specified time. The receiver receives and processes the received word and if the receiver detects no error, it sends back to the transmitter a positive acknowledgment (ACK) signal. Upon receipt of the ACK signal, the transmitter sends the next word.
2. The go-back N ARQ: Here the transmitter sends messages, one after another, without delay and does not wait for an ACK signal. When however the receiver detects an error in a message, a NAK signal is returned to the transmitter. In response to the NAK the

transmitter returns to the codeword of the error message and starts all over again at the word.

3. The selective-repeat ARQ: In this system the transmitter sends messages in succession, again without waiting for ACK after each message. If the receiver detects that there is an error in the codeword, the transmitter is notified. The transmitter retransmits the codeword of the error message and thereafter returns immediately to its sequential transmission. The selective ARQ, as might well be anticipated, has the highest transmission efficiency of the three systems but, on the other hand, it is the most costly to implement.

1.4 Smart Antennas

1.4.1 Definition

A smart antenna is an array of antenna elements connected to a digital signal processor. Such a configuration dramatically enhances the capacity of a wireless link through a combination of diversity gain, array gain, and interference suppression. Increased capacity translates to higher data rates for a given number of users or more users for a given data rate per user.

The smart antenna works as follows. Each antenna element "sees" each propagation path differently, enabling the collection of elements to distinguish individual paths to within a certain resolution. As a consequence, smart antenna transmitters can encode independent streams of data onto different paths or linear combinations of paths, thereby increasing the data rate, or they can encode data redundantly onto paths that fade independently to protect the receiver from catastrophic signal fades, thereby providing diversity gain. A smart antenna receiver can decode the data from a smart antenna transmitter--this is the highest-performing configuration-- or it can simply provide array gain or diversity gain to the desired signals transmitted from conventional transmitters and suppress the interference. No manual placement of antennas is required. The smart antenna electronically adapts to the environment by looking for pilot tones or beacons or by recovering certain characteristics (such as a known alphabet or constant envelope) that the transmitted signal is known to have. The smart antenna can also separate the signals from multiple users who are separated in space (i.e. by distance) but who use the same radio channel (i.e. center frequency, time-slot, and/or code); this application is called space-division multiple access (SDMA).

1.4.2 Types of Smart Antenna Systems

Terms commonly heard today that embrace various aspects of a smart antenna system technology include intelligent antennas, phased array, SDMA, spatial processing, digital beamforming, adaptive antenna systems, and others. Smart antenna systems are customarily categorized, however, as either switched beam or adaptive array systems.

1.4.3 Major Categories

The first category is the switched beam antenna. Its systems form multiple fixed beams with heightened sensitivity in particular directions. These antenna systems detect signal strength, choose from one of several predetermined, fixed beams, and switch from one beam to another as the mobile moves throughout the sector. Instead of shaping the directional antenna pattern with the metallic properties and physical design of a single element (like a sectorized antenna), switched beam systems combine the outputs of multiple antennas in such a way as to form finely sectorized (directional) beams with more spatial selectivity than can be achieved with conventional, single-element approaches.

The second one is the adaptive antenna. Its technology represents the most advanced smart antenna approach to date. Using a variety of new signal-processing algorithms, the adaptive system takes advantage of its ability to effectively locate and track various types of signals to dynamically minimize interference and maximize intended signal reception.

Both systems attempt to increase gain according to the location of the user; however, only the adaptive system provides optimal gain while simultaneously identifying, tracking, and minimizing interfering signals.

1.4.4 What Makes Them Smart

A simple antenna works for a simple RF environment. Smart antenna solutions are required as the number of users, interference, and propagation complexity grow. Their smarts reside in their digital signal-processing facilities.

Like most modern advances in electronics today, the digital format for manipulating the RF data offers numerous advantages in terms of accuracy and flexibility of operation. Speech starts and ends as analog information. Along the way, however, smart antenna systems capture, convert, and modulate analog signals for transmission as digital signals and reconvert them to analog information on the other end.

In adaptive antenna systems, this fundamental signal-processing capability is augmented by advanced techniques (algorithms) that are applied to control operation in the presence of complicated combinations of operating conditions.

For understanding the performance of the smart antenna, two adaptive algorithms should be well known. These algorithms are the normalized least mean-squares (NLMS), and the recursive least-squares (RLS).

1.4.5 NLMS

The most widely used adaptive algorithm is based on the least-mean-square (LMS) algorithm, in which antenna weights are recursively obtained to minimize the mean square error. If the step size is chosen properly, the algorithm guarantees the convergence of the antenna weights. The major benefit of the LMS algorithm lies in its simplicity compared to other adaptive algorithms. The LMS algorithm, however, suffers from a gradient noise amplification problem for large input signals. To circumvent the problem, the normalized LMS (N-LMS) algorithm rather than the original LMS algorithm is usually used in practice. The N-LMS algorithm exhibits a faster rate of convergence than the original LMS algorithm for both uncorrelated and correlated input data.

Since each user sends its pilot signal to the base station periodically, a special channel estimation scheme (such as the weighted multi-slot averaging method) is necessary for the base station to take care of non-continuous pilot signals. In contrast, each user receives the continuous common pilot (CPICH) signal from the base station. This pilot signal can be used for the adaptive combining as well as for a phase reference in the coherent combining. The pilot signal is used in the N-LMS algorithm for our smart antennas at handsets. Like any adaptive combining, the key aspect of the N-LMS algorithm is to compute the weights of antenna signals, and it is explained below for the case of two antennas.

1.4.6 RLS

The RLS is a training base adaptation that is performed to adjust the taps in smart antenna. Main problem with the LMS algorithm is that it takes a long time for the filter coefficients to converge because they are adjusted at an identical rate. The RLS adaptive algorithm goes over this prblem and improves greatly.

1.5 WCDMA

Wideband CDMA (WCDMA) is rapidly emerging as the leading global 3G standard, enabling users to access Mobile Internet services. The first release of the WCDMA specifications fully supports the IMT-2000 requirements, including data rates up to 2 Mbit/s and support of both packet and circuit switched services. Peak data rates in the order of 10 Mbit/s together with lowered roundtrip delays and increased capacity provide a further boost for wireless data access.

In order to provide these high data rates, as well as to improve the system capacity, WCDMA employs three fundamental principles, relying on rapid adaptation of the transmission parameters to the rapidly varying radio conditions: fast link adaptation, fast hybrid ARQ, and fast scheduling of a shared channel.

1.5.1 Fast link adaptation

Fast link adaptation is the process of rapidly adapting the transmission formats to the instantaneous channel conditions. In addition, it provides higher order modulation and improved spectral efficiency in bandwidth limited scenarios. By adjusting the code rate and the modulation scheme used, the momentary data rate can be matched to the instantaneous radio conditions and hence maximize the utilization of the radio channel.

1.5.2 Fast Hybrid ARQ

Fast Hybrid ARQ implies rapidly requesting retransmission of erroneously received data entities and using soft combining of multiple transmission attempts. By combining soft information from multiple transmission attempts, the number of retransmissions needed, and thus the delay, will be reduced. Hybrid ARQ with soft combining also adds robustness against link-adaptation errors and is closely related to the link adaptation mechanism.

1.5.3 Fast Scheduling

Fast scheduling is in control of allocating the shared resources among the different users and operates on the 2 ms TTI basis. It is a key element in the design and to a large extent it determines the overall behavior of the system. In addition to traffic statistics, the scheduler also takes the instantaneous radio conditions into account in the scheduling process. Note that information of the radio conditions is required by the link adaptation scheme as well. Maximum system throughput is obtained by assigning all available radio resources to the user with the currently best radio conditions, while a practical scheduler could include some degree of fairness.

1.6 Summary

In this chapter, the wireless communications was introduced with a brief history. In addition the standards of it were listed. After that, the problems that face the wireless channels were discussed. Then several solutions were stated to solve these problems. Finally, some solutions were introduced and discussed with their main components to understand it well.

CHAPTER TWO

ERROR-RESILIENT H.263 VIDEO CODEC

2.1 Overview

The H.263 standard is used to compress the over in video transmission. Unfortunately, H.263 is error-sensitive in bit errors in wireless channels as well as packets losses in internet channels. This chapter presents an error resilient H.263 video compression scheme over noisy channels. The start codes in the H263 bit stream syntax, because significant error damage if they are incorrectly decoded. Therefore, a video segment regulation algorithm at the decoder is developed to efficiently identify and correct erroneous start codes and block addresses. In addition, the precise error tracking technique is used to further reduce the error propagation effects. Finally, the modified H.263 video decoder, error correction and detection are discussed.

2.2 H.263 compresses video over IP

Whether it's the convenience of holding face-to-face meetings globally or the value of watching remote assets from a central office locally, the potential of video over IP is growing. Physical security and public safety organizations are driving the demand to implement enhanced digital video tools, and technology now allows for migrating those tools onto a communications network.

You don't need to upgrade your network to Gigabit Ethernet to transmit video over IP because video compression, specifically the H.263 standard, provides a cost-effective alternative to increased bandwidth.

In the world of video compression there are two types of choices: proprietary compression algorithms created by several manufacturers, and standards-based technologies such as H.263 and the Moving Pictures Experts Group (MPEG) family, including Motion JPEGs.

2.2.1 Motion JPEG

Commonly known as MJPEG, this compression algorithm is a distant cousin of the MPEG and is commonly seen in digital video recorders. Typically, MJPEG is used in the physical security environment to translate analog video from closed circuit television cameras into a digital stream to be stored onto a hard drive.

MJPEG has the capability to send very high-quality pictures, but it requires an enormous amount of bandwidth - as much as a to produce full-motion video. MJPEG does not use interframe coding, unlike MPEG, and is easier to edit with a nonlinear editor.

However, MJPEG images are among the largest to store on a digital medium and require large amounts of disk space to meet the needs of most businesses today. It is perhaps the most inefficient coder/ decoder (codec) to use in an application such as physical security where the camera needs to be on for extended periods of time. This technology is better for applications when a "snapshot" photo is the desired outcome.

2.2.2 H.263 Video Compression

H.263 lets users scale bandwidth usage and can achieve full-motion video (30 frames per second) at speeds as low as 128K bit/sec. With its flexibility and bandwidth and storage savings, H.263 has a low total cost of ownership and provides a quick return on investment. H.263 was developed to stream video at bandwidths as low as 20K to 24K bit/sec and was based on the H.261 codec, but as a general rule, it requires half the bandwidth to achieve the same video quality.

Originally designed as the standard for videoconferencing over ISDN, H.261 introduced features such as motion prediction and block transformation. This allowed for a smoother picture with good quality, but was limited in the amount of motion it could handle. Also, H.261 used a large amount of bandwidth (64K to 2M bytes) and was targeted primarily at circuit-switched networks.

H.263 has largely replaced H.261. As H.263 became popular because of its high-quality video at low bandwidths, the standard was annexed and updated nine times. IT managers can feel comfortable placing it on their data networks, without increasing bandwidth and storage costs, or interrupting other critical voice and data applications already running on the network.

The H.263 algorithm also can be modified to produce better results and better compression schemes, which in turn gives end users more choices in selecting the implementation that best fits their business applications.

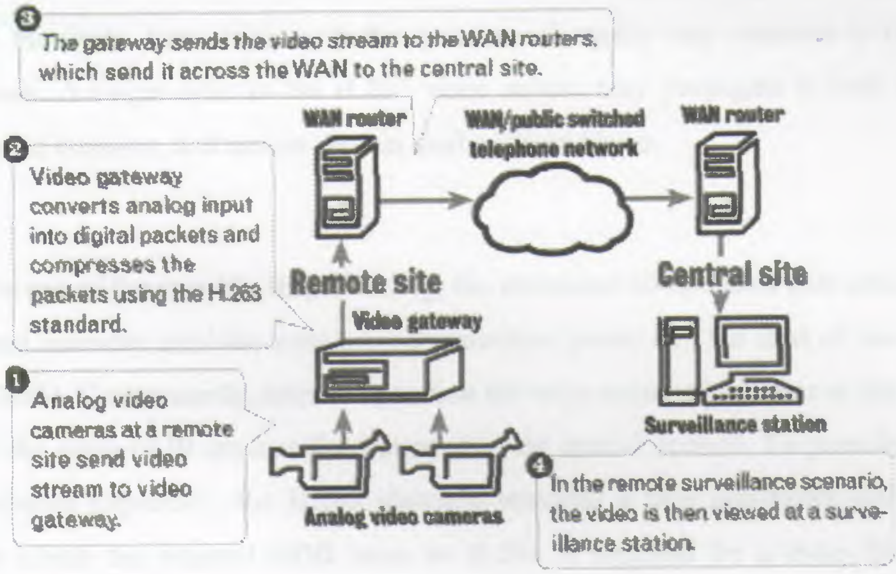

Figure 2.1 H.263 Video Compression Technology

However, without any enhancement H.263 sequence is error-sensitive in bit errors introduced by wireless channels as well as packets losses introduced by internet channels. For instance, if an error corrupts a particular macro block (MB) due to the motion compensated prediction in the encoding algorithm, any forthcoming MB with a motion vector (MV) that references the corrupted MB would be affected. Therefore, the decoder might be forced to discard the subsequent data regardless of their correctness. Hence the transmission power and time might be wasted. Furthermore, this accumulative error may propagate spatially and temporally across the frames. Consequently, in order to mitigate data errors, error resiliency issues must be addressed and error recovery mechanisms need to be introduced.

2.3 H.263 Error-Resiliency

ITLT-T H.263 is a low bit rate video coding standard, which has been successfully used for many applications, such as video conferencing and video telephony. It is developed to improve the compression performance, providing additional features, and supporting various networks more efficiently. By using motion estimation/compensation, discrete cosine transform (DCT), and variable length coding, significant compression gains can be achieved. However, some of these techniques are inherently very sensitive to the channel disturbances. A single error in the H.263 video stream may propagate in both the spatial and temporal domains, and causes serious quality degradation.

Due to the use of the variable length coding, the erroneous compressed data usually cannot be decoded correctly until the next resynchronization point, i.e., the start of next group of blocks (GOB). Consequently, after the position the error occurred, all data in the following blocks of the same GOB are usually destroyed in the spatial domain. To provide enhanced error resilience capability, the H.263 standard provides a new negotiable coding option mode, in which the original GOB layer in H.263 is replaced by a more flexible slice structure. All macro blocks (MBs) of one slice can be independently decoded because no data dependencies such as the prediction of motion vectors (MVs) can be allowed to cross the slice boundaries within the current picture. Therefore, a slice with shorter length can stop spatial error propagation more effectively.

In the temporal domain, since the 1-picture and the P-picture are reference frames in the motion estimation/compensation, errors will propagate to all the following frames until all the erroneous MBs are refreshed by INTRA-mode coding. However, the encoder is not aware of the errors existing in the decoded bit stream in general and therefore becomes asynchronous with the decoder state. Thus, at the decoder the damaged area in the reference picture due to the motion compensation may overwrite a MB in the current decoding picture. Consequently, the reconstructed video quality is deteriorated and the error propagation continues.

Several techniques have been proposed to limit the effects of error propagation, such as the unequal error protection, the automatic retransmission request (ARQ), and the error concealment. The first two techniques, however, are required to modify the bit stream syntax and thus are incompatible with the standard. The error concealment techniques at the receiver have been proved to efficiently improve the image quality and reduce the error damage substantially. Most of the error concealment techniques utilize the information of neighbors in one or more domains to estimate the erroneous block. However, the success of the error concealment relies on the essential assumptions that the locations of erroneous blocks are identified accurately and all neighbors used for the concealment contain no errors. Unfortunately, these assumptions usually do not hold. Therefore, the performances of error concealment techniques degrade and they seldom terminate the temporal error propagation completely.

In H.261 the error detection and error concealment approach was successful for combating transmission errors. The constraints are that all GOBs within a picture can be correctly located. Many error concealment studies also avoid this problem by assuming correct decoding of headers. However, in H.263 headers, the start codes and the group numbers (or MB addresses) provide the information on the locations where to put the decoded bit stream and the basic resynchronization points for error handling. Consequently, an incorrectly decoded header may cause disastrous effects. Unequal error protection or ARQ may solve this problem. However, as we just mentioned; they are not H.263 syntax compatible. The IJG (Independent JPEG Group) provided a default resynchronization method for error recovery in still images, assuming the decoder is unable to back up. Besides, the restart marker has been studied to improve the error robustness.

Those works mainly investigated on the positioning of markers, however, the marker error itself is not considered. The resynchronization regulation algorithm is developed for the progressive JPEG decoding. It successfully regulates the restart interval order and effectively corrects most of the errors in restart markers. Based on this concept, we further develop a segment regulation algorithm to H.263 video coding for reducing such error effects.

A different approach to avoiding the error accumulation is INTRA coding the video with the penalty of rate increase. The usage of a feedback channel for error tracking and recovery after transmission errors can also be used. In the error-tracking strategy, the encoder considers spatial-temporal error propagation caused by motion compensated prediction as well as the delay until the reception of the feedback message. To evaluate the feedback message, the encoder needs to continuously record the MV information during the encoding of each frame. Finally, the evaluated severely contaminated MBs are coded in INTR.A mode for stopping temporal error propagation. With the analysis of temporal dependencies of MBs in successive frames, this feedback approach leads to rapid quality recovery by reconstructing the error propagation effects at the encoder and selecting severely affected regions to be INTRA refreshed. In order to use the minimum number of INTRA MBs to terminate the error propagation, it is necessary to track errors more precisely.

The error robust H.263 coding with the video segment regulation (VSR) and the precise error tracking (PET) techniques is proposed to reduce the error damage from both the headers and video data. VSR reduces the damage from erroneous start codes and PET terminates the temporal error propagation of video signals. Additionally, VSR also provides more reliable feedback acknowledgements on the locations of erroneous MBs for performing PET. With both techniques applied, this system is much robust against errors occurred in all locations.

2.3.1 Architecture of Error Resilient H.263 Video Coding System

Robust error detection and error concealment must rely on correct identification of each GOB (or slice) location and new picture start position. For this reason, we perform the VSR technique in the decoder to identify and correct erroneous start codes and block addresses. A video segment can be a GOB in H.263 or a slice in error resilient H.263. The decoder preprocesses the received compressed bit stream, which is temporarily stored in the receiver buffer, to search for the start codes and their corresponding encoded video segments. Every start code number is examined by a checking procedure to see if it is a correct one. Then, minimizing the distance distortion measure performs the regulation procedure. Therefore, errors occurred in start code numbers can be corrected and consequently all GOB (or slice) locations are uniquely identified. Finally, the embedded error detection mechanism provided by the H.263 decoder can discover the exact locations of erroneous decoded MBs. These negative acknowledgements (NACKs) can be sent back to the encoder for performing error tracking or to the decoder for performing error concealment. With the analysis of temporal dependencies of MBs in successive frames, the feedback NACKs are utilized by the encoder for reconstructing the error propagation effects at the encoding end and selecting severely affected regions to be 1NTRA refreshed. Our proposed strategy, i.e., the PET, uses the pre-stored MVs and traces the motion dependency for each current encoding pixel backward to the previous unsuccessfully de-coded frame informed by the NACK. The encoder thus can exactly evaluate how severely every MB in the current encoding frame is affected by those impaired areas. Finally, all or parts of the contaminated MBs can be selected to refresh by the INTRA-mode coding. To sum up, this algorithm is able to track the actual error propagation and completely terminate it. The earlier the feedback-transmission (NACK} arrives, the sooner the error-propagation stops.

2.3.2 Video Segment Regulation Algorithm

Due to the use of the variable length coding, the erroneous compressed bit stream usually can not be decoded correctly until the next resynchronization point, i.e., the next start code position. Consequently, the start code provided by the H.263 standard plays an important role for the error detection and error recovery in the error-prone environments. The start codes only exist in the picture and GOB (or slice) layers. In other words, the minimum resynchronization unit is one GOB (or slice). A single bit error may corrupt a MB and the subsequent MBs in the same GOB (or slice).

Based on the H.263 standard, there exists a tradeoff in the choice of the occurrence frequency of start codes. The less the start codes are used (or the longer the slice length is used), the longer the errors propagate. However, frequent, start codes result in the substantial overhead in the bit rate and the increased occurrence probability of erroneous start codes. A missing or misinterpreted start code generally results in much more serious image degradation than the errors in image data. For example, one bit error may result in entire GOB 5 of a frame falsely decoded into GOB 1 and thus both GOBs are corrupted. Particularly, a missing or fake picture start code (PSC) may come out with the incorrect number of decoded frames and the inaccurate playback time. Furthermore, the accuracy of the error tracking also relies on the correct feedback addresses of corrupted MBs.

To solve above problems, we develop the video segment regulation algorithm to correct erroneous start codes and block addresses. The proposed scheme mainly comprises two parts, the erroneous video segment allocation and rearrangement. In the procedure of erroneous video segment allocation, the decoder first preprocesses the received bit stream, which is temporarily stored in the receiver buffer, to search for the start code and its corresponding encoded video segment. According to the GOB structure syntax, a start code number is defined as the GOB starts code (GBSC) concatenated with the following group number (GN). If the slice structured mode is in use, the start code number, which contains the slice start code (SSC) and the MB address (MBA), is given in an increment order. A start code is then marked as in a correct video segment if its number is in the consecutive order with the preceding one and the succeeding one. In this case, the decoder continues decoding this video segment as normal. Otherwise, the erroneous video segments are kept in the receiver buffer until the next correct start code number is identified.

After the erroneous video segment allocation, the rearrangement procedure is performed on these erroneous video segments. Here, we denote the number of detected erroneous video segments between two correct ones as *Nfound* and the number of desired consecutive video segments between two correct ones as *Nneed*. Every erroneous segment is classified as one of the following three cases and the corresponding rearrangement procedure is fulfilled.

Forced start code assignment (*Nfound* = *Nneed*): All erroneous start codes are forcedly changed to the corresponding desired start code numbers that are in the consecutive order.

1. Lost start code reconstruction (*Nfound* < *Nneed*): The lost start code is reconstructed by searching the erroneous video segments for the bit pattern that has the minimum Hamming distance to the desired one.

2. Extra start code erasure (*Nfound* > *Nneed*): This case rarely happens in the H.263 as compared to the JPEG because the H.263 start code length is 17 bits long while the JPEG restart marker is 8 bits only. If it happens on occasion, the start codes with the minimum Hamming distance to the desired start code numbers are changed forcedly. Then the rest start codes with their video segments are deleted from the bit stream.

The search directions for the desired start codes can be forward, backward, or even both. In the case of both direction searches, the candidates with the minimum total Hamming distances will be chosen.

After performing the video segment regulation, all GOB (or slice) locations are uniquely recognized and the video sequence can be decoded accordingly. Note that this regulation algorithm cannot be applied to the arbitrary slice ordering submode in error-resilient H.263 if the slice structured mode is in use. Also note that the picture start code, whose MBA is '0,' can also be included in the regulation process. Therefore, this process regulates not only the slice order, but also the picture synchronization.

2.3.3 Precise Error Tracking Algorithm

NACKs with the information on unsuccessfully decoded image blocks are sent back to the encoder via a feedback channel. Once a NACK is received; the encoder performs error tracking to determine if the current encoding MB is contaminated by the erroneous MBs in the past frames. If this happens, this MB becomes a candidate for INTRA-mode refreshing.

The MV of the MB produced in the motion estimation progress indeed provides adequate information for accurately tracing error propagation at the encoder. The following example illustrates how to execute the precise error tracking for the three pixels Q, R and S by employing the pixel-based backward motion dependencies. Any pixel's motion dependency can be found by tracing back the MV of the MB it belongs to. We suppose that the prediction of MB 25 in frame N is obtained from MBs 25 and 26 in frame N-1. MB 26 of frame N-l refers to MBs 15, 16, 26, and 27 in frame N-2. MB 25 of frame N -1 is coded in INTRA mode. If the encoder receives a NACK, which indicates an error occurred at MB 15 of frame N -2, while encoding frame N. At that moment, we can backward trace each pixel in MB 25 of frame N along the corresponding paths, i.e., the corresponding MVs, to see if it refers to the erroneous area, i.e., MB 15 of frame N - 2. The pixel Q is then determined to be a contaminated pixel while pixel R is not. Likewise, the backward motion dependency structure for each MB can be built and the error tracking procedure is performed for all pixels in frame N. it is especially important to note that there exists no MV for an INTB.A MB, such as MB 25 of frame N-1. In such a case, pixel S is certainly claimed to be clean. With this pixel-based error tracking strategy, the degree of damage caused by the error propagation can be calculated.

Although the INTRA-mode coding terminates the temporal error propagation, it usually generates a higher bit rate, too. If the rate increased by the INTRA-mode coding is higher than what we can afford, we need to limit the number of LNTRA MBs in a frame. For instance, we can select only a fixed number of MBs with the largest CR values to perform INTRA refreshing. Of course, in such cases the error propagation may not be stopped completely.

In spite of the fact that this algorithm requires tracing the motion dependency for each pixel, it actually exhibits very low complexity because only simple additions are needed. By setting a maximum round-trip delay, the memory requirement is also low since all we need to store is the required MV information, which has been generated in the motion estimation procedure already.

2.4 Improving the Video Quality

Most of the standard video codec like H.263 are based on motion compensation – discrete cosine transform (MC-DCT) coding scheme, which use the variable length code (VLC), such as Huffman for further compression. The use of VLC in the erroneous compressed data would not allow even the non-corrupted parts to be correctly decoded until after the synchronization point, i.e. start of the following group of blocks (GOB) or slice. Moreover, due to out-of-synchronization between the encoder and decoder states, the error may also propagate into the temporal domain. Due to these reasons, the emerging video coding techniques include provisions for error resilience particularly in H.263 and MPEG.

To limit the effect of error propagation and hence to improve the video quality against the transmission error, several techniques have been proposed in the literature. These can be grouped into four categories: feedback channel or retransmission approach, forward error correction or channel coding approach, error resilience approach and error detection and correction approach. However, these techniques suffer from some of the basic problems. For example, the use of feedback channel introduces additional transmission delay and complexity. The channel coding and use of markers increases the data rate and may not be suitable for low bit rate applications.

The error concealment methods, although don't increase the transmission bandwidth, but yield poor performance under high channel error rate. The error detection and correction at image level is achieved by the frequent use of restart markers. The reversible variable length codes (RVLC) are best suited to detect the errors when bits from erroneous bit onward are not decodable or when invalid codes occur. However, in general all the erroneous bits may not result in invalid codes and bitstreams may be decoded (although erroneously) even in the presence of errors. In addition to all these problems, some errors in the bitstream have almost no or very little impact on the picture quality and it would be waste of resources to correct them.

The error detection is aimed at improving the perceptual video quality after erroneous H.263 coded bitstreams are received. Our error detection algorithm exploits the inherent redundancies within and outside a macroblock to detect and locate the erroneously decoded macroblocks. The redundancies are measured in terms of a set of parameters (based on MB and inner DCT blocks similarities). After the error detection, an iterative re-decoding based correction algorithm is applied to the erroneous slices. The proposed error detection and correction scheme is applied iteratively until no further corrupted macroblock is detected. Finally spatial (for I frame) and temporal (for P frames) concealment techniques are used to conceal the corrupted macroblocks.

2.4.1 Modified H.263 Video Decoder

The standard video coding systems like H.263 envisage various methods to improve their resilience towards channel errors. For example the Annex-K of H.263+ supports the slice-structured mode where all macroblocks of one slice can be decoded independent of the content of other slices by preventing the prediction of motion vectors to cross the slice boundaries. There is however, a need for extra information to decode a slice, because information conveyed in the picture header is not repeated in the slice header. Here the slice-structured mode is used with slight modification in the bitstream syntax.

The error detection and correction technique is incorporated in the modified decoder. Since in the slice structured mode of H.263 bitstream, the quantization information and motion vectors are encoded differentially, if one or more macroblocks in the middle of the slice are omitted, then it is impossible to decode the quantization information and motion vectors of the following macroblocks. In order to avoid this problem, the bitstream syntax is slightly modified from the conventional slice structure mode. The modifications are made only in the slice and macroblock layer, whereas picture and block layers remain unchanged. In the macroblock layer MVD (motion vector difference) prediction is modified.

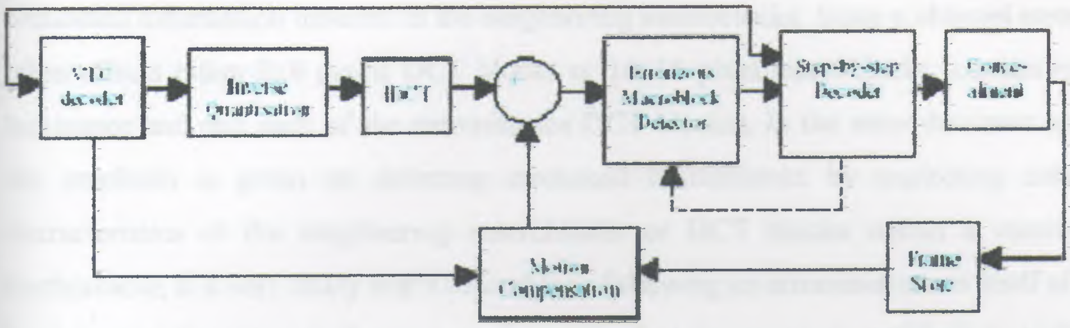

Figure 2.2 Modifications to the H.263 decoder for error detection,

Correction and concealment

2.4.2 Error Detection

In the presence of transmission errors, a decoded frame may contain three types of macroblocks, (1) non-decodable macroblock (fatal error), (2) correctly decodable macroblock and (3) decodable but erroneous macroblock. In our scheme, the first type of macroblock is detected during decoding a slice but the other two types are detected after the decoding. While decoding a corrupted bitstream, the H.263 decoder may detect the slice as erroneous if any of the following conditions occurs:

- Invalid VLC code is detected.
- Quantizer information goes out of range.
- Invalid INTRA DC code is detected.
- Escaped TCOEF with level 0 is detected.
- Coefficient overrun occurred.
- A motion vector refers out of picture.
- The number of bits used in decoding of the slice is different from that in the slice.
- The quantizer information

After a frame is decoded, the visibly erroneous macroblocks will be detected using redundant information inherent in the neighboring macroblocks. Since a channel error most often affects either 8x8 pixels DCT blocks or 16x16 pixel macroblocks (containing four luminance and one each of the chrominance DCT blocks), in the error detection scheme, the emphasis is given on detecting erroneous macroblocks by exploiting either the characteristics of the neighboring macroblocks or DCT blocks within a macroblock. Furthermore, it is very likely that a macroblock following an erroneous one is itself affected by the error; the main challenge is to detect the first erroneous macroblock in a slice. In order to detect corrupted MBs, the following similarity measures are exploited:

a) Macroblock (MB) Characteristics

1) Macroblock boundary

2) Macroblock mean pixel value

b) Inner DCT block characteristics

1) Inner boundary

2) Individual DCT block boundaries

3) DCT block's mean pixel value

For each of these similarity measures, a parameter is calculated and compared with either an absolute or a relative threshold or both. For a macroblock under consideration, if the parameter has a value greater than an absolute threshold, the macroblock is considered as erroneous. However, if the value of the parameter is greater than a relative threshold, it is considered as a candidate of being erroneous (subject to other criteria to be fulfilled).

3.4.3 Error correction

Assuming that the decoder knows the location of the first erroneous macroblock (if any) in the slice, the step-by-step decoding works as follows. The part of the slice preceding the *first erroneous macroblock is conventionally decoded and a pointer is initialized at the* second bit of the first erroneous macroblock, i.e. skipping the first bit. The rest of the bitstream from that point is checked to see whether it is decodable or not. If the bitstream is non decodable, the pointer is incremented by one bit. This process is continued until correctly decodable bitstream is found and decoded. We call this part of the slice as step-by-step decoded part. If the number of decoded macroblocks in the slice is less than the actual number of macroblocks (which is known), the step-by-step decoded part is right aligned. This process is repeated for each erroneous slice (found while decoding the frame) and thus the decoded frame is generated. Although it is expected that the step-by-step decodable part of the slice is obtained when pointer is pointing to the bit of a macroblock boundary, however, in practice it is possible that rest of the slice is decoded from the point different from MB boundary. It is also likely that decodable bitstream for the rest of the slice also contains some erroneous bits. In such cases the decoded frame may still contain erroneous macroblocks. For this reason the error detection and step-by- step decoding is repeatedly performed on the decoded frame until the error detector does not detect any more corrupted MBs, except the gray valued MBs, which are concealed later.

2.4.4 Concealment

After detecting and correcting most of the erroneous macroblocks, the erroneous slices may still have some erroneous macroblocks (at least one in which the error actually occurred) which were filled in with gray values during the step by step decoding. These macroblocks can now be efficiently concealed with any standard error concealment technique. There are two basic approaches for error concealment: spatial and temporal. In the spatial interpolation, which is used for intra coded frames, the pixels of missing macroblocks are reconstructed as the median value of the corresponding pixels in eight surrounding macroblocks.

For inter coded frames, we use simple motion compensated temporal error concealment, in which the motion vectors of the missing macroblock is set to the median value of the motion vectors of the eight neighboring macroblocks. If the surrounding macroblocks are not available, the corresponding motion vectors are set to zero. Then, the macroblock from the previous frame at the spatial location specified by this motion vector is copied to the location of the missing macroblock in the current frame.

2.5 Error-Resiliency by Using Optimal Distribution

The basic two problems in dealing with lossy networks consist of being able to protect data against network losses (error resilience) and to make the error induced artifacts less visible to the human eye (error concealment). For the block based coding schemes traditionally used in coding standards, when predicatively coded blocks (inter blocks) are used, the effect of an error in one frame can propagate to the successive frames. On the other hand, the non-predicatively coded blocks (intra blocks) act as refresh points to stop the temporal error propagation. Intra blocks have a much higher cost in terms of bit rate. Thus video coding for lossy networks is a tradeoff between low bit rate (more inter blocks) and faster error recovery (more intra blocks). Prior work on judicious allocation of intra blocks consisted of using RD optimization with a Lagrangian formulation of minimizing the cost of the modes. In addition to the optimized allocation, we consider the advantages of having a multiple state stream in faster recovery from errors. If one state is lost because of error, the state recovery can use data from the other intact state(s).

A new method is proposed for allocating the intra blocks based upon the past history of the intra/inter distribution for a multiple state stream. This helps in faster recovery than if we just use a forced update of regular intra blocks. This scheme (for a multiple state coding method) has been implemented on H.264 codec. The scheme tries to minimize the cost function with an additional constraint designed to force intra blocks.

2.5.1 Optimal Mode Allocation for Single State Streams

Since intra coded macroblocks are independently coded, these macroblocks naturally become the error propagation stopper. A popular simple way to make use of this feature for error resilience is to update a screen by periodically using intra-coded macroblocks. Examples are using one intra-coded frame per group of pictures as specified by MPEG standards, or using a few intra-coded slices for progressive refreshment, which is also compliant with MPEG and other video coding standards. The current under-going standard H.263 also provides an option of periodically using intra-coded group of blocks to refresh a frame from top to bottom. However, since intra coding does not exploit the temporal redundancy, it generally requires higher bit rate than inter coding does. Furthermore, intra coding does not guarantee a smaller distortion, especially when the quantization step size is not small enough, and hence it may cause penalty on both image quality and bit rate. Thus, optimally choosing the coding modes and allocating intra and inter macroblocks for high error resilience efficiency becomes a critical issue for compressed video over network, especially when the compressed bit stream is transmitted in low bit rate and error prone channels.

To calculate the Dc, the error concealment method used by the decoder must be assumed by the encoder. The packet loss rate p can be also assumed or dynamically calculated based on network feedback. A further improvement of this scheme is presented in [Cote00], which tries to estimate the distortion caused by error concealment based on not only the previous reference frame but also the accumulated effect of the previous multiple frames, starting from a collocated intra-coded macroblock. To do this, a constraint on the motion vectors is also assumed, which may degrade the performance of improved method.

A more precise work is to do the distortion estimation at pixel level by recursively calculating the first-order and second-order moments of the decoder reconstructed pixel values, as presented in [Zhang00].

2.5.2 Multiple State Coding

The multiple state coding schemes was proposed [john1] as a scheme to recover better from error. The basic idea is to divide the data into multiple independent streams called states. If any state is lost, then it can be approximately recovered from the other error-free states. This is due to the inherent redundancies that exist between the states.

This mode of error recovery is the main difference between MSC and Multiple Description Coding (MDC). In MDC the information from one stream can be complemented by information from the other, though the individual streams are independently decodable in either case. We consider a simple case of two state codec, wherein the odd frames are encoded by in one state and the even frames in the other. In case of the single state the lost frame can be regenerated from the previous frames. But in case of MSC, the decoder has access to both the previous and the future frames and can do a much better state recovery.

Various techniques can be used for recovering the lost state from the previous and future states. Some of these like Motion Compensated Interpolation (MC-I) may not be favorable for low complexity decoders. In this project, we have tried simpler low complexity schemes like repetition of previous frame from other stream and averaging of the frames before and after the current frame. The averaging scheme is seen to perform better than the others and was used for the rest of the simulations.

2.5.3 Dynamic mode-tune for faster error recovery

We use the RD framework as presented in [Cote99] as the basic scheme for Lagrangian cost calculation. However, we realize that to make the error recovery fast, we need to have more macroblocks intra-coded, or have "content-important" macroblocks more frequently intra-coded. Here, "content-important" means the macroblock has less correlation with previous frames macroblocks, and should be updated by using intra-coding if it has not been intra-coded for a long time. Therefore, a factor is posed on the Lagrangian cost to bias it for the intra mode with an increasing weight along time. A time counter for macroblocks that are inter coded is used here. This factor also needs to incorporate the same kind of information from the bit stream compressed by the other encoder. And an experimental parameter needs to be set here as a threshold which provides a function similar to refresh rate.

2.6 Summary

In this chapter, the robust error-resilient H.263 was introduced to recover the channel from the errors which damages the system. It is a modification for the H.263 which could not go over this default. In addition the segment regulation and precise error tracking scheme for the error robust H.263 were presented. In addition the error detection and correction technique for H.263 coded video over BSC channels were discussed. The criteria for detecting erroneous macroblocks in a decoded frame were analyzed and it was found that these criteria are sufficient to detect most of the erroneous macroblocks. The information about each macroblock whether erroneous or not, along with the received bitstreams are then used in a step-by-step decoding based correction. Further the spatial and temporal concealment can better be utilized to further improve the quality with this method. Finally, since correction and detection are applied iteratively at slice level, the decoding time is much larger compare to the conventional decoder.

CHAPTER THREE

A NEW ERROR RESILIENT CODING SCHEME for H.263 VIDEO TRANSMISSION

3.1 Overview

For entropy-coded H.263 video frames, a transmission error in a codeword will not only affect the underlying codeword but also may affect subsequent codewords, resulting in a great degradation of the received video frames. In this chapter, a new error resilient coding scheme for H.263 video transmission is introduced to overcome this problem.

3.2 Error Concealment for H.263 Video Transmission

Here both the best neighborhood matching (BNM) algorithm and the spatial error concealment algorithm in TML-9 are employed together to conceal H.263 intra-coded I frames. In the BNM algorithm, each corrupted block of size N×N is extracted from the video frame together with its neighborhood as a range block of size (N + m) × (N + m). Within a range block, all the pixels in the corrupted region belong to the lost part, and the others belong to the good part. After a range block is extracted, an H×L searching range block centralized with the range block within the video frame is generated. Each (N + m) × (N + m) block in the searching range block with no lost pixels may be a candidate domain block for recovery of the lost part of the range block, i.e., the corrupted block. For each candidate domain block, the mean square error (MSE) between the good part of the range block and the corresponding good part of the candidate domain block is evaluated.

The candidate domain block with the minimum MSE will be determined as the best domain block, which is used to conceal the lost part of the range block by copying its corresponding central part to the lost part of the range block. On the other hand, for the spatial error concealment algorithm in TML-9, each pixel value in a corrupted macroblock is formed as a weighted sum of the closest boundary pixels of the selected four-connected neighboring macroblocks.

Here, a corrupted macroblock within an H.263 I frame will be concealed by the spatial error concealment algorithm in TML-9 if the immediately bottom macroblock of the current corrupted macroblock is correctly received. If this condition is not satisfied, the corrupted macroblock within an H.263 I frame will be concealed by using the BNM algorithm.

On the other hand, the BNM algorithm is originally developed for still images and in this study, the "motion-compensated" BNM algorithm is used to conceal corrupted macroblocks in each H.263 P frame.

3.3 New Error-Resilient Scheme

For entropy-coded H.263 video frames, a transmission error in a codeword will not only affect the underlying codeword but also may affect subsequent codewords, resulting in a great degradation of the received video frames. To cope with the synchronization problem, each of the two top layers of the H.263 hierarchical structure, namely, picture and group of blocks (GOB), is ahead with a fixed-length start code. After the decoder receives any start code, the decoder will be resynchronized regardless of the preceding slippage. Although the propagation effect of a transmission error can be terminated when any start code is correctly received, a transmission error may affect the underlying codeword and its subsequent codewords within the corrupted GOB. Moreover, because of the use of motion-compensated interframe coding, the effect of a transmission error may be propagated to the subsequent video frames.

In general, error resilient approaches include three categories, namely: (1) the error resilient encoding approach, (2) the error concealment approach, and (3) the encoder-decoder interactive error control approach. The foregoing error resilient approaches concentrate on limiting error propagation and using the correctly received information to estimate corrupted video data. If the information of neighboring blocks is not available or the video contents between a corrupted block and its neighboring blocks are very different, the concealed results of the foregoing error resilient approaches are not good enough.

Recently, several error resilient coding approaches based on data embedding are proposed, in which some important data useful for error concealment at the decoder can be embedded into video frames, when they are encoded at the encoder. At the decoder, the embedded data for the corrupted blocks are extracted and used to facilitate error concealment performed at the decoder. The block type and edge direction index of a block is embedded within an image into the DCT coefficients of another block of the image by using the odd-even embedding scheme.

The proposed data embedding scheme for error-prone channels, in which some redundant information for protecting motion vectors and coding modes of macroblocks in one frame is embedded into the motion vectors in the next frame. However, if either the next frame of a corrupted frame is also corrupted or the total number of corrupted GOBs in a corrupted frame is larger than 2, the performance of the system will degenerate greatly.

3.3.1 Resilient Coding for H.263 I Frames

Because the human eyes are more sensitive to the luminance component than the chrominance component, in this study, the four quantized DC values for the Y component of each macroblock in an H.263 I frame are extracted as important data, which are identically quantized by a quantization parameter QDC. QDC is set to 64 and 5 bits are required to represent each DC value, i.e., 20 bits are required to represent the four corresponding DC values.

Here, the extracted important data for a macroblock within an H.263 I frame will be embedded into the DCT coefficients of another macroblock, called the masking macroblock, in the same I frame. A macroblock and its masking macroblock should be as far as possible so that both the two corresponding macroblocks will be seldom corrupted at the same time. Here, a macroblock and its masking macroblock should not be in the same

GOB and the masking macroblocks of the macroblocks of a GOB should not be in the same GOB.

To perform data embedding in H.263 I frames, the odd even embedding scheme operates on the quantized DCT coefficients. If the data bit to be embedded is "0," the selected quantized DCT coefficient will be forced to be an even number. If the data bit to be embedded is "1," the selected quantized DCT coefficient will be forced to be an odd number. Additionally, only the quantized DCT coefficients larger than a prespecified threshold, TI, are used to embed data bits.

Then it is noticed that for a macroblock within an H.263 I frame, if the extracted important data of its macroblock cannot be embedded completely into its masking macroblock, the "remaining" important data of the macroblock can be embedded into the corresponding macroblock in the next frame, with the threshold TI being replaced by another threshold TP.

At the decoder, for each corrupted macroblock within an H.263 I frame, its masking macroblock is determined accordingly. If the masking macroblock is correctly received, the embedded data for the corrupted macroblock can be extracted. Then, each corrupted macroblock is firstly concealed by the employed error concealment scheme. The Y component of the firstly "concealed" macroblock is transformed to four sets of DCT coefficients by the 8×8 discrete cosine transform (DCT), and the four firstly "concealed" DC values are replaced by the four corresponding extracted DC values from the corresponding masking macroblock. The resulted four sets of DCT coefficients are transformed back to pixels by the 8×8 inverse DCT to obtain the secondly-concealed macroblock. The resulted secondly concealed macroblocks are processed by a blocking artifact reduction scheme proposed in. It is noticed that if the masking macroblock of a corrupted macroblock is also corrupted, the corrupted macroblock is concealed only by the employed error concealment scheme.

3.3.2 Resilient Coding for H.263 P frames

For H.263 inter-coded P frames, similar to, a rate-distortion (RD) optimized macroblock coding mode selection approach is employed, which takes into account the network condition, including the video packet loss rate, the quantization parameter used in the encoder, the error concealment scheme used at the decoder, and the data embedding scheme used in the encoder.

The maximum intra-coded refresh period, Tmax, is imposed. Inserting intra-coded macroblocks with a maximum refresh period can not only limit the temporal error propagation, but also provide a larger capacity (larger DCT coefficients) for embedding important data. In this study, the error rate of the network is defined as the video packet loss rate and a video packet is equivalently one complete GOB. Hence, the probability of corruption of a macroblock is equivalently the video packet loss rate.

For H.263 inter-coded P frames, two bits are used to represent the coding mode of a macroblock. Because the motion vector for each macroblock includes two components, if the search range for motion vectors is ±15 with half-pixel accuracy, six bits are needed for each motion vector component. Hence, the important data for each macroblock in a P frame will range from two to at most fourteen bits, depending on the mode of the macroblock is intra-coded, inter-coded, or skipped. To reduce the size of the important data required to be embedded, based on the experimental results obtained in this study, for an inter-coded macroblock, if either (1) both components of the motion vector are identically zero, or (2) the MSE (mean square error) between the concealed macroblocks using the actual motion vector and the estimated motion vector is smaller than a threshold TM, the motion vector of the interceded macroblock should not be embedded in its masking macroblock. Here the odd even data embedding scheme is employed, in which the thresholds for intra-coded and inter-coded macroblocks are TI and TP, respectively. The important data extracted from the current P frame will be embedded into the next frame.

Because the smallest synchronization unit in H.263 video frames is a GOB, by adopting the even-odd block-interleaving technique proposed in, a macroblock interleaving GOB-based data embedding scheme is proposed for H.263 P frames, which is described as follows.

Assume that two non-adjacent GOBs in frame k are denoted by GOB A and GOB B, respectively. The important data for the two GOBs are extracted. The data extracted from the even-number macroblocks of GOB A and the data extracted from the odd-number macroblocks of GOB B are interleaved by the even-odd order and concatenated to a mixed bitstream and then the bitstream is embedded into its masking GOB in the next frame (frame $k + 1$) by using the odd-even data embedding scheme. On the other hand, the data extracted from the odd-number macroblocks of GOB A and the data extracted from the even-number macroblocks of GOB B are also interleaved by the even-odd order and concatenated to another bitstream, and the bitstream is embedded into another masking GOB in the next frame. The distance between two masking GOBs in the next frame of the two interleaved GOBs (GOB A and GOB B) in the current frame should be as far as possible so that two or more successive corrupted video packets in the next frame will not induce two corrupted masking GOBs in the next frame.

At the decoder, for each corrupted macroblock in a corrupted GOB, it's corresponding pair of masking GOBs are found first, and the important data of the corrupted macroblock is extracted from the corresponding pair of masking GOBs if the pair of masking GOBs is correctly received. Then if

1. the coding mode of the corrupted macroblock is "skip," the macroblock is concealed by copying the corresponding macroblock in the previous reconstructed frame.
2. the coding mode of the corrupted macroblock is "inter" and its motion vector information is recovered completely from the corresponding pair of masking GOBs, the macroblock is concealed by copying the motion compensated macroblock in the previous reconstructed frame;

3. the coding mode of the corrupted macroblock is "inter" and its motion vector (i) is not embedded, (ii) can not be embedded completely at the encoder, or (iii) cannot be recovered completely, the macroblock is concealed by using the employed error concealment scheme for H.263 P frames; and
4. the coding mode of the corrupted macroblock is "intra," the employed error concealment scheme for H.263 P frames is also employed.

Traditionally, the order of concealing consecutive corrupted macroblocks is in a raster scan manner. If all the eight neighboring macroblocks of a corrupted macroblock are received correctly or well-concealed, the concealed results of the corrupted macroblock will be better. Thus, before concealing a corrupted macroblock, its 8-connected neighboring macroblocks will be checked first. If some of its 8-connected neighboring macroblocks of the corrupted macroblock are also corrupted, and these corrupted neighboring macroblocks can be concealed only with important embedded data extracted from its masking macroblock(s), these corrupted neighboring macroblocks will be concealed first. Finally, the corrupted macroblock can be concealed by using the employed error concealment scheme for H.263 P frames with more neighboring macroblock information.

In this study, for a corrupted GOB, if only one of its masking GOB is corrupted, the even (or odd) macroblocks of the corrupted GOB can be concealed using the important data extracted from the "good" masking GOB first. Then the odd (or even) macroblocks can be concealed by the employed error concealment scheme with more neighboring macroblock information. Because the corresponding two masking GOBs are seldom corrupted simultaneously, the concealed results of the proposed macroblock- interleaving GOB-based data embedding scheme will be better than that of these approaches.

3.4 Summary

In this chapter, a new error resilient coding scheme for H.263 video transmission is introduced to solve the error problem caused by the transmission that will affect the underlying, and subsequent codewords that result in degradation in the received frames.

CHAPTER FOUR

SMART ANTENNAS in WCDMA

4.1 Overview

The multimedia transmission is one of the dominant applications of 3G cdma. However the wireless transmission is very sensitive to the time variant fading channels and the co-channel interference. In this chapter the smart antenna technique is introduced to suppress the unwanted interference.

4.2 Antenna Arrays

Wireless communication has created a continuing demand for increased bandwidth and better quality of service. With the ever-increasing number of mobile network subscribers, available capacity is becoming more of a premium. "Smart" antenna arrays are one way to accommodate this increasing demand for bandwidth and quality. These antenna arrays provide numerous benefits to service providers. However, the processing requirements for smart antenna arrays are many orders of magnitude greater than those for single antenna implementations.

4.2.1 How does it Work

A conventional antenna as omni directional radiates and receives information equally in all directions. This equal distribution leads to power being transmitted to, but not received by, the user. This wasted power becomes potential interference to other users or to other base stations in other cells. Interference and noise reduce the signal-to-noise ratio used by the detection and demodulation operations, resulting in poor signal quality.

To overcome the problems associated with omni directional arrays; smart antennas focus all transmitted power to the user and only "look" in the direction of the user for the received signal. This ensures that the user receives the optimum quality of service and maximum coverage for a base station. An intermediate step to this ideal is using directional antennas that divide the 360-degree coverage into sectors. The four directional antennas can each cover approximately 90 degrees.

Two types of smart antenna arrays are switched-beam arrays and adaptive arrays. The switched-beam arrays comprise a number of predefined beams. The control system switches among the beams and selects the beam that Adaptive antenna arrays, on the other hand, incorporate more intelligence into their control system than do switched beam arrays.

Adaptive antennas monitor their environment and, in particular, the response of the data path between the user and the base station. This information is then used to adjust the gains of the data received or transmitted from the array to maximize the response for the user. With adaptive antenna arrays, the control system has full flexibility and determines how the gains of the arrays are adjusted. By adjusting the gains in this way, the control system can – in addition to maximizing the gain from a particular user – also attenuate the signal from an interfering source, such as from another user or from multipath signals. Therefore, the adaptive arrays maximize the signal-to-interference-plus-noise ratio (SINR) and not just the signal-to-noise ratio (SNR).

This dynamic adaptation of the antenna array response provides focused beams to specific users and a new mechanism for multiuser access to the base station. Conventionally, multiple users are separated when accessing the base station by using different frequencies, as in frequency division multiple access (FDMA). FDMA is used in advanced mobile phone services (AMPS) and total access communications systems (TACS).

FDMA is also used in time, as in time division multiple access (TDMA) for global systems for mobile communications, or code division multiple access (CDMA) which is used in third generation.

4.3 Basics of smart antenna technology

The basic principle behind smart antennas is to control or reduce interference. Typically, this is accomplished through the use of narrow beams at the base site, both on the forward and reverse links. A smart antenna system combines multiple antenna elements with signal-processing capability to optimize its radiation pattern and/or reception pattern in response to the signal environment. The transmit and receive patterns are automatically updated as the subscriber moves through the cell or as signal conditions change. The goal of the smart antenna system is to provide the user with the highest quality uplink and downlink signal.

Smart antennas use an array of antenna elements connected to either an analogue or digital combining network. The size of the array and the number of elements determine the maximum gain and minimum beamwidth of the antenna array. This implies that a trade-off must be made between the size of the antenna array and the antenna gain, and to a lesser degree, the antenna side lobe performance.

Smart antennas form beams by adjusting the amplitudes and phases of the signals received from each of the antenna elements so that, when added together, they form the desired beam. This process is called beamforming. Beamformers can create a wide range of beams: scanned beams, multiple beams, shaped beams, and beams with steered nulls.

Two types of smart antenna arrays are switched-beam arrays and adaptive arrays. The switched-beam arrays comprise a number of predefined beams. The control system switches among the beams and selects the beam that Adaptive antenna arrays, on the other hand, incorporate more intelligence into their control system than do switched beam arrays.

Adaptive antennas monitor their environment and, in particular, the response of the data path between the user and the base station. This information is then used to adjust the gains of the data received or transmitted from the array to maximize the response for the user. With adaptive antenna arrays, the control system has full flexibility and determines how the gains of the arrays are adjusted. By adjusting the gains in this way, the control system can – in addition to maximizing the gain from a particular user – also attenuate the signal from an interfering source, such as from another user or from multipath signals. Therefore, the adaptive arrays maximize the signal-to-interference-plus-noise ratio (SINR) and not just the signal-to-noise ratio (SNR).

4.4 Smart Antennas in WCDMA System

Signal impairments in wireless personal communications are mainly due to intern symbol interference (ISI) and co-channel interference (CCI). The signal delays through the multipath channel cause the ISI, while the multiple accesses cause the CCI. Temporal and/or spatial signal processing is applied to repair the signal impairments. Temporal signal processing reduces the ISI using an equalizer or a rake receiver. Meanwhile, spatial signal processing reduces the CCI using a smart antenna. When spatial signal processing is combined with temporal signal processing, the space-time processing can further improve the impairments to result in a higher network capacity, coverage, and quality. A smart antenna not only suppresses interferences, but also combats multipath fading by combining the multiple antenna signals.

The smart antenna technique has been considered mostly for base stations because of its high system complexity and large power consumption. In addition, two (or multiple) antennas at a handset are in proximity, which may reduce the effectiveness of the antenna system. The feasibility of implementing dual antennas at a mobile handset was investigated in. The 3GPP (third generation partnership project) system requires antenna diversity at base stations and optionally at mobile stations. Recently, the smart antenna technique has been applied to mobile stations. For example, the HDR (high data rate) system of Qualcomm employs dual antennas at a mobile station.

Each antenna signal is applied to its own rake receiver that combines signals from different multipaths. Then, maximal ratio diversity combining is used to combine the two rake receiver signals. A dual antenna system for handsets is also applied to the digital European cordless telephone (DECT) system for indoor radio channel. The dual antenna handset receiver selects one of two signals of the receivers based on the signal-to-interference plus noise ratio (SINR). Each receiver processes a signal that is an equal combination of the signal from one antenna and the phase-shifted signal from the other antenna.

The additional antenna and the circuitry to process multiple antenna signals increase the cost and power consumption of the system. To justify employing multiple antennas at handsets, the performance gain should be large enough to offset the additional cost and power consumption. In this paper we present simulation results on the performance gain (in terms of frame error rate (FER)) of smart dual antennas at handsets for the cdma2000 system, which is one of the third generation (3G) code division multiple access (CDMA) systems proposed by TIA (Telecommunications Industry Association). Three types of the channel model, two levels of diversity combining, and three diversity combining schemes are considered. For the simulation, we used the SPW (signal processing work system) tool of Cadence to model the WCDMA system and to evaluate the performance.

4.4.1 Channel Model

Because the channel model influences the design of receivers and their performance, channel modeling is important when evaluating a system employing an antenna array. In the reverse link of the WCDMA system, each user signal is transmitted asynchronously and traverses different paths from a mobile station to the base station. Thus, the main source of interference is coming from other users' signals within the same cell (inter-cell interference), in which other user signal (interference) may be stronger than the desired user signal. However, in the forward link of the WCDMA system, the signal transmitted from the base station is the superposition of all active users' signals and control signals (pilot, sync, and paging signals). The desired user signal and multiple access interferences (MAIs) traverse the same paths, and they are inherently orthogonal of each other. Meanwhile, the main source of interference is coming from adjacent cells (intra-cell interference). Thus, unlike in the case of the reverse link, the interference is not a severe problem; the mobile station can select the strongest signal from different base stations and the number of adjacent base stations is small. Since a receiver with M antennas can suppress M-1 interferences, the dual antenna is a good candidate for the handsets. Here, we consider the interferences from adjacent cells as additive white Gaussian noises (AWGNs). Thus, only a simple diversity combining technique to process dual antenna signals is applied to obtain the diversity gain at handsets.

The dual antennas at a handset are identical, omni directional, and separated within a wavelength of the carrier. For a wireless channel model, three components are considered for a typical variation in the received signal level. The three components are mean path loss, lognormal fading (or slow fading), and Rayleigh fading (or fast fading). A channel model also considers spreads: i) delay spread due to multipath propagation and ii) Doppler spread due to mobile motion. We consider three types of the channel model for the dual antenna signals: i) uncorrelated fading channel model (Type I), ii) loosely correlated fading channel model (Type II), and iii) spatially correlated fading channel model (Type III).

Each antenna signal is assumed to have independent lognormal and Rayleigh fading in the uncorrelated fading channel model of Type I. In the loosely correlated fading channel model of Type II, each antenna signal is assumed have the same lognormal and Rayleigh fading in the spatially correlated fading channel model of Type III. Thus the two signals are different only in phase due to a nonzero angle of arrival (AOA). A channel model with less correlated dual antenna signals is expected to give higher diversity gain. Hence, Type I is for the best case and Type III the worst case. We believe that the actual channel may be close to Type II. Six multipath signals are considered in the channel model, and a multipath signal is assumed to have the same arrival time for the two antennas. For simplicity, only three multipath signals are presented in the figure. The signal s(t) represents the transmitted signal from the base station in the figure, and signals r1(t) and r2(t) represent the two received antenna signals at the mobile station.

4.4.2 The WCDMA System

The WCDMA is a synchronous CDMA system that was proposed by TIA as a third generation standard to meet the ITU (International Telecommunication Union) IMT-2000 (International Mobile Telecommunications) requirements. One frame of user data bits is randomly generated with a variable traffic data rate of 9600 bps, 4800 bps, 2700 bps, or 1500 bps. The generated data bits are appended with CRC (cyclic redundancy check) and tail bits. The data bits are convolution coded with the rate of ¼ and the constraint length of 9 and block interleaved. Then, data bits are parallelized for QPSK data modulation, and each parallel data bit is spread by Walsh code with the spreading factor of 64 and the chipping rate of 1.2288 Mcps. The resultant data signal is added with the pilot signal, the paging signal, the sync signal, and all the other users' signals. The added signal is quadrature modulated by two short-PN sequences and up-sampled by 8, and then is applied to shaping filters. The shaped signal is transmitted through the channel.

The received signal is shaped back and down-sampled by 8. A four-finger rake receiver dispreads each multipath signal and combines the dispreads multipath signals. The dispread and combined signal is applied to the channel decoder consisting of a block deinterleaver, a Viterbi decoder, and a CRC decoder. In the simulation the decoded data bits are compared with the original data bits to evaluate the system performance in terms of data rate decision error rate (DER), frame error rate (FER), and bit error rate (BER).

4.5 Summary

In this chapter, the smart antenna operating with WCDMA was discussed. In addition, the basics, types, and arrays were discussed in details. At the same time, the work of the arrays was stated.

CHAPTER FIVE
CONCLUSION

The multimedia services, such as video conferencing and video on demand are proposed with the help of the capability that the mobile communications have. Unfortunately, it faces a lot of problems mainly the interference and noise in the wireless channels especially in the implementation of bit-error-rate sensitive application.

First, the H.263, which was standardized by ITU as a low bit rates, was proposed to go over this problem by using motion compensated predictive coding and variable length coding. However, without any enhancement H.263 video sequence video sequence is error-sensitive in bit errors introduced by wireless channels as well as packet losses introduced by internet channel.

Second, the error-resilient H.263 which is the improvement of H.263 is proposed which mitigates the data error. Some research has proposed several protection methods against these channel errors. In this project the approach by explicitly modifying the operation of the video encoder or decoder to enhance robustness is used.

Finally, many research results have shown some improvements in reducing the transmission error. In this project greater performance improvement is achieved by incorporating several such strategies. An error-resilient H.263 video coding scheme is proposed for the wireless communication channel, in addition to the advantage taken by dedicating pilot channel in the WCDMA, in which the video bit streams are sent through an end-to-end WCDMA simulation system consisting of a smart antenna receiver. The smart antenna technique is used to combat the fading channel; also a robust decoder equipped with error recovery mechanisms is built in the receiver. The smart antenna demonstrates its ability to suppress interference and reduce the Bit Error Ratio (BER). The integration of both techniques leads to better video quality and more decoder frames.

Future work will further investigate the combination of smart antenna and error-resilient coding paradigms and joint optimization of the video codec and smart antenna processing based on a feedback channel.

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