IMAGE CODING, PACKETIZATION AND CHANNEL CODING FOR COMMUNICATION OVER WIRELESS NETWORKS – A REVIEW

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ABSTRACT
This paper is an exhaustive literature survey on the various wireless networking standards available and data transmission mechanisms currently employed. We discuss data representation, wireless network standards, image transmission mechanisms and two main channel encoding algorithms – Turbo codes and LDPC codes.

Keywords– Packetization, PSNR, QOS, SPIHT, Wireless Networks

I. INTRODUCTION

With the introduction of 3G wireless communication systems, together with phenomenal growth and popularity of internet, wireless multimedia communication is predicted to grow rapidly in near future. The transmission of multimedia data over wireless channels has become increasingly important during the last decade, especially after Wireless LAN networking has become a global standard for high data rate, low mobility users. Third generation cellular systems, such as IMT – 2000 and UMTS are becoming an important endpoint to provide multimedia data to final users, even at lower data rate. However, representing multimedia data requires a large amount of information, leading to high bandwidth, computation energy consumed in processing information to be transmitted and communication energy consumed when in wirelessly transmitting information.

The large requirements for bandwidth and energy consumption are significant bottlenecks to wireless multimedia communication. Multimedia system incorporates continuous media like voice, video, and images. This implies the need for multimedia systems to handle data with strict timing requirements and at high rate. This multimedia content provides rich information to consumers, but also with information it poses challenging problems of management, delivery, access and retrieval because of its data size and complexity. Most of these representations contain large amounts of redundancy can exist in various forms. It may exist in various form of correlation spatially close pixels in an image are generally also close in value.

Low bandwidth and limited channels are one of the most challenging issues that every nation is facing. Since signals and data to be sent via a channel are available in huge amounts and limited spectrum is available data needs to be compressed at the transmitter and expand at receiver side and sometimes even lead to false reconstruction at the receiver side. This compression and expansion leads to distortion of signals. Thus the
performance of the system decreases and hinders in smooth operation of the system. If the channel is wireless the situation becomes even worse. Different amount of noise enter the channel and corrupt data. If a digital image is sent via wireless channel it has to be compressed at transmitter because of limited bandwidth. At the receiver it may get distorted due to noise and during image expansion it might deviate from the original form and data may be lost while traversing. The key to compressing data is to the distinction between the data and information. In this wireless multimedia framework, the streaming of images requires some care on protection of compressed frames because random bit errors or packet losses introduced by the channel may corrupt critical information for decoding of images up to a point that no data at all can be decoded. Channel coding at code stream level plays a major role in preventing the transmission channel from introducing unrecoverable errors. Forward Error Correction is one of the possible solutions when there is no return channel is available for retransmission of packets.

A variety of error-resilient techniques for image transmission have been recently proposed in literature. Most are based on the state of art SPIHT source coder which generates embedded bit streams in which lower rates are prefixes of higher rates.

II. DATA REPRESENTATION

An image is a positive function on a plane. An image may be defined as a two dimensional function $f(x, y)$, where $x$ and $y$ are spatial or plane coordinates, and amplitude of $f$ at any pair of coordinates $(x, y)$ is called the Intensity or Gray level of image at that point[20].

When $x$, $y$, and intensity values of $f$ are all finite discrete quantities, we call the image as digital image. A digital image $f(x, y)$ described in a 2D discrete space is derived from an analog image $f(m, n)$ in a 2D continuous space through sampling process that is frequently referred to as digitization. Digitizing the amplitude values is called quantization. Digital image is composed of a finite number of elements each of which has a particular location and value. These elements are called picture elements, image elements, pels and pixels. Pixel is the used most widely to denote the elements of a digital image.

III. REPRESENTATION OF DIGITAL IMAGES

Let $f(m, n)$ represent a continuous image function of two continuous variable, $m$ and $n$. We convert this function into a digital image by sampling and quantization. We sample the continuous image into a 2D array $(x, y)$, containing $M$ rows and $N$ columns, where $f(x, y)$, are discrete coordinates. Generally integer values are used for notational clarity for discrete coordinates.

$$x = 0, 1, 2, \ldots, (M-1)$$
$$y = 0, 1, 2, \ldots, (N-1)$$

So in general, the value of image at any coordinates $(x, y)$ is denoted by $f(x, y)$, where $x$ and $y$ are integers. The section of real plane spanned by coordinates of an image is called the spatial domain, with $x$ and $y$ being referred to as spatial variables or spatial coordinates. We write the representation of an $MxN$ numerical array as

$$f(x, y) = \begin{bmatrix}
    f(0,0) & f(0,1) & \cdots & f(0,N-1) \\
    f(1,0) & f(1,1) & \cdots & f(1,N-1) \\
    \vdots & \vdots & \ddots & \vdots \\
    f(M-1,0) & f(M-1,1) & \cdots & f(M-1,N-1)
\end{bmatrix}$$
Each element of this matrix is called an image element, picture element, pixel element or pel. The digitization process requires that decisions be made regarding the values for $M$, $N$ and for number, $L$, of discrete intensity levels. $M$ and $N$ are positive integers. However, due to storage and quantizing hardware considerations, the number of intensity levels. Typically is an integer power of 2.

$$L = 2^k$$  \hspace{1cm} (4)

The discrete level is equally spaced and that they are integer in the interval $[0, L-1]$. The range of values spanned by the gray scale is referred to as dynamic range. Dynamic range of an imaginary system to be ratio of maximum measurable intensity to minimum detectable intensity level in the system. Upper limit of dynamic range is determined by saturation and lower limit by noise.

Generally the number, $b$, of bits required to store a digitized image is:

$$b = M*N*K$$  \hspace{1cm} (5)

When $M = N$, this equation becomes:

$$b = N^2*K$$  \hspace{1cm} (6)

Eight bits of precision for luminance is common in imaging applications. The eight bit precision is motivated by both the existing computer memory structure 1 byte = 8 bits as well as dynamic range of human eye. When an image can have $2^k$ intensity levels, that image is referred to as $k$-bit image.

TABLE 1 shows the multimedia data types and its requirements.

<table>
<thead>
<tr>
<th>Multimedia data</th>
<th>Size/duration</th>
<th>Bits/pixel or Bits/sample</th>
<th>Uncompressed size</th>
</tr>
</thead>
<tbody>
<tr>
<td>Page of text</td>
<td>11” x 8.5”</td>
<td>Varying resolution</td>
<td>16-32 bits</td>
</tr>
<tr>
<td>Gray scale image</td>
<td>512x512</td>
<td>8bpp</td>
<td>2.1Mb/Img</td>
</tr>
<tr>
<td>Color image</td>
<td>512x512</td>
<td>24bpp</td>
<td>6.2Mb/Img</td>
</tr>
<tr>
<td>Medical image</td>
<td>2048x2048</td>
<td>12 bpp</td>
<td>100 Mb/Img</td>
</tr>
<tr>
<td>Full motion video</td>
<td>640x640,10</td>
<td>24 bpp</td>
<td>2.21 Gbits</td>
</tr>
</tbody>
</table>

**IV. STANDARDS FOR WIRELESS NETWORKS**

Wireless networks are among the fastest growing areas in networking research. Wireless communication extends the capabilities of fixed networks to include location independent information storage, transport, retrieval and processing. Future wireless networks allow people on the move to communicate with anyone, anywhere, and at any time using a range of multimedia services and heterogeneous platforms, networks, and devices. Wireless networks play a major role in the application of active networking technology.

Wireless telecommunication history can be classified into different generations of networks. Each generation has been a significant stride which revolutionized the field of mobile or multimedia communication. Era of telecommunication started with 1G in 1980 where all the systems based on analog radio signal technology. Voice was considered to be main traffic. Various 1G standards defined were AMPS, NMT, TDMA and FDMA. In 1990 1G was replaced by 2G which provided rich set of services such as high voice quality and global mobility based on digital radio signal technology. In 2G also voice was main traffic. It includes GSM and GPRS. Both 1G and 2G are based on circuit switched technology for data communication at low speed. 2G was
huge success. Next 2.5 G is replaced by 3G which includes standards from 2.5 G and also some other technology such as Wi Max. It is based on both circuit switched and packet switched technology providing high data rate with low power consumption. It uses infrastructure of GSM and CDMA to provide its services.

Nowadays, more and more multimedia applications integrate wireless transmission functionalities. Wireless networks are suitable for those types of applications, due to their ease of deployment and because they yield tremendous advantages in terms of mobility of user equipment. However, wireless networks are subject to a high level of transmission errors because they rely on radio waves whose characteristics are highly dependent of the transmission environment. In wireless image transmission applications effective data protection is a crucial issue. JPEG 2000, the newest image representation standard, addresses the issues by including predefined error resilient tools in its encoding system. The main characteristics of JPEG 2000 are: lossy or lossless compression modes; resolution, quality and spatial scalability; transmission and progressive image reconstruction; error resilience for low bit rate applications; region of interest functionality, etc.

### 4.1 Characteristics of a Wireless Channel

Due to severe wireless channel conditions, such as path loss, fading, co-channel interference, and noise disturbances, the capacity of wireless channels is much lower than wired channels, and bit error rate (BER) is much higher. A wireless network offers both advantages and disadvantages compared to wired network. Advantages of wireless network include mobility and elimination of unsightly cables. Disadvantages of wireless include the potential for radio interference due to weather, other wireless devices, or obstructions like walls. The throughout may fluctuate due to the time varying characteristics of wireless channels. The severe channel conditions have placed another major obstruction when designing efficient image communication systems over wireless networks.

### V. IMAGE TRANSMISSION OVER WIRELESS NETWORKS

To design an efficient image communication system over wireless networks[12] there still exist many challenges, of which some are caused by resource limitations, such as power supply, processing capability, some wireless channel conditions, some due to the special characteristics of compressed image data. Image compression is minimizing the size in bytes of a graphics file without degrading the quality of the image to an unacceptable level. The reduction in file size allows more images to be stored in a given amount of disk or memory space. It also reduces the time required for images to be sent over the Internet or downloaded from web pages. There are several different ways in which images can be compressed. For internet use, the two most common compressed graphic image formats are the JPEG format and the GIF format. The JPEG method is more often used for photographs, while the GIF method is commonly used for line art and other images in which geometric shapes are relatively simple. In image compression, a small loss in quality is usually not noticeable. When there is some tolerance for loss, the compression factor can be greater than it can when there is no tolerance. For this reason, graphic images can be compressed more than text files or programs. Since image data contains a lot of redundancy, to efficiently utilize limited resources, source compression is always necessary. Wavelets offer an elegant technique for representing the levels of details present in an image. When an image is decomposed using wavelets, the high pass component carry less information. And low pass components carry more information. The possibility of elimination of high pass components gives higher compression ratio in
wavelet based image compression. To achieve higher compression ratios, various coding schemes have been used. Some of the well-known coding algorithms are EZW (Embedded Zero tree Wavelet), SPIHT (Set Partitioning in Hierarchical Tree) and EBCOT (Embedded Block Coding with Optimal Truncation). SPIHT has been one of the popular schemes used for compression. The SPIHT algorithm, developed by Said and Pearlman in 1996 is a fast and efficient image compression algorithm works by testing ordered wavelet coefficients for significance in a decreasing bit plane order, and quantizing only the significant coefficients.

![Figure 1: A Wireless Network Schematic](image_url)

The high coding efficiency obtained by this algorithm is due to group testing of the coefficients of a wavelet tree. The SPIHT algorithm is a refined version of EZW algorithm. It can perform better at higher compression ratios for a wide variety of images than EZW. The algorithm uses a partitioning of trees in a manner that tends to keep insignificant coefficients together in larger subsets.

The SPIHT algorithm groups the wavelets coefficients and trees into sets based on their significant information. The encoding algorithm consists of two main stages, sorting and refinement. In the sorting stage, the threshold for significance is set to as $2^n$, where $n$ is the bit level, and its initial value is determined by number of bits required to represent the wavelet coefficient with maximum absolute value. Significance for trees is obtained by checking all the member detail coefficients.

In compared with general data, the compressed image data has some special characteristics, such as unequal importance, error tolerance and constrained error propagation. In this, we consider wireless image transmission system for images coded using SPIHT with baseline JPEG 2000 and 3G as the medium and propose a FEC scheme based on low density parity check codes.

When an image is transmitted from source to destination over wireless links it will be subject to random bit errors caused by channel impairments. A single bit error will result on the occurrence of a corrupted of neighboring bits also. This stops the image transmission process and whole process has to be repeated again and again until there is no loss. This results in incomplete transmission. An efficient image transmission system over wireless networks is faced by many challenges, of which some are caused by resource limitations such as:

- Wireless channel conditions
• Special characteristics of compressed image data. The characteristics of compressed data is its PSNR value and compression ratio. The PSNR (dB) is calculated with the following formula:

\[
PSNR = 10 \log_{10} \left[ \frac{\text{Max grey level} \cdot MN}{\sum_{xy} \left| g(x,y) - f(x,y) \right|^2} \right]
\]

Where \( g(x,y) \) is the compressed image, \( f(x,y) \) is the raw image, \( M \) is the image width, \( N \) is the image height and max grey level is the maximum value of \( f(x,y) \). A maximum grey level is equal to 255.

• Maintaining a good quality of service (QoS) for multimedia over wireless networks

Multimedia over wireless will certainly need a higher bandwidth. As wireless data transmissions incur the most data loss and distortion, error resilience and error correction become primary concerns. Wireless channels have more interference than wired channels, with specific loss patterns depending on the environment conditions. The bitrates for wireless channels are also much limited as 3G bit rates are most suitable for images. This implies that although a lot of bit protection must be applied, coding efficiency has to be maintained as well. Error resilient coding is important. The QoS parameters specified for wireless multimedia data transmission depends on the following parameters:

• Data rate: A measure of transmission speed, expressed in kilobits per second (kbps) or megabits per second (Mbps)

• Latency (maximum frame/packet delay): Maximum time needed from transmission to reception, measured in milliseconds.

• Packet loss or error: A measure of error rate of the packetized data transmission. Packets get lost or garbled, such as over the internet. They may also be delivered late or in wrong order. Since retransmission is often undesirable, a simple error recovery method for multimedia is to repay the last packet, hoping the error is not noticeable.

• In general, for uncompressed image desirable packet loss is <10\(^{-2}\). For compressed multimedia data desirable packet loss is less than 10\(^{-7}\) to 10\(^{-8}\).

• Jitter (or delay jitter): Jitter is referred to variance of frame/packet delays. It is the worst case of variation in delay.

• Sync skew: A measure of multimedia data synchronization, often measured in milli-seconds. In general, ±200 msec is acceptable.

All types of multimedia information are stored and processed within a computer in a digital form.

<table>
<thead>
<tr>
<th>Network Generation</th>
<th>Sound</th>
<th>Sight</th>
<th>Knowledge</th>
</tr>
</thead>
<tbody>
<tr>
<td>1G-2G</td>
<td>Voice</td>
<td>--</td>
<td>Low speed Data</td>
</tr>
<tr>
<td>3G</td>
<td>Voice, Images</td>
<td>Video</td>
<td>Files (speech, HT, video)</td>
</tr>
<tr>
<td>4G</td>
<td>Voice, speech</td>
<td>Video</td>
<td>Files (speech, HT, video)</td>
</tr>
<tr>
<td>Typical Band width</td>
<td>10-80 kbps</td>
<td>1-20Mbps</td>
<td>0.5-10Mbps</td>
</tr>
<tr>
<td>Required Latency</td>
<td>≤160ms</td>
<td>≤100ms</td>
<td>≤5s</td>
</tr>
<tr>
<td>Principle Application</td>
<td>Communication</td>
<td>Entertainment</td>
<td>Information</td>
</tr>
</tbody>
</table>

VI. TURBO CODES – THE CURRENT CHANNEL CODING ALGORITHM FOR 3G

7.1 ABOUT 3G
Wireless services have the highest demand in wireless and internet world. Every generation technology has some platform for its development. 1G was based on analog signaling whereas 2G on low-band digital data signaling. The 3G technology was developed to overcome the faults of 1G and 2G. 3G finds its applications in wireless technologies, as high-speed transmission, advanced multimedia access. It can support data rates between 128 and 144 kbps for devices that are moving fast and 384 kbps for slow ones. For fixed wireless LANs, the speed goes beyond 2mbps. The spectral efficiency ranges from 1-5Mhz. 3G supports packet switching and its network architecture is wide-area cell based [19].

3G technology provides both circuit design and Packet design. Circuit design, being the oldest, has greater ability to hold connection for a longer duration. On the other hand the packet design is a wireless technology and is the core part of internet data transmission. The combination of these two patterns helps 3G technologies to perform better and faster. 3G uses licensed spectrum. 3G, network based QoS depends on following to provide a satisfactorily service as: Throughput, Packet Loss Rate, reliability and delay. 3G can be used with wireless LAN for better quality of service.

Channel coding is the key element of any digital wireless communication systems [21] since it minimizes the effect of noise and interference of any transmitted signal. In 3G wireless systems channel coding techniques must serve both voice and data users whose requirements considered may vary. Thus in third generation partition project (3GPP) standard offers two coding techniques convolutional coding for voice and Turbo-coding for data services.

Channel coding allows to reduce the power by maintaining the QoS or to improve QoS for a given transmitted power. 3G systems offers different data rates and coding techniques to satisfy the varying latency, throughput and error-performance requirements. Example data services which require lower error rate and high throughput, but can tolerate a larger latency.

7.2 About Turbo Codes

The implementation complexity of encoders in turbo coding is negligible. Both are trellis encoders which map a long input sequence to a coded data stream. They consist of shift registers and an interleaved address generator. The decoders are based on trellis propagation of received input sample sequence to calculate a maximum likelihood sequence or bit detection.

The basic system consists of two identical systematic recursive convolutional encoders connected in parallel with an inter-leaver preceding the second recursive convolutional (RSC) encoder [20]. RSC encoders encode the information bits. The first encoder operates on the input bits in their original order while the second one operates on the input bits as permuted by the inter-leaver.

Forward error correction is enabled by introducing parity bits. For turbo codes, the original information is denoted as systematic information is transmitted together with the parity information. The source for 3GPP consist of two recursive systematic encoders with constraint length $k=4$. Each transmitted block is iteratively decoded. The systematic information and parity information serves as input to first component to decoder. The decoding algorithm involves the joint estimation of two Markov processes one for each constituent code.

The 3G standard specifies the encoder structure and parameters like block size and throughput requirements. The maximum block size of turbo codes including the data, frame quality indicates CRC and two reserved bits is
set to 5114. During encoding an encoder output tail sequence is added which appends another 3 bits the tail bits, for both systematic and parity information of each encoder.

VIII. LDPC CHANNEL CODING FOR 3G

We propose to use LDPC coding as an alternative algorithm for 3G channel coding. LDPC codes were invented by Gallager in early sixties, [25] their importance as capacity approach and performance analysis issues are currently being addressed in coding environments and applications. Basically LDPC codes are linear block codes with a very sparse parity check matrix \( H \) order \( N \times M \). Typically, the matrix \( H \) is generated by applying random perturbations to the zero matrixes until a specified number of ones appear in each column and roughly fixed equal number of ones appear in each row. The associated generator matrix \( G \) can be obtained by Guassian elimination of \( H \), where \( G \) is not necessarily sparse. The pseudo random parity check matrix also leads to LDPC codes that have random like properties with channel coding theorem conditions. LDPC codes could be used to transmit information reliably at the rates close to channel capacity. The decoding can be done by sum product algorithm. This is an iterative probabilistic decoding algorithm that begins with \( H \) and a set of prior probabilistic for the \( N \) bits. It then iteratively updates these on \( M \) parity checks until all of the parity checks are satisfied. Sometimes decoder fails if it cannot satisfy all parity checks. Decoding algorithm works well provided \( H \) does not contain patterns where two columns have two or more check positions in common.

The advantages of using this coding algorithm are as follows [9]:

- LDPC codes with iterative decoding provide very good performance over a variety of channels with reasonably low complexity. In particular irregular LDPC codes outperform turbo codes and regular codes and to approach to channel capacity of several channels at large block lengths.

- Another advantage of LDPC codes over turbo codes is efficient hardware implementation of the decoder.

- These algorithms are parallelizable and can be realized at much faster speed than turbo decoders and thirdly almost all errors are detectable [1]. This is because in turbo coding a large decoding delay is introduced owing to large block lengths and many iterations of decoding required for near-capacity performance and significantly weakened performance at BERs below \( 10^{-5} \).

Due to above advantages of LDPC codes we have chosen these codes for encoding and decoding purpose for wireless image transmission maintaining quality of service.

IX. CONCLUSION

In this paper, we present an overview of the image transmission process over wireless networks. We describe the various techniques for source coding, channel coding and transmission of images through wireless networks. We find that there are research opportunities that exist in channel coding since there are significant advantages of using LDPC codes over Turbo codes.

REFERENCES


