CODING, STREAMING AND WATERMARKING – SOME PRINCIPLES IN MULTIMEDIA SIGNAL PROCESSING

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SUMMARY

In this paper, a review of signal processing for networked multimedia is presented. The emphasis is on three important parts of the area, like coding, streaming and watermarking technologies. Multimedia signal (speech, audio, acoustic, image, video, graphics and data) processing is analyzed, as well as perceptual coding of digital audio signals. Another part of the paper seeks to provide watermarking algorithms, and its application in intellectual copyright protection.

Keywords: *multimedia, signal processing, coding, streaming, watermarking*

1. INTRODUCTION

Signal processing is one of the most relevant disciplines, because diverse algorithms, concepts and applications are interconnected (spectral analysis, sampling theory, different equation theory, coding theory). It is well known that multimedia understands integration of text, audio, images and video, as well as interaction among these media. It creates new systems and new research challenges. The term multimedia represents different concepts, like different audio types (diverse sources), musical presentation, visual experience (representation of the real world as well as its model through a synthetic representation). Multimedia consists of

{multimedia data} + {set of instructions}

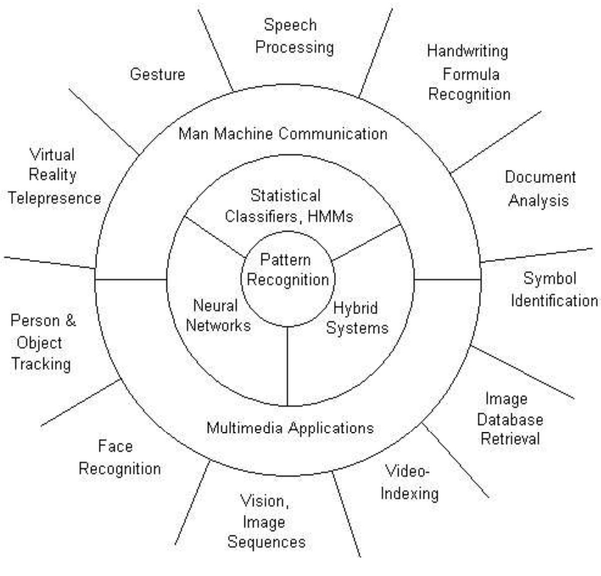

Fig. 1 Signal processing application in different multimedia areas

Multimedia data is collection of three multimedia data: multi-source, multi-type and multi-format data. Set of instructions is simple set of visual, audio and other data [1].

Multimedia signals processing (MMSP) is the representation, interpretation, coding and decoding of multimedia data using signal-processing tools. The goal of MMSP is [2]:

- effective and efficient access
- manipulation
- exchange and storage of multimedia content delivery for various multimedia applications

The revolution in digital technology has increased the ease of manipulation, reproduction, retransmission and distribution of digital images. However, it also offers the potential for illegal use, known as piracy of digital images. Watermarking is a tool for protecting intellectual properties. It attempts to identify the through owner by hiding perceptually invisible information (watermark) inside the digital data. A watermark is a secrete code described by digital signal carrying information about the copyright property of the product.

The application of signal processing in different multimedia areas is shown in Fig. 1.

In the first part of this paper, we will deal with the speech, audio, acoustic and video signal processing, taking into account perceptual coding of digital audio signals. Some streaming issues for speech, audio and video will be processed, too. In the second part, the emphasis will be on watermarking technologies for copyright protection.

2. CHALLENGES OF MULTIMEDIA PROCESSING

Multimedia systems must successfully combine digital video and audio, text animation, graphics, and knowledge about such information units and their interrelationships in real time. Filtering, sampling, spectral analysis and signal representation are basic to all of signal processing. Algorithms for processing m-D signal (m represents the dimension of a signal) can be grouped into four categories [3]:

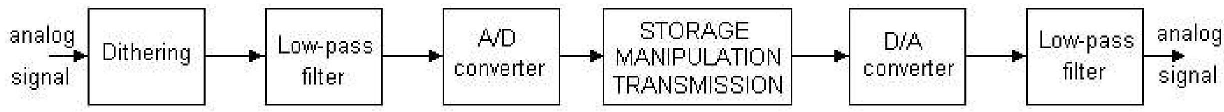

Fig. 2 General scheme of multimedia signal processing

a) separable algorithms that use 1-D operators to process the rows and columns of a multidimensional array,
b) nonseparable algorithms that borrow their derivation from their 1-D counterparts,
c) *m*-D algorithms that are significantly different from their 1-D counterparts,
d) *m*-D algorithms that have no 1-D counterparts.

It is mandatory to use a pre-processing step prior to coding in order to enhance the quality of the fined pictures and to remove the various noises that will affect the performance of compression algorithms. Advances post-processing mechanisms have been studied to improve lip synchronization of head and shoulder video coding at very low bit rate by using knowledge of decoded audio in order to correct the positions of the lips of the speaker. General scheme of multimedia signal processing is given in Fig. 2.

2.1 Speech, audio and acoustic processing

The primary advances that contribute to multimedia applications are in the areas of speech and audio signal compression, speech synthesis, acoustic processing and echo-control, as well as network echo cancellation.

Speech and audio signal compression aim at efficient digital representation and reconstruction of speech and audio signals for storage and playback, as well as transmission. For compression, vector quantization (VQ), marks a major advance. In corporation of knowledge and models of psychophysics in hearing has been proven beneficial for speech and audio processing. Techniques such as noise shaping and explicit use of auditory masking in the perceptual audio coder, have been found vary useful. Excellent speech quality can be obtained at less than 8 kb/s which forms the basis for cellular, as well as Internet telephony. Advances audio coding standards are supporting in Motion Picture Expert Group (MPEG) activities.

Speech synthesis area includes generation of speech from unlimited text, voice conversion and modification of speech attributes, such as time scaling, and articulator mimic. Text-to-speech conversion takes text as input and generates human like speech as output. Key problems in this area include [4]:

- conversion of text into a sequence of speech inputs (in terms of phonemes, dyads, or syllables)
- generation of the associated prosodic structure and intonation as well as methods to concatenate and reconstruct the second waveform
- voice conversion refers to the technique of changing one person's voice to another, from person A to person B, or from male to female, and vice versa
- it is useful to be able to change the time scale of a signal (to speed up or slow down the speech signal changing the pitch) or to change the mode of the speech (making it sounds happy or sad)
- many of these signal processing techniques have appeared in animation and computer graphic applications

The idea of acoustic signal processing and echo control is to allow straightforward high-quality sound pickup and playback in applications, such as a duplex device like a speaker phone, a sound source tracking apertures like microphone arrays, teleconferencing systems with stereo input and output, hands-free cellular phones, home theater with 3-D sound.

Signal processing for acoustic echo control includes modeling of reverberation, design of dereverberation of algorithms, echo suppression, double-talk detection and adaptive acoustic echo cancellation, which is still a challenging problem in stereo full duplex communication environments.

For sound pickup, acoustic processing aims at the design of transducers, or transducers to achieve a durable directionality (beam steering and width control), as well as noise resistance. Various 1-D and 2-D microphone arrays have been shown in teleconferencing and auditorium applications with good results.

Network echo chancellors where invented to correct the problem in the late 1960s, based on the least mean squares (LMS) adaptive echo cancellation algorithm. The network echo delay is on order of 16 ms requiring typically a filter with 128 taps at the sampling rate of 8 kHz.

2.2 Video signal processing

Digital video has many advantages over conventional analog video, including bandwidth compression, robustness against channel noise interactivity and easy of manipulation. Exchange of video signals between TV and personal computers (PCs) requires effective format conversion. As for video filters, they can be classified as interframe/field (spatial), motion-adaptive, and motion-compensated filters. Spatial filters are easiest to implement, but they do not make use of the high temporal correlation in the video signals. Motion compensated filters require highly accurate correspondence estimation between successive views. Other more sophisticated format conversion methods include motion adaptive field rate doubling and deinterlacing, as well as motion compensated frame rate conversion.

Some filters are adaptive to scene content in that they aim to preserve spatial and temporal edges, while removing the noise. Examples include media, weighted media, adaptive linear mean square error, and adaptive weighted-averaging filtering. Reblocking filters can be classified as those that do require a model of the degradation process (inverse, constraint, least square, Wiener filtering), and those that do not (contrast adjucement by histogram specification and unsharp masking). Reblocking filters smooth intensity variations across amount of temporal redundancy.

Assuming that the frames are shifted by subpixel amount with respect to each other, it is possible to exploit this redundancy to obtain a high-resolution reference image (mosaic) of the regions covered in multiple views. One of the challenges in digital video processing is to decompose a video sequence into its elementary parts (shots and objects). A video sequence is a collection of shots, a shot is group of frames and each frame is composed of synthetic of natural visual objects. Temporal segmentation methods added effects as cuts, dissolves, faiths and wipes.

Tresholding and clustering using histogram-based similarity methods have been found effective for detection of cuts. Detection of special effects with high accuracy requires customized methods in most cases, and is a current research topic. Segmentation of objects by means of chrome keying is relatively easy and is commonly employed. Object tracking algorithms that can be classified as boundary region or model based tracking methods, can be based on 2-D or 3-D object representations. Effective motion analysis is an essential part of digital video processing, and remains an effective research topic.

2.3 Storage, archiving, browsing and retrieval

Storage and archiving of digital video in shared disks and servers in large volumes browsing of such databases in real time, and retrieval over switched and packet networks pose many new challenges; one of each is efficient and effective description of content. It is of interest to browse and search for content using compressed data, since almost all video data will likely be stored in compressed forward. Video indexing system may employ a frame-based, scene-based, or object-based video representation. The basic components of a video indexing system are temporal segmentation, analysis of indexing features, and video summarization. The temporal segmentation step extracts shots, scenes, and/or video objects. The analysis step computes content-based indexing features for the extracted shots, scenes, or objects. Content-based features may be generic, or domain dependent. As for content-based browsing, it can be facilitated by a visual summary of the contents of a program, much like a visual table of contents.

Multimedia (MM) signal processing methods allow efficient access to processing and retrieval of content in general, and visual content in particular. This is required across large range of applications in medicine, entertainment, consumer industry, broadcasting, journalism, and e-commerce. Methods originating from numerous research areas, i.e., signal processing, pattern recognition, computer vision, database organization, human computer interaction, and psychology, must contribute to achieve the image retrieval goal. Image retrieval methods face several challenges when addressing this goal. Image retrieval challenges are presented in Tab. 1.

Tab. 1 Image retrieval challenges

Challenges	***Remarks***
Query types	Color-based/shape-based/color and shape-based
Query forms	Quantitative, e.g.: find all images with 30 % amount of red Query by example, e.g.: image region/image/scatch/ other examples
Various content	Natural scenes/head-end-shoulder images
Matching types	Object to object/image to image/object to image
Precision levels	Applications-specific Exact versus similarity-based match
Presentation of results	Application-specific

Content-based image retrieval methods focused on using low-level features such as color, texture, shape layout, because such features can be extracted automatically or semi-automatically.

2.4 Perceptual coding of digital audio signals

The central objective is to generate output audio that cannot be distinguished from the original input, even by a sensitive listener. The introduction of the compact disc (CD) brought all of the advantages of digital audio representation, including high fidelity, dynamic range and a robustness. However, these advantages came at the expense of high data rates [5].

The research interests are lossy compression schemes, which seek to exploit the psychoacoustics principles. Lossy schemes offer the advantage of lower bitrates (less than 1 bit per sample relative to lossless scheme). The lossy compression systems achieve coding gain by exploiting both perceptual

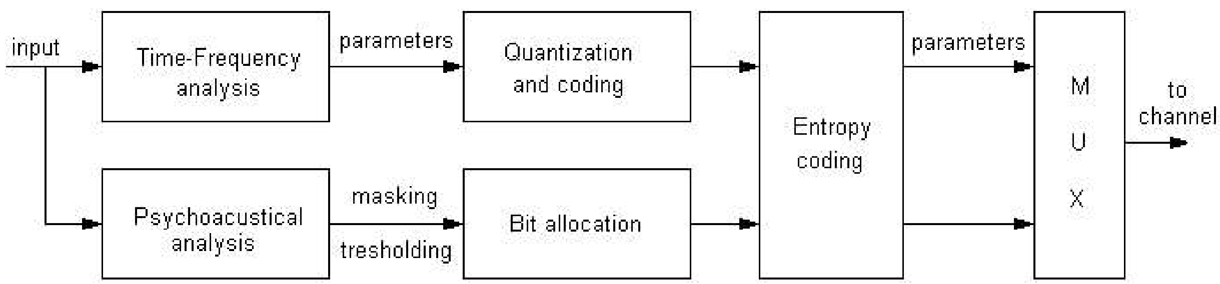

Fig. 3 Perceptual audio coder architecture

irrelevancies and statistically redundancies. Perceptual audio coder architecture is shown in Fig. 3.

The time-frequency analysis approximates the temporal and spectral analysis properties of the human auditory system. It transforms input audio into a set of parameters that can be quantized and coded according to a perceptual distortion metric. Frequency analysis section might contain:

- unitary transform,
- time-invariant bank of uniform bandpass filters,
- time-varying, critically sampled bank of non-uniform bandpass filters,
- hybrid transform/filter bank signal analyser,
- source system analysis (linear predictive coding LPC/multiple excitation).

The quantization and encoding section also can also exploit statistical redundancies through classical techniques, such as Differential Pulse Code Modulation (DPCM), or Adaptive DPCM. Quantization might be uniform, or optimized in the probability density function (PDF) sense (Lloyd-Max quantizer). It might be performed on scalar or vector quantities. Once quantized compact parametric set has been formed, remaining redundancies are removed through run-length and entropy coding techniques (Huffman, arithmetic, Lempel Ziv LZW coding). The study of perceptual entropy suggests that transport coding is possible in the neighborhood of 2 bits per sample for most high fidelity audio sources. Perceptual audio coders seek to achieve transparent quality at low rates with tractable complexity and manageable delay.

3. SIGNAL PROCESSING FOR NETWORKED MULTIMEDIA

Although the range of problems covers large variety of topics, two groups attract the most attention:

- adapting the signal compression techniques to address the spatial requirements imposed by the packet networks, including accommodating for packet losses, delays and jitter, providing capability for multipoint and copying with the heterogeneous nature of today's networks,
- to protecting the intellectual property rights (IPR) associated with the transmitted multimedia data.

The compression and transmission aspects have generally been treated as separate issues [6]. The first problem with this approach is that the resulting compression algorithms usually do not address the needs of networked transmission. Effective error concealment techniques must be present in the receiver to minimize the visual impact of any errors. A straightforward approach to provide a robustness is to insert pointers into the compressed data to make the partially received data usable. Another consideration in designing compression techniques for network use is to identify the impact of loosing different portions of a compressed stream. Considering the relative ease of providing error-free transmission for shorter data segments, it is preferable to have the important of a compressed stream concentrated into a short and identifiable segment [7].

3.1 Streaming issues for speech and audio

Streaming refers to the transmission of multimedia signals for real-time delivery without weighting for entire file to arrive and user terminal. Streaming can be either narrowcast (from the server to just one client) or broadcast (one transmission stream to multiple clients). The real-time information is flowing solely from one source in both cases. There are four main elements to a streaming system [8]:

- the compressed (coded) information content, e.g. audio, video, speech, multimedia, data, etc.,
- the content, which is stored on a server,
- the server, which is connected to the Internet and/or possibly other networks (Integrated Services Digital Network ISDN, Asynchronous Transfer Mode ATM),
- the clients.

Each of these elements can cause impairment.

The access could be through a modem on a line, via ISDN, via a corporate Local Area Network (LAN) running Internet Protocol (IP), or could even include ATM or frame-relay in the access link. The two manifestations of network congestion of the packet stream that represents the real-time signal are highly variable delays and lost packets. There is also problem that lost packets cause the decoder state to loose synchronization with the encoder state. Generally, forward adapted coders can resynchronize the encoder and decoder faster than backward adaptive coders. Hence, forward adaptive coders are preferred for highly congested data network with streaming speech signals.

The server's function in a streaming transaction is to transmit the information (the coded speech or audio) at one average rate designed to maintain real-time decoding of the compressed material. If the server is working too slowly (i.e. heavily overloaded), the real-time signal received by the client will have gaps caused by the decoder running out of bit stream to decode. If the server is transmitting it quickly (i.e. under loaded conditions), the bit stream will build up in the buffers of decoder, eventually overfilling them, and causing a loss of signal, because of a buffer overflow. The server must serve multiple clients, and must respond to changes in the network due to variable traffic and congestion. The principle problem is congestion at a few notes, or over a few links. The congestion results in both variable delays and lost packets.

3.2 Streaming issues for video

Streaming is implemented as part of the application-layer protocols of the transmission, i.e. it uses user data gram protocol (UDP), and transmission control protocol (TCP) at the transport layer. The extent of the losses is a function of the network congestion, which is highly correlated with the time of day and the distance (in terms of number of the routers) between the client and the multimedia source. The practical techniques that have evolved for improving the performance of streaming-based real-time signal delivery can be classified into four broad areas:

a) client-side buffer management determining how much data needs to be buffered both prior to the start of the streaming playback, and during the playback, and determining a strategy for changing the buffer size, as a function of the network congestion and delay and the load on the media server
b) error-resilient transmission techniques increasing client-side resilience to packet losses through intelligent transport techniques, such as using higher priority for transmitting more important parts (headers, etc.) of a stream, and/or establishing appropriate retransmission mechanisms, where possible
c) error-resilient coding techniques using source and perhaps combined source and channel coding techniques that have built-in resilience to packet losses
d) media control mechanisms using efficient implementations of variable coding rate (VCR) type controls when serving multiple clients

None of these techniques is sufficient to guarantee high-quality streaming, but in combination they serve to reduce the problems to manageable levels for most practical systems.

4. WATERMARKING

A watermark is a secrete code described by a digital signal carrying information about the copyright property of the product. The following requirements should be satisfied by a watermarking algorithm [9]:

- alternation introduced in the image should be perceptually invisible,
- watermarking must be undetectable and not removable by an attacker,
- a sufficient number of watermarks in the same image, detectable by their own key can be produced,
- the detection of the watermark should not require information from the original image,
- watermark should be robust as much as possible against attacks, and image processing which preserves desired quality for the image.

Digital data embedding has many applications, like:

- passive and active copyright protection,
- a means to identify the owner or distributor of digital data,
- provides a mechanism for embedding important control descriptive or reference information in a given signal,
- provides different access levels to the data,
- can extract the hidden data from the host signal with no reference to original signal.

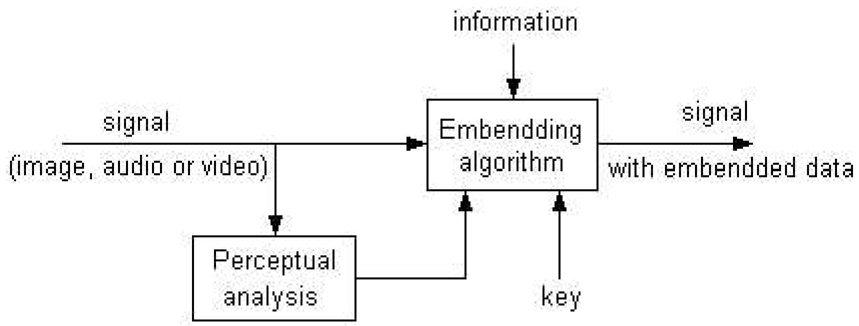

Fig. 4 Block scheme of a data embedding algorithm

Block scheme of a data-embedding algorithm is shown in Fig. 4.

The information is embedded into the signal using the embedded and a key. The algorithm may directly exploit perceptual analysis to embed the information. Embedding data would not be possible without the limitations of the human visual and auditory systems. Data embedding and watermarking algorithms embeds text, binary streams, audio, image or video in a host audio, image and video signal. The embedded data are perceptually inaudible or invisible to maintain the quality of the source data. They can add features to the host multimedia signal, multilingual sound tracks in a movie, or provide copyright protection [10].

4.1 Watermarking technologies for copyright protection

Watermarking is a tool for protecting intellectual properties. It attempts to identify the through owner by hiding perceptually invisible information inside the digital data. Many media urgently need to be protected (digital versatile disc DVD, audio compact disc CD, the audio-based on MPEG-1 layer-3 MP3 standard). There are three approaches designed for protecting the intellectual property and securing the system. These are:

a) data encryption (cryptography),
b) authorization verification (using a signature or password to access the system),
c) watermarking strategies.

Representation of general data security system is shown in Fig. 5.

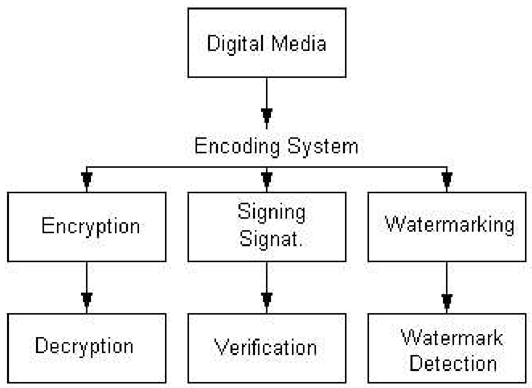

Fig. 5 Representation of general data security system

Only a person who possesses appropriate key (or keys) can decrypt the encrypted data. The weakness from this data protects strategy is that once such data is decrypted, there is no way to protect and track its reproduction.

The second strategy for protecting the intellectual property is an authorized verification system.

The basic concept of watermarking strategy is to directly cast an ideally undeletable watermark or unique signature within the digital media, rather than to encode into a header or wrapper based on this strategy, the signature used for proving the authentication will be permanently affixed with the information, and can extensively fulfill the requirements of copyright protection.

The invisible watermark embedded into the image can be divided into a fragile watermark, and a robust watermark. Fragile watermark scheme is generally desired such that the watermark is sensitive to image content manipulation through many sorts of image processing algorithms and geometric distortion operations. For the robust scheme, watermark has opposing technical properties. The watermark must remain in a watermarked image, including reformatting a watermarked image such as print, scan and photocopy.

Transparency is one of the most significant requirements of digital invisible watermarking schemes. The less energy of a watermark to be embedded, the more vulnerable it is to any kind of attacks.

The characteristics of Human Visual system (HVS) are exploited to adapt the watermark to the image being signed to improve the watermark invisibility and to enhance its robustness, such that watermarks of larger energy content can be embedded. Because of the invisibility constraints of a watermark, techniques have to use signals of relatively lower power than would otherwise be necessary.

Fig. 6 represents image independent watermark embedding process, while Fig.7 shows image dependent watermark embedding process.

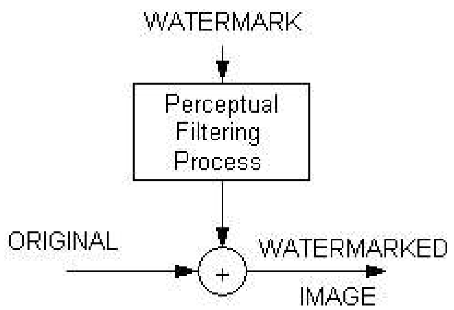

Fig. 6 Image independent watermark embedding process

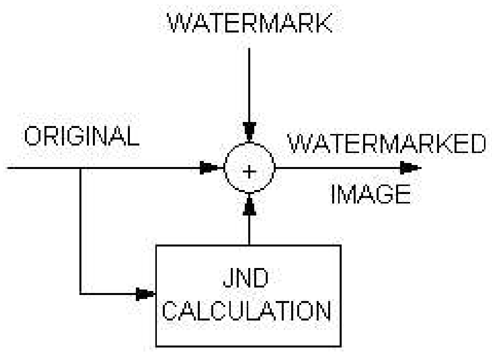

Fig. 7 Image dependent watermark embedding process

To improve the imperceptibility, a good watermarking technique has to adapt to the particular image.

Visual models derived for data compression are ideally suited for the digital watermarking scheme. Models can be directly extended to the watermarking application by providing upper bounds on watermark intensity levels for every pixel of the image, which guarantees transparency, while providing the robustness property.

To improve the robustness of the watermarking algorithms, numbers of techniques based on several transforms, such as Discrete Fourier Transform (DFT), Discrete Cosine Transform (DCT), and Discrete Wavelet Transform (DWT) have been applied. DWT based codec has an excellent performance over the DCT in low bitrate environment. Numbers of algorithms based on watermarking methods in the wavelet transform domain, have been proposed. For example, there is a method based on adding a pseudo-random noise sequence to the large coefficients in the high frequency bands of Embedded Zero Wavelet (EZW) based coder. Structure of wavelet decomposition is presented in Fig. 8.

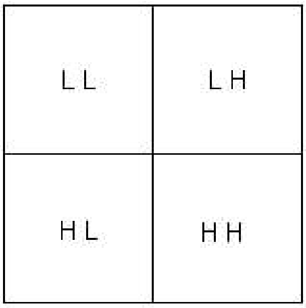

Fig. 8 Structure of wavelet decomposition

The large coefficients in high frequency bands, such as HH, HL, and LH subbands usually indicate edges in an image. Therefore, adding high energy of watermark signal into these large coefficients makes it difficult for the perception by human eyes, and also is hard to remove. By using spatial-frequency characteristics of Discrete Wavelet Transform, some techniques introduce the novel region of interest (ROI) approach watermark algorithm that inserts the watermark into perceptually significant pixels chosen by the multithreshold wavelet codec (MTWC) algorithm.

During the watermark detection process, synchronization of the watermark signal is of the most importance.

There are several advantages to combine at one end the image coding and watermark insertion operations, and at the other hand, the image decoding and watermark extraction.

5. CONCLUDING REMARKS

There are several interesting problems for signal processing research in multimedia communications. The first group concerns adaptive signal compression techniques to address the special requirements imposed by the heterogeneous nature of today's networks, and real-time audio/video streaming. The second group of problems is related to protecting the intellectual property rights associated to multimedia data.

In view of the lack of ability for protecting the current security algorithms, new copyright protection techniques need to be explored. Watermarking is one of new copyright protection algorithms that provide high potential in identifying the ownership of images. The concept is to hide a specific signature (watermark) in an image. Watermarking requires some special properties, such as invisibility and unremovability from the embedded image.

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BIOGRAPHY

Zoran S. Bojkovic received the PhD degree from the University of Belgrade, Serbia, in 1978. Since 1969 he has been with the University of Belgrade. Currently, he is the full professor of Electrical Engineering at the Faculty for Traffic and Transport Engineering, and the Faculty for Electrical Engineering, University of Belgrade. He is the visiting professor in USA, and was the visiting professor in Taiwan, China, Germany, Greece, Poland, Hungary, Romania, Bulgaria, Macedonia and Bosnia-Herzegovina. He is the co-author of the international books: Advanced Topics in Digital Image Processing (Editura Politechnica, Romania, 1997), Packet Video Communications over ATM Networks (Prentice-Hall, 2000), Multimedia Communication Systems (Prentice-Hall, 2002), Introduction to Multimedia Communications: Applications, Middleware and Networking (John Wiley and Sons, to appear). He has published extensively in referred journals, and has been an expert in telecommunications to research institute and academia. He is a Senior Member of IEEE, Senior Member of WSEAS, Member of Eurasip, Member of the New York Academy of Science, IASTED (Calgary, Canada), PRO-MPEG Forum, Research Center for Communications at Politechnica University of Bucharest (Romania). He is also a Member of Serbian Scientific Society and Yugoslav Engineering Academy.

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