A Novel System based on Filtered Multitone Modulation and RCPC codes for Video Transmission

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Abstract

Multicarrier systems are adopted in several standards for their ability to achieve optimal performance in highly dispersive channels. In particular, Orthogonal Frequency Division Multiplexing (OFDM) and Filtered Multitone (FMT) systems are two special cases of multicarrier systems which differs in terms of spectral partitioning. FMT does not require any cyclic prefix and virtual carriers, thus increases the bandwidth efficiency as compared to OFDM systems, but requires effective equalization techniques. Although, FMT suffers from higher computational complexity than OFDM, it yields much better performance in terms of bit error rate (BER). Hence, FMT is a possible alternative to OFDM for broadband wireless applications. In this paper, we have proposed the idea of using FMT for video transmission and we have also implemented Rate Compatible Punctured Convolutional (RCPC) codes to provide unequal error protection of the data according to their relative significance. Moreover, significant gains have also been achieved in terms of transmission efficiency by using RCPC codes for video transmission.

1. Introduction

Multicarrier modulation is the approach to design a bandwidth efficient communication system in the presence of channel distortion in which the available channel bandwidth is subdivided into a number of subchannels. The spectral partitioning in multicarrier systems can be realized in terms of overlapping or non-overlapping subbands. Orthogonal Frequency Division Multiplexing (OFDM) is one of the most widely used multicarrier systems which uses subchannels with overlapping spectra and the modulation filter has an ideal rectangular amplitude characteristic in time domain [1]. It requires a cyclic prefix (CP) and virtual carriers to ensure zero intersymbol interference (ISI) and zero inter carrier interference (ICI). OFDM modulation is adopted by IEEE for the extension of the 802.11 wireless LAN standard to the 5GHz band (IEEE 802.11a), providing data rates up to 54Mb/s. In high speed wireless data applications, although OFDM has been adopted due to its relatively simple receiver structure as compared to the single carrier transmission in frequency selective fading channels, it suffers from loss of efficiency owing to CP. Moreover, another major problem with OFDM is its sensitivity to frequency offset which is also due to large spectral overlap among the subchannel pulses. It has been shown that if frequency offset correction algorithms are not used, then the BER performance degrades severely in practical implementations of OFDM systems. In addition, in the process of ensuring perfect orthogonality, we can only obtain side lobes which are just 13dB below the main lobe, which leads to significant power loss. This also increases the number of virtual carriers required for the OFDM system.

On the other hand, in Filtered Multitone (FMT), the bandwidth of each of the subcarriers is chosen to be quasi orthogonal, which is achieved by using steep roll-off bandpass filters [2, 3]. In fact, we choose a particular case of a uniform filter bank consisting of frequency shifted versions of a low pass prototype filter. Since, the linear transmission medium does not destroy orthogonality achieved in this manner, cyclic prefix is not needed. Though effective equalization techniques are required at the receiver end to remove ISI, it can be a good alternative for OFDM.

In this paper, we have extended the idea of using multicarrier modulation for video transmission by using FMT as the modulation scheme. Inherently, a compressed video stream has varying degrees of importance. By using Rate Compatible Punctured Convolutional (RCPC) codes, one can have a video stream protected with varying levels of redundancy as per the importance of data stream. MPEG-4 ISO/IEC visual standard facilitates this by introducing Data Partitioning mode for effectively separating the header, motion and texture information. Header and Motion information being more important, can be protected with high rate codes, while texture information can be protected with lower rate codes. RCPC codes facilitates this using a single encoder and decoder for the purpose of encoding and decoding a bitstream with varying code rates, thereby lowering the system implementation cost. The paper is organized as follows, Section II describes the

FMT modulation scheme. Section III describes the equalization techniques used for FMT. In Section IV, we discuss application of FMT to video transmission. Section V describes RCPC codes. Section VI elaborates on how MPEG-4 data is generated for transmission over FMT system. Section VII explains the system model for end to end video data delivery. Finally, in section VIII and IX, we discuss simulation results and conclusion.

2. Filtered Multitone Modulation

FMT modulation is the special case of multicarrier modulation in which the spectral partitioning is non-overlapping in nature. This filter bank modulation technique is based on M-branch filters that are frequency shifted versions of a low pass prototype filter (uniform filter bank). The prototype filter achieves a high level of spectral containment such that the interchannel interference (ICI) is almost negligible in the system and the subcarriers can be considered to be close to orthogonal, irrespective of the length of the multipath channel.

Fig. (1) represents the filter bank implementation of the FMT system and Fig. (2) shows the FMT spectrum of first 5 subchannels. The inputs $A^{(i)}(k)$ are BPSK or QAM symbols which can come from different constellations. After up-sampling by a factor of M, each modulation symbol $A^{(i)}(k)$ is filtered at a rate M/T (where T is the FMT symbol period) by the subchannel filter, centered at frequency $f_i = i/T$. The transmit signal $x(n)$ is obtained at the transmission rate M/T by adding together the M filter output signals that have been appropriately frequency shifted. In the notation and figures, we have denoted k as the index for samples with a sampling period equal to T and n for the samples with a sampling period equal to T/M. In the receiver filter bank architecture (shown in Fig.(1)), the receiving filters $\{g^{(i)}(n)\}$ are designed to be matched to the corresponding ones in the transmitter, i.e., $G^{(i)}(f) = (H^{(i)}(f))^*$. $B^{(i)}(k)$ are the symbols which are obtained after the FMT demodulation.

The compromise to have quasi orthogonal system leads to attain high spectral containment having side lobes upto 76dB below the main lobe, resulting in negligible ICI. But, the time domain response of these filters may overlap several successive transmitted symbol periods. So, per subchannel equalization is necessary to reduce the effect of ISI. FMT has already been proposed for both wired [2] and wireless communications [3] as an alternative to most widely adopted OFDM.

3. Equalization in FMT System

A number of receiver algorithms are known which repeat the equalization and decoding tasks on the same set of received data by exchanging soft information iteratively rather than hard information (symbol estimates), where feedback information from the decoder is incorporated

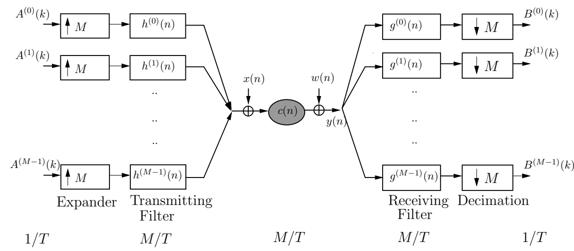

Figure 1: Analysis and Synthesis Filter Bank

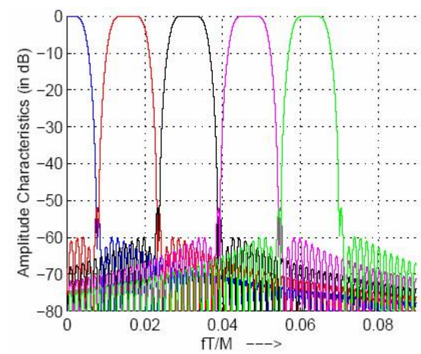

Figure 2: FMT Spectrum with 64 sub-channels: 5 first sub-channels

into the equalization process. Turbo Equalization is one of such iterative equalization and decoding techniques, which can provide impressive gains for communication systems that require data transmission over ISI channels [4, 5]. Typically, in turbo equalization based systems, maximum *a posteriori* (MAP) based techniques, most often a Viterbi algorithm (VA) producing soft output information, are used exclusively for both equalization and decoding. But they suffer from high computational load for the channels with long memory. Recently, reduced complexity structure of soft-in soft-out (SISO) equalizer were introduced using linear equalizer (LE) and decision feedback equalizer (DFE) based on minimum mean squared error (MMSE) criteria [6]. The soft information is in the form of log likelihood ratios (LLR) and is exchanged iteratively between the equalizer and decoder. A suitably chosen termination criterion stops the iterative process. Fig. (3) depicts the basic turbo FMT system [7].

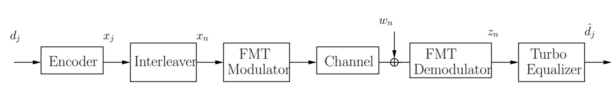

Figure 3: Basic Turbo-FMT system

The binary data d_j is encoded with a convolutional encoder yielding code symbols x_j, which are mapped to the alphabet $\mathcal{B}$ of the signal constellation. In this model, for simplicity we have used binary phase shift keying (BPSK), i.e., $\mathcal{B} \in \{-1, +1\}$. The interleaver permutes the x_j symbols and after FMT modulation, the data is transmitted over the ISI channel. The noise is modeled as additive white Gaussian noise (AWGN), i.e., the noise samples are independent and identically distributed (i.i.d.) with normal probability density function (pdf)

$$f_w(w) = \frac{1}{\sqrt{2\pi\sigma_w^2}} e^{-w^2/2\sigma_w^2} = \mathcal{N}(0, \sigma_w^2) \qquad (1)$$

and independent of the data. After demodulation the data z_n is passed through the turbo equalizer block which equalizes and decodes it to give the binary data d_j.

In the next section, we describe concept and the implementation of using FMT for video transmission.

4. Application of FMT to Video Transmission

One of the main goals in the near future is the development of multimedia systems efficient in terms of data coding, compression and transmission techniques, which could handle real-time communications. Moreover, the systems have to be adaptive according to the changing channel conditions. We have extended the idea of using multicarrier modulation for video transmission and replaced OFDM with Filtered Multitone (FMT).

The design of an error correction coding system usually consists of selecting a fixed code with a certain rate and the correction capability, which is matched to the protection requirement of the data to be transmitted. In many cases however, one would like to be more flexible because the data to be transmitted has different error protection needs, the channel is time varying or the channel has insufficiently known parameters. Therefore, there is a need of a flexible channel encoder and an adaptive decoder. So, in addition to introducing FMT for video transmission, we have also applied unequal error protection (UEP) using Rate Compatible Punctured Convolutional (RCPC) codes to have proper protection of the data. UEP has been shown to provide good performance in the case of transmission of compressed sources, where the bits produced have a different significance [8]. Before we proceed, we discuss some basic concepts of RCPC codes.

5. Rate Compatible Punctured Convolutional codes

Unequal error protection is classically performed at the channel coding level through convolutional codes and, more recently, using turbo codes. Some recent studies propose to perform unequal error protection in the modulation domain, exploiting the characteristics of multicarrier modulations [9]. In our simulations, we have exploited the former way of implementing UEP. Basically, while transmitting compressed digital signals, it is often needed to transmit some of the information bits with more redundancy than others. So, we wish to change the code rate and hence the correction power of the code during the transmission of a piece of information according to source and channel needs. For practical purposes, we would like to have not just switching between a set of encoders and decoders, but one encoder and one decoder which can be modified without changing their basic structure. This can be achieved by not transmitting certain code bits, namely, by puncturing the code. Hagenauer [8] was the first to propose punctured codes for variable transmission redundancy. In order to accommodate soft decisions and channel state information (CSI) at the receiver, a maximum likelihood decoder is required. This motivates the use of convolutional codes and the Viterbi algorithm for decoding.

The concept of punctured convolutional codes is modified for the generation of a family of codes by adding a rate compatibility restriction to the puncturing rule. The restriction implies that all the code bits of a high rate punctured code are used by the lower rate codes. This allows transmission of incremental redundancy and continuous rate variation to change from low to high error protection within a data frame.

In our simulations, we have assumed that the channel conditions are already known at the receiver. We have used two separate streams of data bits and applied the puncturing according to their relative significance.

6. Generation of MPEG-4 video data

In order to evaluate the performance of video transmission with the proposed technique, we focused on MPEG-4 [10], the recent MPEG ISO/IEC standard for video compression. The MPEG-4 standard utilizes the concept of object-based coding, allowing interactivity, and layered coding. As most video compression standards, it extensively relies on prediction and entropy coding and it is consequently very sensitive to channel errors.

With the goal of transmission over error prone channels, some error resilience tools have been added to the MPEG-4 standard: in particular, with the use of Resync markers, the MPEG-4 bit stream can be created with packets which are of almost the same length, separated by start codes. Start codes are unique words, recognizable from any sequence of variable length codewords, but not robust to channel errors. The data partitioning tool allows the separation of data with different significance within the packet. Regardless of these tools, MPEG-4 video transmission over wireless channels is still critical: for this reason, studies aimed at efficiently transmitting MPEG-4 video over wireless channels are currently being undertaken. If properly exploited, error resilience tools can produce a further improvement of the received video quality. In particular, the data partitioning tool can be usefully exploited with the purpose of performing unequal error protection: information bits contained in each packet are separated in three partitions, viz. header, motion and texture, each of which has a different sensitivity to channel errors.

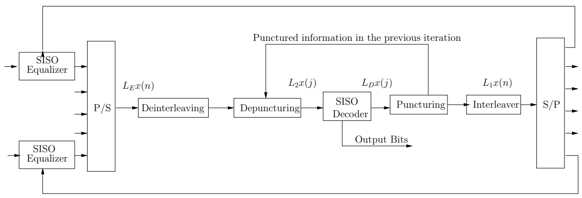

Figure 4: Modified Turbo Equalization system to incorporate unequal error protection

7. System model

In this work, we have coded the first 35 frames of a video sequence ("foreman" test sequence in QCIF format) at different bit rates ranging from 64kbps to 192kbps. The MoMuSys MPEG-4 codec has been used, with some modifications in the decoder, in order to improve the robustness to errors. Additional standard-compatible error resilience techniques have also been adopted. We used a packet size of 400 bits. Each packet is divided into two parts, first part containing Header and Motion data and the second part containing Texture data. Since the former part is more important, we transmit all the bits of it without any puncturing. On the other hand, the latter part containing texture, which is relatively less significant, has been transmitted after applying puncturing at the rate 4/6. After that, the data is arranged into FMT blocks of 64 symbols each containing some portion of texture and some portion of header and motion. Finally, the FMT modulated data is transmitted over the channel. At the receiver end, after the FMT demodulation, we applied turbo equalization with incorporation of depuncturing of bits. Fig. (4) explains the overall structure of the improvised turbo equalization technique at the receiver end [7]. We stick to the flat-fading assumption for the transmission channel which can therefore be equalized by one tap per subchannel. Since, overall channel impulse response is designed to be real, therefore we will have real tap coefficients.

A turbo equalizer consists of a SISO Equalizer and a SISO decoder, operating in an iterative manner. The MMSE-LE criteria is used as developed in [5, 6] for the SISO equalization task and the Bahl Cocke Jelinek and Raviv (BCJR) algorithm [11] has been used for the decoding block. The BCJR decoding algorithm is known to yield the optimal symbol estimate with minimum symbol-error rate. The essential part is the BCJR algorithm's ability to yield soft information in the form of *a posteriori* LLRs for both coded and information bits. The L-value operator $L(x)$, called log likelihood ratio (LLR), is applied to quantities $x \in \{-1, +1\}$ and is given by

$$L(x) \equiv \log \frac{P(x=+1)}{P(x=-1)} \tag{2}$$

The SISO equalizer inputs *a priori* $L_1(x_n)$ LLR, computes estimates $\hat{x}_n$ of transmitted symbol x_n from received symbols z_n by minimizing the cost function $\mathbf{E}(|x_n - \hat{x}_n|^2)$ and outputs *a posteriori* LLR minus the *a priori* LLR called the *extrinsic* LLR $L_E(x_n)$.

$$L_E(x_n) \equiv \log \frac{P(x_n=+1|\hat{x}_n)}{P(x_n=-1|\hat{x}_n)} - \log \frac{P(x_n=+1)}{P(x_n=-1)} \tag{3}$$

$$= \log \frac{p(\hat{x}_n|x_n=+1)}{p(\hat{x}_n|x_n=-1)} \tag{4}$$

The *a priori* LLR, which is $L_1(x_n)$ represents prior information on the occurrence probability of x_n and is provided by the decoder. For the initial equalization step, no *a priori* information is available and hence we have $L_1(x_n) = 0$. $L_E(x_n)$ is considered to be independent of $L_1(x_n)$. This and the concept of treating feedback as *a priori* information are the two essential feature of any system applying the turbo principle. The equalizer output after de-interleaving and de-puncturing is considered to be *a priori* LLR $L_2(x_j)$ for the decoder. The decoder also computes *extrinsic* LLRs $L_D(x_j)$ which after interleaving and puncturing is given to the equalizer as input. The SISO decoder is a MAP decoder, which computes the *a posteriori* probabilities (APP's) $P(x_j = x|L_2(x_1), \cdots, L_2(x_{K_c}))$, $x \in \mathcal{B}$ given K_c code bit LLRs $L_2(x_j),\ j = 1, 2, \cdots, K_c$, and computes the difference as

$$L_D(x_j) \equiv \log \frac{P(x_j=+1|L_2(x_1), \cdots, L_2(x_{K_c}))}{P(x_j=-1|L_2(x_1), \cdots, L_2(x_{K_c}))} - \log \frac{P(x_j=+1)}{P(x_j=-1)} \tag{5}$$

The SISO decoder also computes the data bit estimates.

$$\hat{d}_j \equiv \underset{d \in \{0,1\}}{\arg\max}\, P(d_j = d|L_2(x_1), \cdots, L_2(x_{K_c})) \tag{6}$$

In the first iteration, we de-puncture the received information by fixing the soft information LLR to be zero, which is then passed to the decoder. The decoder as discussed decodes the input data according to BCJR algorithm and sends the new soft information to the equalizer. But since the equalizer requires data after puncturing, the soft information from decoder is punctured using the same rate. The punctured information is stored and inserted back as the input data to the decoder in next iteration while de-puncturing. The results obtained by inserting the saved soft information are much better in comparison to inserting zero as the LLR values in successive iterations.

After the turbo equalization, the received header, motion and texture data is multiplexed to form the video packets and decoded using MPEG-4 decoder. Finally, the performance evaluation in terms of PSNR (peak signal-to-noise ratio) for MPEG-4 video transmission for wireless data service is addressed, showing the large gain that can be obtained, especially at high signal-to-noise ratios.

8. Simulation Results

MoMuSys MPEG4-SP video codec is used to generate data streams at seven different bit rates of 64, 96, 128, 160, 192 and 256kbps. The data partition mode is used with packets of size 400 bits each. 35 frames of video have been transmitted to perform the simulations. The GOP size for video bitstream has been set to 30, resulting in two I frames in the entire video data. For unequal error protection (UEP), the header and motion data are separated from the texture data and then transmitted without any puncturing at rate $1/2$. However, the texture data has been punctured and transmitted at rate 4/6. It was observed that, any error introduced in an I frame causes the PSNR to drop sharply. Therefore, entire I frame was transmitted without any puncturing. The simulations have been carried out at three different channel SNRs of 6, 8 and 10dB to evaluate the performance of both the UEP and equal error protection (EEP) schemes. Fig. (5), Fig. (6) and Fig. (7) show the PSNR plots against video bit rate for both equal and unequal error protection cases at 6dB, 8dB and 10dB channel SNR, respectively. From the plots, we can deduce following conclusions:

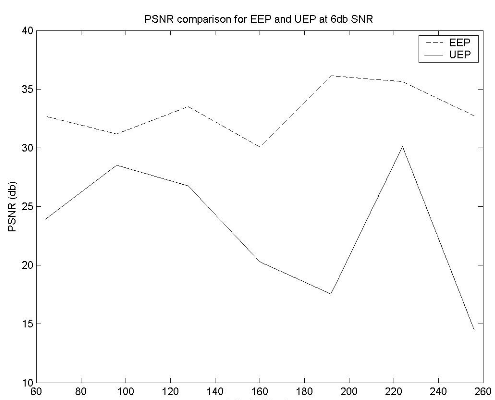

Figure 5: PSNR comparison for UEP an EEP at 6db channel SNR

1. The performance of the EEP is better than UEP at low channel SNR (6dB), which is as expected. So we should go for maximum protection at low SNR.
2. It can also be seen that at lower SNR, the performance curves goes down for high bit rate. As the number of bits per frame increases with bit rate, at low channel SNR, there are more errors per I frame. This results in significant degradation of overall video quality due to propagation of errors into subsequent P frames.

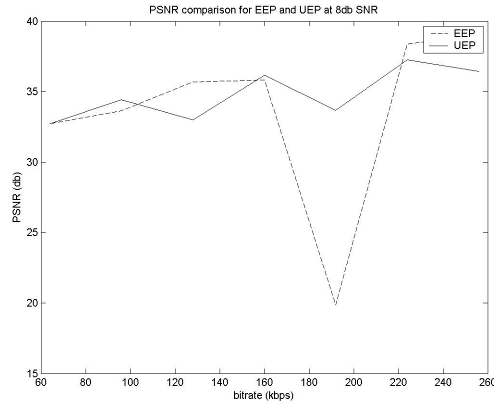

Figure 6: PSNR comparison for UEP an EEP at 8db channel SNR

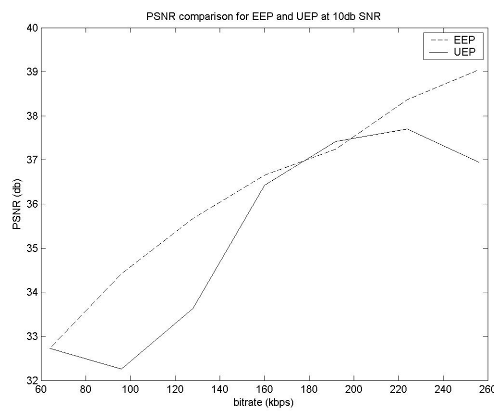

Figure 7: PSNR comparison for UEP an EEP at 10db channel SNR

3. It is also observed that, UEP gives better or at least comparable video quality as that of EEP at higher channel SNR of 10dB, thus increasing the transmission efficiency by 15%.
4. An isolated incidence of drop in PSNR is observed for EEP case at 8dB channel SNR and 192 kbps bit rate. This is due to large number of errors in first I frame.
5. The UEP scheme can be extended to frequency domain by using the feedback from receiver about channel condition and allocating bits to different sub-channels according to their respective SNRs.

9. Conclusion

The system proposed is suitable to counter the effects such as frequency and timing offsets due to high spectral containment and effective equalization with reasonable complexity. We have developed an application of FMT for video data transmission. In addition, we have applied unequal error protection according to relative importance of the data bits. In the process, we have increased the transmission efficiency of the system by 15%. A novel technique for utilizing RCPC codes with turbo equalization and FMT has been developed.

10. References

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