TECHNICAL RESEARCH REPORT

Image and Video Transmission over Wireless Channels: A Subband Modulation Approach

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IMAGE AND VIDEO TRANSMISSION OVER WIRELESS CHANNEL: A SUBBAND MODULATION APPROACH

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ABSTRACT

A new approach of reliable image and video transmission over wireless channel is proposed. The subband modulation which combines source coding and channel modulation scheme achieves high compression efficiency and preferable quality. Further performance gain is obtained by multiresolution modulation and bits re-mapping scheme that assign efficient mapping from each source codeword to channel modulation points. We show that bits re-mapping scheme performs nearly same as the optimal mapping design scheme [4] but with much lower complexity. The simulations are carried out on *Additive White Gaussian Noise* (AWGN) channel and slow Rayleigh fading channel.

1. INTRODUCTION

Reliable image and video transmission over noisy channel has been a challenge in multimedia communication, especially the transmission of large volume of data over the unreliable and bandwidth limited channels. In video conference applications the bandwidth assigned to each user is quite limited and if a scheme can reduce the bandwidth usage of each user while maintaining preferable quality, the original bandwidth can support more users and more profit can be made. Therefore, developing reliable transmission under the bandwidth constraint has become an important research issue. For subband decomposed image and video, one solution is the joint optimization of source and channel units among all the subbands that allocates different rates among source and channel units to minimize the overall distortion.

We observe that the approaches proposed in the literature [1, 2, 3, 4] are suited for either low SNR or high SNR channels. When the SNR is high enough that the error probability of BPSK modulation is nearly saturated, but still not high enough to cover the modulations with large number of constellation points, these approaches will not achieve their best performance. In this paper we present an new approach that handles the image and video transmission over noisy channel with middle range SNR (8-24dB) under tightly limited bandwidth (0.3-1.0bpp) constraint. Such condition is one of the most important performance region in practice, and yet few approaches provide good solution so far. Based on subband decomposition the proposed system combines source coding and channel modulation that optimally partitions the transmission bandwidth among source quantizers and modulators for all subbands. The proposed bit re-mapping scheme performs nearly same as the optimal mapping design scheme proposed in [4] but with much less computational complexity. The simulations are based on AWGN channel and slow Rayleigh fading channel.

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2. SUBBAND MODULATION

2.1. The Motivation

The combined source and channel coding approach is a classical solution to the transmission over noisy channel. Mathematically, it is equivalent to distributing the given bit budget efficiently among a set of given admissible source coding unit choices and channel coding unit choices, to minimize overall distortion. For subband decomposed source data, this shows more promising advantages. The source data is divided into a set of subbands with different importance. Subbands with higher energy are more important since they carry more "information". The source coding unit and channel coding unit may not be same for all subbands. More important subbands should have higher source rate and more channel protection.

Source coding units and channel coding units must be selected according to transmission requirements and channel conditions. Particularly, if transmission bandwidth is limited, both the source coding and the channel coding units must carry efficient compression capability as well as the error protection capability. For source coding unit, we propose fixed length Vector Quantization which is robust against error and easy implemented. For channel unit, adding channel protection codes is only required for very noisy channel at cost of less compression. Indeed, for wireless communications, the channel bandwidth is the most precious resource. For channel with middle range SNR and low bandwidth, we recommend multilevel modulations which offer different compression capabilities as well as different error protections[4, 5]. In addition, appropriate mapping from source codeword to channel constellation points is needed to reduce the noise effect.

2.2. Mathematic Formulation

Using Mean Squared Error (MSE) as the distortion measure, the overall distortion can be divided into source distortion $D_{i,s}$ and channel distortion $D_{i,c}$ for each subband i, $i \in [1, M]$ which can be interpreted as the function of source coding rate $R_{i,s}$ and channel modulation rate $R_{i,c}$. For subband decomposed data, the overall distortion and rate are the sum of that of all the independent subbands given as $D_{total} = \sum_{i=1}^{M} D_{i,s} + D_{i,c}$ and $R_{total} = \sum_{i=1}^{M} R_{i,s}/R_{i,c}$. Given a bandwidth constraint R_{budget} to code M subbands, the objective is turned to a constrained optimization problem defined as

$$\text{Min } D_{total} \text{ subject to } R_{total} < R_{budget}. \qquad (1)$$

We develop an efficient solution similarly to that of [6]. For real time applications, the images and videos should be first divided to classes and the optimal rate distributions can be obtained for each particular class.

2.3. MR modulation

The modulation design aims to find modulation schemes with efficient power consumption, high compression and reliable error protection. The multiresolution (MR) modulation proposed by [1] showed great difference in error protection of each bit at different locations of modulation points. This observation becomes a major inspiration to our system. Figure 1 illustrates the MR modulation constellation graphs and mappings of PAM-4, QAM-16, and QAM-64. Here $d2$ and $d1$ represent intracloud and intercloud distances, respectively. We refer to each constellation point as a symbol. Since MSBs of a constellation symbol represent clouds and LSBs represent satellites, MSBs receive better protection than LSBs. Under the condition that source codewords are arranged properly so that the two nearest codewords are also the nearest in source distortion sense, the MR modulations will arrange the distance among the constellation symbols to reduce the total distortion.

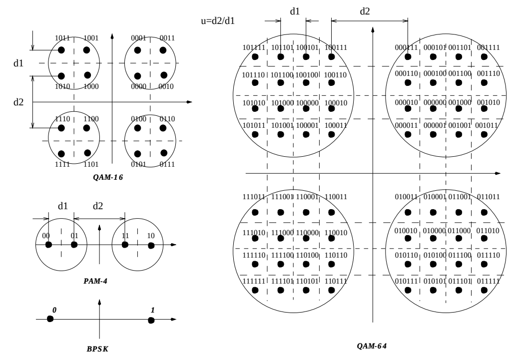


Figure 1: *MR modulations*

The distance between the symbols with large distortion is increased, and to maintain the same power the distance between two symbols with small distortion is decreased. Such balance can be adjusted by adapting $\mu = d2/d1$ or changing the modulation rate.

2.4. Mapping Design

In our system, the mapping design is different from the MR modulation system in [1] because the source coding rate $R_{i,s}$ and the modulation rate $R_{i,c}$ are not equal for most of the time. Usually $R_{i,s} > R_{i,c}$, so that one codeword is mapped to several constellation points, called constellation vector. In addition to the proper design of constellation set, real applications need an efficient, intelligent mapping from source codeword to constellation vector in order to reduce the noise effect.

- **The Optimal Mapping Design**.
  Mathematically, the optimal mapping design can be interpreted as another optimization problem defined as follow.
  Given $p(u|u')$ $, u, u' \in \Omega,$ $x, x' \in \mathcal{C}$, find $F : \mathcal{C} \longrightarrow \Omega^m,$ $F(x) = \{u_1, u_2, .., u_m\}$ such that

$$\begin{aligned} & \text{Min } \{\sum_{x,x' \in \mathcal{C}} P(x'|x) d(x', x)\} \\ = & \text{ Min}\{\sum_{x,x' \in \mathcal{C}} \prod_{i=1}^{m} p(u'_i|u_i) d(x', x)\}, \qquad (2) \end{aligned}$$

  where Ω represents the set of constellation points, Ω_m is the m dimensional vector extension of Ω, and $\mathcal{C}$ is the set of source codewords with cardinality of $2^{R_{i,s}}$.

  Dynamic programming and simulated annealing can be used to find the optimal solution[4], but difficult and computationally expensive. Also the optimization depends on the source data, thus has to be done for every type of source data.

- **Bit Re-mapping Scheme**.
  It is based on the property that MSBs of each

modulation symbol carry better error protection than LSBs in multilevel modulation. If we map the source codeword to constellation points following the original order, the order of bits inside each constellation point may be totally random unless $R_{i,s}$ and $R_{i,c}$ are integer multiple of the other one. This will cause degradation since the MSBs of the source codeword could map to the LSBs of the constellation point, thus receive less protection and cause more distortion. To prevent this, the bits order of source codeword or the mapping has to be rearranged according to $R_{i,s}$ and $R_{i,c}$. Below, we describe the bits re-mapping scheme:

- Case 1: For $R_{i,s} < R_{i,c}$, follow the original order.
- Case 2: For $R_{i,s} > R_{i,c}$, divide the source codewords of length $R_{i,s}$ to $R_{i,c}$ group containing $\lfloor R_{i,s}/R_{i,c} \rfloor$ bits. Select one bit from each group and map to the same location of each group consecutively. The left residue ($R_{i,s}$mod $R_{i,c}$) bits as the LSBs of the source codeword, are transmitted separately from the other bits. Otherwise the bits order becomes random.

For example, when $R_{i,s}$ equals to 9 and $R_{i,c}$ equals to 2, the source data is from bit 8 to bit 0 with bit 8 as the MSB and bit 0 as the LSB. There is one residue bit: bit 0. Modulated following the original order, the bits order turns out to be 87, 65, 43, 21,08, where last 8 represents the bit 8 from next codeword. This is not appropriate since bit 8 and 7 are more important than bit 6,4,2 and 0. After the bits re-mapping the order will be 84, 73, 62, 51. Bits 8, 7, 6, and 5 receive higher protection at the cost of less protection for bits 4, 3 ,2, 1; and the effect of bit 0 is removed. Another advantage is that bit re-mapping does not depend on source data type and therefore is practical in real time. As we will show in simulation, the bit re-mapping scheme performs nearly same as optimal mapping design scheme defined by (2), but with much lower complexity.

3. SIMULATION RESULTS IN AWGN CHANNEL

In the simulation, we use subband based encoder operating in "intraframe only" mode, which aims to prevent the error propagation. The image is 2-D subband decomposed using orthogonal wavelet filters, while every two consecutive video frames are 3-D subband decomposed together using Harr filtering in time domain and wavelet filtering in spatial domain. Targeted average bit rates range from 0.3 bit per pixel(bpp) to 1.0 bpp. The lowest subband is quantized using Lloyd Quantizer (LQ) while the other subbands are Vector Quantized (VQ). The AWGN channel is modeled as adding noise with double sided spectral density $N_0/2$. The available modulations are BPSK, MR-PAM4, MR-QAM16 and MR-QAM64 as same as those illustrated in Figure 1.

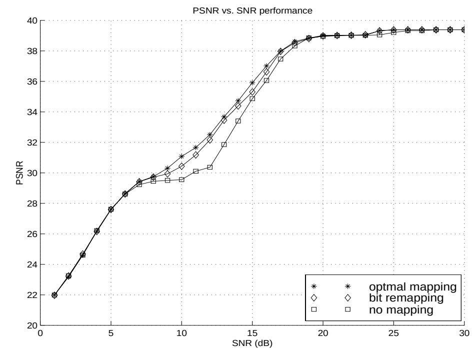


Figure 2: *PSNR vs. SNR performance comparison of the systems without mapping design, with bits re-mapping design and with optimal mapping design based on "Salesman" sequence frame 3.*

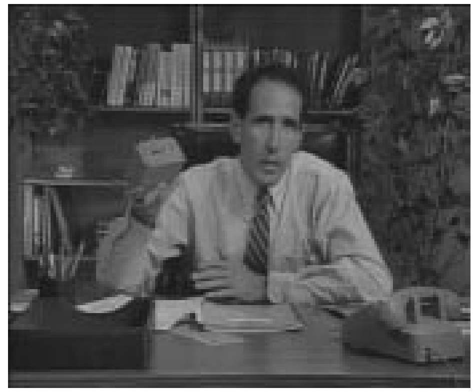
Figure 2 plots a comparison of our system without mapping design, with bits re-mapping and with optimal mapping design. Here the system without mapping design represents the direct mapping but still transmitting the residue bits separately. The optimal mapping is obtained as the optimal solution of (2). As can be seen, the bits re-mapping system performs nearly the same as the optimal mapping system and is about 0.2 - 2.0 dB better than the system without mapping design, particularly in the middle SNR range. Examining the visual quality of our system and the BPSK modulation system (use BPSK modulation for every subband) from Figure 3 and Figure 4, our resulted image is clearer and more detailed with only a negligible number of defective points.

We also compare our system to the other two existing approaches [3][4]. Figure 5 shows the proposed system outperforms the one-to-one direct mapping system of [4] for 512×512 "Lena" image coded at 0.5bpp and 0.4bpp at SNR ranging from 10 to 24 dB. When SNR is above 24 dB, our system saturates due to the upper bound on source coding rate. For the comparison in very low SNR scenario such as 1-8dB, we also plot the performance of the joint source and channel coding system with RCPC code on the "Salesman" video sequence [3]. Figure 6 illustrates that for very low SNR (below 6dB), using channel coding with BPSK generate quite preferable result. When SNR is above 6dB, Our system shows great improvement by assigning modulation at different rates to different subbands. It

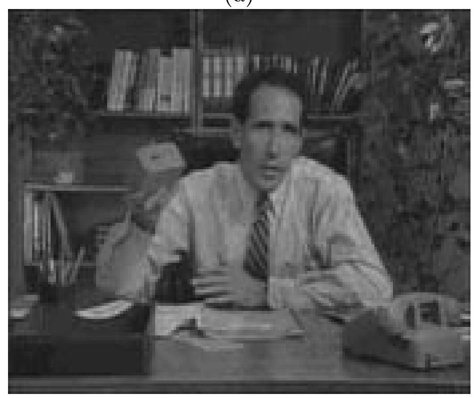
(a)

(b)

Figure 3: *Result "Lena" image of (a) our system and (b) fixed BPSK modulation system at 0.3bpp, SNR=15dB*



(a)



(b)

Figure 4: *Result "Salesman" frame 3 of (a)our system and (b) fixed BPSK modulation system at 0.5bpp, SNR=15dB*

should be pointed out that low rates such as 0.4bpp and 0.3bpp cannot be achieved by using BPSK modulation with channel codes. The above two comparisons further prove that our system is most suitable for channel with middle (8-24dB) SNR value and low bandwidth.

4. EXTENSION TO SLOW RAYLEIGH FADING CHANNEL

Although above simulation only involves AWGN channel, our system can be generally applied to wireless channel such as slow Rayleigh fading channel. In [7], the adaptive modulation scheme based on switching levels is used for slow fading channel. The transmitter and receiver are able to adjust the modulation type according to channel condition. This adaptive scheme is suitable for our system since the the quantizations and modulations will not change abruptly if SNR changes. The optimal switching levels are computed based on the received SNR value. The objective function is given by

$$\begin{aligned} \mathrm{D} &= \int_0^{l_1} D_0(\gamma)F(\gamma,\,,\,)d\gamma + \int_{l_1}^{l_2} D_1(\gamma)F(\gamma,\,,\,)d\gamma \\ &+ \int_{l_2}^{l_3} D_2(\gamma)F(\gamma,\,,\,)d\gamma + \int_{l_3}^{\infty} D_3(\gamma)F(\gamma,\,,\,)d\gamma \quad (3) \end{aligned}$$

where $F(\gamma,\,,\,)$ is the probability distribution function (PDF) of the received channel SNR, , is the average channel SNR and D_i is the distortion corresponding to the quantization and modulation scheme (R_s, R_c) selected for the ith SNR range. For each range, (R_s, R_c) can be selected as the selection used by the middle SNR value. Or the one that minimizes the total distortion summing over all SNR within this range. For slow Rayleigh fading channel[8], the channel SNR has the PDF of

$$F(\gamma,\,,\,) = \frac{1}{,}\exp\frac{-\gamma}{,}, \quad (4)$$

where , is the average channel SNR.Since our optimization is only based on integer SNR (dB) value, denoted as k, the probability of received SNR being k dB is

$$P(k) = \int_{k-0.5}^{k+0.5} F(\gamma,\,,\,)d\gamma. \quad (5)$$

Given the average channel SNR , , the optimization function is defined as

$$\mathrm{D} \;=\; \sum_{0}^{l_1-1} D_0(k)P(k) + \sum_{l_1}^{l_2-1} D_1(k)P(k)$$

$$+ \quad \sum_{l_2}^{l_3-1} D_2(k)P(k) + \sum_{l_3}^{\infty} D_3(k)P(k). \qquad (6)$$

The optimal levels l_1, l_2, l_3 can be solved by dynamic programming. Table 1 and 2 shows the optimal levels that are computed for "Salesman" sequence at rate 0.5bpp with different average channel SNR values noted as , , based on our system and the BPSK modulation system mentioned before.

As can be seen from Table 1, generally the switch level (8dB,12dB,25dB) works well for all average channel SNR values. And our system performs better than BPSK modulation system especially for middle average SNR values (10-20dB).

5. CONCLUSION

We have proposed a combined source coding and channel modulation scheme for image and video transmission over wireless channel. The bits re-mapping scheme performs nearly same as the optimal mapping design scheme but at much less computational cost. In addition to AWGN channel, our system performs better in slow Rayleigh fading channel than the fix modulation systems. Compared to conventional schemes, our scheme need 3 or 4 modulators/demodulators. Since we use common modulations such as BPSK, PAM and QAM which are quite the same in terms of implementation, the increase in complexity is ignorable compared to the performance improvement.

6. REFERENCES

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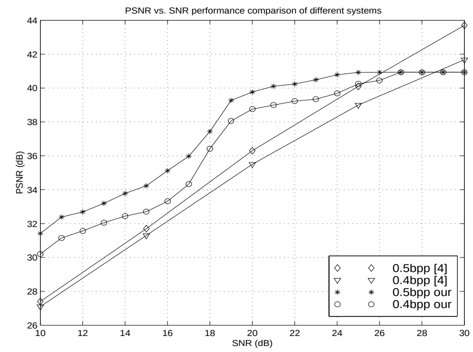

Figure 5: *Comparison of our system and intelligent mapping system of [4] at different rates.*

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Table 1. Simulation Result of Our Adaptive System

,	level 0	level 1	level 2	MSE	PSNR
5	2	5	18	154.28	26.25
10	10	12	15	92.37	28.48
10	8	12	25	97.37	28.25
12	8	12	25	77.19	29.26
15	8	12	25	51.77	30.99
18	8	12	25	33.67	32.86
20	8	12	25	25.35	34.09

Table 2. Simulation Result of BPSK Adaptive System

,	level 0	level 1	level 2	MSE	PSNR
5	2	4	11	155.30	26.22
10	2	4	19	102.91	28.00
12	2	6	13	92.26	28.48
15	2	8	11	83.16	28.93
18	2	4	17	77.92	29.22
20	2	5	13	76.08	29.32

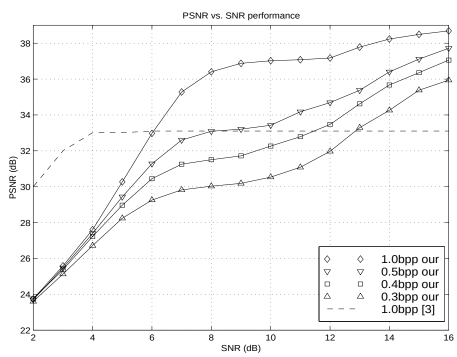

Figure 6: *Comparison of our system and joint source/channel coding system of [3]*