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ALPHABETIZED ACCORDING TO STUDENT LASTNAME

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Progress Report on MIMO Research: Winter 2005

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I. Introduction

Achieving high throughput while maintaining robustness is one of the biggest challenges in wireless communications. In general, practical systems sacrifice the crucial diversity that is required for high performance in exchange of higher spatial multiplexing.

It is known that multi-input multi-output (MIMO) systems provide significant capacity increase [1]. MIMO systems also achieve a high diversity order. A technique that provides high diversity and coding gain with the help of channel state information (CSI) at the transmitter is known as beamforming. Beamforming separates the MIMO channel into independent subchannels. Therefore, multiple streams of data can be transmitted easily. Single beamforming (i.e., sending one symbol at a time) was shown to achieve the maximum diversity in space with a substantial coding gain compared to space time codes [2]. If more than one symbol at a time are transmitted, then the technique is called multiple beamforming. For uncoded multiple beamforming systems, it is shown that while the data rate increases, one loses the diversity order with the increasing number of streams used for flat fading channels [3].

Bit interleaved coded modulation (BICM) was introduced as a way to increase the code diversity [4]. We showed that with the inclusion of BICM to the system, one does not lose the diversity order with multiple beamforming even all the subchannels are used. That is, we showed that multiple beamforming with BICM (BICM-MB) can achieve full diversity order of NM, and full spatial multiplexing of $\min(N, M)$ for flat fading channels using N transmit and M receive antennas. In order to guarantee full diversity, we found a design criterion for the interleaver.

If the channel is frequency selective, then OFDM can be used to combat intersymbol interference (ISI). By using a cyclic prefix (CP) of an appropriate length, OFDM converts the frequency selective channel into parallel flat fading channels. Hence, by combining BICM-MB with OFDM, one can achieve full spatial multiplexing of $\min(N, M)$ with full spatial and frequency diversity order of NML for L-tap frequency selective channels.

Some of our findings are submitted for a publication to Globecom 2005 [5].

II. Simulation Results

The industry standard 64-state 1/2-rate $d_{free} = 10$ convolutional code is used in the simulations. When the interleaver of IEEE 802.11a standard [6] with an interleaver depth of S OFDM symbols is used, the consecutive coded bits of length corresponding to S OFDM symbols are transmitted over the same subchannel. Therefore, the error event is dominated by the worst subchannel, and hence the performance is poor. If the interleaver of [7] is used (labeled as [7] in the figures) with an interleaver depth of S OFDM symbols, the consecutive coded bits on the trellis are transmitted over different subchannels.

As can be seen from Figure 1-(a) BICM-MB with interleaver of [7] performs substantially better than BICM-MB with 802.11a interleaver, and zero forcing with BICM. **The performance gain at 10^{-5} bit error rate is more than 20 dB for 2×2 system, about 14 dB when 2×3 systems are compared, and about 18 dB for 3×3 BICM-MB with 2 streams**.

Figure 1-(b) illustrates the results comparing BICM-MB-OFDM with BICM-OFDM using ZF. Both systems use the interleaver of [7]. As can be seen from the figures BICM-MB-OFDM achieves a higher diversity order than the zero forcing system, and hence provides a substantial increase in the performance. **For a 2×2 system, BICM-MB-OFDM outperforms ZF with BICM-OFDM more than 20 dB for flat fading channel, and more than 10 dB for rms delay spread 50 ns (11 taps) channel at 10^{-5} BER level**.

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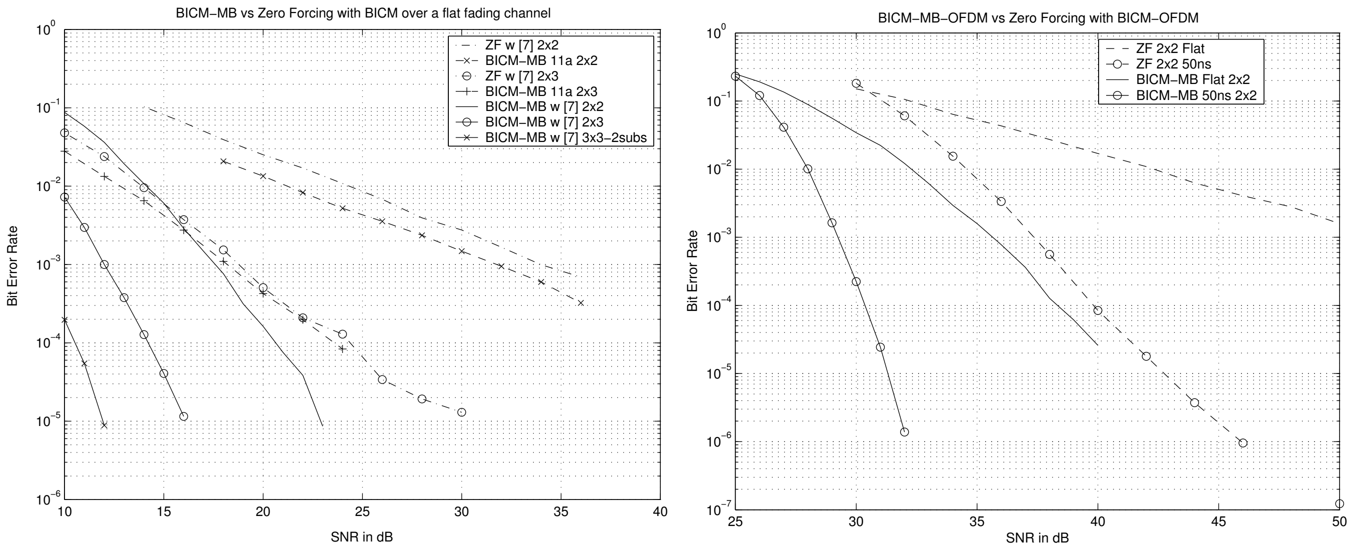

(a) BICM-MB (b) BICM-MB-OFDM

Fig. 1. All the systems transmit 2 symbols at a time. 16 QAM and 64 states 1/2 rate convolutional code are used.

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Cross Layer Design for Real Time Wireless Video

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Abstract

High quality real-time wireless multimedia services will become a reality in the future. A major application area is the video tele-presence service, which requires real-time interactive video support over the wireless channel. Video coding and wireless physical layer have been largely developed with little consideration for each other. Video encoder design aims to provide the highest possible encoding quality at the cost of degrading error resilience capability, while wireless physical layer technologies aim to maximize channel capacity. However, since conventional video rate control schemes cannot respond to the sudden capacity fluctuation of the wireless channel, any increase in wireless channel capacity goes unutilized by the video application. In this paper we address this issue by specifically developing a cross-layered solution. Specifically we utilize MIMO technology to guarantee a certain QoS and employ a video layer technology that capitalizes on the available capacity at the physical layer in real-time.

1. Introduction

High quality multimedia services will be a major application supported by the wireless communications industry in the near future. Commercial wireless video codec systems can already support compressed video with maximum frame size of 352288 (CIF), frame rate of 30 fps, and a bit rate of 382 kbps [1]. Multimedia (MM) capable handhelds such as the Pocket PC, Palm devices, and some high-end cellular phones, provide up to 320x320 screen resolutions and have the processing power for video services. However, the issue of how to effectively transport such high quality multimedia streams across fluctuating radio channels remains a challenging hurdle because highly compressed video data is very susceptible to low quality fluctuating channel conditions [1][2][3][4].

A typical class of wireless video communications is the video tele-presence service, which requires the support of real-time interactive video over the wireless channel. Video tele-presence services are particularly important for emergency health services, such as stroke victims, where the immediate diagnosis by a physician as to the condition of the patient is a matter of life and death. The responses of the patient to the doctor's orders are assessed and serve to determine the medication that must be administered on the spot. The importance of tele-presence in rescue and crisis situations is also paramount. The ability to convey the severity of a situation and report on emergencies requires the ability to accurately relay that information to command centers where decisions on crew deployments are made. In addition, the support of real-time interactive video should be in huge demand in the consumer and entertainment markets. Early signs of its popularity with consumers have been witnessed with the success of video based 3G cell phones released in Japan.

2. Motivation for cross-layer design

In the area of video over wireless channels, the video coding techniques and the wireless physical layer technology are largely developed with little consideration for each other. On the one hand, the video encoder design is pursuing the highest possible encoding quality at the cost of error resilience capability degradation. On the other hand, most wireless physical layer technologies aim at maximizing the channel capacity. The increased wireless channel capacity however, cannot be immediately utilized by the video application, because the conventional video rate control schemes can hardly respond to the sudden capacity fluctuation of the wireless channel. Some cross-layer schemes [29][31] are proposed to change the strength of channel coding such that the error correction capability can adapt to the instant bit error rate (BER), which is normally caused by the channel capacity fluctuation. Nevertheless, it is obvious that we may make better use of the wireless channel if we are able to fine-tune the transmission rate of the generated video bits (i.e., the information bits instead of the channel coding bits) to match the fluctuating wireless channel capacity. By enabling the interaction between application layer and physical layer, we expect to seamlessly integrate the real-time video encoder and the wireless channel. The whole approach includes the application layer design of the error resilient video encoder that can easily adapt to the channel capacity fluctuation, the provision of the wireless channel quality of service using Multiple Input Multiple Output (MIMO) technology, and the interaction between the application layer and the physical layer.

3. Issues addressed in the paper

In this paper we address the issue of efficient cross-layer design for wireless multimedia traffic. It is based on the wireless physical layer and the video processing application layer. The wireless channel provides an extremely challenging situation for handling multimedia traffic with delay constraints, due to the changing fading channel conditions over time. Fading conditions can be differentiated into short-term and long term fading. We propose to make use of the channel condition information available at the physical layer to design efficient application layer video encoding via cross layer optimization and efficient mechanisms to transport the data by MIMO physical layer techniques.

A. Physical layer techniques:

At the physical layer, bandwidth and power are the scarce resources. Given that the bandwidth utilized is fixed, the power has to be used efficiently. Moreover one has to limit the peak power requirement that cannot be exceeded. At first we develop a MIMO physical layer solution to maintain multimedia traffic at a constant BER. The technique utilizes adaptive modulation for MIMO system and allocates maximum possible bits under a power constraint. In typical video communications, one needs to satisfy a minimum quality of service (QoS) requirement by the multimedia traffic. In our case we take the minimum QoS to be a guaranteed minimum of bits transmitted for all periods outside outage. We set the peak power required so that atleast a minimum amount of bits would be transmitted under all non-outage periods. It is possible that the wireless channel becomes extremely bad that under the power constraint not even a minimum QoS can be maintained. These periods are called outage periods and we constrain them to a required value by proper peak power requirements. Now not all channel conditions would utilize all of the peak power to send the minimum bits. If the minimum power requirement is lower than the peak power for sending the minimum number of bits, the remaining power could be utilized to send more bits. This is what is precisely the goal of our physical layer technique. This technique is for maximizing the channel capacity and providing a Variable Bit Rate (VBR) at a constant BER while satisfying at least a minimum rate which can be utilized to transmit the base layer of a layered video.

As a specific example, this technique will be suitable for handling FGS (Fine Grained Scalability) video. In FGS there are two layers - the base layer whose rate is nearly constant and an enhancement layer whose rate is variable. A total constant maximum power for each transmission based on the base layer rate with a certain outage, is assumed. This available power will then be based on worst case for a given outage. Since most transmission periods outside outage will require only part if this maximum power, the remaining available power will be used to transmit a certain number of bits from the enhancement layer that will maximize channel capacity in the long run. For instance, if the channel conditions are extremely bad but still lie within the non-outage region for the minimum bit rate requirement under a constant BER, more bits from the enhancement layer will very well be transmitted under peak power constraint.

B. Application layer techniques:

With the available physical layer techniques, we propose to develop an adaptive encoding strategy based on the channel fading conditions for layered video. The strategy is to efficiently assist the physical layer in adjusting to short term fading conditions. As mentioned above, the physical layer can transmit only a certain number of bits based on maximizing channel capacity. However the physical layer does not know which bits to be dropped (that is, it does not know the semantics of the enhancement layer). Hence a novel encoding strategy which will place the bits in a gradual order of decreasing priority is proposed. This strategy will assist the physical layer in selecting the bits to be picked without knowing the semantics of the enhancement layer. The physical layer could just pick the required bits from the stream, beginning with the base layer and drop bits at the end of the stream, most often from the enhancement layer. Since the physical layer technology maintains atleast a minimum QoS for all periods outside outside atleast the base layer is guaranteed to be transmitted under BER constraints.

Impact on FEC: As explained above the physical layer tries to maintain a channel with a constant BER under a variety of fading conditions. The importance of the technique is that it can simplify the Forward Error Correction mechanism. Actual video quality depends on PER (Packet Error Rate). In video encoding even if a single bit in a packet is lost, the entire packet shall not be used and is assumed lost. Assuming a packet size of 1000 bits and a BER of 10^{-3}, then on an average only 1 bit in a packet will be lost. A very simple but high rate coding mechanism (Reed Solomon or BCH code) with low complexity that can correct a few bits will be sufficient to further bring down the PER to an acceptable value. Hence very high rate codes can be used, making the system bandwidth efficient.

4. MIMO solution to guarantee QoS

In order to support a minimum QoS, we provide a physical layer solution using MIMO technology that tries to guarantee the maximum bit rate possible under a target BER and under peak power limitations. Typically one would like to get a minimum rate transmitted equal to the base layer rate under all periods outside outage. This would determine the minimum QoS. The peak power is determined under this criterion. Hence under the peak power constraints we can guarantee at least the base layer bits maintaining an acceptable level of outage.

We consider the channel model as in V-BLAST. Let the transmitter have M antennas and the receiver have N antennas with $N \geq M$. We assume a standard channel model as in [7],[8]. The channel is given as a NXM matrix C with complex gaussian random variables of mean 0 and variance 1. We use a M-QAM constellation for each transmit antenna, whose size can be varied as desired. The transmitted signal is a vector S_t of size $MX1$ whose entries correspond to a QAM signal from each antenna. The received signal vector S_r and noise vector N are of size $NX1$. The elements of the noise vector are zero-mean gaussian random variables with noise density N_o. The desired target BER is BER_{target}. Let E_{tx} be the peak power constraint.

From [7],[8], the received signal is modeled as,

$$S_r = CS_t + N \tag{1}$$

At the detector a zero-forcing linear detector is assumed. Conceptually the detector produces an estimate of the transmitted vector, S_{det} by the following operation

$$S_{det} = (C^\dagger C)^{-1} C^\dagger S_r \tag{2}$$

($C^\dagger$ is the conjugate transpose)

which gives,

$$S_{det} = S_t + (C^\dagger C)^{-1} C^\dagger N \tag{3}$$

For each transmit antenna, with transmit power E_{s_i} with k_i bits, constellation size $L_i = 2^{k_i}$, the BER is tightly bounded, according to [5],[6], [13], by,

$$BER_i \leq 0.2 \cdot e^{(\frac{-1.5}{L_i - 1} \cdot \frac{E_{s_i}}{N_o} \frac{1}{[(C^\dagger C)^{-1}]_{ii}})} = BER_{target} \tag{4}$$

Rewriting the above equation provides,

$$k_i = \log_2(1 + \frac{-1.5 E_{s_i}}{N_o \ln(5\text{BER}_{target})[(C^\dagger C)^{-1}]_{ii}}) \tag{5}$$

For convenience denote $D_i = \frac{-1.5}{\sigma_n^2 \ln(5\text{BER}_{target})[(C^\dagger C)^{-1}]_{ii}}$ and hence $k_i = \log_2(1 + D_i E_{s_i})$.

Here we seek to maximize the sum of bits transmitted over all the antennas which is $\sum_{i=1}^M k_i$ under the constraint that power utilized over all the antennas, which is $\sum_{i=1}^M E_{s_i}$, is constrained.

Formulating the lagrangian provides,

$$L = \sum_{i=1}^M k_i + \mu \sum_{i=1}^M E_{s_i} \tag{6}$$

Here μ is the Lagrangian multiplier. Taking partial derivative of (6) with respect to each E_{s_i} gives

$$\frac{\partial L}{\partial E_{s_i}} = 0 = \frac{D_i}{(1 + D_i E_{s_i}) \ln(2)} + \mu \tag{7}$$

This gives,

$$E_{s_j} = -\frac{1}{D_j} - \frac{1}{\mu \ln(2)} \tag{8}$$

Since $\sum_{i=1}^M E_{s_i} = E_{\text{tx}}$, we get $E_{\text{tx}} = -\frac{M}{\mu \ln(2)} - \sum_{i=1}^M \frac{1}{D_i}$. This gives,

$$\mu = \frac{-M}{(E_{\text{tx}} + \sum_{i=1}^M \frac{1}{D_i}) \ln(2)} \tag{9}$$

Substituting (9) in (8) gives power level for each Tx antenna i as,

$$E_{s_j} = -\frac{1}{D_j} + \frac{E_{\text{tx}} + \sum_{i=1}^M \frac{1}{D_i}}{M} \tag{10}$$

The bits allocated to each antenna i can be found by substituting (10) in (5).

Max Power to transport Base layer for a given outage

We mentioned in the beginning of the previous section that the peak power is constrained so that a minimum rate is guaranteed for all periods outside a given percentage of outage. Any excess power is undesirable. Here we describe a technique to find the peak power necessary. We know that for each Tx antenna i,

$$E_{s_i} = -\frac{\sigma_n^2 \ln(5\text{BER}_{target})}{1.5}(2^{k_i} - 1)[(C^\dagger C)^{-1}]_{ii} \tag{11}$$

Since each antenna on an average transports $\frac{TCBR}{M}$ bits, the quantity $-\frac{\sigma_n^2 \ln(5\text{BER}_{target})}{1.5}(2^{k_i} - 1) = -\frac{\sigma_n^2 \ln(5\text{BER}_{target})}{1.5}(2^{\frac{TCBR}{M}} - 1) = T$ is fixed.

To achieve the BER_{target} the power level in (11) is the minimum necessary for a given $[(C^\dagger C)^{-1}]_{ii}$ at antenna i. Hence any power above this level could be used to transport the Base Layer bits. Suppose the outage probability desired is $p\%$, then the probability of outage is given by,

$$Prob(E_{s_i} \geq T[(C^\dagger C)^{-1}]_{ii}) = p\% \tag{12}$$

Or,

$$Prob([(C^\dagger C)^{-1}]_{ii} \leq \frac{E_{s_i}}{T}) = p\% \tag{13}$$

Therefore for all antennas,

$$Prob(\sum_{i=1}^M [(C^\dagger C)^{-1}]_{ii} \leq \frac{1}{T} \sum_{i=1}^M E_{s_i}) = p\% \tag{14}$$

The maximum power required depends implicitly on the TCBR or Base Layer bits and the statistics of the channel. It has been shown in [5] that $\frac{1}{[(C^\dagger C)^{-1}]_{ii}}$ is chi square distributed. Hence each $[(C^\dagger C)^{-1}]_{ii}$ is inverse gamma distributed with even degrees of freedom. The distribution of the summation of inverse gamma distributed random variables is unknown[17][18][19]. Hence the peak power required is derived from simulations and collecting the statistics of $\sum_{i=1}^M [(C^\dagger C)^{-1}]_{ii}$. The peak power is found from the density function of $\sum_{i=1}^M [(C^\dagger C)^{-1}]_{ii}$ and the value corresponds to the portion of the curve under $1 - p\%$ area. Now since anything above this value occurs for only $p\%$ of time, therefore the outage probability will be $p\%$.

5. Background on application layer used

The application layer of a video system consists of a video encoder and decoder. Basically, the underlying layers provide transmission information to the application layer and the video encoder must adapt to the situation instantly.

Given the optimal wireless channel capacity, one needs to adjust the video encoder such that the bit rate of the output stream matches the wireless channel capacity. However, because the wireless channel capacity changes frequently, it is a challenging task for the output bit rate to adapt to this change.

The video encoder control should adjust the bit rate of its generated stream to fulfill the requirement of the desired bit rate. There is usually a latency before the bit rate can be adjusted to the desired value. This is because that the rate control mechanism normally depends on the encoding buffer status. If the buffer is near full, the quantization parameter (QP) is increased to drop video quality of the next frame such that the bit usage is reduced. Conversely, if the buffer is near empty, QP is decreased to consume more bit budget and to boost the video quality of the coming frame. The buffer status cannot accurately reflect the instant bit rate change. Instead, the buffer aims to smooth the encoding quality by absorbing the bit rate fluctuation. The consequences of the mismatching on the transmission bit rate may either induce more bit error rate and unnecessary transmission power consumption or waste of available bandwidth.

On the other hand, even given the same bit rate requirement, the rate control mechanism will not be able to follow that value constantly. This is because the accuracy of the rate control also relies on the frame type. For the generic video system, there are three types of frames: intra-frame (I-frame), predictive-frame (P-frame), and bi-directional predictive frame (B-frame). The I-frame does not use the temporal redundancy among neighboring frames and has the least encoding efficiency, but the I-frame is important to stop the temporal error propagation which is important for video transmission over error prone environments [27]. The P-frame technique utilizes the previous I/P-frame to predict the current frame, which leads to significant encoder performance gain over the I-frame. The P-frame, however, spreads the error: once a frame is in error the frame that depends on it may be affected. The B-frame uses bi-directional prediction but it is never used by other frames as a reference. Removing all of the B-frames will not affect the validity of the video sequence. Hence, in our study, we do not consider B-frames. Given the same encoding quality, the bits used by the I-frame is fairly large. Therefore, it is virtually impossible to control the bit budget of an I-frame to a P-frame level and to maintain the quality smoothness simultaneously. We will use PGOP in our design to help control the bit rate.

PGOP alternative for efficient rate control: In [24][28], we proposed a scheme named progressive group of picture (PGOP), by which we may replace the I-frame with a sequence of P-frame in which a number of macro blocks are intra-coded. In [28], we further compare the attributes of the PGOP scheme and the I-frame based GOP structure. We stated that PGOP can aid the rate control mechanisms (both frame-level rate control and macro block-level rate control) to achieve the bit rate smoothness with the equivalent error resiliency capability as an I-frame.

FGS - a scalable video architecture: With the growing need on the streaming video over the Internet/Wireless network, the scalable video coding is proposed to better adapt to the network channel capacity. Like the single layer video transmission, the scalable video encoding also favors the smoothing bit rate in order to keep the modest quality variation for the CBR channel.

FGS has recently been proposed to MPEG-4 video coding standards. Aimed at streaming application, video streams of FGS can be flexibly truncated at a very fine granularity to adapt to the available network resources. With FGS encoding the video is encoded into a base layer (BL) and an enhancement layer (EL). Similar to conventional scalable video coding, the base layer must be received completely in order to decode and display a basic quality video. In contrast to conventional scalable video coding, which requires the reception of the complete enhancement layer to improve upon the basic video quality, FGS coding allows a fast and arbitrary bit-level cutting on the enhancement layer.

Combined PGOP and FGS performance: Figure 1 demonstrates the performance comparison of PGOP and GOP in both scalable and non-scalable modes. The base layer coding rate is set to be 180 kbps. We adjust the enhancement layer coding rate from 40 kbps to 320 kbps. Thus we have the total occupied channel rate ranging from 220 kbps to 500 kbps. It is shown from the column chart that the PGOP based FGS outperforms GOP based FGS. As a comparison, the line chart illustrates the gain of PGOP using fixed non-scalable MPEG-4 coding. The big quality gap between the scalable coding and the non-scalable coding clearly shows that FGS scheme trades the encoding efficiency with the scalability flexibility.

6. Cross Layer Solution

As we mentioned earlier, our proposed physical layer design enables a constant error bit rate by varying the transmission rate in conjunction with an optimized power allocation among different antennas. This is beneficial to the error control. As mentioned above in section 3, for a given bit error rate, we might be able to correct all of the errors in a single packet using a simple FEC code. In [29], we introduced a FEC that may respond to the varying bit error rate instantly, which exactly matches our requirements on the error correction capability. Our sustained bit error control can guarantee a constant bit error rate by adjusting power levels and bit levels in each antenna optimally. However, for that period, it is a challenging task to adapt the encoder bit rate to the varying maximal channel bit rate.

In equation(14), choosing the TCBR value corresponding to the Base Layer bits(made smooth by PGOP), makes a constraint on the peak power available. This constraint guarantees the reception of the Base Layer for all periods outside outage that satisfying a minimum QoS.

Although FGS is not specifically designed for the real-time video encoding, the FGS technology does not induce much encoding complexity. In our cross layer design we employ FGS due to its flexibility of arbitrary bit rate truncation at the enhancement layer. This can be utilized to an advantage to adapt to the fluctuating channel capacity.

In the proposed cross layer solution for rate control, the encoder does not choose the rate. The encoder provides all the necessary bits (both base layer and enhancement layer) in a frame in a buffer in a continuous fashion for the physical layer to access. The physical layer based on the maximal number of bits possible after optimization provided in section 4, selects only the necessary bits and truncates the next. Since the maximum power constraint guarantees that the base layer will be transported for all periods outside outage, the truncation for all non-outage periods happens at the enhancement layer. This will not be a problem as video can be provided with partial reception the Enhancement Layer. This provides real-time rate adaptation for every coherence period.

7. Demonstration of the solution

In our simulation, we use Microsoft version MPEG-4 FGS reference code [25] with Test Model 5 rate control [26]. We plot performance comparison results using *Foreman* QCIF sequence. The sequence lasts 300 frames and the frame coding rate is 30 fps.

PGOP for FGS: PGOP is able to maintain the consistent quality of each frame. In the non-scalable coding mode, each frame is the reference of its next neighbor during the inter-prediction process. In [28], we have shown that PGOP has much smoother quality than GOP. This quality consistency also leads to the consistent residual errors of the frame to be predicted, which may contribute to the average quality gain. For the scalable encoding mode, the base layer provides reference for the enhancement layer. For the similar reason, the fluctuation of residual errors to be encoded in the enhancement layer is also reduced, which leads to the average quality gain shown in Figure 1. Figure 2 shows the bit rate fluctuation comparison of the base layer between GOP and PGOP; Figure 3 shows the combined bit rate fluctuation comparison between GOP and PGOP.

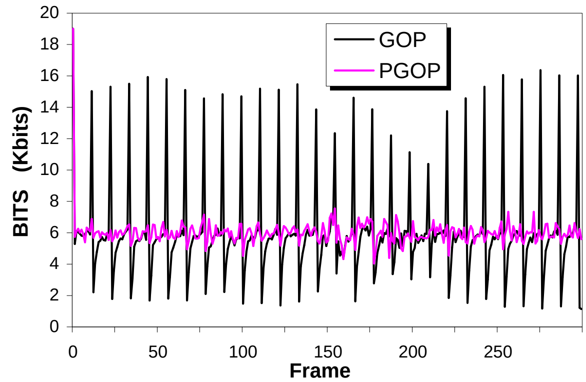

Fig. 2. Base layer bit rate for scalable coding in GOP and PGOP.

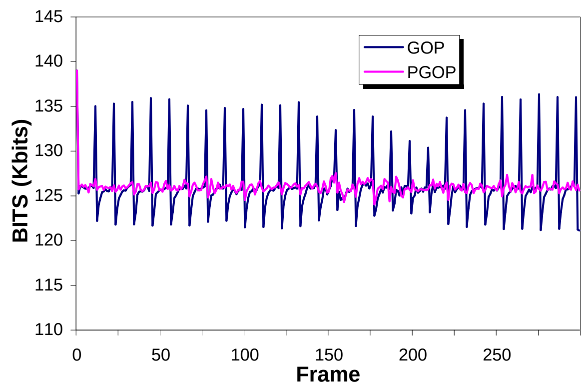

Fig. 3. Combined bit rate for scalable coding in GOP and PGOP.

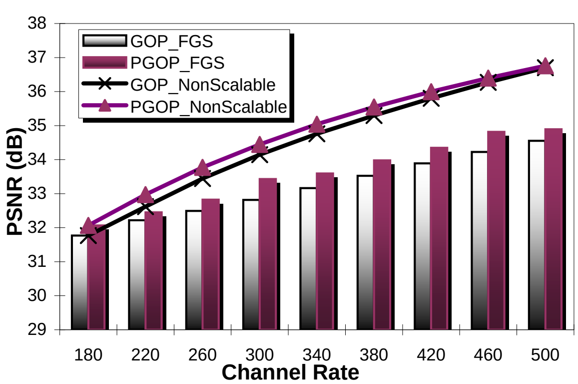

Fig. 1. Quality comparison.

Demonstration of the cross layer scheme: In this subsection we demonstrate our scheme. Figure 4 shows the number of bits that were originally in each frame frame and the number that were truncated. One could notice that a significant proportion of each frame was truncated. However the base layer rate average was around 12kbits per frame implying that, during the simulation, the truncation happened only at the enhancement layer. Figure 5 shows a plot of the received signal quality with the cross layer scheme. The figure shows that using our technique (PGOP in base layer + enhancement layer truncation) matches the wireless channel capacity provision. Even though we terminate the bits at the enhancement layer to match the available channel capacity, our scheme provides a good PSNR value. From the above results and also the fact mentioned in section 6 about the ability to truncate in real time at the physical layer it is clear that the proposed cross layer scheme is a very practical cross layer solution for real-time video satisfying QoS.

8. Conclusion

In this paper we proposed a new cross layer scheme for real-time video that satisfies a minimum QoS. We derived the conditions of the power requirement and optimal energy and bit allocation to each antenna of a MIMO system. Then we introduce FGS and PGOP combined scheme that matches the channel bit rate well. The scalable video provides the ability to truncate video at the enhancement layer. This makes it ideal for implementation at the physical layer and hence provides an efficient cross layer scheme for real time video.

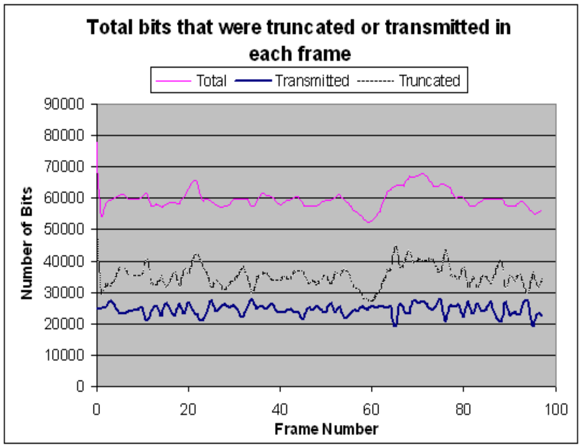

Fig. 4. Chart showing the number of bits truncated and transmitted in each frame with the cross layer scheme.

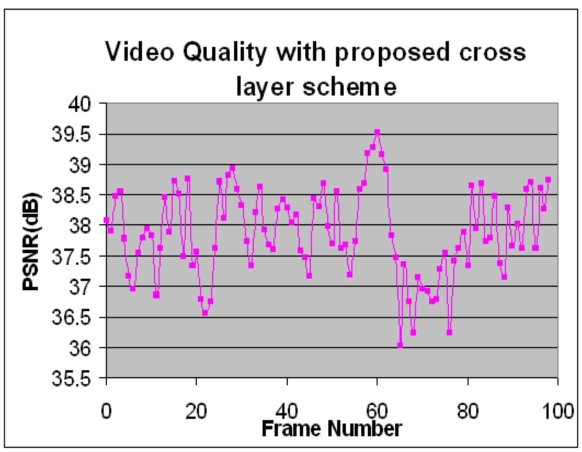

Fig. 5. Demonstration of the cross layer scheme.

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Design, Implementation and Verification of an IEEE 802.11e HCF Simulation Model in ns-2.28

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I. Introduction

The IEEE working group 802.11 is about to ratify a QoS-enhanced standard for wireless LANs, called IEEE 802.11e[1]. The standard proposes improved MAC functions which support prioritized and parameterized QoS services. The MAC model includes a Hybrid Coordination Function (HCF) composed of the legacy Distributed Coordination Function (DCF) of 802.11[2], the HCF Contention Access Function (EDCA), and the HCF Controlled Access Function (HCCA).

The public domain network simulator, ns-2[3], only supports DCF in the IEEE 802.11 MAC layer. In Fall'04, we extended the MAC layer functionalities of ns-2.27 with IEEE 802.11e MAC algorithms[4]. The accomplishments of winter quarter are:

- We have now completed the 802.11e simulation tool in ns-2. Almost all of the functionality of 802.11e is now captured in the tool.
- We have run extensive simulations with the tool and are confident that it functions properly.
- We ported the HCF module to the new ns-2.28 release.
- We have added traffic stream generation, classification and buffering capability.
- We have implemented example HCCA scheduler of the standard. We are working on scheduling algorithms that will improve efficiency under various traffic scenarios.
- Via simulations, we have shown that DCF performs poorly in a saturated scenario, while within EDCA using TXOPs, throughput still can be increased by sacrificing reasonably from delay. We are currently also working on call admission control algorithms which will tune the network load and parameters efficiently.

II. HCF Module MAC Architecture

Figure 1 depicts a block diagram of current MAC layer implementation in ns-2.28 that facilitates the operation of the QoS channel access functions specified by the 802.11e standard.

Traffic arriving at the MAC form the upper layers is buffered at the corresponding interface queue depending on its priority and classification. If it is served by the EDCA, it is first mapped to one of EDCA access categories (AC). The access of an individual EDCA queue to the channel is based on CSMA/CA. If a traffic stream is generated which is to be served by the HCCA, the station should wait for scheduled poll grants from AP to access the channel. Our module currently includes the example scheduler and call admission controller of the 802.11e draft[1].

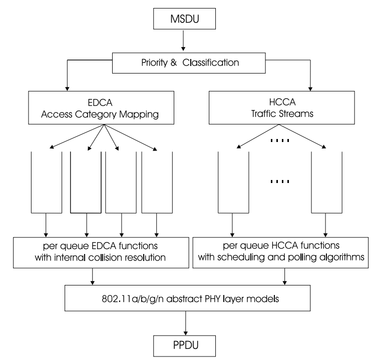

Fig. 1. HCF module MAC architecture

III. Simulation Results

When simulated over IEEE 802.11n Scenario 1[5], with each STA supporting the 802.11n PHY layer of WWiSE proposal[6] at a fixed data rate of 106 Mbps, 2x2 MIMO mode, and 20 MHz protection, the EDCA resulted in a MAC efficiency of %35. For the same simulation scenario, the HCCA provides a MAC efficiency of %50. Although WWiSE proposal reports higher values for both access functions, it should be noted that their MAC implementation also supports yet unstandardized MSDU and PPDU aggregation which increases efficiency considerably. The simulation results prove our HCF module functions properly. At the time of writing, we are also improving the ns-2 HCF module with an MSDU aggregation algorithm.

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Date: April 8, 2005

Introduction:

In the Winter Quarter 2005, we completed a significant part of our research on the use of the Pre-Distortion approach for linearization of the High Power Amplifier (HPA) to mitigate PAPR (Peak-to-Average-Power Ratio) in OFDM. Furthermore, we have worked on combining this approach with the approach to PAPR mitigation based on adaptive input power control. A brief description of the specific accomplishments follows.

Summary of accomplishments:

First, we finalized a journal paper describing the algorithm for PDs (Pre-Distorters) for SSPA/TWTA (Solid State Power Amplifier/Traveling Wave Tube Amplifier) in detail [1]. In this paper, we designed new Pre-Distorters for both TWTA and SSPA, which significantly mitigate the HPA nonlinear problem by pre-compensating the distortion. They are based on accurate analytical representations of the amplifier and pre-distorter characteristics. These use very few parameters, which, due to their sparseness, can be captured and tracked in real time.

Second, we tested the validity of polynomial approximation using the LMS algorithm for PD of SSPA. The polynomial approximation fits well with our PD in the low amplitude range. However, in the high amplitude range, a lot of polynomial coefficients are needed to get relatively close accuracy with our PD. Since the amplitude of the OFDM signal is Rayleigh distributed (which means that only very few samples are distributed in the high amplitude range) we conclude that polynomial approximation for PD can also constitute a good implementation method.

Third, we are almost ready to submit one more paper publication. This paper describes implementation issues of PD for SSPA in detail [2]. In this paper, based on the results in [1], we provide the design of a tracking algorithm for the case in which the practical HPA is unknown and varying. Simulation results are presented to investigate the performance improvement of the pre-distorter, and to study the distortion effects caused by saturation, overflow, and quantization with different bit-widths, since the bit-width of OFDM base-band (OFDM BB) and DAC/ADC is limited by cost and design constraints in real systems.

Fourth, we are preparing one more paper which is an extended version of [1]. Even though in [1], we clearly described the validity of our algorithm, we didn't consider how to optimize the algorithm in real systems with relatively less complexity, which is closely related to the implementation issue. We assumed that the change of parameters in HPA is Gaussian distributed and tracked the variation of parameters using the LMS algorithm. Initial simulations of PD for SSPA look promising.

Future work:

As future work, we will try to develop a novel universal PD not only for HPA but which will include also the channel. This means it will include channel "pre-equalization". Forward channel estimation would be one of the core techniques to be developed as part the PD design effort. Also, since a PD for HPA only works on a limited range (which leads to clipping at large signal amplitudes), we will continue to investigate how to best combine pre-distortion with adaptive input-power control so that both PAPR and clipping may be mitigated. Our ultimate goal is to extend these techniques for application to Multi-User (code) MIMO-OFDM wireless communication systems.

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UC IRVINE CENTER FOR PERVASIVE COMMUNICATIONS AND COMPUTING
CPCC Fellowship Second Quarter Progress Report, April 2005

Project Name:	**SOC Power Optimization Framework**
Graduate Students:	**Yun Long**, EECS (on CPCC Fellowship for WQ05)
	Sudeep Pasricha, ICS (on CPCC Fellowship for WQ05)
CPCC Affiliate Professors:	Fadi J. Kurdahi and Nikil D. Dutt

Project overview

The long term goal of the proposed project is to develop a system level methodology for power optimization for SoCs. In the immediate term, the proposed project will investigate techniques for efficient power modeling of SOC bus architectures, as well as of system-level IP blocks, and their use in the architectural exploration of IP-based SOC designs. On the basis of such power models of the system, we will be able to explore the architectural design space and scheduling schemes by estimating system performance and power consumption. We propose a two-stage exploration flow, coupling both the modeling of IPs, as well as power-performance exploration at the transaction level of modeling. In the HAIM (Hierarchically Abstracted IP modeling) stage, the data access pattern on the interface of each module is extracted, removing the computation detail inside the modules while keeping the flexibility and constraints on data accesses. In DOSE (Data Organization Space Exploration) stage, scheduling schemes are explored and performance, power consumption, and area cost are evaluated so as to selecting the most efficient design. The proposed exploration flow will be based on the COMMEX transaction-level communication architectural framework, on which we will study the H.264 application (the latest video coding standard), and JPEG2000 (the latest still image coding standard).

Progress

During WQ05, we engaged six Ph.D. students in this project and made significant progress. Yun Long and-Sudeep Pasricha focused on SystemC modeling of H.264 decoder, and also started an ASIC design task with four other students on the same system in RTL level, aiming at providing data of performance and power consumption for SystemC model. We also engaged an undergraduate student on SystemC modeling on JPEG2000 encoder, for purpose of providing reference information on a different multimedia application. To conclude, we have achieved the following:

- Designed a SystemC based framework of H.264 decoder, including the H.264 standard reference code and COMMEX transaction-level model of communication architecture. This framework facilitates component configuration, such as different IP cores and different communication channels, and eases testing different scheduling schemes. The framework migrates easily to other applications. The implementation of this framework is smooth and is over 50% finished.
- Started ASIC design task of H.264 decoder. Finished documenting design specifications on the four major functional modules, the communication channel (bus and arbiter), and memory architecture. The overall architecture, state machine and major datapath of each module are also documented and ready to be coded in structural HDL languages.
- Started to build SystemC model of JPEG2000 encoder. Different from the aforementioned H.264 model, this model uses TLM library of SystemC, and it will explore the tiling (possibly overlapped) problem presented in the system. This model will use the implementation data acquired from a former research project by Yun Long and another Ph.D. student. This task will expose more interesting problems to us on broader applications, and will probably provide feasible solutions.

Going forward, our goal is to finish the System model, integrating in the performance and power data acquired from the ASIC design task, and study power optimization techniques with the help of easy design configuration of the model and accurate power modeling. The model will be able to easily and rapidly migrate to any multimedia application systems.

Project report for CPCC Sheng Yen

Since having a meeting with Professor Heydari regarding the project conducted in the future for CPCC funding research, I started to get involved into the research of exploring the application of ultra-wide band front end analog circuit design with focusing on designing frequency synthesizers. After reviewing related system level papers and information from FCC website, I was getting involved into the goal and the developing situation of ultra-wide band communication circuit design and system requirement more. To understand the key concept and fundamental knowledge of ultra-wide band, communication front end design, and frequency synthesizers, I went through papers, books, and reference reports to realize the history and what has been done so far. In the beginning, because there are two categories in ultra-wide band circuit design: impulse radio which has been used in radar detection and multi-band OFDM which is modified and extended from narrowband RF circuit design, we studied these two topics at the same time. Some pulse shaping and low-power application has also been studied and proposed for sensor networks system from the idea of ultra-wide band impulse radio. Later on, we decided to concentrate on designing a multi-band frequency synthesizer. In order to demonstrate the performance of designed circuits, Compute aided design software "Cadence" is adapted to use. I learned to simulate circuits in Cadence for time-domain transient and DC analysis. I started to construct a conventional design of a CMOS phase lock loop circuit to be familiar with simulation performed in Cadence and get more sense on operation mechanism of frequency synthesizers. I have successfully simulated frequency detectors, charge pump with low pass filter circuits. Then, the main focus would be put on voltage controlled oscillator and frequency divider circuits. First, I utilized idea components to have VCO oscillate and get to know its operating principle and design tricks. After that, I designed LC-VCO oscillating at 10GHz with non-ideal components including PMOS-NMOS structure and pure-NMOS structure. Phase noise are also been tested between with and without bias current source designs. In addition, divide-by-2 frequency divider has also been completed. Since UWB circuits need to work up to 10GHz, traditional CMOS D-flipflop could not achieve such high frequencies. CML CMOS family is utilized to achieve 20GHz without incorporating. Future work would be put on simulating more different design of frequency divider and VCO circuits to optimize the performance of lower phase noise, higher operating frequencies and lower power consumption. In the mean time, the fractional-N frequency divider concept is worth to explore.

Progress Report Winter 2005
Project: Design Techniques Toward a Full-Rate 40Gb/s Transmitter in 0.18μm CMOS

Ahmad Yazdi, CPCC Fellow
Prof. Michael Green, Research Advisor

Introduction: Serial data communication systems are operating at the throughputs up to 40Gb/s. Up to now, communications integrated circuits operating at such high speeds were engineered using GaAs, InP, or SiGe bipolar technologies. Aggressive technology scaling in CMOS processes, however, bring a number of advantages over these compound semiconductor technologies. In particular, CMOS technology exhibits a high integration density, thereby making it possible to design many building blocks on the same die. This enables economical system-on-a-chip solution for high-performance circuits. Despite all these advantages, one major concern, namely lower unity-gain frequency, makes CMOS IC design particularly challenging for high-speed IC design. Many bandwidth enhancement techniques have been developed to increase the speed of the CMOS ICs. This work features combination of the distributed and lumped techniques to achieve higher speed. Also due to the non-ideal clock and mismatch in different circuits of the transmitter, the 40 Gb/s output data commonly carries periodic jitter (PJ) and duty-cycle distortion (DCD). A retiming circuit running at 40 GHz clock greatly reduces the PJ and DCD from the output data. This is the main advantage of the full-rate transmitter over the half-rate transmitter.

Accomplishments for Winter 2005

- *Design of a new CML based retiming circuit for 40Gb/s application:*
  This circuit relies on holding the data by parasitic capacitances during the half of the clock period (12.5ps). The circuit features a new type of the shunt-peaking techniques with excess overshoot in the time domain.

- *Retimer and output buffer corner analysis:*
  Corner analysis shows that the output buffer at 40Gb/s are very sensitive to the process variation. The resistor are changing 16% over the corners while the 40Gb/s output buffer can only tolerate 5% changes over the corners. The retimer is less sensitive over the process variation

- *Design of a 40GHz "push-push" VCO:*
  In this design a 20GHz waveform is generated by conventional cross-coupled VCO. By adding the two differential outputs the fundamental frequency will cancel out and the second harmonic appears at the output (40GHz).

Future work during spring 2005

1. Design an oscillator in 0.18µm CMOS that provides a differential 40GHz clock signal with sufficiently low random jitter

2. Optimizing the output buffer for being more robust over process variations.