Channel Controlled Payload Length &Rate in MIMO Networks Using Cross-Layer

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Abstract—**Today multimedia traffic is congested due to heavy data traffic. If traffic becomes flooded, congestion is more and Data delivery is less. The source can be optimized based on the channel condition. Hence, Shannon's theorems can be enhanced to include channel state information (CSI). We propose cross-layer design which optimizes data rate and payload length based on CSI in MIMO networks. Video summary frames are transmitted via MIMO transmit/receive diversity with AMC at the physical layer and optimal payload length algorithm and ARQ at the data link layer and source coding at the application layer. We, assume first model the finite-state Markov chain (FSMC) for the physical layer service and based on that, we then characterize the Lagrangian relaxation and Dynamic programming to find the optimal shortest path. The source coding as well as retransmission requests are based on the channel-state information (CSI) and the system controller is implemented in all the four physical, MAC layer and application layers. The numerical results revealed that our proposed cross-layer design can efficiently achieve the distortion gain and payload adaptation at worst channel condition.**

***Index Terms*—Cross-layer design, quality-of-service (QoS), adaptive modulation and coding (AMC), multiple input multiple output (MIMO), automatic repeat request (ARQ), mobile wireless networks.**

About four key words or phrases in alphabetical order, separated by commas.

I. INTRODUCTION

The 4G cellular standards work towards multi-user environment. The users are provided with different data rates and at different terrain profiles. So, the information theory bounded by Claude Shannon in his work "A Mathematical Theory of Communication" may no longer be optimal. This division of coding theory into compression and transmission is justified by the information transmission theorems, source-channel separation theorems that justify the use of bits as the universal currency for information in many contexts. However, these theorems only hold in the situation

where, one transmitting user wishes to communicate one receiving user.In scenarios with more than one transmitter(multiple-access channel), more than one receiver(broadcast channel) or intermediary "helpers"(relay channel),or more general networks, compression followed by transmission may no longer be optimal. Network information theory refers to these mmulti-agent communication models.

In multi-user scenario, user can access the network unaware of channel conditions. This will make the network more congested. If congestion becomes predominant, QoS will be degraded. This will results in loss of data, delay and distortion full reception at the receiver. If the channel's bandwidth is fixed, then user may suffer. So this problem can be overcome by optimizing the physical layer with higher layers.

The effective throughput is affected by a number of parameters, including transmission rate, payload and header size, constellation size, transmitted power and received noise characteristics. Previous results revealed that careful payload length adaptation significantly improves the throughput performance at low signal to noise ratios (SNRs), while at higher SNRs, rate adaptation with higher payload lengths provides better throughput performance. These payload sizes cover a wide range of applications from various voice codecs to H.264 video conferencing applications along with various data applications like web browsing, FTP, etc. In [12], payload length was considered as an optimization parameter and tight coupling between payload length and data rate to maximize the single-user throughput on the AWGN and different fading channels was required.

In general, the video summarization algorithm will generate a still-image storyboard, which is composed of a collection of salient images extracted from the underlying video sequence. Some earlier work considered the packet loss factor due to unsatisfactory wireless channel, where some does not consider. The source coding has not been optimized in the framework, which might directly impact the perceptual quality of the results. In addition, the algorithm does not guarantee a good content coverage aspect of the selected frames because potential packet loss penalty heavily biases the selection process.

Here, we propose Cross-layer design that jointly optimizes the physical layer, MAC layer and application layer. Transmission of video summary at application layer is based on MIMO transmit/receive diversity with AMC at the physical layer and payload length and rate adaptation at the MAC layer and the source coding at the application layer. The rest of the paper is organized as follows. In section II, we derive the Problem Formulation. System Overview is shown in Section III. We have analyzed the Related Work in Section IV. Experimental Results are shown in Section V. Finally we concluded in Section VI.

II. PROBLEM FORMULATION

In this section, we provide a brief explanation about the various techniques adopted in this paper. The summary video frames are fragmented into packets of smaller payload. The link adaptation tables are used to identify the SNR operating regions and corresponding data rates to optimize the video summary packets. The optimization parameters are based on channel state information. The physical layer is designed adaptive modulation coding with MIMO in order to achieve better throughput.

The summary is fragmented into NP packets. If the size of summary frame is less than the packet length, there is no fragmentation to be done. Then the transmitted Packet Error Rate (PER) is PTrans = PL1/L and La = LS. If the frame size is larger, then the packet length Np as

$$Np = \left\lceil \frac{L_s}{L_f} \right\rceil$$

Where, La = actual packet length.
LS = summary frame size.
Lf = fragmentation packet size.

If the actual length of the first packet size is equal to the fragmentation size, then the last packet should be less than the length of the previous packets. So, the PTrans [3] is

$$\text{PTrans} = 1-\left(1-P_L^{\;1/L}\right)^{1/N_P} \tag{8}$$

Where, PL is the probability that L consecutive summary frames are lost simultaneously.

$$= \frac{1}{P_k}\int_{\Gamma_k}^{\Gamma_{k+1}}\left[1-\left(1-a_k\exp(-g_k g)\ \right)^{La}\right] P_\Gamma(g)\ dg$$

$$= \frac{1}{P_k}\cdot\frac{\mathrm{M}}{\Gamma(mL)}\sum_{i=0}^{M-1}(-1)^i\binom{M-1}{i}\sum_{j=0}^{i(mL-1)}\xi_{ji}\left(\frac{bm}{\bar{g}}\right)^{j+mL}\left[PER_{k+1}(g)-PER_k(g)\right] \tag{4}$$

Thus the average PER can be expressed as

$$\text{PER}_k = \frac{\sum_{k=2}^{K} R_k P_k \overline{PER_k}}{\sum_{k=2}^{K} R_k P_k} \tag{5}$$

Where Rk denotes the spectral-efficiency of the kth mode which is listed in TABLE II

Pk denotes the probability that the SNR falls into mode k which is determined by

$$\text{P}_k = \int_{\Gamma_k}^{\Gamma_{k+1}} \text{P}_\Gamma(g)\ dg \tag{6}$$

The packet success rate is defined as the probability of receiving a packet correctly corresponding to AMC mode k, if the SNR γ falls in to the range $\Gamma_k \le \gamma < \Gamma_{k+1}$.

The bits inside the packets have the same bit error rate related with packet error rate. So we can write packet error rate P as for packet containing La bits.

$$\text{P} = 1-\left(1-BER\right)^{L_a} \tag{10}$$

In order to guarantee the upper bound P_{AMC}, the required BER to achieve for any AMC mode is

$$\text{BER}_{AMC} = 1-\left(1-P_{AMC}\right)^{1/L_a} \tag{11}$$

The packet-error rate (PER) for the k[th] AMC mode for k= 2, 3..K can be approximated [8] by

$$\text{PER}_k(g) = \begin{cases} 1, & \text{if } 0<g<g_k \\ a_k\exp(-g_k g), & \text{if } g \ge g_k \end{cases} \tag{2}$$

The AMC is in mode k if the SNR γ falls in to the range $\Gamma_k \le \gamma < \Gamma_{k+1}$.

γ_k can be expressed as below

$$g_k = \frac{1}{\text{g}_k}\ln\left(\frac{PER_k(g)}{a_k}\right) \tag{3}$$

The average packet error rate $\overline{PER_k}$ for the AMC combined with MIMO[10] is

$$\overline{PER_k} = \frac{1}{P_k}\int_{\Gamma_k}^{\Gamma_{k+1}} PER_k(g)\text{P}_\Gamma(g)dg$$

$$\text{PSRk} = 1-\text{PERk} \tag{7}$$

The effective throughout can be defined as the number of payload bits per second received correctly. The payload overhead should also be taken into account and for the initial analysis; we consider the acknowledgements are error free.

The payload overhead takes into account the CSMA/CA channel access time and the header overheads as specified by the IEEE 802.11 protocol. Based on the data rate, the overhead is varying .So the transmit time of packet including MAC, PHY headers and DCF protocol overheads can be defined as [12]

$$P_k = D_k * P_{to} \tag{8}$$

Here, P_{to} = Total protocol overhead
D_k = Data rate corresponding to AMC mode.

The total protocol overhead is calculated as specified by the IEEE 802.11 protocol. Each MAC frame consists of MAC header or MAC protocol data unit (MPDU), variable length frame body and frame check sequences (FCS) as shown in Fig.1. The MAC header and FCS consist of 28 bytes and the ACK is 14 bytes long.

MAC Header 24bytes	IP Header 20bytes	RTP/UDP Header 20bytes	Payload Data	FCS 4bytes

Fig.1. Frame format of a data frame MPDU

The effective throughput corresponding to the AMC mode is

$$T_k = (L_a / L_a + P_k)^* D_k^* PSR_k \quad (9)$$

III. System Overview

The Cross-layer Design is shown in Fig.2.This consists of a structure of physical layer, MAC layer and application layer. The MIMO diversity schemes consists of Nt transmit and Nr receive antennas. As shown in Fig.2 at the transmitter side, we store the video summary frames at the buffer. Source coding is done at application layer. The video summary frames are fragmented into multiple packets for transmission at lower layers. The fragmented packet length which is nothing but the payload length at the MAC layer is adapted as per the channel state information received. At the physical layer according to the channel condition, MIMO combining with AMC method is adopted. If the fragmented packets are received correctly, then the frames are stored at the buffer and the video clip frames are reconstructed.

A. MIMO Diversity Scheme

The selected modulation is transmitted and received via Nt transmit and Nr receive antennas. The different diversity techniques [10], are adopted as per the channel condition as shown in TABLE I. If perfect CSI is available at both sides of the wireless link, maximal-ratio transmission (MRT) also known as beamforming and maximal-ratio combining (MRC) are known as the optimal transmit- and receive-diversity schemes respectively. When the CSI is not available at the transmitter side, space-time block coding (STBC) is good choice to achieve transmit diversity. The selection combining (SC) at either the transmitter or receiver side is a good tradeoff both performance and complexity level.

TABLE I

PARAMETERS FOR MIMO DIVERSITY

MIMO Diversity Schemes	M	L	β
Tx-MRT/Rx-1	1	N_t	1
Tx-STBC/Rx-MRC	1	$N_t N_r$	N_t
Tx-SC/Rx-MRC	N_t	N_r	1
Tx-MRT/Rx-SC	N_r	N_t	1
Tx-STBC/Rx-SC	N_r	N_t	N_t
Tx-SC/Rx-SC	$N_t N_r$	1	1
Performance Upper-bound	1	$N_t N_r$	1

The probability density function (pdf) of the combined signal-to-noise ratio (SNR) denoted by $P_\Gamma(g)$ can be derived as a unified expression [8] as follows:

$$P_\Gamma(g) = \frac{M}{\Gamma(mL)} \sum_{i=0}^{M-1} (-1)^i \binom{M-1}{i} \exp\left(-(i+1)\frac{bm}{\bar{g}} g\right) \sum_{j=0}^{i(mL-1)} \xi_{ji} \left(\frac{bm}{\bar{g}}\right)^{j+mL} g^{j+mL-1} \quad (1)$$

Where, Γ (.) represents the Gamma function, $\bar{g}$ denotes the average SNR of the combined signal, m denotes the fading parameter, ξji,the multinomial expansion coefficients determined by $\xi_{ji} = \sum_{p=a}^{b} x_{p(i-1)/[(j-p)!]}$ with a = max { 0, j - (M-1)},
b = min{ j, (i-1) (M-1) }, ξj0 = ξ0i = 1 , ξj1 = 1/ (j!), and ξ1i = i ! . The parameters M, L and β depends on MIMO diversity schemes as specified in the TABLE I. Here M denotes the selection diversity order and L denotes the combining diversity order β is varied only when STBC scheme is used. The total diversity order is determined by M x L i.e., Nt x Nr.

B. Combined AMC with MIMO

AMC is the powerful technique to increases the spectral-efficiency. The specific AMC modes are selected from the TABLE II based on the cross-layer optimization parameters ak and gk which are the modulation order and the corresponding coding rate of a particular mode. Each mode consists of a specific modulation and FEC code pair as in 3GPP, HIPERLAN/2, IEEE 802.11a and IEEE 802.16 standards. SNR range is divided into 7 non-overlapping intervals as $\Gamma_1 < \Gamma_2 < \ldots \Gamma_{k+1}$ with $\Gamma_1 = 0$ corresponds to outage mode and $\Gamma_2 \ldots \Gamma_7$ correspond to BPSK, QPSK, QPSK, 16-QAM, 16-QAM and 64-QAM respectively. If the SNR increases, it will choose higher modes and if the channel is worst, lower modes are selected.

The packet-error rate (PER) for the kth AMC mode for k= 2, 3...K can be approximated by

$$PER_k(g) = 1, \quad \text{if } 0 < g < g_k$$
$$= a_k \exp(-g_k g), \quad \text{if } g \geq g_k \quad (2)$$

The AMC is in mode k, if the SNR γ falls in to the range $\Gamma_k \leq \gamma < \Gamma_{k+1}$. g_k can be expressed as below:

$$g_k = \frac{1}{g_k}\ln\left(\frac{PER_k(g)}{a_k}\right) \quad (3)$$

C. Source Coding

The video summary frames, which are going to be transmitted, are fragmented into multiple packets for transmission at lower layers. If n number of frames of a video clip{ f0,f1,…..fn-1} of m number of frames of its video summary $\{g_0, g_1, ..., g_{m-1}\}$ are to be transmitted then, the lossy source coding produce the resultant consumed bits of ith summary frame as, i = 0,1,…,m-1. Let Si and Bi be the coding parameters of the lossy coding, which are used to optimize the design. Let, Qi denote the number of fragmented packets of ith summary frame and Ni,q, denote the number of transmissions and Fi,q, denote the packet size for the qth packet of the ith summary frame

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Next we listed the operating assumptions adopted in this paper.

(i).We consider = 2 such that, PL is10-2 so, no neighbouring summary frames can be lost together during transmission.

(ii).We assume fading parameter m= 2 indicating the Rayleigh fading channel and the average SNR $\bar{g}$ = 10 dB. The channel remains time Invariant during transmission of a packet, but varies from packet to packet. Thus the channel can be estimated from the received SNR, g per packet which is a random variable with a probability density function (pdf):

$$P_\Gamma(g) = \frac{1}{\bar{g}}\exp\left(-\frac{g}{\bar{g}}\right)$$

(iii).We assume fixed channel transmission rate, r = 6 * 106 sym/sec and Fixed round trip time TRTT = 100 ms and maximum retransmission number Nmax = 3.

TABLE III OFDM PHY CHARACTERISTICS

Parameter	Value	Notes
tSlot	9µs	Slot Time
tSIFS	16µs	SIFS Time
tDIFS	34µs	DIFS=SIFS+2*Slot
CW_{min}	15	min.contention window size
CW_{max}	1023	max.contention window size
tPLCP_Preamble	16µs	PLCP preamble duration
tPLCP_SIG	4µs	PLCP SIGNAL Field duration
tsymbol	4µs	OFDM symbol interval

(iv).The OFDM PHY characteristics of IEEE 802.11a are summarized in TABLE III.The Payload overhead is taken as 40 bytes as the MAC header and FCS consists of 28 bytes and the ACK is of 14 bytes long.

(v). We have taken the SNR of 2dB. We consider the different modes of AMC as the data rate of 6Mbps, 12Mbps, 18Mbps, 24Mbps, 36Mbps and 54Mbps. The payload length is system varied from 2500 bytes to 5000 bytes.

(v). The MIMO parameters are M= 2, L=2 and B =1 which in nothing but Tx-SC/Rx-MRC

IV. Related Work

High bit rate requirements of a video may congest the network significantly. It is imperative to account for the potential impact of each video user on the network statistics and guarantee that the network is not operating beyond its capacity. Unfortunately, most network designs do not provide mechanisms for protocol layers to optimally adapt to underlying channel conditions and specific application requirements. While protocol layering is an important abstraction that reduces network design completely, it is not well suited to wireless networks since the nature of the wireless medium makes it difficult to decouple the layers. Moreover, meeting the end-to-end performance requirements of demanding applications is extremely challenges without interaction between protocol layers. Video streaming over wireless networks can benefit substantially from a cross-layer design. In this design, interdependencies between layers are

characterized and exploited by adapting to information exchanged between layers and building the appropriate amount of robustness into each layer. For example, routing protocols can avoid links experiencing deep fades, or the application layer can adapt its transmission rate based on the underlying network throughput and latency. The following subdivisions explore cross-layer framework that incorporates adaptation across application layer with other layers to meet the requirements of QoS capabilities for video transmission over wireless networks

Hierarchical video coding is a smart solution to handle the heterogeneity of receivers in multimedia multicast transmission over the wired internet such as in RLM-based schemes and SARC. Basically in a hierarchical encoding scheme which is analyzed in [2], [3], [4], [5],[6], [7], [11], the most relevant elements of the video sequence are included in a base layer, while less relevant pieces of information are put into a second level also denominated as enhancement layer. Usually base layer receives a high priority treatment, while the other layer is delegated to a second plane. While designing hierarchical video codecs, some amount of overhead is introduced due to breakpoint used when splitting the encoded video bit stream into the base and enhancement layers and the ability of assigning different priorities.

One of the main advantages of hierarchical coding is that, this technique can be applied to all encoding schemes, such as H.261, H.263, MPEG-1,MPEG-2,MPEG-4,H.264 among others. The authors considered H.264 video encoding standard, also known as MPEG-4 AVC which is highly efficient by offering perceptually equivalent video quality at about 1/3 to ½ of the bit rates offered by the MPEG-2 format [3], [6], [7].In a nut shell,H.264 consists of different layers. First, the Video Content layer (VCL) contains the specification of the core video compression engines that achieve basic functions such as motion compensation, transform coding and entropy coding. This layer is transport-unaware and its highest data structure is the video slice, a collection of the coded Macroblocks(MBs) in scan order. Second, the Network Abstraction layer(NAL) is responsible for the encapsulation of the coded slices into transport entities of the underlying protocols. Each slice header acts as a resynchronization marker, which allows the slices to be independently decodable, and to be transported out of order and still be decoded correctly at the decoder. A set of error resilience techniques such as data-partitioning, which is an effective application-level framing technique, which divides the compressed data into separate units of different importance had been proposed. Data partitioning creates more than one bit string (partition) per slice and allocates all symbols of a slice into an individual partition with a close relationship [6], [7].The application layer passes its traffic information (the priority of the stream) with their QoS requirements to the MAC layer, which maps these partitions to different traffic categories to improve the perceived video quality. In [6], they have presented a preliminary evaluation of ARSM, which proved effective in adapting the channel rate taking into account the varying channel conditions. ARSM (Auto Rate Selection for Multicast) is an adaptive mechanism in which the AP selects the PHY data rate to be used for the multicast service. The PHY data rate to be used is determined by taking into account the channel conditions perceived by each and every MT (Mobile Terminal) belonging to a given multicast group.

IV .EXPERIMENTAL RESULTS

In this section, we present experimental results for cross layer design with MIMO Channels. Experiments are designed using H.264/AVCJM 10.2 for the video clip called “Glassgow” which is a typical test clip. For ease of comparison we summarize the first 300 frames into 30 frames. For QP adaptation we use different values of QP can be chosen from 10, 20, 30, 40, 50 when delay budget is set equal to the delay time according to different network conditions. For without QP adaptation we use the fixed values like 10 or 20 or 30 or 40 etc.

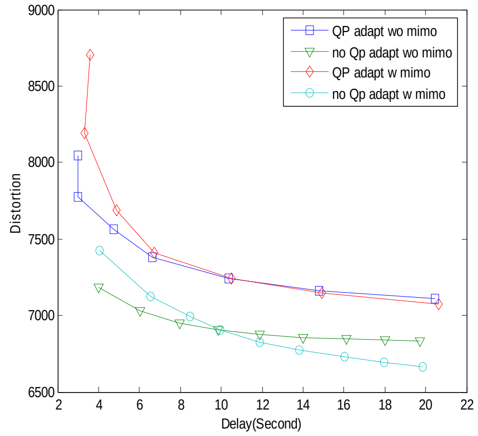

Fig.3. Distortion vs Delay comparison for system model with and without MIMO

Fig.3 plots the system based on MIMO transmit/receive diversity. The system almost works similar for QP adaptation in both with and without MIMO. The performance which is shown in rounded nodes work better compared with all other systems as the distortion gain upto 16% is achieved. In Fig.4, we have shown the effective throughput for data rate of 6 Mbps. Initially increase in payload length increases the effective throughput but it reaches an optimum value from 4000 bytes payload as further increase would increase the packet error rate.

Fig.4.Payload Length Adaptation for date rate of 6Mbps using MIMO Channels

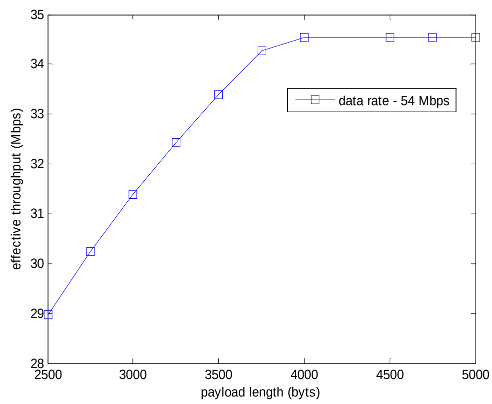

Fig.5.Payload Length Adaptation for date rate of 6Mbps using MIMO Channels

In Fig.5, the effective throughput is 34.5Mbps for 54Mbps date. In Fig.6 we compare the data rate for of 6Mbps, 9Mbps and 12Mbps.Selecting the lowest data rate is too conservative while higher data rate can significantly reduces the effective throughput.

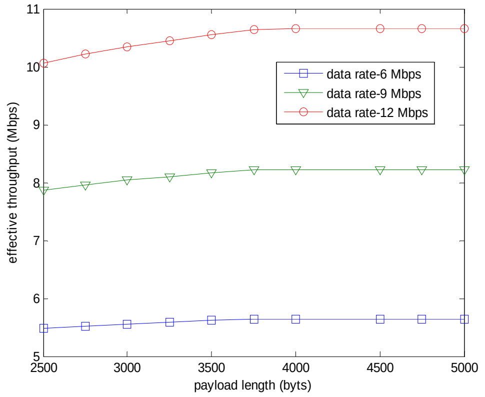

Fig.6. Payload Length Adaptation for different date rates using MIMO Channels

The expected distortion E[D]of the video clip which is shown in Fig.3 can be calculated [10]by

$$E[D] = \sum_{k=0}^{n-1} E\left[D\left(f_k, \tilde{f}_k\right)\right] =$$

$$\sum_{i=0}^{m-1}\sum_{j=l_i}^{l_{i+1}-1}\sum_{b=0}^{i}\{(1-r_{i-b})d\left[f_j, \tilde{g}_{i-b}(S_{i-b})\right]\prod_{a=0}^{b-1} r_{i-a} \quad \} \tag{10}$$

The expected distortion of the video clip should be with in tolerable time delay. So time delay should be bounded with in some delay budget Tmax as

$$\text{Min E[D]}, \quad \text{s.t} \quad T \le T_{max} \tag{11}$$

So, eqn (10) can be written as

$$\text{Min E[D]}, \quad \text{s.t} \quad T \le T_{max} \text{ and}$$
$$\text{Max } (G_i, G_{i-1} \ldots G_{i+1-L}) = 1, \quad i \in [0, m-1] \tag{12}$$

Where, T_{max} is the given delay budget for delivering the whole video clip.

T is the delay in transmitting the whole summary frame which can be expressed as

$$T = \sum_{i=0}^{m-1}\sum_{q=1}^{Q_i}\sum_{n=1}^{N_{i,q}} \left[\frac{F_{i,q}(S_i, B_i)}{R_{i,q,n}(A_{i,q,n}, C_{i,q,n}) * r} + T_{RTT}\right] \tag{13}$$

Where, TRTT is the maximum allowed RTT to get the acknowledgement packet via the feedback channel before a retransmission trial.

V CONCLUSIONS

In this paper, we have considered wireless video streaming application. The goal is to deliver a video sequence with minimum delay and distortion constraints using minimum required transmission energy to the receiver through multipath fading channel. Our formulation considers the tradeoffs in the selection of source coding parameters like source coding, payload adaptation, and physical layer adaptation (AMC with MIMO).

One of the main lessons in the paper is that payload length adaptation at MAC yields higher throughput based on the CSI information obtained. The MIMO employed at the physical layer works well compared with other systems. So, the networks employed with MIMO perform better in order to yield higher throughput. Our system can be readily extended to multi-user scenario like CDMA system. In addition to optimization parameters from the three layers, other layers information can also be included in order to give better results.

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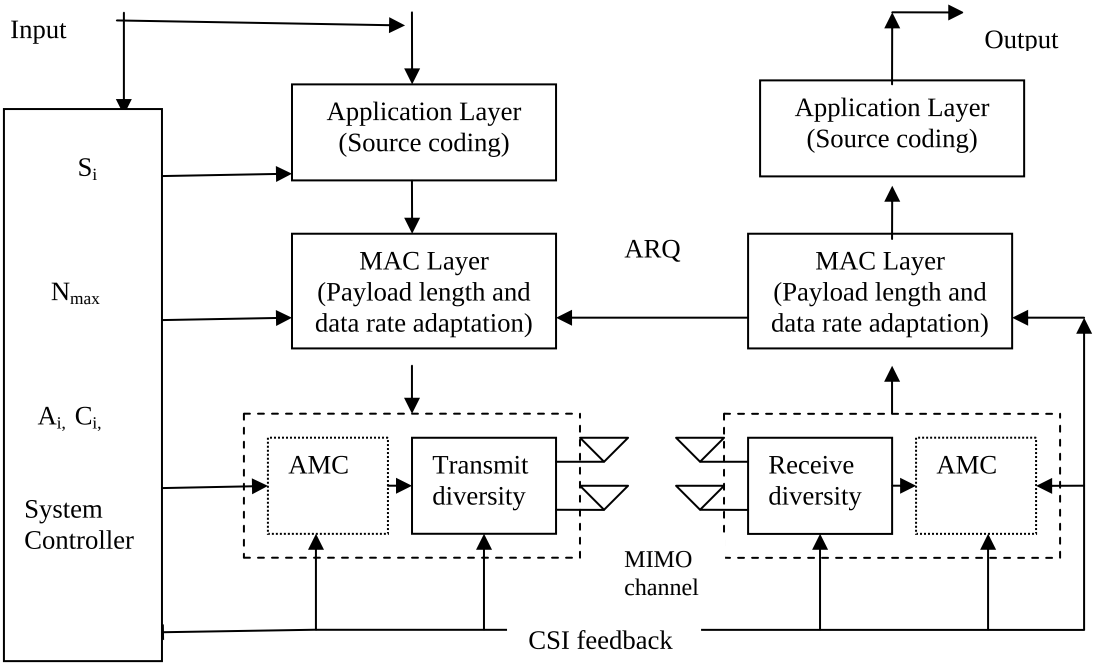

Fig.2. Cross-layer Design Mod

TABLE II AMC MODES

Mode	Mode1	Mode2	Mode3	Mode4	Mode5	Mode6
Modulation	BPSK	QPSK	QPSK	16-QAM	16-QAM	64-QAM
Coding Rate (C_k)	1/2	1/2	3/4	1/2	3/4	3/4
Data Rate D_k (Mbps)	6	12	18	24	36	54
R_k (bytes/symbol)	3	6	9	12	18	27
a_k	1.1369	0.3351	0.2197	0.2081	0.1936	0.1887
g_k	7.5556	3.2543	1.5244	0.6250	0.3484	0.0871