SoftHM: A Software-Based Hierarchical Modulation Design for Wireless System

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***Abstract*— Hierarchical Modulation (HM) is quite efficient for multimedia broadcast in wireless networks. It exploits the wireless broadcast advantage and allows a transmission to reach different users with various qualities. However, existing HM related schemes are based on specially designed hardware, leading to extra hardware and deployment cost in practice. As a remedy for this, we design and implement a software-based HM scheme, named as SoftHM. By carefully tuning the Physical Layer input of a packet (a sequence of "0" or "1" bits, known as PHY payload), a transmitter simply broadcasts the packet with a single rate (e.g., 54Mbps, the highest rate in 802.11g WLAN), while diverse receivers can decode and extract different amount of information (with different reception rate, e.g., 12Mbps or 54Mbps) from it based on their link qualities. Reverse engineering techniques are adopted so that SoftHM works in a plug-in mode, offering it high potentials for wireless systems without any changes in existing modulation and coding modules. We implement SoftHM based on Universal Software Radio Peripheral (USRP) B210 and Commercial of the Shelf (COTS) Wi-Fi devices. Both rigorous analysis on its performance and comprehensive implementation-based experiments show that compared with the traditional time-sharing approach, to support low-quality users with the same bandwidth, high-quality users in SoftHM achieve 111.1% to 616.7% higher throughput. Likewise, to serve high-quality users with the same bandwidth, low quality users in softHM achieve 176.9% to 390.9% higher throughput.**

***Index Terms*— Hierarchical modulation, signal emulation, mobile communication system, PHY layer payload design, modulation and coding scheme.**

I. Introduction

WITH the rapid development of Internet and mobile computing, wireless communications are dominating our network traffic. Cisco Visual Networking Index report shows that by 2022, 71% of the total IP traffic are through wireless and mobile devices with seven-fold increase in volume. Different from its wired counterpart, wireless communications offer an inherent advantage of the broadcast medium. A transmitter can reach multiple receivers in vicinity by a single transmission without any additional cost.

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In practice, to take this advantage is not straightforward. Wireless channels are highly dynamic and different receivers may experience different link qualities [1], making a transmission hard to satisfies all receivers simultaneously. An aggressive data rate may please the receivers who are experiencing high channel quality (named as good-quality receivers). However, the receivers with poor channel quality (named as low-quality receivers) may not be able to decode any thing and are sacrificed. To the opposite, a conservative rate may serve all receivers but waste the bandwidth of high-quality receivers, resulting in low transmission efficiency.

To address these issues, many research works have been devoted for efficient broadcast in wireless networks. One category of works attempt to access different receivers by different channels, such as the bandwidth sharing approach [2] (e.g., different link quality users access the channel at different time). It is still based on the one-to-one communication paradigm that does not really take the wireless broadcast advantage. Another category of works make cross-layer designs by designing source and channel coding jointly, e.g., SoftCast [3], ParCast [4], and FlexCast [5]. All these approaches re-design the PHY layer completely and thus require tremendous hardware implementation. Their deployment cost is prohibitively high.

Among them, one promising technology is HM (Hierarchical Modulation) [6]–[11]. It embeds different data streams, e.g., a high and a low priority stream, in the same transmission with different Modulation and Coding Schemes (MCS). High-quality users will receive both streams and decode high-quality images or videos. Meanwhile, low-quality users can still receive the high priority stream and enjoy some acceptable images or videos. HM provides a graceful modulation solution for diversified users, and thus can exploit the scarce spectrum resources efficiently.

Existing HM approaches, however, are neither implemented nor supported by modern wireless communication systems. This is partially because they are not compatible with the COTS (Commercial of the Shelf) Wi-Fi [12] or LTE devices [13]. As shown in Fig. 1(a), existing HM needs multiple encoders (decoders) and the a specially designed modulator (de-modulator) to embed different data streams. Current COTS Wi-Fi and LTE devices, however, only have one simple coding and modulation module. New hardware thus is needed for HM implementation, which is too costly for deployment as well.

To overcome these limitations, a fundamental yet open question is, "can HM be realized with existing coding and modulation module?" To answer this question well, in this

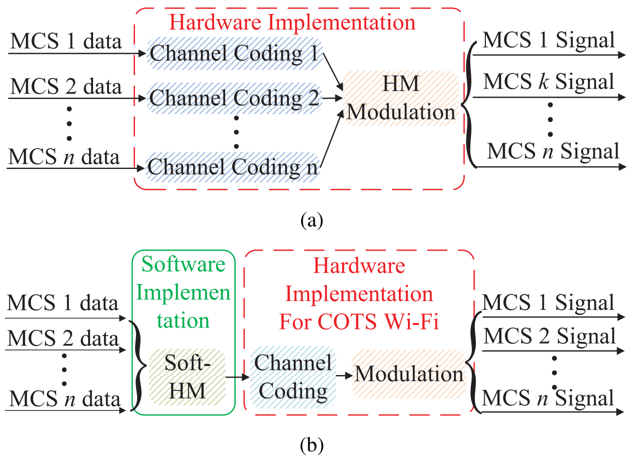


Fig. 1. An illustration of HM and SoftHM coding and modulation module implementation: (a).HM; (b). SoftHM. In SoftHM implementation, there is no hardware modification in the channel coding and modulation modules for COTS Wi-Fi devices.

paper, we design and implement SoftHM, a software-based HM scheme. As shown in Fig. 1(b), SoftHM works in an add-on mode. It is plugged in on top of the existing Wi-Fi or LTE PHY layer protocol stack. Via dedicating all efforts to manipulate the PHY payload of a packet, it can provide different data streams with different degrees of protection though all the data streams in the packet use the same MCS. Consequently, SoftHM requires no coding nor modulation module changes and can be easily extended to modern COTS Wi-Fi or LTE devices.

As a software based approach, SoftHM is highly flexible. The number of data streams and the MCS of each stream can be easily modified by passing a few parameters into SoftHM without any hardware changes. It also requires no channel state information and can work in a "fire and forget" manner. The transmitter simply transmits the multimedia content (e.g. an image) with a pre-defined MCS, and then it can go. Receivers will receive it with various MCSs based on their link qualities. Different users thus can obtain the same content with different versions. To summarize, main contributions are the following.

1) We propose a generic software based HM approach named as SoftHM. Different from traditional HMs, SoftHM does not require multiple coding modules or modulation modifications, instead it targets PHY layer payload design so that a single packet can be decoded gracefully by users with various channel quality. SoftHM thus is compatible to the coding and modulation modules in COTS Wi-Fi or LTE devices.
2) To realize SoftHM, we introduce an add-on module on top of current 802.11g PHY standards. To our best knowledge, the proposed paradigm is the first in literature that can enable this feature without modifying 802.11 PHY layer coding and modulation modules.
3) We conduct rigorous analysis on the performance of SoftHM. Theoretical results show that compared with the traditional one-to-one dedicated communication paradigms (i.e., low- or high-quality users are served in a time-sharing approach), to support low-quality users with 6Mbps, 9Mbps, 12Mbps or 18Mbps data rate (corresponds to BPSK $\frac{1}{2}$, BPSK $\frac{3}{4}$, QPSK $\frac{1}{2}$ or QPSK $\frac{3}{4}$ MCS), SoftHM can provide 700%, 466.7%, 250% or 166.7% higher throughput for high quality users respectively. Similarly, to support high-quality users with 42Mbps, 48Mbps or 51Mbps data rate (as can be achieved in SoftHM with QPSK $\frac{1}{2}$+64QAM $\frac{3}{4}$, QPSK $\frac{3}{4}$+64QAM $\frac{3}{4}$, BPSK $\frac{1}{2}$+64QAM $\frac{3}{4}$ or BPSK $\frac{3}{4}$+64QAM $\frac{3}{4}$ MCS combination), SoftHM can provide 350%, 800% or 1600% higher throughput for low quality users respectively.
4) We implement SoftHM based on a USRP (Universal Software Radio Peripheral) B210 [14] and Savvius Wi-Fi adapters with ominipeak [15] to evaluate its performances in real environments. USRP B210 functions as the single transmitter and the Wi-Fi adapters are independent receivers. JPEG images are employed as representatives of multimedia contents. Lab and hall experiment results show that to serve low quality users with the same bandwidth, high quality users in SoftHM achieve 616.7%, 383.3%, 187.5% or 111.1% (with BPSK $\frac{1}{2}$, BPSK $\frac{3}{4}$, QPSK $\frac{1}{2}$ or QPSK $\frac{3}{4}$ MCS) higher throughput than that of the time sharing approach. Similarly, to serve high quality users with the same bandwidth, low quality users in SoftHM achieve 390.9%, 414.3%, 176.9% or 237.5% higher throughput (with BPSK $\frac{1}{2}$, BPSK $\frac{3}{4}$, QPSK $\frac{1}{2}$ or QPSK $\frac{3}{4}$ MCS).

The rest of the paper is organized as follows. The background and motivations are discussed in Sec. II. Sec. III propose SoftHM paradigm with great details. Discussions are included in Sec. IV. Sec. V analyzes evaluation results. Literature reviews are provided in VI. In Sec. VII, conclusions and future work are drawn.

II. Background and Motivation

In this section, we provide some background knowledge and use a simple example to motivate our work.

A. 802.11g PHY Coding Procedure

Fig. 2 gives the block diagram of 802.11g coding and modulation modules, which shows how payloads are encoded and modulated in 802.11g protocol stack. Notice that our design is transparent to 802.11 based coding and modulation modules, requiring no changes to these modules, and thus it can be simply extended to any 802.11 legacy protocols such as 802.11n and 802.11ac. Likewise, since LTE system has similar coding and modulation structure, our design can be extended to LTE systems too. Our designs are mainly towards the Orthogonal Frequency Division Multiplexing (OFDM) mode [12] as it is more efficient and widely applied.

According to Fig. 2, given a PHY payload of binary sequence, say $\boldsymbol{I}_0$ ("101101" in Fig. 2), it is firstly randomized by a scrambler to avoid the occurrence of a long period of consecutive "0" or "1" bits. The scrambling output $\boldsymbol{I}_S$ ("110100" in Fig. 2) is then encoded with the Block Convolution Code (BCC) to provide Forward Error Correction (FEC)

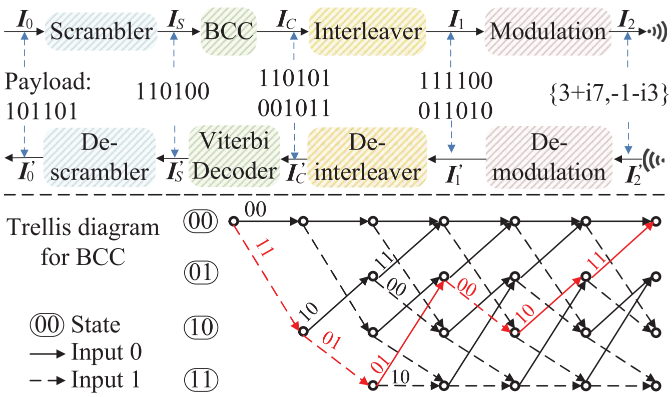


Fig. 2. 802.11g PHY layer coding and modulation procedure.

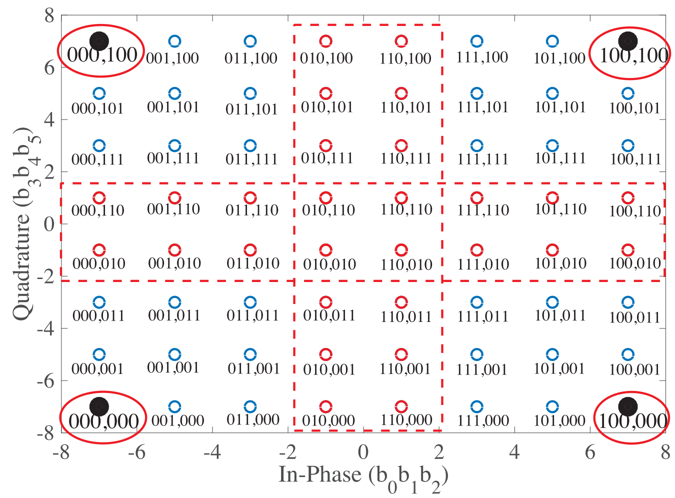


Fig. 3. Constellation map for 64QAM modulated signal. The black dots in four corners refer to normal QPSK modulated signals.

capabilities. Different coding rates are supported with different FEC capabilities such as $\frac{1}{2}$, $\frac{2}{3}$, and $\frac{3}{4}$. In Fig. 2, we assume $\frac{1}{2}$ coding rate is applied, where the new sequence $\boldsymbol{I}_C$ becomes 12 bits of "110101001011", i.e., one redundant bit after every single bit. The new $\boldsymbol{I}_C$ is then interleaved to different sub-carriers, and $\boldsymbol{I}_1$ is obtained ("111100011010" in the Figure). This is to guarantee that once a sub-carrier fails, only interleaved but not a block of bits are lost. And these interleaved lost bits can be recovered by BCC. After interleaving, each sub-carrier will have a subset of the data to transmit. This subset of data are then mapped to real numbers and converted to signals according to the modulation scheme. For example, when 64QAM modulation scheme is applied, 6 consecutive bits are mapped into a signal, three for real and three for imaginary part. The 64QAM modulation constellation diagram is shown in Fig. 3. Suppose sub-carrier 1 and 2 has the bit stream "111,100" and "011,010" respectively. The bit stream will map them to the complex number 3+i7, and $-1-$i3 (refer to the constellation diagram Fig. 3), and load them to sub-carrier 1 and 2 accordingly. For QPSK modulation, two bits are modulated at a time, one in the real and the other in the imaginary part of the signal. Refer to Fig. 3, QPSK modulated signals correspond to the four dark spots in the diagram.

The receiver takes a reverse procedure of above. It demodulates the received signals, maps them into "0" and "1" bits, de-interleaves and decodes the BCC encoded bit stream (with Viterbi decoder). It then de-scrambles the data to obtain the PHY payload $\boldsymbol{I}_0'$. Ideally, $\boldsymbol{I}_0'$ equals to $\boldsymbol{I}_0$.

B. Traditional Hierarchical Modulation Scheme

Traditional HM (Hierarchical Modulation) uses multiple MCSs (Coding and Modulation Schemes) in the same packet. Different receivers can then demodulate and recover the desired data with different MCSs. When jointly designed with progressive image or scalable video encoders [16], HM is able to provide different levels of resolution of the same image or video to users. In particular, given the system requires two data streams, it first separates the data into high- and low-priority streams. The high priority stream is modulated with a low MCS to enhance its reliability. The low priority stream is modulated with a high MCS. Notice that, the modulations schemes used in HM and the proposed SoftHM framework, two prerequisites should be met.

1) There should be both coarse- and fine-grained modulation schemes used in HM (or the proposed SoftHM), and different modulation schemes carry different amount of information;
2) Modulation schemes used in HM (or SoftHM) are not orthogonal. In other words, different demodulation schemes can demodulate the same signal with meaningful though different amount of information.

Since QAM modulation schemes, such as QPSK, 16QAM, 64QAM and etc, can easily meet the above two requirements, and they are commonly used in modern WLAN and LTE networks, QAM modulation schemes are adopted for both HM and SoftHM framework. However, as will be discussed in Sec. IV-D, other modulation schemes can also be applied to HM and the proposed SoftHM scheme.

To implement the traditional HM, the high- and low-priority streams need to be encoded and modulated separately. Moreover, to guarantee the performance of the higher priority stream (lower modulated stream), a hierarchical constellation diagram is required for modulating two streams. Refer to Fig. 4(a), different from an ordinary 16QAM modulation scheme, the distance between different points are unequal. For instance, the four points in the left lower corner locate close together and form a group. There are four groups of points in the figure, with the minimum inner group distance denoted as d_1. The points in a group are treated as the same point for the high priority stream., i.e., in the left lower group, they are represented as (0,0) (the first bit in the real and imaginary part of the constellation map). These points also carry information for the low priority stream via the second bit in the real and imaginary part, i.e., the points in the left lower group are represented as (0,1), (1,1), (0,0) and (1,0). The minimum inner group distance d_1 is the minimum distance between two modulated signals for the low priority stream data. Likewise, as different group of points carry different information for the higher priority stream, the minimum inter

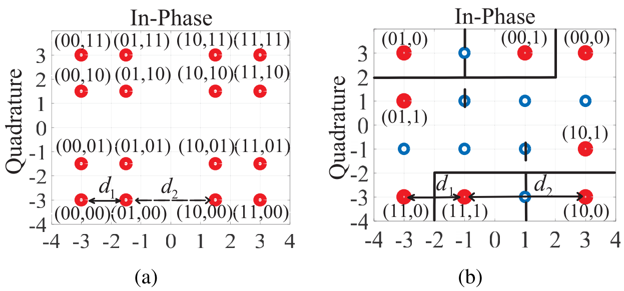


Fig. 4. An illustration of the constellation diagram of HM and its extension scalable modulation. (a).HM; (b). Scalable modulation. The blue dots in the diagram are not used for mapping bit streams.

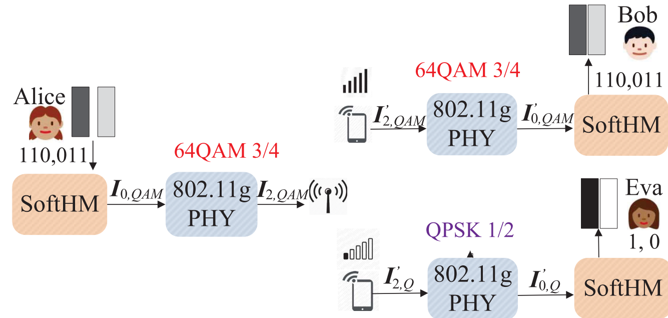


Fig. 5. An example as how SoftHM works for different users. Here $\boldsymbol{I}_{0,QAM}$ = "101, 101", $\boldsymbol{I}'_{0,QAM}$ = "101, 101", $\boldsymbol{I}'_{0,Q}$ = "0, 0", $\boldsymbol{I}_{1,QAM}$ = "111, 100, 011, 010", $\boldsymbol{I}'_{1,Q}$ = "1, 1, 0, 0", $\boldsymbol{I}'_{2,QAM}$ = $\{3+i7, -1-i3\}$, $\boldsymbol{I}'_{2,Q} = \{1+i1, 1-i1\}$.

group distance, denoted as d_2, implies the minimum distance between two modulated signal for high priority stream data. The ratio $\frac{d_1}{d_2}$ can be used to control the trade-off between the error probability of the QPSK and 16QAM signal. E.g., the lower the ratio $\frac{d_1}{d_2}$ ($\frac{d_1}{d_2} \leq 1$), the lower the bit error rate for the high priority stream signal, and the higher the bit error rate for the low priority stream signal.

To remove the complexity of implementing a specialized modulation module (with a special constellation diagram), existing modulation has been discussed in prior work [9] for hierarchically embedding different data streams in the same transmission. In this category of schemes, the different streams are modulated jointly but differently based on their priority. Different degrees of protection for different data streams are obtained by redesigning the constellation mapping strategy of the existing QAM modulation (e.g., 16QAM or 64QAM). E.g., refer to Fig. 4(b), a new constellation map is designed for scalable modulation [9], an extension of the traditional HM. In scalable modulation, only part of the 16QAM points (red dots in the figure) are used for jointly mapping two encoded data streams. In this way, 16QAM can only be used for carrying 3 bits of information (with 50% capacity loss for the lower priority stream). Recall that traditional 16QAM signals can carry 4 bits of information in each symbol. With the new mapping, two points are grouped together, and there are 4 groups of points. For example, points (11,0) and (11,1) belong to the same group. Similar to traditional HM, different points in the same group are used to carry information for the low priority stream. The minimum inner group distance is denoted as d_1. Information for high priority stream is embedded with different groups, with the minimum inter group distance dented as d_2. For example, in Fig. 4(b), the ratio $\frac{d_1}{d_2}$ is 0.5.

In summary, there are two major practical issues for implementing traditional HM based approaches. Firstly, different data streams are encoded (decoded) separately, which requires multiple hardware coding modules. Recall that 802.11g or LTE PHY standard only support one coding module for a packet. Secondly, either a brand-new modulator or constellation mapping modifications for existing modulation scheme is required.

C. Motivation Example

In this part, we will use an illustrative example in Fig. 5 to show the basic idea of SoftHM. Suppose we are transmitting the color information of two blocks, one is dark gray which can be represented as "110" and the other is light gray "011". The transmitter Alice simply feeds the signals of 6 consecutive bits "110011" to SoftHM. SoftHM will transform the bit sequence into "101101" (denote as $\boldsymbol{I}_{0,QAM}$) as the PHY payload bits. After 64QAM $\frac{3}{4}$ MCS, complex number "3+i7", and "$-1-$i3" (denote as $\boldsymbol{I}'_{2,QAM}$) will be loaded to sub-carrier 1 and 2 respectively.

The high-quality user Bob has a sufficient Signal-to-Noise Ratio (SNR) and will use 64QAM $\frac{3}{4}$ MCS to demodulate signals and retrieve the PHY payload bit sequence "101101" ($\boldsymbol{I}'_{0,QAM}$). For the low-quality user Eva, her SNR is not strong enough to demodulate the 64QAM signals. Nevertheless, Eva's SNR is still able to demodulate QPSK $\frac{3}{4}$ signals and just use QPSK to demodulate the signal as "00" (denoted as $\boldsymbol{I}'_{0,Q}$). By passing $\boldsymbol{I}'_{0,Q}$ to SoftHM decoding module, Eva can recover the desired signal "10". With a single SoftHM Bob and Eva both receive something meaningful. In this case, Bob has the finer color information of 3-bit colors, while Eva is still OK with 1 bit colors, black and white.

D. Design Insight

Though the proposed SoftHM shares the similar goal of embedding multiple data streams in a single transmission as done in HM, it is fundamentally different from the traditional HM designs. To be more specific, SoftHM does not need multiple modulation and coding modules to load different data streams to the transmitting signals directly. Instead, an add-on encoding/decoding module is plugged in before/after PHY layer. The add-on module is implemented in a software-based approach, and employs the reverse engineering technique so that by manipulating the PHY layer payload, different data streams can be loaded to different set of bits in the transmitting signals accordingly. E.g., refer to Fig. 3, high quality stream is loaded to the second and third significant bits of the constellation diagram ("b_1 b_2" or "b_4 b_5" with higher bit error rate); low quality stream is loaded to the first significant bits of the constellation diagram ("b_0" or "b_3" with lower bit error rate). This is the key factor to enable transmitting multiple streams in a packet while leaving the hardware implementation intact.

E. Design Challenges

In a COTS Wi-Fi device, all coding and modulation modules, from the scramblers to the modulator, are implemented in the hardware and fixed when they are manufactured in the factory. In other words, we can only control the PHY payload $\boldsymbol{I}_0$ and get $\boldsymbol{I}'_0$ back. These hardware limitations bring a lot of practical challenges.

The first challenge is how to embed different data streams in the transmitted frequency domain signal $\boldsymbol{I}_{2,QAM}$. In contrary to traditional HM schemes, SoftHM is constrained from encoding, modulating different streams differently and merging them into a single PHY layer signal. Nevertheless, if a lower MCS is used to decode a higher MCS packet directly chances are that the decoded payload is far from the desired payload. According to Fig. 5, suppose the transmitter Alice simply passes the payload (“110,011”) to the 802.11g PHY without SoftHM processing. Since the transmitting bits $\boldsymbol{I}_{1,QAM}$ is encoded exclusively for 64QAM $\frac{3}{4}$ MCS, its QPSK version $\boldsymbol{I}'_{1,Q}$, however, is not encoded with QPSK $\frac{3}{4}$ MCS. Of course, Eva cannot expect to receive a correct payload $\boldsymbol{I}_{0,Q}$ by decoding $\boldsymbol{I}'_{1,Q}$ with QPSK $\frac{3}{4}$ MCS.

Moreover, even if $\boldsymbol{I}_{2,QAM}$ and $\boldsymbol{I}_{1,QAM}$ have been designed correctly, how to derive the real payload $\boldsymbol{I}_{0,QAM}$ is not an easy task. Recall that we have no access to the data in intermediate components in 802.11g PHY standards, and there is no simple connection between the original payload $\boldsymbol{I}_{0,QAM}$ and the transmitted bits $\boldsymbol{I}_{1,QAM}$. In the example in Fig. 2, the original payload $\boldsymbol{I}_{0,QAM}$ is “101101”, while after scrambling, BCC encoding, and interleaving, the transmitted signals $\boldsymbol{I}_{1,QAM}$ is “111100011010”. The original payload are very different from the transmitted ones.

The second challenge is that as we are not able to modify the constellation map of the modulator. It is thus impossible to adopt different minimum distances for different layers of information as has been done in HM (refer to Fig. 4). This makes it very difficult to protect the lower modulated signal in some cases. For instance, if the PHY payload is modulated with 64QAM, some signals are likely to be mapped to points within the red cross area shown in Fig. 3. Since those points are close to the original point, for the lower modulated signal (e.g., QPSK signal), they are error prone, resulting in high bit error rate.

All these challenges will be addressed in our SoftHM design, which will be detailed in Sec. III.

III. SoftHM Design

In this section, we present the detailed design of SoftHM. For ease of presentation, we suppose two kinds of users are to be served throughout this section, the low-quality users that can support QPSK $\frac{1}{2}$ MCS, and the high-quality users that can support 64QAM $\frac{3}{4}$ MCS. Our designs are generic that can support other combinations of MCS with similar principles (refer to IV for the implementation of other MCS combinations).

A. SoftHM Overview

Fig. 6 depicts the module diagram of SoftHM. It adds an encoding module before $\boldsymbol{I}_0$ and a decoding module after $\boldsymbol{I}'_0$ so

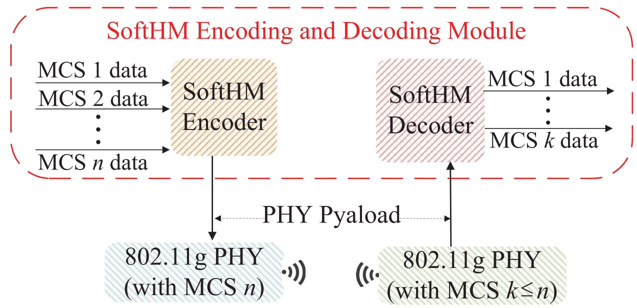


Fig. 6. The proposed SoftHM Architecture. Here MCS n data stream is the data stream with the highest MCS embedded in the packet. A transmitter uses the highest MCS (MCS n) to encode and modulate the packet. Different receivers can then use different MCS k (with $k \leq n$) to extract different data streams.

that the payloads, $\boldsymbol{I}_0$ and $\boldsymbol{I}'_0$, can be manipulated. As we aim at manipulating PHY payload, all modules in 802.11g standards are left unchanged.

Suppose we have two versions of an image. One has a low resolution and is designed for QPSK low-quality users, which we call QPSK users and QPSK streams. The other has a high resolution and is for 64QAM high-quality users, which we call 64QAM users and 64QAM streams. The two streams of payloads are independent to each other. These payloads of two streams are fed into SoftHM encoder, one from each, and are merged as a single payload $\boldsymbol{I}_0$. $\boldsymbol{I}_0$ will pass through the 802.11g protocol stack, be transformed as $\boldsymbol{I}_1$, modulated and transmitted as the frequency domain signals $\boldsymbol{I}_2$. The low-quality user will identify these signals as QPSK ones, while the high-quality users will identify them as 64QAM ones. They will process the signals by their own 802.11 protocol stack, and obtain the two different streams respectively.

We also assume the MCSs used by different data streams are fixed and known at both the transmitter and the receivers. The receiver, after receiving the packet, uses the high rate to decode the packet first. If the high rate packet can be decoded successfully, the information will be retrieved accordingly. Otherwise, the receiver will use the low rate to decode the packet. Moreover sophisticated rate mechanism, such as rate adaptation, can also be supported by introducing some infrequent feedback from receivers to the transmitter. For instance, each user can count the number of high and low rate packets it has received for a period of time, and report this information to the transmitter from time to time. The transmitter will then decide whether to increase, decrease or keep the same rate based on such information. The details about how to facilitate such adaptation will be included in our future work.

Moreover, throughout this section, as $\boldsymbol{I}_2$ can be derived directly given I_1, we only discuss how to establish the connection between $\boldsymbol{I}_1$ and $\boldsymbol{I}_0$. In the above process, there are three key issues to deal with.

- Given an $\boldsymbol{I}_1$ binary sequence, how to modulate the signals so that both QPSK and 64QAM can be used to demodulate it? We will address this issue in Sec. III-B.
- Given two separate streams of data, how to derive a single payload $\boldsymbol{I}_0$ such that after 802.11g protocol stack,

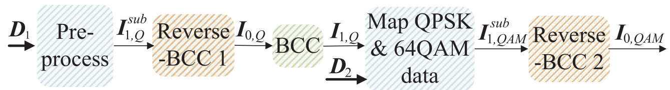


Fig. 7. Pre-coding implementation for SoftHM. Here $\boldsymbol{D}_1$ and $\boldsymbol{D}_2$ represent the QPSK $\frac{1}{2}$ and 64QAM $\frac{3}{4}$ MCS data.

the corresponding $\boldsymbol{I}_1$ still carry the information of both? We address this issue in Sec. III-C.

- How to provide better protection for the low-quality users, and enhance their noise-resistance capability will be presented in Sec. III-D.

B. Modulation Scheme

Fig. 3 shows the constellation map for 64QAM modulated signals. Six bits, denoted as $b_0b_1b_2b_3b_4b_5$ are called one symbol and will be transmitted as a unit via one sub-carrier at a time. In traditional QPSK modulation, payloads are organized with symbols of 2 bits: "00", "01", "10" and "11". From the constellation point of view, "00", "01", "10" and "11" corresponds to "000000", "000100", "100000" and "100100" in 64QAM modulation (labeled as black dots).

In our design, 64QAM modulation is the same as the original, while the QPSK modulation is different. All the 16 points in a quadrant from the 64QAM constellation diagram is considered as one valid symbol in QPSK modulation. For example, besides "000100", all symbols of "0**1**" (e.g., "001100" and "011101") are considered as the QPSK symbol "01". Notice that, when QPSK demodulator is used, these other symbols than "000100" will also be demodulated and decoded by the traditional 802.11g protocols and they are simply considered as a noisy version of "000100". To avoid the dilemma discussed in Sec.II-E, the key feature here is that all higher MCS signals can be decoded by lower MCS decoder with meaningful information. This key feature is obtained by the following designs.

C. Distributing Two Data Streams to the PHY Payload of the Packet ($\boldsymbol{I}_{0,QAM}$)

The pre-coding paradigm is shown in Fig. 7. In this subsection each component in the figure will be discussed with greater details. Notations are listed in Table I.

*1) **"Pre-Process QPSK Data"**:* This module is used to load QPSK signals to a set of QPSK sub-carriers. Refer to Fig. 2, let $n_{0,Q}$ and $n_{1,Q}$ be the number of bits in PHY payload and after interleaving for one QPSK $\frac{1}{2}$ encoded Wi-Fi symbol. Here a Wi-Fi symbol refers to the bit stream loaded to all of the data sub-carriers.

According to 802.11g standards, bit streams will be encoded symbol by symbol. Hence in the rest of this subsection, we will show how to determine the payload of a Wi-Fi symbol. Since there are 48 data sub-carriers in 802.11g, for QPSK $\frac{1}{2}$ MCS, $n_{0,Q} = 48$ and $n_{1,Q} = 96$. Let $\boldsymbol{I}_{0,Q}$ and $\boldsymbol{I}_{1,Q}$ be the original and the encoded PHY payload. As $n_{1,Q} < n_{0,Q}$, we can only find $n_{0,Q}$ linearly independent bits from $\boldsymbol{I}_{1,Q}$ (denoted as $\bar{\boldsymbol{I}}_{1,Q}$). The $n_{0,Q}$ linearly independent bits are selected from $n_{0,Q}$ independent rows in $\boldsymbol{I}_{1,Q}$. With $\bar{\boldsymbol{I}}_{1,Q}$, we can easily map the QPSK data (the input, denoted as $\boldsymbol{D}_1$, in Fig.2) to $\bar{\boldsymbol{I}}_{1,Q}$.

TABLE I

NOTATIONS

$n_{0,QAM}/n_{0,Q}$	The number of bits in PHY payload for one 64QAM / QPSK modulated Wi-Fi symbol.
$n_{1,QAM}/n_{1,Q}$	The number of bits after interleaving for one 64QAM / QPSK modulated Wi-Fi symbol.
$\boldsymbol{I}_{0,QAM}/\boldsymbol{I}_{0,Q}$	The input/ output PHY payload of one 64QAM / QPSK modulated Wi-Fi symbol.
$\boldsymbol{I}_{S,QAM}/\boldsymbol{I}_{S,Q}$	The scrambled PHY payload of one 64QAM / QPSK modulated Wi-Fi symbol.
$\boldsymbol{I}_{1,QAM}/\boldsymbol{I}_{1,Q}$	The BCC encoded output of one 64QAM /QPSK modulated Wi-Fi symbol.
$\bar{\boldsymbol{I}}_{1,Q}/\bar{\boldsymbol{I}}_{1,QAM}$	A subset of $n_{0,Q}/n_{0,QAM}$ independent bits from $\boldsymbol{I}_{1,Q}/\boldsymbol{I}_{1,QAM}$.
$\boldsymbol{M}_{S,Q}/\boldsymbol{M}_{S,QAM}$	The scrambling vector of one 64QAM / QPSK modulated Wi-Fi symbol.
$\boldsymbol{M}_{C,Q}/\boldsymbol{M}_{C,QAM}$	The BCC encoding matrix of one 64QAM / QPSK modulated Wi-Fi symbol.
$\boldsymbol{M}_{I,Q}/\boldsymbol{M}_{I,QAM}$	The interleaving matrix of one 64QAM / QPSK modulated Wi-Fi symbol.
n_g	The number of guarding bits.

*2) **"BCC" and "Reverse-BCC 1"**:* These modules used to determine the PHY encoded payload $\boldsymbol{I}$. We first introduce the following notations.

- $\boldsymbol{M}_{S,Q}$: the scrambling vector of one QPSK modulated Wi-Fi symbol. $\boldsymbol{M}_{S,Q}$ is a $n_{0,Q} \times 1$ vector;
- $\boldsymbol{M}_{C,Q}$: the BCC encoding matrix of one QPSK modulated Wi-Fi symbol. $\boldsymbol{M}_{C,Q}$ is a $n_{1,Q} \times n_{0,Q}$ matrix with rank equals $n_{0,Q}$;
- $\bar{\boldsymbol{M}}_{C,Q}$: a submatrix of $\boldsymbol{M}_{C,Q}$ composed by $n_{0,Q}$ independent rows denoted as $\boldsymbol{R}_Q$, from $\boldsymbol{M}_{C,Q}$. Hence $\bar{\boldsymbol{M}}_{C,Q}$ is a $n_{0,Q} \times n_{0,Q}$ full rank matrix;
- $\boldsymbol{M}_{I,Q}$: The interleaving matrix of one QPSK modulated Wi-Fi symbol. $\boldsymbol{M}_{I,Q}$ is a $n_{1,Q} \times n_{1,Q}$ full rank matrix;
- $\bar{\boldsymbol{M}}_{I,Q}$: a submatrix of $\boldsymbol{M}_{I,Q}$ composed by columns selected from $\boldsymbol{M}_{I,Q}$, and the corresponding rows that have one non-zeros entry. $\bar{\boldsymbol{M}}_{I,Q}$ is a $n_{0,Q} \times n_{0,Q}$ full rank matrix.

Since $\bar{\boldsymbol{M}}_{C,Q}$ and $\bar{\boldsymbol{M}}_{I,Q}$ have $n_{0,Q}$ independent rows, we have

$$\boldsymbol{I}_{S,Q} = \boldsymbol{I}_{0,Q} \oplus \boldsymbol{M}_{S,Q} \tag{1a}$$
$$\bar{\boldsymbol{I}}_{1,Q} = \bar{\boldsymbol{M}}_{I,Q} \otimes \bar{\boldsymbol{M}}_{C,Q} \otimes \boldsymbol{I}_{S,Q} \tag{1b}$$

Here $\otimes$ and $\oplus$ are the multiplication and addition operation in Galois Field. Moreover, (1a) is the scrambling operation. With some manipulation of (1a-1b), $\boldsymbol{I}_{0,Q}$ and $\boldsymbol{I}_{1,Q}$ can be derived as

$$\boldsymbol{I}_{0,Q} = (\bar{\boldsymbol{M}}_{C,Q}^{-1} \otimes \bar{\boldsymbol{M}}_{I,Q}^{-1} \otimes \bar{\boldsymbol{I}}_{1,Q}) \oplus \boldsymbol{M}_{S,Q}; \tag{2a}$$
$$\boldsymbol{I}_{1,Q} = \boldsymbol{M}_{I,Q} \otimes \boldsymbol{M}_{C,Q} \otimes (\boldsymbol{I}_{0,Q} \oplus \boldsymbol{M}_{S,Q}). \tag{2b}$$

Equations (1b-2a) refers to the "Reverse-BCC 1" operation, and (2b) refers to "BCC" operation in Fig. 7.

*3) **"Map QPSK and 64QAM Data"**:* Let $n_{0,QAM}$ and $n_{1,QAM}$ be the number of bits in PHY payload and after interleaving for one 64QAM $\frac{3}{4}$ encoded Wi-Fi symbol. Let $\boldsymbol{I}_{0,QAM}$ and $\boldsymbol{I}_{1,QAM}$ be the original and the encoded PHY payload. Hence $\boldsymbol{I}_{0,QAM}$ and $\boldsymbol{I}_{1,QAM}$ are a $n_{0,QAM} \times 1$ and a $n_{1,QAM} \times 1$ vector respectively. E.g., for 64QAM $\frac{3}{4}$ MCS,

$n_{0,QAM} = 216$ and $n_{1,QAM} = 288$. Moreover, $n_{1,QAM} = 3n_{1,Q}$.

Recall that to demodulate a 64QAM signal as a QPSK signal, for each sub-carrier the receiver will decode 6 bits (denoted as $b_0\ b_1\ b_2\ b_3\ b_4\ b_5$) as 2 bits ($b_0b_3$). Let $\boldsymbol{I}_{1,QAM}(i)$ and $\boldsymbol{I}_{1,Q}(j)$ be the ith bit of $\boldsymbol{I}_{1,QAM}$ and $\boldsymbol{I}_{1,Q}$ respectively. A receiver can decode a 64QAM packet with QPSK $\frac{1}{2}$ MCS correctly only if

$$\boldsymbol{I}_{1,QAM}(3i) = \boldsymbol{I}_{1,Q}(i), \quad \forall 0 \leq i \leq n_{1,Q} - 1. \tag{3}$$

As to any $i \in [0, n_{1,Q}]$, $\boldsymbol{I}_{1,QAM}(3i+1)$ and $\boldsymbol{I}_{1,QAM}(3i+2)$ can be either 0 or 1. This provides the possibility to embed more information in a 64QAM MCS packet.

In particular, as it takes $n_{1,Q}$ bits to load QPSK $\frac{1}{2}$ MCS to a 64QAM $\frac{3}{4}$ MCS packet, and there are $n_{0,QAM}$ available bits from $\boldsymbol{I}_{1,QAM}$, we can use the remaining $n_{0,QAM} - n_{1,Q}$ bits to carry data for the 64QAM $\frac{3}{4}$ stream. To this end, we find $n_{0,QAM}$ independent rows from $M_{C,QAM}$ (denoted as $\boldsymbol{R}_{QAM}$), where $M_{C,QAM}$ is the BCC encoding matrix used for 64QAM $\frac{3}{4}$ MCS. To satisfy (3), for any $i \in [0,]n_{1,Q}-1], 3i \in \boldsymbol{R}_{QAM}$. The QPSK and 64QAM stream data is then mapped to $\bar{\boldsymbol{I}}_{1,QAM}$ accordingly.

*4) **"Reverse-BCC 2":*** The 64QAM payload $I_{0,QAM}$ is derived via Reverse-BCC 2. Reverse-BCC 2 can either use a similar component shown in "Reverse-BCC 1" or adopt the robust coding component to be discussed in Sec.III-D.

*5) **Decoding Procedure:*** Since all the useful information are loaded directly to $I_{1,Q}$ (for QPSK $\frac{1}{2}$ MCS) or $I_{1,QAM}$ (for 64QAM $\frac{3}{4}$ MCS) and they are not accessible directly, after decoding the PHY payload, receivers need to redo the encoding procedure to retrieve the data from $I_{1,Q}^{sub}$ or $I_{1,QAM}^{sub}$.

D. Improved Robustness for the Lower Modulated Stream

This subsection discusses algorithms to improve the robustness of the QPSK stream. It can be considered as an advanced design for "Reverse-BCC 2" component in Fig. 7. Refer to Fig. 3, a normal QPSK signal is transmitted with the points in the four corners highlighted with red circles, and the normalized minimum distance between two QPSK points is 2. However, with SoftHM, it is possible that the PHY payload is encoded with the points in the red cross area (the red circles in Fig. 3, denoted as error-prone points), e.g., "010010" and "010110". In this case, the normalized minimum distance between the two QPSK points is $\frac{2}{\sqrt{21}}$. Obviously, compared to a normal QPSK signal, low-quality users in SoftHM are much more prone to be ruined by noise. To address this issue, the basic idea of the advanced algorithm is to **prevent the error-prone points from being chosen**.

In this regard, we introduce n_g guarding bits, and use only $n_{0,QAM} - n_{1,Q} - n_g$ bits to carry data for 64QAM stream in each Wi-Fi symbol. Recall that, 64QAM stream can carry $n_{0,QAM} - n_{1,Q}$ bits of information in each symbol without guarding bits. By sacrificing n_g bits for data transmission, n_g degrees of freedom can be exploited for avoiding choosing the points in the red cross area. Hence, given $\bar{\boldsymbol{I}}_{1,QAM}$, we will focus on finding $\boldsymbol{I}_{0,QAM}$ with a pre-determined n_g so that the number of error-prone points is minimized.

Before delving into the details, we first take a second look at BCC. BCC encoder can either be represented with matrix multiplication (as has been discussed previously) or be considered as a state machine [17]. Regarding to the state machine view, i.e., as shown in Fig. 2, a state contains 2 bits, and there are 4 different states: "00", "01", "10", "11". If the input is $\{110100\}$ and the initial state is "00", with $\frac{1}{2}$ coding rate, the output will be $\{11, 01, 01, 00, 10, 11\}$ (the red path in Fig. 2). The state machine view will be used to design the advanced version of the "Reverse BBC 2" module.

In detail, we first manipulate the PHY payload a little bit, and define $\boldsymbol{I}'_{0,QAM}$ and $\boldsymbol{I}'_{1,QAM}$ as:

$$\boldsymbol{I}'_{0,QAM} = \boldsymbol{I}_{0,QAM} \oplus \boldsymbol{M}_{S,QAM}; \tag{4a}$$
$$\boldsymbol{I}'_{1,QAM} = \boldsymbol{M}_{I,QAM} \otimes \boldsymbol{I}_{1,QAM}. \tag{4b}$$

Here $\boldsymbol{M}_{S,QAM}$ and $\boldsymbol{M}_{I,QAM}$ are the scrambling vector and the interleaving matrix for 64QAM $\frac{3}{4}$ MCS packet respectively.

According to the BCC module in 802.11g PHY standard, the output of the kth bit in $\boldsymbol{I}'_{1,QAM}$ (denoted as $\boldsymbol{I}'_{1,QAM}(k)$) only depends on the set of bits $\{\boldsymbol{I}'_{0,QAM}(f(k)-6), \cdots, \boldsymbol{I}'_{0,QAM}(f(k))\}$, but not any bit before $f(k)-6$ or after $f(k)$, where $f(k)$ is a function determined by the coding rate of the MCS. For $\frac{3}{4}$ coding rate, $f(k)$ is

$$f(k) = \lfloor \frac{k}{4} \rfloor * 3 + \begin{cases} 1, & k \mod 4 = 1, 2; \\ 2, & k \mod 4 = 3; \\ 0, & k \mod 4 = 0. \end{cases} \tag{5}$$

The operation $\lfloor x \rfloor$ takes the integer part of x. The memoryless property of BCC provides an opportunity to design a dynamic programming algorithm for minimizing the number of error-prone points. The algorithm works in the following way. Denote

- $\bar{\boldsymbol{I}}'_{1,QAM}$: the set of bits in $\boldsymbol{I}'_{1,QAM}$ that will carry either QPSK $\frac{1}{2}$ (QPSK $\frac{3}{4}$) or 64QAM $\frac{3}{4}$ data. Note that $\bar{\boldsymbol{I}}'_{1,QAM} \subset \boldsymbol{I}'_{1,QAM}$ and $|\bar{\boldsymbol{I}}'_{1,QAM}| = |\boldsymbol{I}'_{1,QAM}| - n_g$ where $|\bar{\boldsymbol{I}}'_{1,QAM}|$ is the cardinality of $\bar{\boldsymbol{I}}'_{1,QAM}$.
- $\boldsymbol{S}_{f(k)} = \{\boldsymbol{I}'_{0,QAM}(f(k)-6), \cdots, \boldsymbol{I}'_{0,QAM}(f(k)-1)\}$: the state at the $f(k)$th input.
- $\boldsymbol{V}_k$: a set of bit positions whose values will affect error-prone points generated after input bit $f(k)$.
- $h(k)$ and $h^{-1}(k)$: the mapping from the kth bit in $\boldsymbol{I}_{1,QAM}$ to the $h(k)$th bit in $\boldsymbol{I}'_{1,QAM}$, and the reverse mapping from $\boldsymbol{I}'_{1,QAM}$ to $\boldsymbol{I}_{1,QAM}$. For instance, if $h(3k'+2) > k > h(3k'+1)$, and $3k'+1 \notin \boldsymbol{I}'_{1,QAM}$, $h(3k'+1)$ belongs to $\boldsymbol{V}_k$.
- $\boldsymbol{U}(\boldsymbol{V}_k)$: the set of values of all the bits in $\boldsymbol{V}_k$.

When the kth output bit is generated, according to Fig. 3, an error-prone point appears only when $b_1 = 1$ and $b_2 = 0$; or $b_4 = 1$ and $b_5 = 0$. That is

$$y_k = 1, y_{k_1} = 0, h(k) = 3k' + 1, h(k_1) = h(k) + 1; \text{or} \tag{6a}$$
$$y_k = 0, y_{k_1} = 1, h(k) = 3k' + 2, h(k_1) = h(k) - 1. \tag{6b}$$

Here y_k is the output bit in $\boldsymbol{I}_{1,QAM}$. Designate d_k as the desired value to be loaded to the kth bit in $\boldsymbol{I}_{1,QAM}$. We then introduce a penalty function $C^{\boldsymbol{U}(V_k)}_{\boldsymbol{S}_{f(k)}}$ as follows

$$C^{\boldsymbol{U}(V_k)}_{\boldsymbol{S}_{f(k+1)}} = \begin{cases} C^{\boldsymbol{U}(\boldsymbol{V}_{k-1})}_{\boldsymbol{S}_{f(k)}} + 1, & (6) \text{ and } \\ & T(\boldsymbol{S}_{f(k)}, \boldsymbol{S}_{f(k+1)}) = 1; \\ \infty, & d_k \neq y_k \text{ or } \\ & T(\boldsymbol{S}_{f(k)}, \boldsymbol{S}_{f(k+1)}) = 0; \\ C^{\boldsymbol{U}(\boldsymbol{V}_{k-1})}_{\boldsymbol{S}_{f(k)}}, & \text{otherwise.} \end{cases} \tag{7}$$

In (7), $T(\boldsymbol{S}_{f(k)}, \boldsymbol{S}_{f(k+1)}) = 1$ if state $\boldsymbol{S}_{f(k)}$ can be transferred to state $\boldsymbol{S}_{f(k+1)}$ in trellis diagram and $T(\boldsymbol{S}_{f(k)}, \boldsymbol{S}_{f(k+1)}) = 0$ otherwise. Equation (7) represents the number of error-prone points generated so far when the first $f(k)$ input bits have been encoded. $C^{\boldsymbol{U}(\boldsymbol{V}_k)}_{\boldsymbol{S}_{f(k+1)}} = \infty$ when there is any mismatch between the encoded output bit and the information to be carried on that bit (state $\boldsymbol{S}_{f(k+1)}$ cannot be reached from state $\boldsymbol{S}_{f(k)}$). The payload ($\boldsymbol{I}'_{0,QAM}$) is then designed to minimize (7). Notice that $C^{\boldsymbol{U}(\boldsymbol{V}_k)}_{\boldsymbol{S}_{x_{f(k)}}}$ depends on $\boldsymbol{S}_{x_{f(k+1)}}$ and $\boldsymbol{U}(\boldsymbol{V}_k)$, but not any other bits before $\boldsymbol{I}'_{0,QAM}(f(k-6))$ or after $\boldsymbol{I}'_{0,QAM}(f(k))$. Hence, $C^{\boldsymbol{U}(\boldsymbol{V}_k)}_{\boldsymbol{S}_{f(k+1)}}$ can be derived easily.

E. System Performance Analysis

We analyze the performance of SoftHM here. We first let r^H_0 and r^L_0 be the bandwidth of the higher and lower MCS (e.g., 64QAM, and QPSK); r_H and r_L be the effective bandwidth of a higher and lower MCS stream; and $\hat{r}_H = r_L + r_H$ be the overall effective bandwidth of the high MCS user.

1) ***The Effective Bandwidth of SoftHM****:* With 64QAM and QPSK MCS combination, r_H and r_L becomes

$$r_H = r^H_0 * \frac{(n_{0,QAM} - n_{1,Q} - n_g)}{n_{0,QAM}}; \tag{8a}$$

$$r_L = r^L_0. \tag{8b}$$

The effective bandwidth (r_H, r_L and $\hat{r}_H$) of different MCS combinations are shown in Table II. In summary, compared to the traditional time sharing approach, e.g., only low- or high-quality users can be served at one time, to support low quality users with the same bandwidth (BPSK $\frac{1}{2}$ MCS for 6Mbps data rate, BPSK $\frac{3}{4}$ MCS for 9Mbps data rate, QPSK $\frac{1}{2}$ MCS for 12Mbps data rate or QPSK $\frac{3}{4}$ MCS for 18Mbps data rate), high quality users in SoftHM achieve 616.7%, 383.3%, 187.5% or 111.1% higher throughput respectively. Similarly, to serve high quality users with the same bandwidth, low quality users in SoftHM achieve 390.9%, 414.3%, 176.9% or 237.5% (BPSK $\frac{1}{2}$, BPSK $\frac{3}{4}$, QPSK $\frac{1}{2}$ or QPSK $\frac{3}{4}$ MCS) higher throughput respectively.

2) ***Computation Complexity of SoftHM****:* To evaluate the computation complexity of SoftHM, we need to consider the complexity of each component in the pre-coding procedure shown in Fig. 7. As has been discussed previously, all the components before "Reverse-BCC 2" only involves several matrix multiplications and additions. The matrixes, such as $\boldsymbol{M}_{C,Q}, \boldsymbol{M}_{I,Q}$ etc., are sparse matrixes with at most 7 non-zeros entries in each row. Hence the computation complexity of those components are $O(m)$, where m is the payload size.

TABLE II

EFFECTIVE BANDWIDTH WITH VARIOUS n_g

QPSK $\frac{1}{2}$ + 64QAM $\frac{3}{4}$					
n_g	0	20	30	40	50
r_L (Mbps)	12	12	12	12	12
r_H (Mbps)	30	25	22.5	20	17.5
$\hat{r}_H$ (Mbps)	42	37	34.5	32	29.5
QPSK $\frac{3}{4}$ + 64QAM $\frac{3}{4}$					
n_g	0	20	30	40	50
r_L (Mbps)	18	18	18	18	18
r_H (Mbps)	30	25	22.5	20	17.5
$\hat{r}_H$ (Mbps)	48	43	40.5	38	35.5
max_v	0	0	1	3	3
BPSK $\frac{1}{2}$ + 64QAM $\frac{3}{4}$					
n_g	0	10	20	30	40
r_L (Mbps)	6	6	6	6	6
r_H (Mbps)	42	39.5	37	34.5	32
$\hat{r}_H$ (Mbps)	48	45.5	43	40.5	38
BPSK $\frac{3}{4}$ + 64QAM $\frac{3}{4}$					
n_g	0	10	20	30	40
r_L (Mbps)	9	9	9	9	9
r_H (Mbps)	42	39.5	37	34.5	32
$\hat{r}_H$ (Mbps)	51	48.5	46	43.5	41
max_v	0	0	1	1	0

Next we analyze the computation complexity of component "Reverse-BCC 2" implemented with the proposed dynamic programming algorithm. We observe that the number of steps in the algorithm is the size of the payload. In each step, the number of states for BCC is a constant $k_1 = 2^6$. For each state, the maximum number of bit combinations need to evaluate is a constant $k_2 = 2^{max_v}$ (refer to (7)). Here $max_v = \max_k |U(V_k)|$ is the maximum number of bits whose values will affect error-prone points generated after input bit $f(k)$ (Refer to Sec.III-D. According to Table II, max_v is a very small value, e.g., when $n_g = 50$, $max_v = 3$ for QPSK modulation and when $n_g = 30$, $max_v = 1$ for BPSK modulation scheme. Since k_1 and k_2 are constant, the complexity of the component "Reverse-BCC 2" is $O(m)$ with moderate n_g. Put everything together, the complexity of SoftHM is $O(m)$ which grows linearly with the payload size.

IV. DISCUSSION

A. SoftHM Protocol Design for 802.11g WLAN

802.11g PHY define a signal field before PHY payload to indicate the MCS of the packet and the length of the payload. The signal filed occupies one Wi-Fi symbol with BPSK $\frac{1}{2}$ MCS. In this symbol, it uses 4 bits ($2^4 = 16$ combinations) to indicate the MCS, and there are only 8 MCSs in 802.11g. Hence, SoftHM mode can be defined with the unused combinations. E.g., "0000" can be used to indicate QPSK $\frac{1}{2}$ + 64QAM $\frac{3}{4}$ MCS combination, and "1000" can be used for QPSK $\frac{1}{2}$ + 64QAM $\frac{3}{4}$ MCS combination, and etc. After decoding the signal filed, the receiver will recognize if the packet is a normal packet or if it is in SoftHM mode. If the packet is in SoftHM mode, a user will decode both streams or the low MCS stream based on its channel condition.

Notice that, minor hardware modifications are necessary for implementing SoftHM. E.g., Wi-Fi cards need to provide an interface so that the signal filed can be modified. Though it is not supported in current COTS Wi-Fi cards, the modifications are easy to implement. Moreover, compared with SoftHM's potential benefits, these minor modifications are well deserved.

B. Generalizing Other MCS Combinations for SoftHM

Though only QPSK modulation has been discussed as the lower modulation scheme in Sec. III, SoftHM is more general and can adopt other MCS combinations, e.g., BPSK or 16QAM. To achieve it, we only need to design new mappings from 64QAM signal to BPSK or 16QAM signals. In particular, if BPSK is used as the lower MCS, the points in the left most column in Fig. 3, denoted as "000,100", "000,101",…,"000,001", "000,000" are represented as BPSK signal "0" and the points in the right most column in the figure can be used as BPSK signal "1". With the new mapping, similar coding procedure can be used to load BPSK information to 64QAM signals.

As to 16QAM MCS, 64QAM modulation is not suitable for emulating 16QAM signals as it will introduce additional noise to 16QAM signals. However, 256QAM modulation used in 802.11ac system is quite capable to emulate 16QAM signals. Similarly, 1024QAM modulation can be used for emulating 64QAM signals.

C. Supporting Higher Number of Data Streams in One Transmission

SoftHM is not restricted to 2 data streams. In general, the number of data streams to be supported by SoftHM depends on the highest modulation scheme used in the system. The higher the modulation scheme used in a system, the more data streams it can transmit. To be more specific, when the highest modulation scheme is 2^kQAM, the number of available data streams is $\lfloor\frac{k-4}{2}\rfloor+1$, where $\lfloor x\rfloor$ represents the maximum integer that does not exceed x. E.g., when 64QAM is used as the highest modulation scheme (for 802.11g system), it can support $\lfloor\frac{6-4}{2}\rfloor+1=2$ data streams, that are QPSK and 64QAM; when 256QAM is used (as in 802.11ac system), it can support $\lfloor\frac{8-4}{2}\rfloor+1=3$ data streams, QPSK, 16QAM and 256QAM; when 1024QAM is used (as in 802.11ax system), it can support $\lfloor\frac{10-4}{2}\rfloor+1=4$ data streams, QPSK, 16QAM, 64QAM and 1024QAM.

D. Applying SoftHM to Other Modulation Schemes

Generally speaking, softHM is not restricted to QAM modulation schemes and can also be applied to Frequency Shift Keying (FSK), Phase Shift Keying (PSK) and etc as long as those modulation schemes meet the two prerequisites listed before (refer to Sec. II-B). For instance, we can use 8-PSK and BPSK for softHM. According to Fig. 8, in 8-PSK, all the points are represented by three bits. If the point is at the left side of the Q-axis, its first bit is "0". If the point is at the right side of the Q-axis, its first bit is "1". In addition, when BPSK is used, the point at the left side of the Q-axis is also "0", and "1" otherwise. Therefore, a transmitter can use 8-PSK to modulate the signals to be transmitted. Notice that, to avoid the decoding ambiguity of the BPSK receiver,

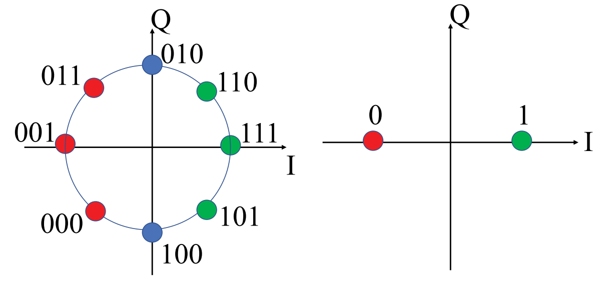


Fig. 8. The constellation diagram of (a). 8-PSK; (b).BPSK. Here three points "011", "001", and "000" in 8-PSK can be demodulated as "0" in BPSK; and three points "110","111","101" can be demodulated as "1" in BPSK.

two points "010" and "100" in 8-PSK cannot be used. A high SNR receiver will then demodulate the signals with 8-PSK and decode $log_2 6$ bits of information (two points "010" and "100" are disabled). A low SNR user can use BPSK to demodulate the same signal and decode 1 bit of information (all points on the left side of Q-axis is decoded as "0", and "1" otherwise).

E. SoftHM Extension to Other Systems

SoftHM can be easily extended to 802.11n, 802.11ac and LTE networks as they also adopt similar coding strategies. In addition, SoftHM is orthogonal to MIMO technologies. When SoftHM and MIMO are used together, further system performance can be expected.

In particular, for LTE network, Multimedia Broadcast Multicast Service (MBMS) has been discussed for broadcasting or multicasting information to users. Since the coding and modulation procedure in LTE system follows a similar idea as the one employed for 802.11g WLAN, by designing a pre-coding module on top of the LTE PHY layer, SoftHM can also manipulate the PHY payload of an LTE packet, and thus provide higher data rate for cell center users while guarantee the basic rate for cell edge users. We will leave those extensions as our future work.

SoftHM, however, cannot be applied to LoRa system directly. The reason is the following. LoRa uses !!!Chirp Spread (CCS) as its modulation scheme. Though CCS can adjust its transmission rate by changing the Spreading Factor (SF) in the modulation scheme, transmissions with different SF are orthogonal to each other. A Lora packet with one SF (and the same bandwidth) cannot be decoded with another SF. SoftHM is thus not applicable to LoRa system. How to realize transmitting a packet with multi-rates for LoRa system will be left as our future work.

F. Applying SoftHM to Other Applications

SoftHM can be extend to complicated multimedia contents, such as video transmission. It can also be working with SVC. SoftHM provides an efficient and implementation friendly way to realize HM for COTS systems, such as Wi-Fi or LTE. Since traditional HM can be used for video transmission with/without SVC and etc, softHM can also be applied to those scenarios. We will leave its video implementation with SVC as our future work.

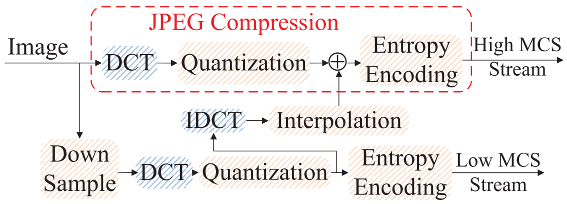


Fig. 9. An illustration of modified JPEG compression algorithm. Here DCT refers to Discrete Cosine Transformation, and IDCT is Inverse DCT.

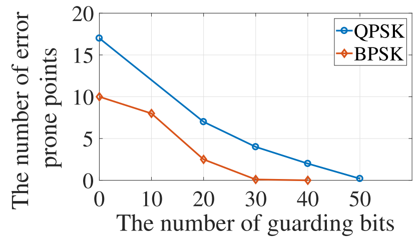


Fig. 10. The number of guarding bits n_g versus the number of error-prone points (per symbol).

V. Evaluation Results

A. Experiment Environment

1) Modified JPEG Compression Algorithm: We use a modified JPEG algorithm to evaluate SoftHM implementation, and leave the video transmission implementation as our future work. Recall that SoftHM is a generic PHY layer pre-coding paradigm, and it is compatible to other upper layer coding algorithm, such as salable coding.

Fig. 9 shows the flow diagram of the traditional [18] and the modified JPEG compression algorithm. For traditional JPEG transmission, to compress an image, it first performs Discrete Cosine Transform (DCT) to the original picture to decouple the correlation among adjacent pixels. Quantization is then adopted to quantize the real DCT components into integer numbers. Entropy coding achieves additional compression by encoding the quantized DCT component more compactly based on their statistical characteristics. For the modified JPEG algorithm, we first down sample the original picture and use the standard JPEG compression algorithm to encode the low MCS stream. Meanwhile, the original picture will also be DCT-transformed and quantized to a higher quality picture. We then decompressed the down-sampled image, interpolate it to the original size, and encode the differential information between the interpolated and the higher quality compressed picture. The differential information are loaded to the higher MCS stream. Notice that the information carried by the low and high MCS stream are decoupled. A 64QAM user can expect a correct high resolution image with the decoupled information given the observation that if a user can receive a higher MCS stream, it can also receive a lower MCS stream.

In addition, the image quality at different users can be determined as follows. We first calculate the maximum achievable rate for high- and low- quality users respectively. The Quantization matrixes approach the maximum achievable rate for high- and low- quality users are then selected.

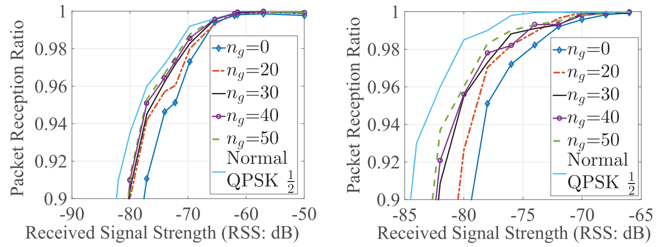
2) Experimental Settings: We use a USRP B210 as the SoftHM transmitter, so that the signal field can be modified to the desired value. Savvius Wi-Fi adaptor and Omnipeak are used to receive packets with various Received Signal Strength (RSS). To let a packet be decoded in different ways, we let the transmitter send two consecutive packets with the same PHY payload. The signal field of the first packet is set to the low MCS (i.e., QPSK $\frac{1}{2}$) while the signal field of the second packet is set to the high MCS (i.e., 64QAM $\frac{3}{4}$). Since the coherent time of a wireless channel is about several milliseconds, and a 802.11g packet is only several hundred microseconds, the channel states for the two consecutive packets are almost the same. We test the performance of SoftHM with four MCSs combinations: BPSK $\frac{1}{2}$ MCS + 64QAM $\frac{3}{4}$ MCS, BPSK $\frac{3}{4}$ MCS + 64QAM $\frac{3}{4}$ MCS, QPSK $\frac{1}{2}$ MCS + 64QAM $\frac{3}{4}$ MCS and QPSK $\frac{3}{4}$ MCS + 64QAM $\frac{3}{4}$ MCS combinations. The test is performed for both lab and hall environment as indicated in the Figures.

B. Experimental Results

1) The Impact of n_g on the Number of Error-Prone Bits: Fig. 10 shows the number of guarding bits n_g versus the number of error-prone points in a Wi-Fi symbol for QPSK+64QAM and BPSK+64QAM modulation combinations. The number of error-prone bits in a symbol drops quickly with the increase of n_g. Hence by increasing n_g, the receiver can decode the packet using a lower MCS with higher probability. In addition, for QPSK+64QAM combination, when the number of error-prone bits drops from 17 to 0.2 the effective bandwidth of the higher MCS stream is reduced from 30M to 17.5M; for BPSK+64QAM combination, when the number of error-prone bits drops from 10 to 0.1 the effective bandwidth of the higher MCS stream is reduced from 42M to 32M. This implies the trade-off between packet robustness (for the lower MCS stream) versus the achievable bandwidth (for the higher MCS stream) in the network. Even though, for all different MCS combinations, low MCS stream (i.e., QPSK or BPSK stream) will never experience any bandwidth loss.

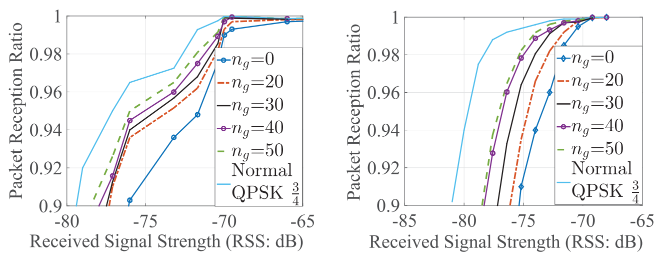
2) The Impact of n_g on the Packet Reception Ratio of the Low MCS Stream: Fig. 11-Fig. 14 show the Packet Reception Ratio (PRR) versus different Received Signal Strength (RSS) in both lab and hall environment for four MCS combinations. Here we only inspect the PRR for the lower MCS stream. For the higher MCS stream, it is expected to achieve the same reception ratio as that of a normal Wi-Fi packet.

According to the figures, it is obvious that the higher n_g, the higher the PRR with a given RSS. For 64QAM $\frac{3}{4}$ + QPSK $\frac{1}{2}$ combination, with PRR equals 90%, the average RSS of packets with $n_g = 30$ is 2.5dB to 3dB lower than the case when $n_g = 0$. However, as n_g keeps increasing, the increase of the PRR is marginal. For 64QAM $\frac{3}{4}$ + QPSK $\frac{3}{4}$ combination, with PRR equals 90%, the average RSS of packets with $n_g = 40$ is 2dB to 3dB lower than the case when $n_g = 0$. Similar conclusions can be applied to 64QAM $\frac{3}{4}$ + BPSK $\frac{1}{2}$ and 64QAM $\frac{3}{4}$ + BPSK $\frac{3}{4}$ combinations. E.g., with 64QAM $\frac{3}{4}$ + BPSK $\frac{1}{2}$ MCS combination, for the same PRR about 2.5dB lower average RSS can be expected from



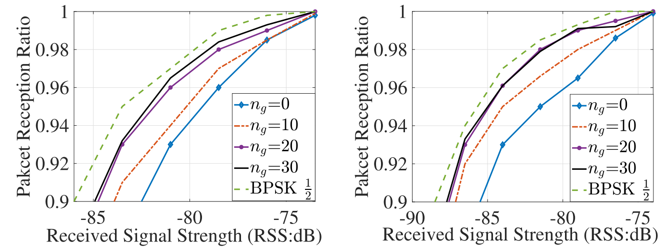
(a) (b)

Fig. 11. Packet reception ratio (QPSK $\frac{1}{2}$) versus various received signal strength for: (a). lab environment; (b). hall environment.



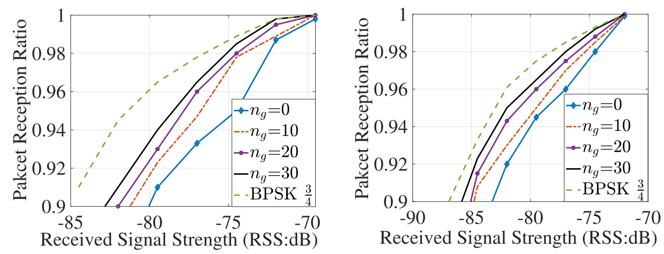
(a) (b)

Fig. 12. Packet reception ratio (QPSK $\frac{3}{4}$) versus various received signal strength for: (a). lab environment; (b). hall environment.



(a) (b)

Fig. 13. Packet reception ratio (BPSK $\frac{1}{2}$) versus various received signal strength for: (a). lab environment; (b). hall environment.



(a) (b)

Fig. 14. Packet reception ratio (BPSK $\frac{3}{4}$) versus various received signal strength for: (a). lab environment; (b). hall environment.

the curve $n_g = 20$ compared with the case when $n_g = 0$. With 64QAM $\frac{3}{4}$ + BPSK $\frac{3}{4}$ MCS combination, 2dB to 3dB lower average RSS can be expected from the curve $n_g = 30$ compared with the case when $n_g = 0$. In general, the PRR of both QPSK $\frac{3}{4}$ MCS and BPSK $\frac{3}{4}$ are more sensitive to the error-prone bits than that of QPSK $\frac{1}{2}$ and BPSK $\frac{1}{2}$ MCS. This is due to the reason that $\frac{3}{4}$ coding rate is less robust than $\frac{1}{2}$ coding rate.

*3) **The Average PSNR of SoftHM for Image Transmission**:* We use SoftHM to transmit a modified JPEG image, and evaluate its performance with various RSS. We test the following four settings: 64QAM $\frac{3}{4}$ and QPSK $\frac{1}{2}$ MCS with the number of guarding bits $n_g = 30$; 64QAM $\frac{3}{4}$, and QPSK $\frac{3}{4}$ MCS with $n_g = 40$; 64QAM $\frac{3}{4}$ and BPSK $\frac{1}{2}$ MCS with $n_g = 20$; and 64QAM $\frac{3}{4}$ and BPSK $\frac{3}{4}$ MCS with $n_g = 30$.

The image to be transmitted is then divided into multiple packets with packet duration equals to $600\mu s$. A synchronization mark is added in front of each packet so that once a packet is lost, the remaining packets can still recover something meaningful. Also as multimedia transmission are sensitive to packet loss, we add Reed Solomon (RS) code ((255,231)) for the transmitted packet for further protection. Based on the above operations, the maximum achievable effective transmission rates for those setting are: 31.3Mbps and 10.9Mbps; 34.4Mbps and 16.3Mbps; 39.0Mbps and 5.4Mbps; 39.4Mbps and 8.2Mbps. The resulting achievable PSNR are: 31dB and 22dB; 31.5dB and 27.8dB; 31.7dB and 16.7dB; 31.8dB and 19.2dB respectively.

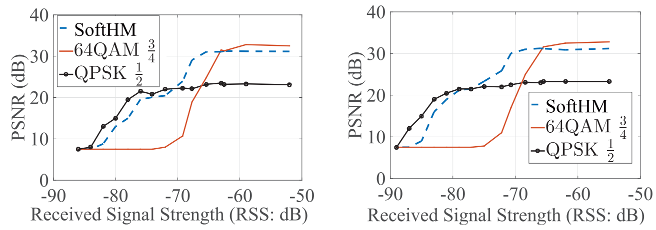
Fig.15-Fig.18 show the average PSNR (Peak Signal to Noise Ratio) of the received image under various RSS for both lab and hall environments with different MCS combinations. The figures imply that SoftHM can effectively alleviate the cliff effect and provide graceful PSNR to a large range of RSS. E.g., in hall environment, with QPSK $\frac{1}{2}$ + 64QAM $\frac{3}{4}$ MCS combination, and when RSS ranges from -66dB to -79dB, SoftHM can provide higher PSNR than a normal QPSK $\frac{1}{2}$ or 64QAM $\frac{3}{4}$ MCS packet. Moreover, as can be observed, compared with QPSK $\frac{3}{4}$ + 64QAM $\frac{3}{4}$ configuration, QPSK $\frac{1}{2}$ + 64QAM $\frac{3}{4}$ tends to provide lower PSNR at higher RSS regime, but higher PSNR at lower RSS regime. It also indicates that we can choose different MCS configurations based on users' link quality conditions.

Still, at the very high RSS (e.g. RSS≥ -66dB) regime SoftHM is not able to achieve as good PSNR as that of 64QAM $\frac{3}{4}$ MCS since the overall high quality link users' effective bandwidth ($\hat{r}_H$) is lower than that of the normal one. At the very low RSS (e.g., RSS≤ -79dB for outdoor environment) regime, the PSNR of SoftHM is also lower than that of the normal QPSK $\frac{1}{2}$ MCS packet. This is caused by imperfect signal emulation for the lower MCS signal: the minimal distance between the emulated QPSK points is smaller than that of the normal ones due to the coding constraints.

Similar observations can be expected for BPSK $\frac{1}{2}$ + 64QAM $\frac{3}{4}$ and BPSK $\frac{3}{4}$ + 64QAM $\frac{3}{4}$ MCS combinations. When RSS rangs from -90dB to -55dB, SoftHM is able to provide graceful PSNR and alleviate the cliff effects significantly.

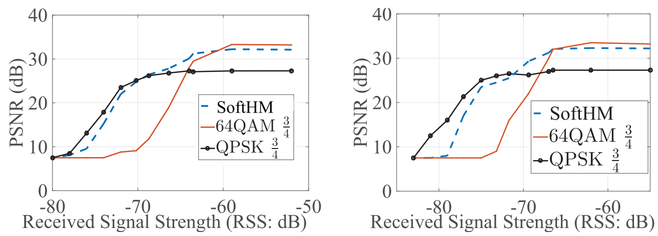
VI. Related Work

Wireless Multimedia broadcasting has drawn extensive research attentions from both industry and academic area. From application layer point of view, scalable coding [16] intended to encode video streams with various qualities (e.g., resolution, number of frames, and etc.). However, scalable coding itself cannot fully resolve the cliff effect as wireless



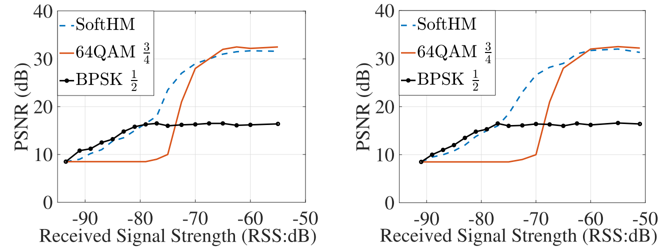
(a) (b)

Fig. 15. Average Peak SNR (QPSK $\frac{1}{2}$) versus various received signal strength for: (a). lab environment; (b). hall environment.



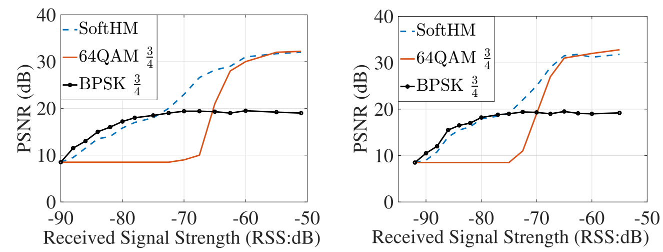
(a) (b)

Fig. 16. Average PSNR (QPSK $\frac{3}{4}$) versus various received signal strength for: (a). lab environment; (b). hall environment.



(a) (b)

Fig. 17. Average Peak SNR (BPSK $\frac{1}{2}$) versus various received signal strength for: (a). lab environment; (b). hall environment.



(a) (b)

Fig. 18. Average PSNR (BPSK $\frac{3}{4}$) versus various received signal strength for: (a). lab environment; (b). hall environment.

link is unreliable, and packet stalling has huge impact on the overall PSNR for multimedia transmission.

To better confront with the unreliable wireless channel, a naive approach is to send multiple packets with different transmission rates. However, as has been discussed in [2], this approach is not able to fully exploit the achievable spectrum efficiency in wireless networks. Moreover, in a Carrier Sensing Multiple Access (CSMA) based WLAN, separating information into different packets also doubles the associated control overhead (from PHY layer to TCP layer), causing higher network congestion and longer packet delay due to the random back-off mechanism. Last but not the least, the time-sharing approach is incapable of guaranteeing tight synchronization among users with different link qualities.

SoftCast [3] and ParCast [4] propose approaches to design source coding and channel coding jointly for video broadcasting in both Single Input and Single Output (SISO) and MIMO channel respectively. In SoftCast and ParCast, since the transmitted signals are linearly related to the original pixel's luminance, different receivers can recover the video with different levels of resolution. To maintain the linear relationship between the transmitted signal and the original image, images can only be compressed with analog compression, and the potential compression gain from standard video compression approach such as H.265 and entropy encoding cannot be fully exploited. Flexcast [5] tries to offer different degrees of protection to video data according to their importance. In other words, the more important the data is, the higher the degree of protection it gets. Nevertheless, Flexcast are suitable to handle channel variations of a wireless link but cannot benefit multiple users simultaneously. Moreover, though significant system performance are expected from the aforementioned approaches, both channel coding and modulation module in PHY layer needs to be redesigned, resulting in significantly high implementation cost.

Hierarchical Modulation is first introduce to digital broadcast systems to upgrade the transmission rate of new digital broadcast receivers while maintaining backward compatibility to old receivers [6]. When jointly designed with progressive image and scalable video encoders [7], [16], HM provides different levels of resolutions of the same image or video to users. Its MIMO version is discussed in [8]. The prior work [10] extends the idea to a multi-hop network to minimize the overall system energy consumption.

To reduce the modulation/demodulation complexity of the traditional HM scheme, [9] propose to use part of the 64QAM modulation constellation map to modulate two layers of data streams jointly. Still, separate coding modules are required for each data streams. Moreover, to combine two layers of data stream in one packet, a special modulation scheme designed exclusively for scalable modulation is a prerequisite. Of course, the coding modules and the modulation strategy are not applicable to modern broadband communication systems such as 802.11 based WLAN or LTE based cellular network. The authors in [11] employ a similar idea. A special constellation map is designed to embed (extract) different layers of information within one packet. Different streams of data still need to be encoded (decoded) separately, which implies multiple encoding modules are required. In summary, traditional HM schemes need tremendous hardware modifications if to be applied to modern broadband communication systems. Hence they are constrained from being used for those systems.

On the other hand, recent advances in Cross Technology Communications (CTC) [19] show that by emulating Zigbee signal via Wi-Fi signal, Wi-Fi devices can talk to Zigbee devices directly with high efficiency. Inspired by the signal emulation technology, we exploit the possibility of emulating signals for multiple layers of MCS within one 802.11g packet.

VII. Conclusion and Future Work

In this work, we propose SoftHM, a new coding paradigm, to facilitate multiple MCSs transmission with one packet. Different from traditional HM, SoftHM is a software based approach, and thus does not require multiple hardware based coding and modulation modules for embedding different streams of data in one packet. In particular, SoftHM is designed on top of 802.11 based PHY payload, it follows 802.11 based PHY standards and can be easily extend to COTS Wi-Fi devices without changing their coding or modulation modules. Moreover, as the packet contains multiple layers of MCS, users with different link qualities can interpret and retrieve different amount of information from the same packet.

According to real experiment results, compared with the traditional time sharing approach, e.g., only low- or high-quality users can be served at one time, to support low quality users with the same bandwidth (BPSK $\frac{1}{2}$ MCS for 6Mbps data rate, BPSK $\frac{3}{4}$ MCS for 9Mbps data rate, QPSK $\frac{1}{2}$ MCS for 12Mbps data rate and QPSK $\frac{3}{4}$ MCS for 18Mbps data rate), high quality users (64QAM $\frac{3}{4}$ MCS) in SoftHM achieve 616.7%, 383.3%, 187.5% and 111.1% higher throughput respectively. Likewise, to serve high quality users with the same bandwidth, low quality users in SoftHM achieve 390.9%, 414.3%, 176.9%, and 237.5% (with BPSK $\frac{1}{2}$, BPSK $\frac{3}{4}$, QPSK $\frac{1}{2}$ and QPSK $\frac{3}{4}$ MCS) higher throughput.

In future work, we will devote our effort to fully exploit the benefit of SoftHM for MIMO systems.

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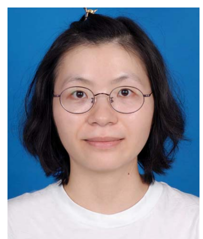

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