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Multi-destination aggregation with binary symmetric broadcast channel based coding in 802.11 WLANs

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Abstract

In this paper we consider the potential benefits of adopting a binary symmetric broadcast channel paradigm for multi-destination aggregation in 802.11 WLANs, as opposed to a more conventional packet erasure channel paradigm. We propose two approaches for multi-destination aggregation, i.e. superposition coding and a simpler time-sharing coding. Theoretical and simulation results for both unicast and multicast traffic demonstrate that increases in network throughput of more than 100% are possible over a wide range of network conditions and that the much simpler time-sharing scheme yields most of these gains and have minimal loss of performance. Importantly, these performance gains are achieved exclusively through software rather than hardware changes.

Keywords Multi-destination aggregation · Binary symmetric broadcast channel · Time-sharing coding · Superposition coding · 802.11 WLANs

1 Introduction

Increasing the PHY rates used in a WLAN leads to faster transmission of the packet payload of a frame, but the overheads associated with each transmission (PHY header, MAC contention time etc) typically do not decrease at the same rate and thus begin to dominate the frame transmission time. To maintain throughput efficiency at high PHY rates, 802.11n [12] uses packet aggregation, whereby multiple packets destined to the *same* receiver are transmitted together within a single large frame. In this way, the overheads associated with a single transmission are amortised across multiple packets and higher throughput efficiency is achieved, e.g. see [15].

A logical extension is to consider aggregation of packets destined to *different* receivers into a single large frame. Such multi-destination aggregation is currently the subject of much interest because with the increasing number of

WiFi hotspots and other accessing technologies available, for a single WLAN AP, there simply may not be enough traffic to an individual destination to allow large packets to be formed in a timely manner and so efficiency gains to be realised. One of the key issues in multi-destination aggregation is the choice of modulation and coding scheme (MCS) for aggregated packets. Although multi-destination aggregation allows simultaneous transmission to multiple receivers, the channel quality between the transmitter and each receiver is generally different, and thus the optimal MCS which matches the channel quality of each receiver is also different. The current 802.11 standard constrains transmitters to use the same MCS for all bits within a frame, and the state of the art is to send multicast/broadcast packets (which contain messages for multiple receivers) at the highest MCS rate which the receiver with the worst channel quality can support [11]. While this ensures that every receiver is capable of decoding the received packet, clearly it is highly inefficient.

In this paper we consider an alternative approach to multi-destination aggregation, which still uses the same MCS for every symbol within an aggregated frame (and so does not require hardware changes) but encodes packets

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destined to different receivers with different levels of protection by using higher-layer coding techniques.

The approach builds on an experimental observation that packets discarded at 802.11 MAC layer due to CRC errors actually contain a high proportion of correct bits, and thus potentially provide a useful channel through which information can be transmitted. Recently [5] indicates that this channel can be accurately modeled as a binary symmetric channel. Based on this, multi-destination aggregated packets from the AP form a binary symmetric broadcast channel between the transmitter and multiple receivers. Then by using appropriate BSBC-based error correction coding bits within a single frame can be transmitted to different destinations at different information rates while still using the same MCS. To our knowledge, we present the first detailed analysis of multi-user coding for aggregation in 802.11 WLANs.

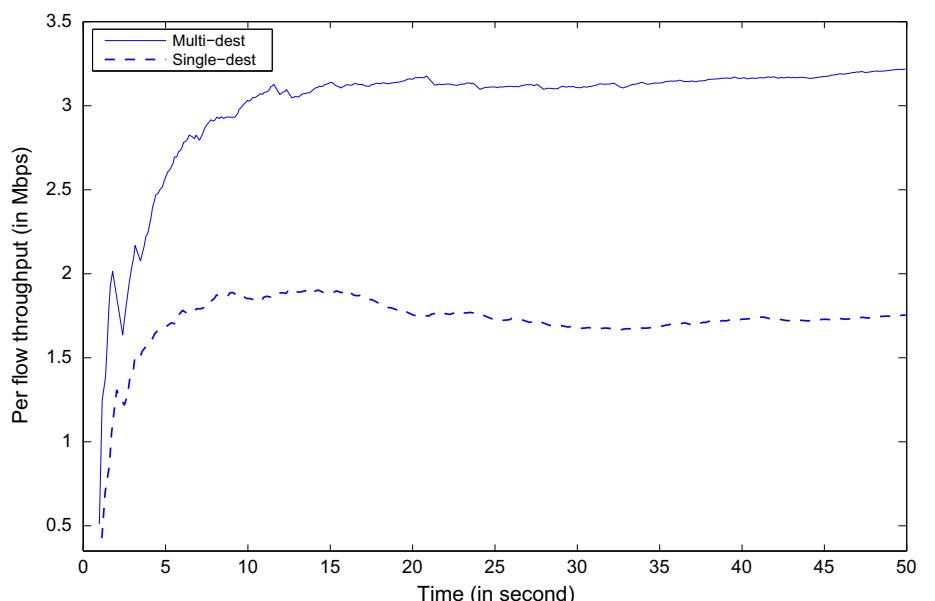
We demonstrate in Sects. 6 and 7 that by using this coding approach for multi-destination aggregation increases in network throughput of more than 100% are possible over a wide range of channel conditions. This is illustrated, for example, in Fig. 1 which presents throughput measurements for downlink transmissions in a WLAN containing 10 downlink flows and 10 competing uplink flows. When single destination aggregation is used, on average insufficient packets are available for each destination to allow a full sized frame (65,535 bytes) to be assembled. On average only 36 packets are assembled in each single destination aggregated frame, resulting in a substantial loss of network efficiency. At each transmission opportunity, the AP first checks the destination address of the first packet in the queue, and then searches through the queue to assemble packets destined to the same receiver. With

multi-destination aggregation, full-sized frames can be assembled at every transmission opportunity. On average 117 packets are assembled in each multi-destination aggregated frame. Since the coding proposed here is introduced above the MAC layer, there is no need for any hardware changes and these performance gains therefore essentially comes for “free”.

2 Related work

The concept of Multiple Receiver Aggregate (MRA) was first proposed by the TGnSync group in [18]. The idea of aggregating multiple packets into a single large frame, and then multicasting/broadcasting it to distinct receivers became the subject of much interest soon for delay-sensitive and short-packet applications such as VoIP [13, 14, 21, 23]. For example, [23] proposes a voice multiplex–multicast (M–M) scheme of multiplexing packets from several VoIP streams into one multicast packet for downlink transmissions to overcome the heavy overhead of VoIP traffic over WLANs. Similarly [14] proposes a congestion-triggered downlink aggregation scheme by stretching the 802.11n A-MPDU format [12] to carry MPDUs addressed to different destinations. Aggregation is performed only when there is congestion. When an aggregation is triggered, the VoIP packets queued at MAC layer are put into the aggregated frame in the same order as in the queue, with no sorting and no packaging for per destination. The aggregation complexity and overhead is thus reduced compared to the per-destination grouping strategy as proposed in [18]. Apart from the downlink multi-user aggregation, [21] presents a complimentary

Fig. 1 Access point (AP) downlink throughput with single destination aggregation and with multi-destination aggregation. The AP is sending traffic to 10 client stations, one flow per station. Meanwhile each station has a competing uplink flow to the AP. Each flow has Poisson distributed packet arrivals at rate 2000.802.11g WLAN (see Table 1 for PHY/MAC parameter values), AP buffer size is 200 packets, ns2 simulation



uplink aggregation technique that effectively serializes channel access in the uplink direction. The combination of uplink and downlink aggregation mechanisms simultaneously improves VoIP call quality while preserving network capacity for best-effort data transfer.

All of the above works only consider homogeneous networks, i.e. stations in a WLAN have the same channel qualities and thus use the same data rate. In a heterogeneous network where stations have different optimal transmission rates, multicasting or broadcasting the entire aggregated frame at the low enough rate to ensure all the stations can receive it will result in a significant loss in throughput. This problem is addressed in [16]. This paper proposes a scheme called data rate based aggregation (DRA) which groups packets in the MAC queue in terms of data rates, and then aggregates packets for all links that have the same data rate and allows packet reordering. Such a way mitigates the performance demotion caused by aggregating across data rates. But the grouping strategy does not always provide the best performance. [16] also proposes a scheme data rate based aggregation with selective demotion (DRA-SD) which allows a cross rate merge of two DRA frames under some conditions. The simulation results show evidence of better performance in terms of transmission time.

Packet aggregation is considered together with network coding in [20]. This paper proposes a scheme that uses length aware packet aggregation and network coding to improve the throughput of single relay multi-user wireless networks. At the relay node, upload and download packets are exclusive ORed and then broadcast to the next hop. Aggregation is performed before coding if packets in both directions do not have similar sizes. The network coding is a packet-level coding scheme.

To the best of our knowledge, this is the first work that uses bit-level coding schemes to solve the problem of multi-rate throughput compromise in multi-destination aggregation. The proposed method could benefit from both aggregation and bit-level channel capacity improvement.

3 Preliminaries

3.1 Multi-destination aggregated frames form a binary symmetric broadcast channel

In a binary symmetric channel (BSC) each received packet is considered as a binary vector in which an unknown subset of bits have been independently “flipped” with crossover probability p . It is shown in [5] that, after some pre- and post-processing, this accurately models the behavior of the channel provided by 802.11 corrupted frames. In a binary symmetric broadcast

channel (BSBC) [7], n receivers overhear a transmission. Each receiver obtains a separate copy of the transmission, with received bits being flipped independently with probability p_i at receiver i . The crossover probability p_i embodies the link quality between the transmitter and receiver i , and in general is different for each receiver and varies with the MCS used for the transmission. This is illustrated schematically in Fig. 2.

3.2 Running example: two-class WLAN

We will use the following setup as a running example. Namely, an 802.11 WLAN with an AP and two classes of client stations, n_1 stations in class 1 and n_2 in class 2. Stations in class 1 are located relatively far from the AP and so have lossy reception with crossover probability p which depends on the MCS used. Stations in class 2 are located close to the AP and experience loss-free reception (the crossover probability is zero) for every available MCS. Our analysis can, of course, be readily generalised to encompass situations where each station has a different crossover probability, but the two-class case is sufficient to capture the performance features of heterogeneous link qualities in a WLAN.

3.3 Coding for binary symmetric broadcast channels

The binary symmetric broadcast paradigm allows transmission of a multi-destination aggregated frame at different information rates to different destinations while using a single MCS. We consider two main approaches for achieving this, namely superposition coding and time-sharing coding.

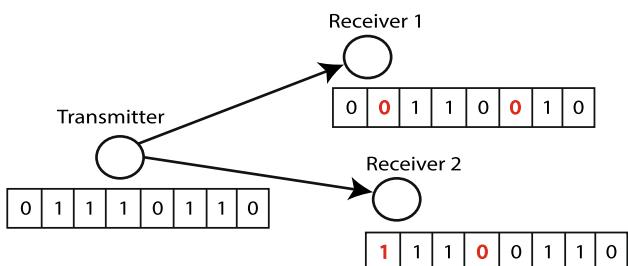


Fig. 2 Illustrating binary symmetric broadcast channel. A binary vector broadcast by the transmitter is overheard by two stations. Reception may be lossy, with bits being received flipped at receiver i with probability p_i (example bit flips are indicated in red bold) (Color figure online)

3.3.1 Superposition coding

Superposition coding works as follows. Encoding is straightforward: binary vectors destined to different receivers are simply added together, modulo 2, and transmitted as a single binary vector. Receiver i then receives its binary vector with bits flipped by (1) the physical channel and (2) by the addition of the messages for other receivers. Let p_i denote the physical channel crossover probability at receiver i and q_j , $j \neq i$ denote the effective crossover probability induced by adding the message intended to receiver j . Letting r_i denote the probability that a bit is flipped, the channel capacity to receiver i is then $C_i = 1 - H(r_i)$, where $H(r_i) = -r_i \log_2 r_i - (1 - r_i) \log_2 (1 - r_i)$ is the binary entropy function.

For example, with $n = 2$ receivers, $r_1 = q_2(1 - p_1) + (1 - q_2)p_1$ and $r_2 = q_1(1 - p_2) + (1 - q_1)p_2$. Provided messages to receiver i are sent at less than this information rate, they can be successfully decoded. Specifically, this rate can be achieved using the following nested decoding procedure: (1) order the n receivers by increasing crossover probability (decreasing channel quality), with ties randomly broken, (2) set $i = 1$, decode¹ the message for receiver i and subtract it from the received binary vector and (3) set $i \leftarrow i + 1$ and repeat until i equals the index of the current receiver.

Although the capacity of general binary broadcast channels remains unknown, for many important special cases (e.g. for stochastically degraded binary broadcast channels), it is known that superposition coding is capacity-achieving [3].

With superposition coding, the achievable sum-capacity of a binary symmetric broadcast channel with n receivers is $\sum_{i=1}^n C_i$ with $C_i = 1 - H(r_i)$ and r_i the effective cross-over probability of the binary symmetric broadcast channel between the transmitter and receiver i . For our running example of a WLAN with two classes of stations, with $n_1 = 1 = n_2$ the effective cross-over probability r_1 for the class 1 station is $r_1 = \beta(1 - p(R)) + (1 - \beta)p(R)$ where $p(R)$ is the crossover probability of the physical binary symmetric broadcast channel between the transmitter and the class 1 station (which depends, of course, on the MCS rate R selected), and β is the crossover probability determined by the binary addition with the message destined to class 2. $H(\beta)$ is the information rate at which data is transmitted to the class 2 station.

3.3.2 Time-sharing coding

From the discussion above it can be seen that superposition decoding can be a relatively complex operation. A simpler but demonstrably near-optimal choice is time-sharing coding [7]. In time-sharing, the transmitted binary vector is partitioned into n subsets of bits, where n is the number of receivers, and the i 'th subset of bits contains the message intended for receiver i and this message is encoded at a rate which is matched to the channel between the transmitter and receiver i . This approach is akin to packet aggregation, but with each packet carrying a payload that is separately encoded by the application layer¹. The application layer encoding adds appropriate redundancy that allows the intended receiver to decode the embedded information message even when the packet is received with bits flipped. For the two-class WLAN example, in time-sharing coding each transmitted frame is partitioned into two parts, the first intended for class 1 stations and the second intended for class 2 stations. The portion intended for class 2 will be received error-free and thus does not need further protection. The portion intended for class 1 is protected by a suitable BSBC error correcting code that allows information to be extracted even when some bits are corrupted; the information rate is obviously reduced compared to a noise-free channel.

4 Unicast throughput modelling

In this section we develop a detailed theoretical throughput performance analysis for three multi-destination aggregation approaches: (1) uncoded frame aggregation in a packet erasure channel paradigm; (2) aggregation with superposition coding in a broadcast BSBC paradigm; (3) aggregation with time-sharing coding in a broadcast BSBC paradigm. We focus on the two-class setup introduced in Sect. 3.2, the extension to more than two classes being straightforward.

4.1 802.11 MAC model

We consider a WLAN consisting of an access point (AP), n_1 class 1 stations and n_2 class 2 stations. We assume that all stations are saturated (unsaturated operation is considered later). The AP transmits $n_1 + n_2$ downlink unicast flows. Namely, one flow destined to each of the n_1 class 1 stations and one flow destined to each of the n_2 class 2 stations. When transmitting, the AP aggregates these downlink flows into a single large MAC frame which is sent at a single PHY rate. Each client station also transmits an uplink flow to the AP.

¹ Any coding approach for a binary symmetric channel can be used to encode the messages to receiver i here e.g. capacity achieving codes for BSBCs are described in [1, 2, 9].

Following Bianchi [4, 17], time is divided into MAC slots (which can be idle, success or collision slots). Let τ_0 denote the probability that the AP attempts a transmission in a MAC slot, τ_1 the probability that a class 1 station attempts a transmission and τ_2 the probability that a class 2 station attempts a transmission. Transmissions by class 1 stations are subject to collisions with transmissions by the other stations in the WLAN and, in the packet erasure paradigm, are also subject to noise-related erasures. The probability that a transmission from a class 1 station fails (due to collision and/or loss) is

$$p_{f1} = 1 - (1 - p_{c1}^U)(1 - p_{e1}^U) \quad (1)$$

where p_{e1}^U is the probability that an uplink transmission by a class 1 station is erased due to noise, and p_{c1}^U is the probability that it collides with another transmission, with

$$p_{c1}^U = 1 - (1 - \tau_1)^{n_1-1}(1 - \tau_2)^{n_2}(1 - \tau_0) \quad (2)$$

Class 2 stations do not suffer from noise caused erasures. Although sub-frames destined to class 1 in the packet erasure paradigm are subject to noise-related erasures, sub-frames destined to class 2 are error-free. The transmission failures from the AP or a class 2 station are only caused by collisions. Hence class 2 and the AP share the same station attempt probability, i.e. $\tau_0 = \tau_2$, and the same probability that a transmission fails, which is

$$p_{f0} = p_{f2} = 1 - (1 - \tau_1)^{n_1}(1 - \tau_2)^{n_2} \quad (3)$$

The usual Bianchi [4] expression gives a relation between the station transmission attempt probability τ and the probability p_f that a transmission fails. However, here we make use of expression (4) that extends the Bianchi expression to take account of a finite number of retransmission attempts and losses due to decoding errors [19].

$$\tau = \begin{cases} \frac{2(1 - 2p_f)(1 - p_f^{m+1})}{M} & m \leq m', \\ \frac{2(1 - 2p_f)(1 - p_f^{m+1})}{M + W2^{m'}p_f^{m'+1}(1 - 2p_f)(1 - p_f^{m-m'})} & m > m'. \end{cases} \quad (4)$$

in which $M = (1 - p_f)W(1 - (2p_f)^{m+1}) + (1 - 2p_f)(1 - p_f^{m+1})$, $W = CW_{min}$, m denotes the 802.11 retry limit number, and m' represents the number of doubling the CW size from CW_{min} to CW_{max} .

4.2 Network throughput

The network throughput is

$$S = \frac{X_0 + n_1 X_1 + n_2 X_2}{E_T} \quad (5)$$

where $X_0 = \tau_0(1 - p_{f0})(E_1^D + E_2^D)$, $X_1 = \tau_1(1 - p_{f1})E_1^U$, $X_2 = \tau_2(1 - p_{f2})E_2^U$ with E_1^D the expected payload delivered from the AP to class 1 stations and E_2^D to class 2 stations, E_1^U the expected payload delivered from a class 1 station to the AP, E_2^U the expected payload delivered from a class 2 station to the AP, and E_T is the expected MAC slot duration. It is important to stress that the expected payload delivered need not equal to the raw frame payload due to the impact of corruption of the frame payload during transmission across the radio channel and due to the overhead of any error-correction coding. Calculations of the expected payloads delivered and of the expected MAC slot duration are discussed in detail below for each of the three multi-destination aggregation schemes considered.

4.3 Fairness

Before proceeding to the calculation of the flow throughputs for the three multi-destination aggregation approaches, we note that to ensure a fair comparison amongst different schemes it is not sufficient to simply compare the sum-throughput. Rather we also need to ensure that schemes provide comparable throughput fairness, as an approach may achieve throughput gains at the cost of increased unfairness. In the following we take a max-min fair approach and impose the fairness constraint that all flows achieve the same throughput. Extension of the analysis to other fairness criteria is, of course, possible.

4.4 Expected payload

We begin by calculating the expected payload in a MAC slot for the three multi-destination aggregation approaches.

4.4.1 Uncoded

Similarly to the approach used in 802.11n A-MPDUs [12], we consider a situation where messages addressed to distinct destinations are aggregated together to form a single large MAC frame. We do not present results here without aggregation since the throughputs are strictly lower than when aggregation is used [15].

We need to calculate the expected delivered payloads E_1^D , E_2^D , E_1^U and E_2^U .

We proceed as follows. The expected payload delivered by an uplink packet of a class 1 station is

$$E_1^U = x_1^U (1 - p_u(R))^{L_1^U(R)} \quad (6)$$

where $p_u(R)$ is the first-event error probability of Viterbi decoding [22] for convolutional codes used in 802.11 standards when transmissions are made at PHY rate R ,

$$L_1^U(R) = DBPS(R) \left\lceil \frac{(x_1^U + L_{machdr} + L_{FCS}) \times 8 + 22}{DBPS(R)} \right\rceil \quad (7)$$

is the class 1 uplink frame size in bits. $DBPS(R)$ represents data bits per symbol at PHY rate R . L_{machdr} is the MAC header in bytes, and L_{FCS} is the FCS field size in bytes. x_1^U is the class 1 uplink frame payload in bytes. As transmissions by class 2 stations are erasure-free at all supported PHY rates, the expected payload of an uplink packet from a class 2 station is

$$E_2^U = x_2^U \quad (8)$$

where x_2^U is the payload size in bytes of class 2 transmissions. Turning now to the AP, similar to the approach used in 802.11n, the aggregated MAC frame consists of $n_1 + n_2$ unicast packets. The length of a MAC frame is

$$L = n_1 L_1^D + n_2 L_2^D \quad (9)$$

in which $L_1^D = x_1^D + L_{subhdr} + L_{FCS}$ and $L_2^D = x_2^D + L_{subhdr} + L_{FCS}$ are respectively the sub-frame size for class 1 and class 2 in bytes. L_{subhdr} is the sub-header length. x_1^D , x_2^D denote, respectively, the AP payload size in bytes destined to class 1 and class 2 stations. Note that the downlink PHY rate is determined by the client which has the worst link quality, and so equals the class 1 PHY rate R . The expected payload delivered to a class 1 station by an AP frame packet is therefore

$$E_1^D = x_1^D (1 - p_u(R))^{8 \cdot L_1^D} \quad (10)$$

while the expected payload delivered to a class 2 station is

$$E_2^D = x_2^D \quad (11)$$

For max-min fairness we need to equalize the throughput of each flow. That is, we require

$$x_2^U = x_2^D = x_1^D (1 - p_u(R))^{8 \cdot L_1^D} \quad (12)$$

$$\tau_1 (1 - p_{f1}) (1 - p_u(R))^{L_1^U} x_1^U = \tau_2 (1 - p_{f2}) x_2^U \quad (13)$$

For a given PHY rate R and AP frame size L we can solve (12) and (9) to obtain x_1^D and x_2^D . As p_{f1} depends on the payload size x_1^U due to noise-related erasures, we need to solve (13) jointly with the MAC model (4) to obtain x_1^U , τ_1 and τ_2 . We can then obtain E_1^D , E_2^D , E_1^U , E_2^U from (10), (11), (6), (8) as required.

4.4.2 Time-sharing coding

For the binary symmetric broadcast paradigm we start by considering the simpler time-sharing coding scheme. As in the erasure channel case, MAC frames are constructed by

aggregating two portions: one intended for class 1 stations and protected by an application layer error correction code (with coding rate matched to the channel quality between the AP and class 1 stations), the second intended for class 2 stations and uncoded (since the PHY layer MCS provides adequate protection). Each portion is further sub-divided into packets intended for the different stations. We also apply similar coding to protect uplink transmissions from class 1 stations to allow information to be recovered from corrupted uplink frames.

Let x_1^D denote the downlink information payload size for a class 1 station and x_2^D for a class 2 station. Suppose a downlink PHY rate R is chosen and the crossover probability for class 1 stations is $p(R)$. The number of coded bytes to ensure reception of x_1^D information bytes is $x_1^D / (1 - H(p(R)))$. The expected downlink payload delivered to class 1 and class 2 are $E_1^D = x_1^D$ and $E_2^D = x_2^D$. To equalize the downlink throughputs of stations in both classes (i.e. for max-min fairness), we therefore require

$$E_1^D = E_2^D \quad (14)$$

The AP frame size is $L = n_1 L_1^D + n_2 L_2^D$ where $L_1^D = (x_1^D + L_{subhdr} + L_{FCS}) / (1 - H(p(R)))$, $L_2^D = x_2^D + L_{subhdr} + L_{FCS}$.

To equalize the uplink and downlink throughputs we require

$$E_2^U = E_1^U = E_1^D \quad (15)$$

The expected uplink payload delivered from class 1 and class 2 are $E_1^U = x_1^U$ and $E_2^U = x_2^U$. Hence we have $x_2^U = x_1^U = x_1^D = x_2^D$ (where we are making use here of the fact that since frames are not erased in the binary symmetric broadcast channel paradigm, $p_{e1}^U = 0$ and thus $\tau_0 = \tau_1 = \tau_2$). Therefore given a specified AP frame size L we can solve for τ_1 and x_1^D in the similar way and obtain E_1^D , E_2^D , E_1^U , E_2^U .

4.4.3 Superposition coding

With superposition coding the MAC frames are constructed in two steps. Once a value of β has been determined, binary vectors are generated by aggregating IP packets of each class, and these are then summed, modulo 2, to generate the MAC frame. Despite the coding scheme being more complicated, the throughput analysis is similar to the time-sharing case. The main difference lies in the calculation of the downlink payload size.

Letting R denote the downlink PHY rate used by the AP and $p(R)$ denote the corresponding BSC crossover probability. The downlink BSBC capacity in bits per channel use between the AP and a class 1 station is $1 - H(\beta \circ p(R))$, where $\beta \circ p(R) = \beta(1 - p(R)) + (1 - \beta)p(R)$, and that

between the AP and a class 2 station is $H(\beta)$. The AP frame payload is formed by superimposing n_2 packets destined to class 2 stations to n_1 packets destined to class 1 stations. Hence, the AP frame size is $L = n_1 L_1^D + n_2 L_2^D$ where

$$L_1^D = \frac{x_1^D + L_{subhdr} + L_{FCS}}{(1 - H(\beta \circ p(R)))} \quad (16)$$

$$L_2^D = \frac{x_2^D + L_{subhdr} + L_{FCS}}{H(\beta)} \quad (17)$$

$E_1^D = x_1^D$ is the expected downlink payload for a class 1 station, and $E_2^D = x_2^D$ is the expected downlink payload for a class 2 station. To equalize the downlink throughputs of stations in both classes, we require

$$E_1^D = E_2^D \quad (18)$$

The ratio n_1/n_2 then fixes the value of β . With the value of β determined, given a specified AP frame size L we can solve to obtain E_1^D , E_2^D . To equalize the uplink and downlink throughputs we then require

$$E_2^U = E_1^U = E_2^D \quad (19)$$

4.5 Expected MAC slot time

Now we calculate the expected MAC slot duration. Let T_{AP} denote the duration of a transmission by the AP, T_1 the duration of a transmission by stations in class 1 and T_2 the duration of a class 2 transmission. As we have seen previously, we cannot adopt the usual approach of assuming that these transmissions are all of equal duration. However, we can still make use of the ordering in frame durations $T_{AP} \geq T_1 \geq T_2$. With this ordering, there are four possible types of MAC slot:

1. *AP transmits*: the slot duration is T_{AP} (even if other stations also transmit). The event occurs with probability τ_0 .
2. *Class 1 transmits*: the slot duration is T_1 if the AP does not transmit and at least one class 1 station transmits. This event occurs with probability $p_{T_1} = (1 - (1 - \tau_1)^{n_1})(1 - \tau_0)$.
3. *Only class 2 transmits*: the slot duration is T_2 if only class 2 stations transmit. This event occurs with probability $p_{T_2} = (1 - (1 - \tau_2)^{n_2})(1 - \tau_0)(1 - \tau_1)^{n_1}$.
4. *Idle slot*: the slot duration is the PHY slot size σ if no station transmits. This event occurs with probability $p_{Idle} = (1 - \tau_1)^{n_1}(1 - \tau_2)^{n_2}(1 - \tau_0)$.

The expected MAC slot duration is therefore

$$E_T = p_{Idle}\sigma + \tau_0 T_{AP} + p_{T_1} T_1 + p_{T_2} T_2 \quad (20)$$

4.6 MAC overheads

The duration of a class 1 station transmission is $T_1 = T(x_1^U) + T_{oh}$ where x_1^U is the payload in bytes of a class 1 station frame, and of a class 2 station transmission is $T_2 = T(x_2^U) + T_{oh}$ where x_2^U is the payload in bytes of a class 2 station frame. The duration of an AP transmission is $T_{AP} = T(L) + T_{oh} + T_{phyhdr1} - T_{phyhdr}$ where L is the payload in bytes of an AP frame and $T_{phyhdr1}$ the PHY/MAC header duration for an aggregated frame. Here, $T_{oh} = T_{difs} + 2T_{phyhdr} + T_{sifs} + T_{ack}$ is the PHY and MAC signalling overhead, with T_{phyhdr} the PHY header duration in μs , T_{ack} the transmission duration of an ACK frame in μs , T_{difs} a DIFS and T_{sifs} a SIFS. $T(x) = 4 \cdot \lceil \frac{(x+L_{machdr}+L_{FCS}) \times 8 + 22}{DBPS(R)} \rceil$ is the transmission duration, including MAC framing, of a payload of x bytes at PHY rate R .

In these calculations we assume that uplink transmissions by client stations are immediately acknowledged by the AP (rather than, for example, using a block ACK proposed in 802.11e [10]). Similarly, we assume that downlink transmissions are immediately acknowledged by client stations and, to make our analysis concrete, we adopt the approach described in [8] which uses the orthogonality of OFDM subcarriers to allow a group of client stations to transmit feedback signals at the same time, and thereby ACK collisions are avoided. However, we stress that these assumptions regarding ACKing really just relate to the calculation of the MAC overheads and our analysis could be readily modified to account for alternative acknowledgement mechanisms.

Similarly, to keep our discussion concrete, we assume the frame format shown in Fig. 3 is used for multi-destination aggregation in the packet erasure paradigm and with time-sharing coding. Again, it is important to stress that this just relates to the calculation of the MAC overheads. In Fig. 3 a sub-header is prefixed to each IP packet to indicate its receiver address, source address and packet sequence information. An FCS checksum is used to detect corrupted packets in packet erasure paradigm. Since the sub-header already contains the receiver address, source address and sequence control, the MAC header removes these three fields, but keeps other fields unchanged from the standard 802.11 MAC header. We assume that the MAC header is transmitted at the same PHY rate as the PLCP header and thus is error-free.

Although the sub-header of each time-sharing segment contains the receiver address as depicted in Fig. 3, this field is not reliable due to channel noise. And as the length of each coded segment depends on the current channel quality, it varies over time. Therefore, we need to notify each receiver in the common MAC header to locate its segment. We use the spare field in the MAC header to map the initial

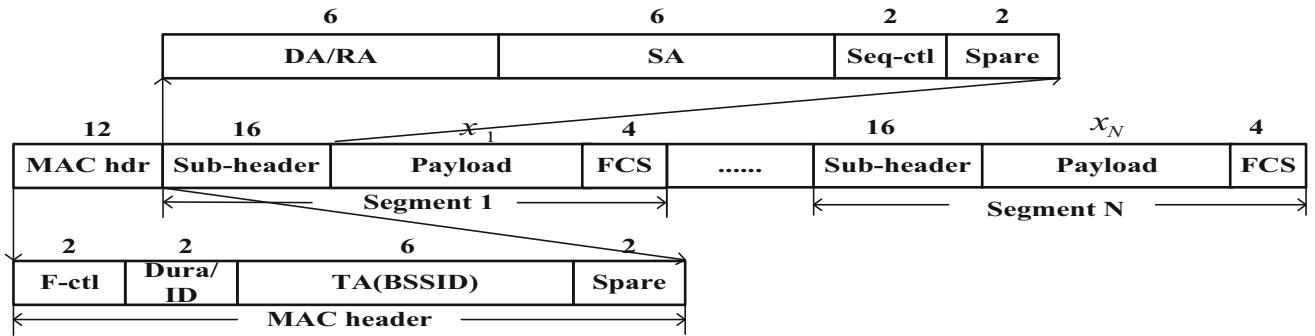


Fig. 3 Erasure channel frame format

position of each segment to its destination. Each receiver is allocated with a unique ID when associated with the AP. In the mapping field, receivers are identified using this ID number instead of their MAC addresses to save space.

5 Multicast throughput modelling

The foregoing unicast analysis can be readily extended to encompass multicast traffic. The AP now multicasts two downlink flows which are aggregated into a single MAC frame. Flow 1 is communicated to the n_1 class 1 stations and flow 2 is communicated to the n_2 class 2 stations. When there are no competing uplink flows we can compute the throughput using the analysis in Sect. 4 by selecting the following parameter values: $n_1 = n_2 = 1$; $x_1^U = x_2^U = 0$; $p_{e1}^U = p_{c1}^U = 0$; $\tau_1 = \tau_2 = 0$; $\tau_0 = 2/(W + 1)$. The expected payload and MAC slot duration can now be calculated using the same method as the unicast analysis, but for a multicast network the per-station multicast saturation throughput is $S_1 = \frac{\tau_0 E_D}{E_T}$ for class 1 stations and $S_2 \frac{\tau_0 E_D}{E_T}$ for class 2 stations. The network sum-throughput is $S = n_1 S_1 + n_2 S_2$.

6 Theoretical performance

We use the models developed in the previous sections to compare the throughput performance of the uncoded and binary broadcast schemes. The models yield the throughput as a function of the channel error rate, i.e. the packet erasure rate for uncoded operation and the bit crossover probability when using coding. Combining these with data on channel error rate as a function of SNR/RSSI and PHY rate allows us to determine the optimal transmission PHY rates for downlink and uplink flows and obtain the maximum network throughput for a range of SNR/RSSIs. For

Table 1 MAC protocol parameters

T_{sifs} (μ s)	16	L_{subhdr} (bytes)	16	T_{ack} (μ s)	24
T_{phyhdr} (μ s)	20	L_{FCS} (bytes)	4	T_{difs} (μ s)	34
$T_{phyhdr1}$ (μ s)	36	L_{machdr} (bytes)	24	Retry limit	7
Idle slot σ (μ s)	9	CW_{max}	1024	CW_{min}	16

this purpose we use the experimental channel measurements shown in Fig. 4, which are taken from [5]. The experimental PEC capacity shown in Fig. 4 is for a packet length of 8640 bits. To obtain the PEC capacity for any other values of packet length, we need to first derive the first event error probability of Viterbi decoding for convolutional codes, which is given by

$$P_u = 1 - (1 - FER)^{(1/l)} \quad (21)$$

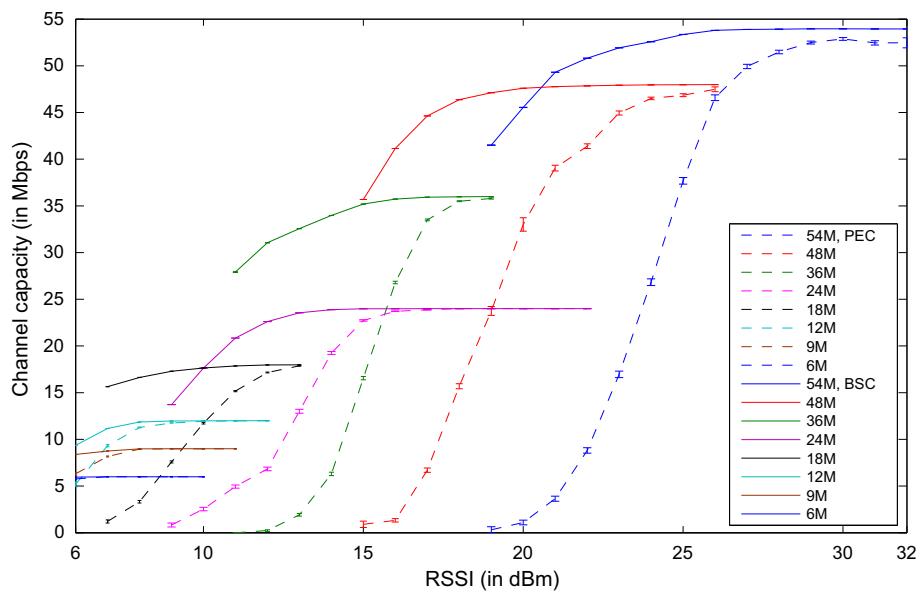
where FER is the measured packet erasure rate at a given RSSI, and l is the packet length used in the experiment, i.e. 8640 bits. Using this first event error probability P_e , the packet erasure rate for a packet length of L is, in turn, given by $1 - (1 - P_u)^L$.

The MAC parameters used are detailed in Table 1.

6.1 Unicast

We first consider unicast traffic. We compare the throughput performance for four different approaches: (1) uncoded; (2) time-sharing coding with the entire packet transmitted at a single PHY rate; (3) superposition coding; (4) time-sharing coding with segments transmitted at different PHY rates, i.e. segments destined to stations in class 2 are transmitted at the highest PHY rate available, which is 54Mbps in 802.11a/g, and the downlink PHY rate for class 1 segments is selected to maximise the network throughput. Figure 5(a) shows the sum-throughputs achieved by these different approaches for a network consisting of 20 client stations, 10 in class 1 and 10 in class 2. This is quite a large number of saturated stations for an

Fig. 4 Experimental measurements of outdoor BSBC capacity provided by 802.11 corrupted frames [5]. For comparison, also shown is the channel capacity when a packet erasure paradigm is adopted (packets failing a CRC check are discarded)



802.11 WLAN and suffers from a high level of collision losses. Comparing it with Fig. 4, it can be seen that the throughput is significantly reduced due to the various protocol overheads and collisions that have now been taken into account. Nevertheless, the relative throughput gain of the coding-based approaches compared to the uncoded approach continues to exceed 50% for a wide range of RSSIs. Time-sharing coding achieves very similar performance to the more sophisticated superposition coding. The approach of using different PHY rates for different time-sharing coding segments naturally achieves higher throughputs than using the same PHY rate. The gains are especially high at low RSSIs. This is because when the entire packet is transmitted using the same PHY rate, the optimal PHY rates for the uncoded and coded schemes are usually not very different, e.g. it is impossible that the uncoded scheme chooses 6Mbps but a coded scheme chooses 54Mbps. However if segments destined to distinct receivers are allowed to use different PHY rates, the optimal PHY rates for both schemes can be quite different, e.g. in our two-class example, the portion for class 2 always uses a quite high PHY rate of 54Mbps, while the portion for class 1 could use a very low PHY rate, especially at low RSSIs.

Figure 5(b) shows the corresponding results for a smaller number of client stations, 5 in class 1 and 5 in class 2. The overall throughput is higher than that with 20 stations because of the lower chance of collisions, and the gain offered by the coding approaches is even higher i.e. more than 75% over a wide range of RSSIs.

Figure 6(a) illustrates how the number of stations affects these results. The decrease in network throughput with

increasing number of stations is evident, as is the significant performance gain offered by the coding schemes. For smaller numbers of stations (which is perhaps more realistic), the throughput gain offered by the coding approaches is larger e.g. nearly up to 70% for 2 stations and falling to around 30% with 20 stations. The proportion of class 1 and class 2 stations can be expected to affect the relative performance of the uncoded and coded schemes. This is because we now have multiple transmitting stations, and each station defers its contention window countdown on detecting transmissions by other stations. Since class 1 transmissions are of longer duration than class 2 transmissions, we expect that the network throughput will rise as the number of class 1 stations falls and indeed we find that this is the case. See, for example, Fig. 6(b) which plots the network throughput versus the varying ratio of the number of class 2 stations over the total number while maintaining the total number of client stations constant as $n_1 + n_2 = 10$.

6.2 Multicast

For multicast, we compare the per-station throughput for the four aggregation approaches. Figure 7(a) shows the per-station throughput for a network with $n_1 = 10$ class 1 stations and $n_2 = 10$ class 2 stations. The throughput is much higher than the unicast case as shown in Fig. 5 because of the absence of collisions with uplink flows. Nevertheless, both of the coded schemes (time-sharing and superposition coding) continue to offer substantial performance gains over the uncoded approach, increasing throughput by almost 100% over a wide range of RSSIs. The superposition coding scheme performs slightly better

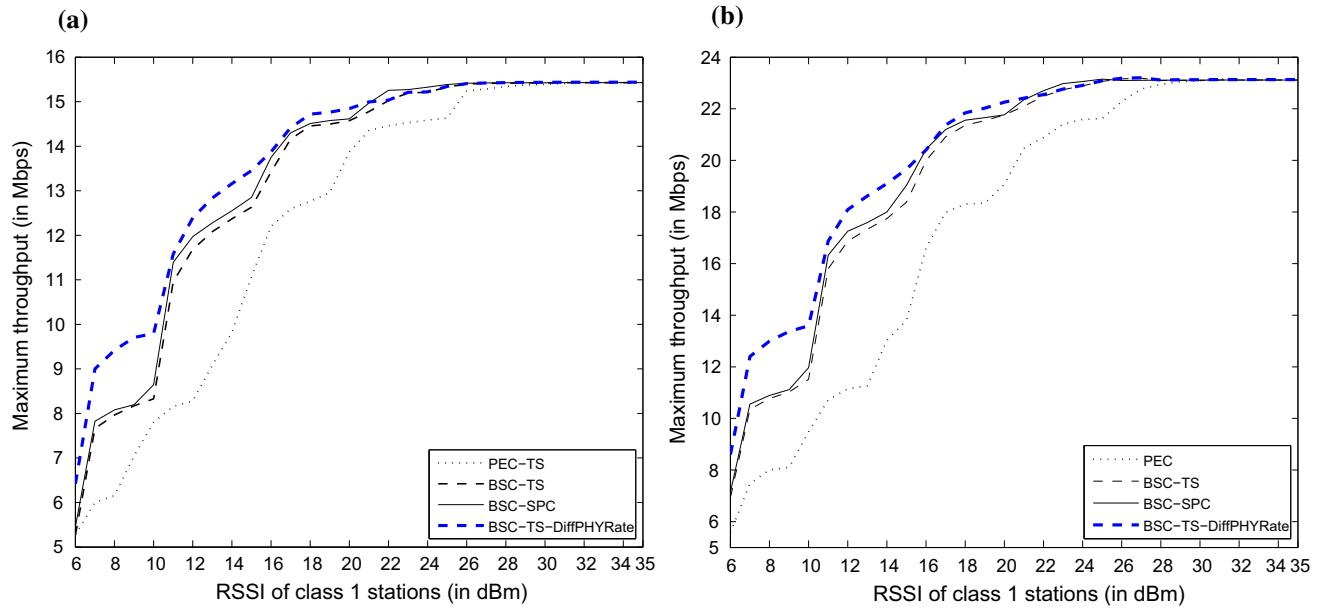


Fig. 5 Unicast maximum network throughput versus RSSI of class 1 stations, $L = 8000$ bytes. TS and SPC indicate time-sharing coding and superposition coding respectively. **a** $n_1 = n_2 = 10$ stations. **b** $n_1 = n_2 = 5$ stations

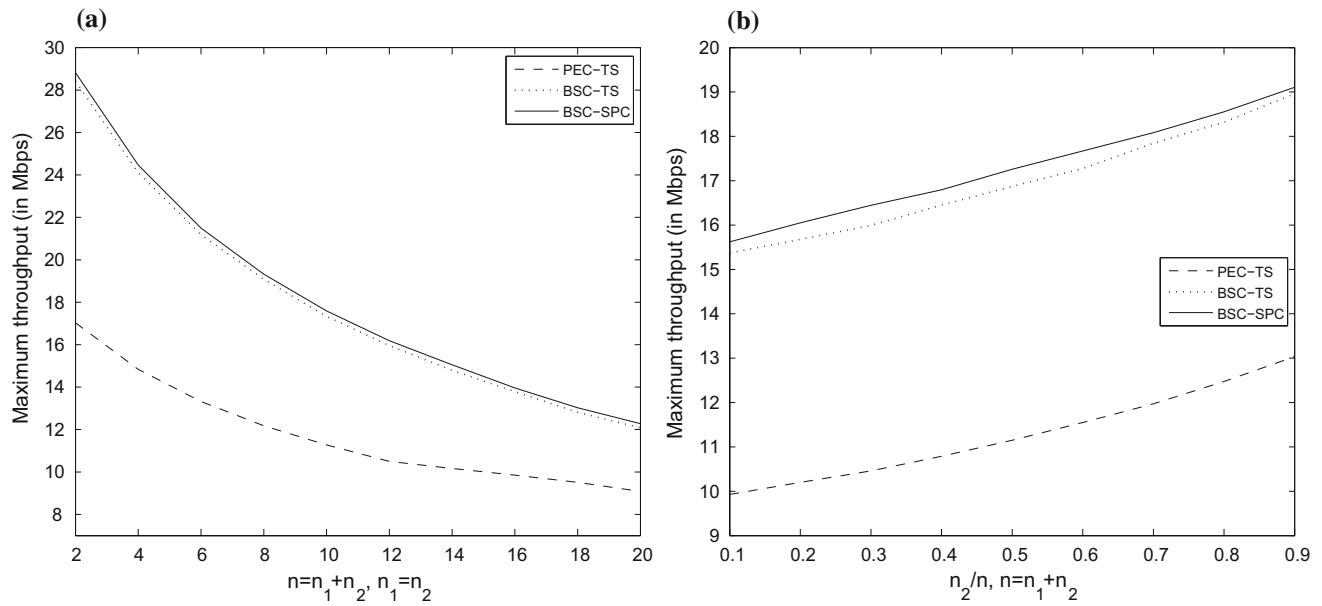


Fig. 6 Unicast maximum network throughput, $L = 8000$ bytes. TS and SPC indicate time-sharing coding and superposition coding respectively. **a** $n_1 = n_2$, $\text{RSSI} = 13$ dBm. **b** $n_1 + n_2 = 10$, $\text{RSSI} = 12$ dBm

than the time-sharing scheme, but the difference is minor. Figure 7(b) shows the corresponding results with a larger MAC frame size of 65536 bytes, which is the maximum frame size allowed in the 802.11n standard [12]. The performance gain offered by the coded approaches increases as the frame size is increased. Since the per-station multicast throughput is independent of the number of stations, we show results for only one value of n_1 and n_2 .

7 NS-2 simulations

The theoretical performance results presented in Sect. 6 consider the scenario where stations are saturated, and so there are always enough packets available to form maximum-sized aggregated packets. It can be expected that the impact of traffic arrivals and queueing strongly affects the availability of packets for aggregation. In some

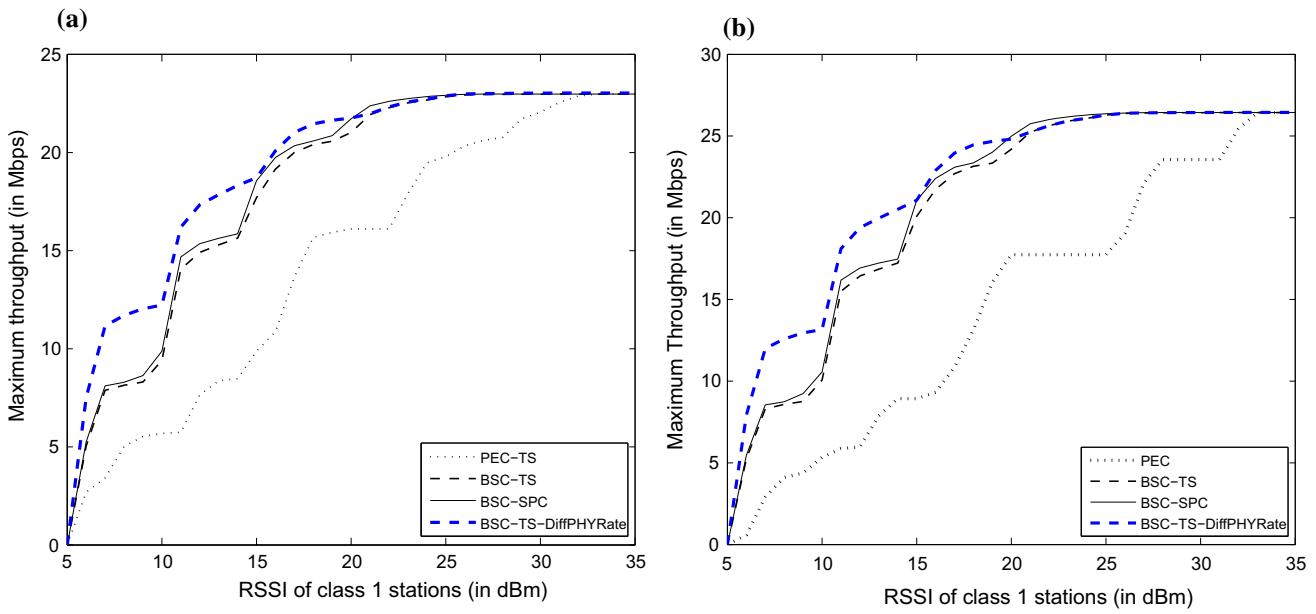


Fig. 7 Multicast per-station maximum throughput versus RSSI of class 1 stations, $L = 8000$ bytes, $n_1 = n_2 = 10$ stations. TS and SPC indicate time-sharing coding and superposition coding respectively. **a** $L = 8000$ bytes. **b** $L = 65,536$ byte

circumstances, stations may not have enough packets to allow the maximum-sized aggregated frames to be formed (achieving the highest aggregation efficiency). In this section we use the network simulator 2 (NS-2) to evaluate the benefits of the proposed schemes in unsaturated situations. It is worth noting that the NS-2 simulations in this section are not aimed to verify the throughput models and the performance results presented in Sects. 4, 5 and 6, as the theory analysis is based on the widely recognized Bianchi model which has already been thoroughly verified.

We use two metrics to evaluate the performance:

1. *Per downlink flow throughput (in bits/s)*: Let m_i denote the number of received packets of downlink flow i during the simulation duration t . The packet length is L in bytes. The throughput of flow i is thus $8Lm_i/t$. The per downlink flow throughput is the mean over all n downlink flows, which is $\sum_{i=1}^n 8Lm_i/(tn)$.
2. *Mean downlink delay (in seconds)*: We define the delay of a packet as the period from when it arrives at the InterFace Queue (IFQ) of the transmitter until it arrives at the MAC layer of the receiver. The mean downlink delay is the mean over all downlink packets. We use DropTail FIFO queues in our simulations.

7.1 Single-destination vs multi-destination aggregation

We begin by comparing uncoded multi-destination aggregation with single-destination aggregation. We consider a WLAN with an AP and N stations. The AP has n downlink

unicast flows individually destined to each of the N stations, and meanwhile each station has a competing uplink flow destined to the AP. Different from the two-class example described in Sect. 3.2, as we would like to emphasize the impact of packet availability to the two aggregation schemes, in this example we assume that all links are error-free. Downlink transmissions are large aggregated packets and uplink transmissions are normal 802.11 packets. As aggregated packets are quite long, we use the RTS/CTS exchange before data packets in our simulations. Again, we assume that the multi-destination aggregation uses the SMACK [8] scheme to allow receivers to send acknowledgments simultaneously, and hence there is only one ACK packet duration after each aggregated data packet. To ensure a fair comparison, for the single-destination aggregation, we assume that the receiver sends one ACK after each aggregated packet to acknowledge reception of data packets aggregated in that packet. The traffic is real-time stream data and follows a Poisson process with mean arrival rate of λ . The RTP/UDP/IP header is 40 bytes (IP = 20 bytes; UDP = 12 bytes; RTP = 8 bytes). The maximum aggregated frame size is 65535 bytes. The PHY data rate is 54 Mbps.

Figure 8 plots the per downlink flow throughput and the mean downlink delay versus the mean packet arrival interval ($1/\lambda$) for a WLAN with 10 stations. The traffic packet size is 500 bytes. The queue size is 100 packets. It can be seen that as expected the multi-destination aggregation achieves strictly higher throughput and lower delay than single-destination aggregation. When the mean packet inter-arrival time is above 0.008s (corresponding to a light

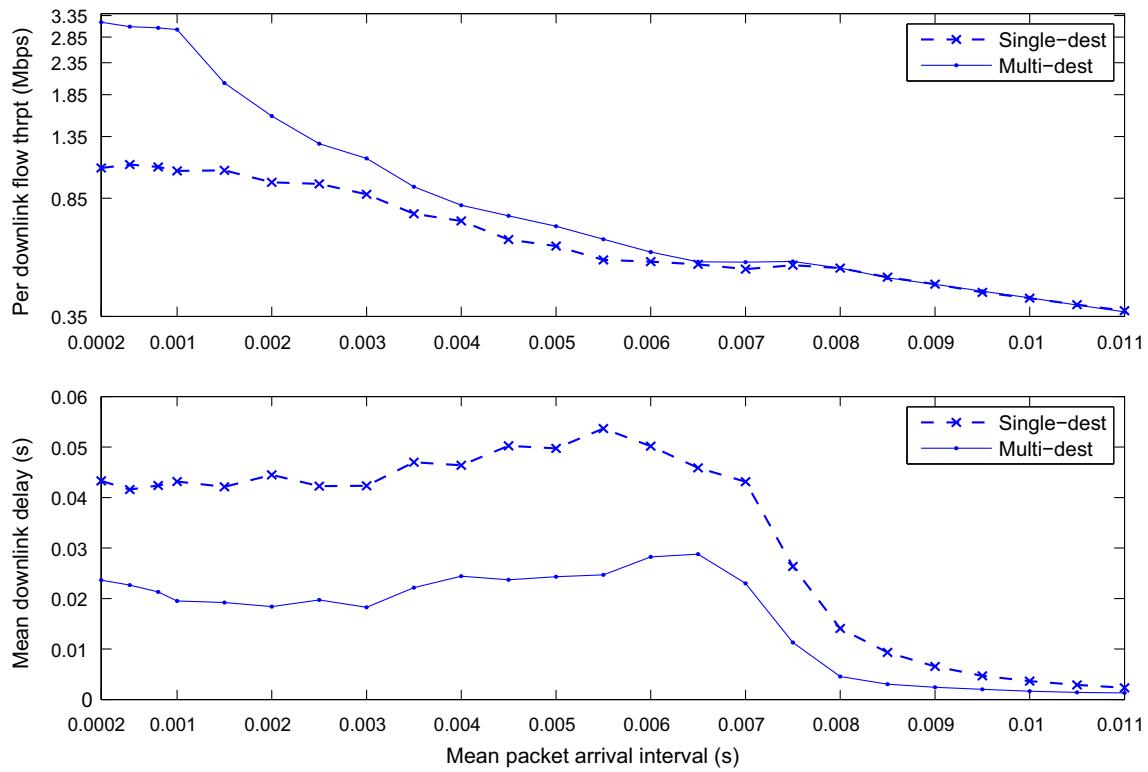


Fig. 8 Per downlink flow throughput and mean downlink delay vs. mean Poisson packet arrival interval, 10 stations, flow packet size 500B, queue size 100 packets

traffic load), both schemes achieve similar throughput. This is because in this range the AP is unsaturated, i.e. there is typically only one packet available to be aggregated. As the mean inter-arrival time is decreased (and so the traffic load increases), the AP queue starts to build up. The measured delay then includes the period awaiting in the queue. The multi-destination aggregation scheme tends to aggregate more packets in each transmission, and hence obtains higher throughputs and lower delays. When the mean inter-arrival time is decreased to 0.001s, the AP is saturated (the queue is persistently backlogged) and it can be seen that multi-destination aggregation achieves about a 200% increase in throughput over single-destination aggregation.

Figure 9(a) illustrates how the queue size affects the multi-destination aggregation throughput. Again there are 10 stations and the traffic packet size is 500 bytes. We compare three queue sizes of 50, 100 and 200 packets. It can be seen that when the AP becomes saturated, a larger queue provides a higher throughput because there are more packets available to be aggregated. When the queue of 200 packets is filled up, the aggregated packet reaches the maximum size limit and thus less than 200 packets are aggregated in one transmission.

Figure 9(b) plots the downlink flow throughput versus the mean packet inter-arrival time for two packet sizes of 500 and 1000 bytes. There are 10 stations and the queue

size is 200 packets. With single-destination aggregation, the throughput with a packet size of 1000 bytes is around twice that with a packet size of 500 bytes. This is because with the fixed queue size and packet arrival rate, the expected number of packets available to be aggregated is also fixed. However, with multi-destination aggregation, when the queue fills, both packet sizes obtain almost the same throughput because both reach the maximum aggregated frame size limit.

7.2 Uncoded versus coded approaches

In this section we compare uncoded multi-destination aggregation with the binary broadcast time-sharing coding scheme. We consider the two-class WLAN where both classes have the same number of stations. The AP has n downlink flows individually destined to each of the stations. There are no competing uplink flows. Flows are constant B it rate (CBR) traffic with a fixed packet size of 1500 bytes. The queue size is 500 packets.

Similarly to the theoretical performance analysis, we use experimental channel data shown in Fig. 4. We assume that class 1 has a RSSI of 12 dBm, and class 2 has a RSSI of 35 dBm. The uncoded multi-destination aggregation approach uses a PHY rate of 18 Mbps, while the coded approach uses a PHY rate of 36 Mbps (Note that this choice of PHY rates

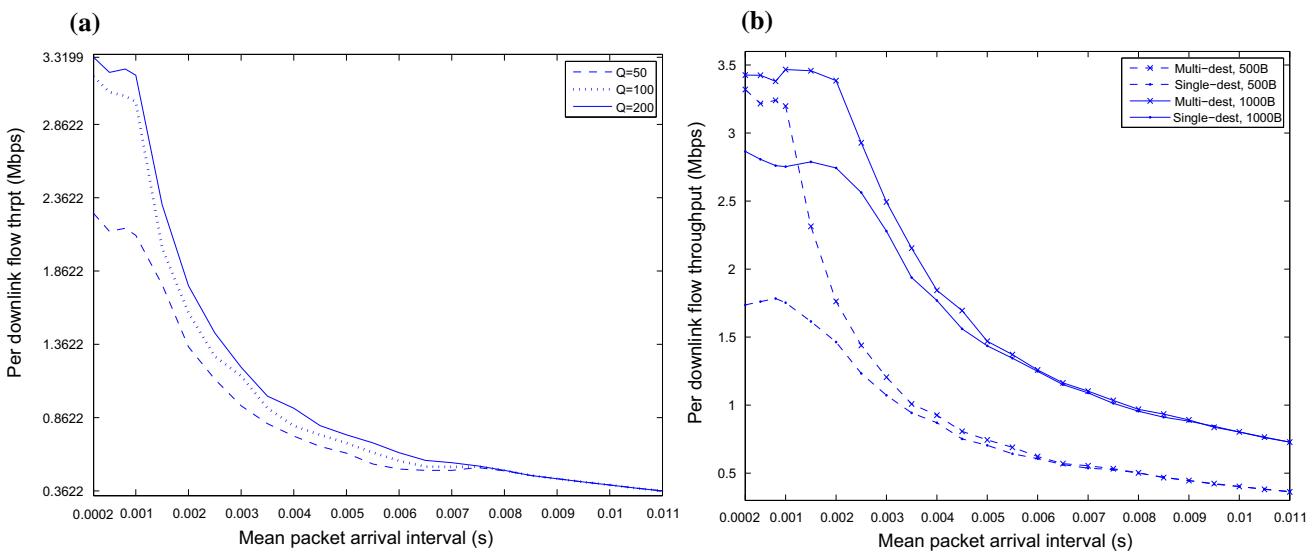


Fig. 9 Per downlink flow throughput versus mean Poisson packet arrival interval for **a** different queue sizes, **b** different packet sizes, 10 stations. **a** Flow packet size 500B. **b** Queue size 200 packets

is not necessarily optimal). The CBR traffic arrival rate is 1 Mbps.

Figure 10 plots the measured per downlink flow throughput as the number of stations is varied. As expected, the time-sharing coding scheme is strictly better than the uncoded scheme, achieving higher throughput and lower delay. For larger numbers of stations, it can be seen that the time-sharing coding scheme offers an almost 100% increase in per-flow throughput, and the mean delay is half that of the uncoded scheme. When the number of stations is below 28, the mean delay of time-sharing scheme is

extremely small. This is because when using the coded scheme with such small numbers of flows, the queue in the AP is mostly empty. As the number of stations increases above 28, the queue becomes backlogged and the mean delay (which includes the packet waiting time in the queue) starts to increase. In addition to the substantial increase in throughput and decrease in delay, we also find that with time-sharing coding the AP can support significantly more stations. It can be seen from Fig. 10(b) that with uncoded multi-destination aggregation the AP queue starts to become backlogged when the number of stations rises

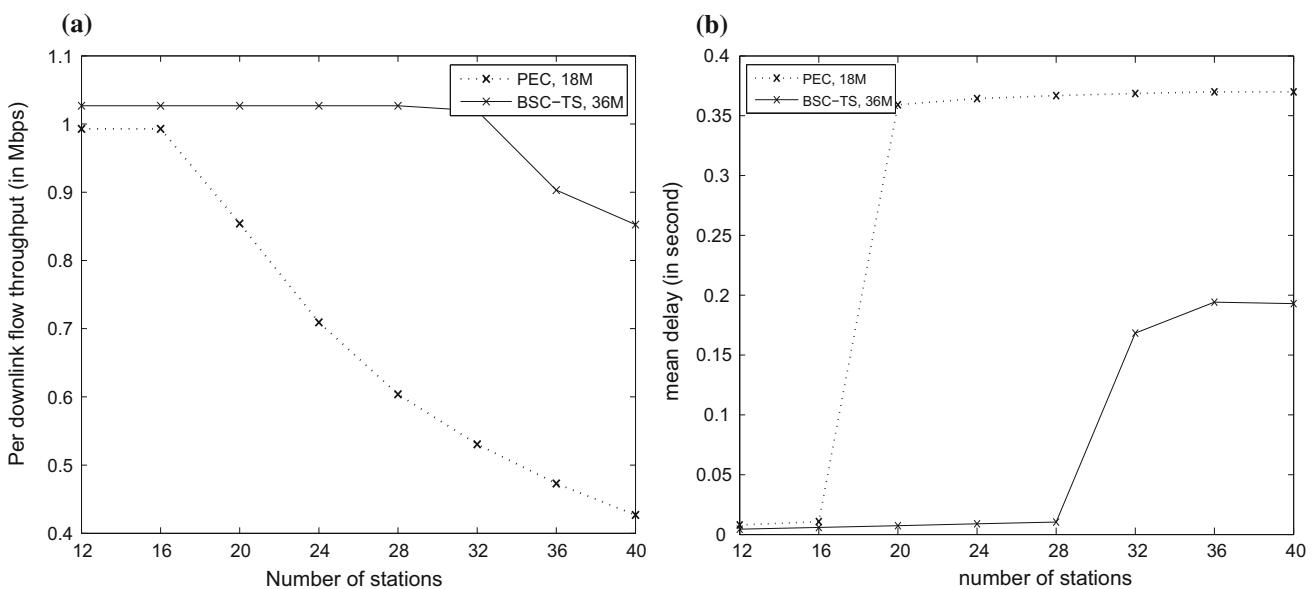


Fig. 10 Per downlink flow throughput and mean downlink delay vs number of stations, outdoor measurement channel data, RSSI of class 1 18 dBm, RSSI of class 2 33 dBm. **a** Per downlink flow throughput. **b** Mean downlink delay

above 16. In comparison, with time-sharing coding AP queue does not start to become backlogged until the number of stations increases above 28.

8 Discussions

8.1 Generalisation to a uniformly distributed error-prone WLAN

The analytical work in Sects. 4 and 5 can be generalised to a universal scenario where n stations are uniformly distributed over the area in a WLAN, and each of them has an independent error-prone channel. In the PEC paradigm, a transmission from the AP fails (i.e. AP doubles its contention window) only if none of the sub-frames is acknowledged by one of the multiple receivers. This could be caused by either a collision or noise-related erasures for all of the receivers. Similarly, a transmission from an ordinary station fails also due to collisions or noise-caused erasures. Thus, for both the AP and ordinary stations, the probability of a transmission fails is given by

$$p_{f_i} = 1 - (1 - p_{ci})(1 - p_{ei}), i = \{0, 1, 2, \dots, n\} \quad (22)$$

in which $p_{ci} = 1 - \prod_{j=0}^n (1 - \tau_j)$ and $p_{ei} = 1 - (1 - p_{ui})^{L_i}$ with p_{ui} the first-event error probability in Viterbi decoding and L_i the length of frame in bits. In the BSBC paradigm, since there are no noise-related packet erasures, packet losses are only caused by collisions, the transmission failure probability is $p_{f_i} = p_{ci}$.

Apart from the difference in the MAC throughput model, the calculation of the expected payload for each flow and the expected MAC slot duration is similar. If max-min fairness is considered, that is to equalise the flow throughput, following the same methodology for our two-class running example, the packet size for each downlink or uplink flow can be solved by combining the MAC model relationship Eqs. 4 and 22 with the specific packet organisation requirement in each scheme.

8.2 Extension to other fairness criteria

The proposed BSBC coding multi-destination aggregation schemes can be considered along with other fairness criteria, e.g. proportional fairness [6]. The analysis will be established on a utility function in terms of the fairness requirement and specific constraints. An analytical or numeral solution to achieve the fairness objective can be obtained by using some optimisation method. The analysis for other fairness criteria is beyond the scope of this paper. To implement the proposed schemes in more general

WLAN scenarios, we will consider different fairness criteria in the future.

8.3 Implementation on standard hardware

The present paper focuses on fundamental theoretical aspects. The experimental demonstration of a fully working system is out of scope. We nevertheless comment briefly on the compatibility of the proposed coded multi-destination aggregation schemes with existing 802.11n hardware. To implement multi-destination aggregation with time-sharing coding on standard hardware, a fairly direct approach would be to aggregate MPDUs destined to different receivers into an A-MPDU frame. Many 802.11 chipset drivers (e.g. atheros, broadcom) can be easily modified so as not to discard corrupted frames e.g. see [5]. Encoding/decoding of the MPDU payload could then be carried out by a shim within the driver, and this would be transparent to higher network layers. The 802.11 Block ACK functionality could be used to manage generation of MAC ACKs, or alternatively the 802.11 standard supports transmission of unicast packets with a “No ACK” flag set in the header and by using this ACKs could then be generated at a higher layer. A less efficient user-space approximation to this scheme that requires no driver changes could be to encode packet payloads in user-space and use TXOP bursting to send these packets in a back-to-back burst (albeit with higher overhead than A-MPDU aggregation). At the receiver, recent versions of the pcap API (or tcpdump) allow corrupted frames to be collected, where decoding could then take place in user-space. The channel state information (CSI) which is used for adaptive rate control at the physical layer needs to be passed upwards to the application layer for the AP to update the coding rate for each channel.

9 Conclusions

In this paper we consider the potential benefits of viewing the channel provided by an 802.11 WLAN as a binary broadcast channel, as opposed to a conventional packet erasure channel. We propose two approaches for multi-destination aggregation, i.e. superposition coding and a simpler time-sharing coding. We develop throughput models for these coded multi-destination aggregation schemes. To our knowledge, this provides the first detailed analysis of multi-user coding in 802.11 WLANs. Performance analysis for both unicast and multicast traffic, taking account of important MAC layer overheads such as contention time and collision losses, demonstrate that increases in network throughput of more than 100% are possible over a wide range of channel conditions and that the much

simpler time-sharing scheme yields most of these gains and have minimal loss of performance. Importantly, these performance gains involve software rather than hardware changes, and thus essentially come for “free”.

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