GRATIS: Free Bits In The Network

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Abstract

Conventional signal-to-noise (SNR) based rate adaptation algorithms in wireless networks provide limited flexibility in terms of the maximum achievable data rates. Transmitters are often forced to step down to a lower data rate due to the lack of an intermediate rate despite having a higher SNR than the minimum required for the lower rate. In this paper, we propose a novel scheme that provides increased flexibility to the MAC layer by providing multiple step down rates, while utilizing client channel diversity to transmit independent packets to multiple users: this is an example of *simultaneous transmission*. We achieve this by having a programmable baseband modulator producing constellations that provide increased network throughput while maintaining acceptable error rates. We call this scheme Group Rate Adaptation and Transmission with Intertwined Symbols, or GRATIS for short.

We have implemented this method using a testbed of reconfigurable radio nodes and demonstrated that this method improves average throughput while maintaining similar or lower packet loss compared to the conventional contention based transmission mechanism as well as other multiuser techniques such as superposition coding and hierarchical modulation.

1 Introduction

Modern wired, wireless and optical communication systems use different *modulation schemes* to balance data rates against error rates. Each modulation scheme encodes a varying number of bits in a physical representation of the data. Low rate modulations are used in a noisy channel and high-rate modulations are used when the signal to noise ratio is higher. For example, wireless systems such as 802.11 or WiMAX use the following modulations: BPSK (2 states, 1 bit), QPSK (4 states, 2 bits), 16QAM (16 states, 4 bits) and 64QAM (64 states, 8 bits).

In digital communication, these modulations are usually represented visually using a "constellation diagram", which is a collection of points in a 2-D plane representing the two components of a complex signal (the I and Q channels). Logically, a transmitter sends data by moving between the distinct states in the constellation plot and adjusting the I and Q components accordingly. A receiver attempts to map the received I and Q components to specific points in the constellation plane to decode the signal. Actual samples of a signal emerge as "point clouds" around the points in the constellation because channels are noisy. When decoding each modulation type, the constellation point that minimizes the error-vector magnitude is used.

The adherence of the received signal to the constellation is based on the signal to noise ratio (SNR). If there is either less signal (*e.g.*, through attenuation) or more noise (*e.g.*, through interference), a given modulation may have too many errors. Modulation rate control algorithms select amoung the different modulations to choose one that provides the highest data rate for a given SNR. The wide variety of SNR among different nodes in a wireless network make such discretized SNR based rate adaptation



Figure 1: Comparison with concurrent technologies

inaccurate and leads to non-optimal use of the medium. Thus, a more fine-grained control over the possible data rates can maximize the channel utilization.

In this paper, we show that by explicitly injecting "noise" in the message destined for a primary receiver, we can encode a secondary message for a secondary receiver, increasing the network overall bandwidth. In doing so we can also introduce multiple data rates to provide an even gradation of SNR based rate adaptation. Group Rate Adaptation and Transmission with Intertwined Symbols, or GRATIS as we call this scheme, provides "free bits" in the network while maintaining acceptable error rates over a wide range of SNR. Consider the three nodes in Figure 1; Charlie needs to send similarly sized packets to Alpha and Beta. The SNR for Alpha is higher (better) than that for Beta. In the standard 802.11 PHY and MAC, transmissions to Beta use a low-rate modulation scheme (e.g., BPSK) while those for Alpha are able to use a high-rate modulation (e.g., QPSK or 16QAM). The message for Beta takes longer to transmit than the message for Alpha; each message also needs to acquire the media using the 802.11 CSMA/CA method. In a pure TDMA protocols, much of the MAC overhead can be avoided, but the messages are still encoded using the two distinct packets. When using GRATIS, the message for Alpha is intertwined with that for Beta; when Charlie encodes the message for Beta, small deviations from the actual BPSK constellation are added. Those deviations actually represent the message for Alpha. Beta is only expecting a BPSK message – the rate adaptation algorithm always uses the highest rate modulation possible to optimize the overall bit error rate. Beta uses a maximum likelihood estimator to determine what states are being transmitted while the signal intended for Alpha is treated just as any other noise source. Since Alpha has a better SNR (because it is closer to Charlie), it can reliably decode those "errors" into distinct states.

For example, Charlie could be transmitting a message using a programmable constellation as shown in figure 8(a), which is a derivative of a 16QAM constellation. Alpha can clearly decode the four states – similar to QPSK, but Beta will collapse the four distinct states into two BPSK states. Figure 2(a) shows the theoretical BER over a noise channel (using an Additive White Gaussian Noise (AWGN) model) for such intertwined transmission. The error rate of the primary message sent to Beta is as good as BPSK (best effort rate for Beta), while that of Alpha is between the error rates for 16QAM and 64QAM. With sufficient diversity in the network, we can intertwine the message for Alpha, as free bits, while maintaining the maximum data rates achievable for Beta. Alpha may actually be able to decode an even higher rate modulation, such as 64QAM, but receives the message at a lower-than-optimal rate. The real benefit from this is that the message can be sent to Alpha at the same time as a message is sent to Beta, meaning no additional air time is needed. This technique can not be used to improve



Figure 2: Theoretical BER and throughput comparison for *Mode1* of GRATIS with conventional 802.11 data rates.(b) denotes the base layer and (g) denotes the GRATIS layer.

the throughput to Beta alone; if Beta was able to successfully decode a QPSK message, that would be the modulation that Charlie would use. The technique is possible because there are a discreet or fixed number of modulation schemes each of which encodes a fixed number of bits of data. Each scheme has *range* in which it offers the highest throughput based on the SNR and the resulting bit error rate (BER); we add noise that pushes the received signal to the periphery of the acceptable range and rely on channel diversity to provide more data rates to a wide range of user. The technique works best when stations have a diverse range of SNRs because that allows high-rate receivers to receive messages intertwined with low-rate messages. This has the added benefit of lessening a throughput unfairness that occurs in multi-rate systems such as 802.11 [8].

Using different analysis techniques we show that this scheme is both practical, profitable and implementable when compared to other multiuser schemes like superposition coding and hierarchical modulation. We summarize the key contributions of this paper as follows:

- By programming (reinterpreting) the constellations already available for 802.11, we present 6 new modes of transmitting data with varied degree of error performance giving more flexibility to rate adaptation algorithms. This is a distinct advantage over superposition coding based schemes which provides 2 or 3 discrete rates within the operating range of wireless nodes.
- Theoretical gain analysis of this scheme has been done to ascertain the benefits in a real networks. We have analyzed various 802.11 traces from SIGCOMM conferences to determine how diverse the SNR range of receivers are in practice and the potential benefit of this technique. We use that to estimate the potential performance improvement when the technique is applied. Performance improves both because two messages are sent at once and because fewer MAC media accesses are needed.
- GRATIS has been shown to outperform a competing method (superposition coding) both in theory and practice. We have implemented GRATIS using a testbed of SDR nodes and validated the theoretical gains using actual over-the-air packet transmissions. This also demonstrates the power of programmable hardware in providing unforeseen gains in wireless networks.

• The GRATIS method is easier to implement, making use of existing hardware modulation and de-modulation methods. Much of the demodulation can be handled by simple software, rather than more complicated fixed-function hardware.

In §2, we discuss prior work in this domain and place our work amongst contemporary techniques. In §3, we describe the GRATIS protocol in more detail. In §4 we conduct a basic bit-error analysis for the GRATIS protocol showing under what conditions the technique can be used. We then describe the hardware used to implement the technique in §5 and the results from our implementation in §6. §7 describes a (potential) gain analysis for GRATIS using 802.11 packet traces. The ability to send two messages at once can be used in a number of ways; we describe some of these in §8.

2 Related Work

In this paper we discuss a form of multiuser communication which utilizes information embedded within a modulation constellation. By defining the maximum likelihood decoding regions in the constellation, a signal can be decoded differently under different received SNR. We compare this scheme to other multiuser communications methods currently used in wireless networks.

The transmitted signal contains bits for two users and each user can receive, equalize and decode exactly in the same way as done for e.g., conventional 802.11a/g packets. We also utilize the exiting constellations in 802.11a/g and reprogram (reinterpret) them to encode multiple packets in one data stream. This ensures co-existence with commercial WiFi technologies.

Our implementation and analysis of GRATIS uses Orthogonal Frequency Division Multiplexing, although it is not dependent on OFDM. Simultaneous transmissions can also be detected by the multi-user detection scheme in CDMA. To reduce the complexity in correlating for different user codes, the authors in [22] use various heuristic methods to obtain a suboptimal solution. However, this complexity is absent in an OFDM based system, which is the basis for GRATIS. Multiuser communication using Orthogonal Frequency Division Multiple Access (OFDMA) is also prevalent in WIMAX [10] technology. OFDMA has also been introduced in cellular networks as a simultaneous communication mechanism, where separate contiguous sets of subcarriers are assigned for carrying multiuser data [23, 15]. Packets are decoded by multiple users by using a 2-dimensional map in time and frequency. In this way a significant amount of time is saved in contending for the shared wireless channel. However, GRATIS can still be useful for multi-user systems such as WiMAX as well our implementation based on WiFi. In both WiMAX and WiFi, low rate modulation schemes are used to transmit data to distant stations; our scheme simply uses the difference in signal-to-noise experienced by multiple receivers to encode additional data.

By intertwining two messages in GRATIS, we not only eliminate the media access time, but also save time over back-to-back transmissions by merging two packets into a single transmission. Thus, comparing to concurrent technologies our method uses already available but unexplored resources to provide enhanced performance, while keeping the design completely backward compatible with 802.11a/g nodes.

Another domain where constellations are used to transmit to multiple users is digital video broadcast. The authors in [17] discuss a layered modulation technique, often termed as *hierarchical modulation* or *multiresolution modulation*. The higher order bits in a dense constellation are used to decode a poorer quality signal that can be decoded by a user with low SNR while users with high SNR will be able to decode all the bits in the symbol. In the DVB digital television standard, 16-QAM and QPSK are used for hierarchical modulation. In [25], the authors put forward techniques to minimize error since

the euclidean distance for higher constellations gets smaller and requires higher SNR or stronger error correcting codes. Most of the work involving hierarchical modulation, including [24] and [21], finds its application in a multicast or broadcast environment of digital video.

Hierarchical modulation is used for transmitting the same information (*e.g.*, video signals) to multiple users. GRATIS can transmit completely different information to multiple users with similar or greater reliability. While prior works in related fields calls for optimizing the signal structure and the signal constellations, our effort is focused on harnessing the strengths embedded in unexplored areas of existing specifications, which makes it novel and yet compatible with coexisting technologies. Also, the wide range of operation of our method (more number of transmission modes over a range of SNR) puts our scheme in an unique position amongst its peers and predecessors.

Other packet mixing techniques in wireless networks either employ superposition coding [16, 18, 2, 7] or network coding [13, 11, 12, 14]. Superposition coding relies on iterative decoding by decoding the base layer (lower order bits) first and then re-modulating it to extract the higher layer (higher order bits); this is more complicated to implement, which is why it is difficult to find experimental data for the performance of superposition coding. Also, superposition coding offers less flexibility by limiting the SNR ranges where it can be used with acceptable error performance and often requires very high SNR as the constellation gets denser, which may not be available for any wireless node. In comparison GRATIS provides a simpler decoder structure and offers more flexibility by providing more data rates for the MAC to select among to merge multiple packets for receivers within realistic SNR ranges. Also under similar operating conditions, GRATIS provides higher network throughput compared to superposition coding. Network coding relies on overhearing of a packet which it used to decode a second packet encoded using network coding techniques. However, network coding fails if the first packet is not overheard by the intended receiver – under such a case the relay node falls back to multiple packet transmissions, which can be avoided by using GRATIS. Most 802.11 based communication is in the form of an AP with multiple clients. The AP acts as a router between the wireless clients and the wired back-end network. Network coding cannot be used in such environments since the routing medium is not common and packets cannot be overheard. In such cases GRATIS can be still used to merge downlink packets.

3 GRATIS: The Concept of Free Bits

In this section, we describe GRATIS. Discrete steps in the SNR requirements for each data rate forces the rate adaptation algorithm to fall back to a lower rate even if the node is reachable at a SNR higher than minimum required. In this scenario, we use GRATIS to introduce calculated deviations around a low rate modulation to encode additional bits for another node. This is done in such a way that the noise thus introduced in the signal does not decrease the SNR to a level that it is undecodable at the intended receiver. This extra modulation is a minor variation of constellation points, which is only decodable at a higher SNR. In this way, a node can combine two packets: without compromising the modulation rate of one of the nodes, while transmitting another packet to a node reachable at a higher SNR. We call the two layers the *Base Layer* (the layer decodable at lower SNR) and the *GRATIS Layer* (the layer decodable at higher SNR). Consequently, the second packet is transmitted without any extra airtime, and comes as free bits to the the receiver with a higher SNR – these free bits increase the aggregate throughput of the network.

We propose an encoding and decoding technique that requires very little change to a stand alone 802.11g transceiver. In order to introduce deviations to a constellation, we rely on identifying preconfigured clusters within the standard set of 802.11g constellations, *viz.* BPSK, QPSK, 16QAM and



Figure 3: Encoding and decoding of programmable GRATIS clusters derived from standard 802.11g constellations

64QAM. Depending on the amount of deviations or cluster size, the error performance of the base layer and the GRATIS layer varies. Typically, the clusters are derived from higher order constellations (16QAM and 64QAM), so that there is sufficient amount of deviation available to encode bits for the GRATIS layer. At the same time it has to be ensured that the deviations introduced do not reduce the performance of the base layer receiver. Therefore, gauging the error performance of any such cluster is of utmost importance before using it to merge packets. We discuss this in §4. Once identified, any arbitrary cluster can be realized by using only the constellations in the cluster, while suppressing other constellation points. These changes are easy to make on a software defined radio, and the overall complexity of the design is reduced.

In the 802.11 PHY layer, each message must be individually acknowledged. One major hindrance in using multiuser communications such as GRATIS is the need for those acknowledgments. For this we rely on a *simultaneous acknowledge mechanism* [5] to gather acknowledgments from multiple recipients of the merged packet.

3.1 Encoding Packets using GRATIS

We introduce *six* distinct "modes", or methods of combining packets, as shown in Figure 3. At the transmitter, two packets are encoded independently to the modulation subsystem. At this point, we combine the bits of the two packets to form a symbol in the chosen cluster. The modulator is programmed to map (modulate) these symbols to the I/Q plane using the modulation constellation from which the cluster has been derived.

For example, Figure 3(a) shows the constellation points used in *Mode1* and *Mode2*, which are derived from a 16QAM constellation. In *Mode1*, resultant cluster points are modulated to carry two bits of useful information, one bit for each layer. To reduce the probability of error in the base layer, the points are arranged similar to that of BPSK, with small deviations, which carries another BPSK packet in the GRATIS layer. Out of the 4 bits available for every constellation point in the cluster, bits b_0 is used to encode base layer, b_2 contains the GRATIS layer. The other two bits, b_1 and b_3 , remains constant at 0 and 1, respectively for correct cluster mapping. *Mode2* uses all the constellation points of a 16QAM

PLCP Header SIGNAL Symbol Service Field							>		
RATE	Reserved	LENGTH 1	Parity	TAIL	Scrambler Init	Reserved	Data of Base Layer	Dat	ta of Base Layer
4 bits	1 bit	12 bits	1 bit	6 bits	7 bits	9 bits	Variable bits		Variable bits
,	Merged 1 bit				Reser 5 bi	ved GRA	TIS Mode 4 bits	LENGTH 2 12 bits	+ Data of GRATIS Layer Variable bits
First OFDM Symbol					Second OFDM Symbol			Third OFDM Symbol	
Coded BPSK, R=1/2					Coded Rate = RATE			Coded Rate = RATE M	

 Merged: 1 indicates merged packet;
 RATE: Base Rate or Decode Rate of Far Node

 LENGTH 1: Length of Base Layer;
 LENGTH 2: Length of GRATIS Layer

RATE M: Rate after Merge; this rate is used by the near node to decode from third symbol to the end of packet,

then useful bits are extracted according to GRATIS Mode Information

Figure 4: Modified 802.11 PLCP header

constellation, and provides 2 bits of information per subcarrier, as is done in QPSK, to two nodes. Bits b_0b_2 and bits b_1b_3 are used to encode the information of base layer and GRATIS, respectively. As a result of such mapping when a cluster point that is closest to an axis overshoots the axes due to noise, symbol error occurs for the base layer, as seen in QPSK modulations but this event does not incur any error in the GRATIS layer. Hence, this type of mapping provides some error protection to the GRATIS layer.

Mode3 and *Mode4* utilizes 64QAM constellation to encode the two layers as shown in figure 3(b). *Mode3* provides 16QAM data rate to the base layer using bits $b_0b_1b_3b_4$, while *Mode4* provides QPSK data rate to the base layer using bits b_0b_3 . The remaining bits are used to encode the data of GRATIS layer. As in *Mode2*, the GRATIS in these cases have higher protection from error as crossing the base layer boundaries does not introduce any error in the GRATIS layer. *Mode5* and *Mode6* uses a cluster derived from a 64QAM constellation as shown in figure 3(c). *Mode5* uses bits b_0b_3 for the base layer and bits b_1b_4 for the GRATIS, providing QPSK data rate to both the packets. Bits b_2b_5 are modulated as 1, to generate the desired cluster as shown in dash-dotted lines. *Mode6* uses bits b_0 and b_3b_5 to encode the information of the base and GRATIS layers, respectively, while modulating bits b_1 as 0 and b_2 as 1 leading to the desired cluster points.

To successfully decode a GRATIS packet, a node needs to know the encoding information of the packet. We propose to utilize the reserved bits of the PLCP Header of IEEE 802.11a/g packet to provide the encoding information as shown in figure 4. The 1-bit reserved in SIGNAL symbol denotes whether the packet contains a GRATIS layer. The RATE field indicates the modulation rate at which the base layer is encoded. We use 4 bits out of 9 reserved bits in the Service field to include the mode of the GRATIS layer encoding, which can have values from 1 to 6. As in any unmerged packet of IEEE 802.11, the first symbol is modulated in BPSK with 1/2 rate coding. The second symbol is modulated in the rate specified in the RATE field of SIGNAL symbol. In this symbol, GRATIS layer information is embedded, which is used to demodulate from the third symbol onwards. The first 12 bits of the third symbol in GRATIS layer carries the length of the GRATIS packet.

3.2 Decoding The Free Bits

The encoding procedure ensures that decoding of the base layer is exactly same as decoding any generic 802.11g packet. The demodulation for the GRATIS layer changes from the third symbol onwards based on the information received in the 'GRATIS Mode' field in the second symbol of the packet, as shown in figure 4. Decoding the constellations to information bits is done using pre-defined thresholds called

decision boundaries. For the base layer the decision boundaries are governed by the RATE field while that of the GRATIS layer is defined by 'GRATIS Mode'. Once the constellation points are mapped to binary symbols, the corresponding bits are extracted by the GRATIS layer. For example, if the 'GRATIS Mode' field contains 6, the receiver programs the demodulation block to act as a 64QAM demodulator, which produces 6 bits per I/Q sample set. As encoded, the receiver extracts bits b_3b_5 to get back the bits intended for the GRATIS layer. In this way, by defining the cluster boundaries we can decode the extra packet while completely backward compatible with a legacy node that is unaware of any GRATIS packet transmission.

4 GRATIS: In Theory

In this section we evaluate the bit error rate (BER) for various modes in GRATIS, introduced in 3, in presence of AWGN. Constellation points in the I/Q plane are mapped to corresponding bits by using maximum likelihood (ML) decoding at the receiver. Constellation points are defined by a modulation dependent decision boundary to ensure error free decoding. The BER for such a scheme is given by 1.

$$P_B(E) = \frac{1}{2} \operatorname{erfc}\left(\sqrt{\frac{\Delta_E(i,j)}{4N_0}}\right) \tag{1}$$

The bit error rate for an arbitrary modulation scheme and ML decoding boundaries is upper bounded by 2, where,

$$P_{WUB}(E) = \sum_{j=1}^{M-1} \sum_{i \neq j} \frac{1}{2M} \operatorname{erfc}\left(\sqrt{\frac{\Delta_E(i,j)}{4N_0}}\right)$$
(2)

$$=\sum_{k=1}^{N} \frac{A_d(k)}{2M} \operatorname{erfc}\left(\sqrt{\frac{\Delta_E(k)}{4N_0}}\right)$$
(3)

Where,

- $\Delta_E(ij)$ = Squared Euclidean distance between two distinct constellation points *i* and *j*.
- $N = N \le M(M-1)/2$ possible different squared Euclidean distances in the *decoded* constellation.
- M = Total number or *decoded* constellation points.
- N_0 = Additive white noise power.
- $\Delta_E(k)$ = Distinct pairwise Euclidean distance in the *decoded* constellation.
- $A_d(k)$ = Number of signal pairs having squared Euclidean distance of $\Delta_E(k)$.

Using equation 2 we can compute the BER for any constellation and ML decision boundary. An example BER computation for the base layer of *Mode2* has been shown in section A.

In the theoretical BER computation we consider that the constellations are mapped using gray code and encoded using a half rate convolution code as done in standard 802.11 transmissions [9]. This analysis has been done to ascertain the operating range of different modes and their suitability to merge



Figure 5: BER for the base layer of all the modes with AWGN channel, Gray code mapping and half rate convolution coding

multiple packets using GRATIS. The BER performance for the GRATIS layer is as important as the base layer because it allows the MAC to identify users with suitable SNR that can decode the bits from the enhanced GRATIS layer. The BER computation for the GRATIS layer is similar to that of the base layer and can be easily calculated using equation 2. We discuss the performance of the enhanced layer using a testbed in §6.

From Figure 5 we find that the base layer of *Mode1* and *Mode5* requires a SNR between QPSK and BPSK. Hence these modes can be used to modulate signals for nodes whose SNR are lower than that required by QPSK. Similarly, *Mode6* and *Mode2* offers two step down rates for nodes not reachable with 16QAM but have higher SNR than QPSK. While *Mode3* offers similar flexibility by stepping down to an intermediate data rate instead of 16QAM, the benefits from *Mode4* can be seen when used in conjunction with the GRATIS layer : providing a combined bit rate equal to that of 64QAM, which none of the two nodes would have been able to achieve with independent packet transmissions. Thus, the primary node is always transmitting at the highest possible rate, while providing some extra information (a complete packet) to another node without any extra overhead. We exploit this property in GRATIS to increase network throughput. We validate the theoretical results using a testbed of wireless nodes in §6 and we apply the different modes of operation in realistic network traffic in §7.

5 GRATIS: Implementation in SDR

To implement GRATIS, we have used a programmable hybrid software defined radio (SDR) using a Virtex-IV FPGA [6, 4]. The radios are capable of transmitting and receiving generic 802.11a/g packets as given in the physical layer specification [9]. The baseband processing engine is connected to a front-end radio operating in 2.4GHz ISM band as shown in fig. 6. Since we have implemented GRATIS using a 802.11g physical layer, backward compatibility is of utmost importance. The high degree of programmability of the underlying architecture allows for creating arbitrary programmable constellations



Figure 6: FPGA board with radios and antennas

Figure 7: Demodulator block in the receiver

that are easily derived from constellations supported by 802.11g. Similarly at the receiver, it is our goal to make the receiver of the base layer to be oblivious of any enhanced layer. It is the GRATIS layer receiver who needs to have additional controls to decode its packet.

As detailed in section 3.1 the PLCP header has been modified to accommodate the extra OFDM symbols to transmit the information required by the enhanced layer. By looking for Overlap bit and the Mode in the signal symbol the receiver decodes GRATIS layer. If the Overlap bit is asserted then the packet length information for the enhanced layer is found in the next OFDM symbol which is decoded using the rates specified by the *Mode* bits, otherwise the packet is decoded using the base rate provided in the *rate* field of the signal symbol. Depending on the GRATIS mode used to encode the second packet, the demodulator either needs to change its ML decision thresholds or extract the additional bits: typically termed as bit slicing or bit bashing. Figure 7 shows the basic structure of the modified demodulator. The bit basher unit is responsible for slicing the bits of the GRATIS layer. Since the two packets are encoded and packetized using identical but independent MAC processes, it is important that the receiver, performs the decoding with the correct parameters: de-interleaver and FEC decoder. Other receiver subsystems prior to the demodulator, the e.g., synchronizer and equalizer remains unchanged for implementing GRATIS. The equalizer always aims to restore the original transmitted constellation, while it is the decoder that decides either to interpret it as a standard packet or a combined packet using GRATIS. Apart from decoding packets, the receiver also reports the average receive SNR of a packet and also performs MAC CRC checks in the hardware to measure packet loss.

Figure 8 shows the programmability of the SDR to produce different constellations that supports multiuser communication using GRATIS. Figure 8(a) shows the constellation for *Mode1*, which is derived from a 16QAM constellation by suppressing all other constellation points. Similarly, figure 8(b) and 8(c) shows a modified 64QAM constellation that provides a combined network throughput of 4 bits/OFDM subcarrier (2 for base and 2 bits for GRATIS layer) and 3 bit/OFDM subcarrier (1 for base and 2 bits for GRATIS layer) respectively. Instead of designing specific I/Q mappers (modulators) at the transmitter to change the I/Q components of a multiuser constellation, we use a programmable modula-

Figure 8: Programmable constellations using SDR

tor that allows us to choose the appropriate set of I/Q vectors required by GRATIS protocol from a set of already available vectors (802.11 constellations). This requires no change to the underlying hardware and makes the system completely backward compatible.

One key advantage of using SDR in this scenario is that the binary operations that follows the demodulator can be efficiently performed in software as well as allowing the underlying hardware to remain unchanged for this kind of an implementation. This is a distinct advantage over other multiuser decoding techniques that rely heavily of complex signal processing algorithms, best performed using custom hardware. Hence, GRATIS provides an example of harnessing the power of programmable hardware in innovating new and improved protocols.

6 GRATIS: Putting it to Work

In this section we present experimental results using GRATIS and evaluate how this protocol can be used to increase the aggregate throughput of the network. We have implemented a testbed of three nodes with one transmitter and two receivers in indoor laboratory environment. The results shown are the average throughput and packet error rate (PER) of various node arrangements used to create sufficient variability in SNR and multipath fading. Each of the modes proposed in §3 have been compared to the standard rates available in 802.11g.

Figure 9 shows the results of the base and GRATIS layer of 6 Modes discussed in §3, along with standard modulations, BPSK, QPSK, 16QAM and 64QAM. For each modulation, we plot the physical layer throughput and mark the minimum SNR required for a PER of 2%. Maintaining acceptable error rates while maximizing throughput is important, as it might lead to unwanted retransmissions consuming additional airtime and reducing the throughput of the network. These marks are the SNR requirements for each modulation, below which it cannot be used reliably. The base and GRATIS layers are denoted by suffixes '(b)' and '(g)' respectively.

Mode1(b) and Mode6(b) are two modes that provide BPSK rate, and has SNR requirement between BPSK and QPSK. Thus, when a node becomes unreachable in QPSK, we can use these two modes to combine packets, and send a packet to another node reachable at higher SNR: at BPSK rate to node reachable at 17dB or higher using *Mode1* and at QPSK rate to a node reachable at 21dB or higher if using *Mode6*.

Figure 9: Experimental results showing throughput

Mode2 and *Mode5* provides QPSK data rate to both the layers, and have SNR requirements in between QPSK and 16QAM. The presence of these two modes provides more flexibility to choose the GRATIS mode in a wider range of SNR. If SNR of a node is more than 7.5dB but less than 10.5dB, and that of another node is more than 23.5dB, *Mode5* can be used effectively, but *Mode2* cannot be used in this SNR range. Similarly, if a far node has SNR between 10.5dB and 13dB, while another near node has SNR between 15dB and 23.5dB, we cannot use *Mode5* for the near node, but can successfully encode the packets using *Mode2*.

Mode3(b) provides data rate equal to that of 16QAM, and has SNR requirement in between 16QAM and 64QAM. Evidently, this mode can be used whenever the SNR of a node falls below the SNR requirement of 64QAM providing the the best effort data rate, 16QAM in this case, for that node. *Mode4(b)* provides throughput equal to that of QPSK, but requires more SNR than 16QAM to be decoded correctly. Apparently it might seem that this mode does not provide the best effort rate to the far node. Careful observation reveals that *Mode4(g)* provides a throughput is equal to that of 16QAM, and has lower SNR requirement than 64QAM. So, we can transmit a packet in *Mode4(g)* to a node when its SNR falls below 64QAM, and *Mode4(b)* then can be used to transmit any extra bits as a GRATIS layer.

Combining packets using *Mode2* is similar to superposition coding (SC), where both the layers can receive QPSK data. To compare our results, we have also implemented SC using our hardware and measured the throughput and PER with the same node arrangement. 'SC-70%' in figure 9 denotes the performance of SC which allocates 70% of the energy to the first layer, and remaining 30% of the energy to the second layer. This is most commonly used [2] energy allocation ratio used in SC. *Mode2* of GRATIS outperforms 'SC-70%' in terms of minimum SNR requirement by 6.5dB for the base layer and 3dB for the enhanced or the GRATIS layer. However, we noticed that if the energy allocation ratio is 80 : 20 to the two layers, the mapping thus generated resembles that of 16QAM, which is used to encode *Mode2*. Even using the same mapping we notice that *Mode2* performs better than 'SC-80%'. This improvement is attributed to the mapping of two layers onto a 16QAM constellation. If the 16QAM constellation is mapped using Gray Code and then packets are merged using *Mode2*, the bits of the GRATIS layer are protected from errors due to symbols crossing either axes due to channel noise. This unique mapping technique provide noticeable improvement over SC. Therefore we see that, GRATIS provides a more fine-grained control over the SNR-throughput space while outperforming other contemporary multilayer techniques.

7 GRATIS: Practical Gains

Experimental results in §6 show the potential benefits of using GRATIS in modern networks like 802.11 and WiMax. We are interested to know what gains GRATIS can achieve over TDMA, or in applications like WiMax. Figure 10 shows the gain in aggregate throughput of two users in the network, whose SNR varies from 5dB to 40dB. Aggregate throughput is calculated based on a packet size of 1024 bytes, and the minimum SNR required for a particular data rate is ascertained from the results in §6. For GRATIS, we have considered that a merged packet can be transmitted after a DIFS interval, while for TDMA-like transmission, two packets are transmitted one after another with an initial DIFS time. Results show improvement of up to 47% in aggregate throughput using GRATIS. The gain is obtained from frequent packet merging when the SNR of two users are varied, which leads to higher gains in the throughput.

It is sometimes difficult to understand the benefit of a particular wireless optimization from BER plots and bench experiments. How often do stations have sufficient rate diversity to exploit GRATIS? How useful is the combined coding efficiency and reduced MAC access time? To determine the *system* benefit of GRATIS, we analyzed captured packets in a SIGCOMM 2008 dataset [20], containing packets

Figure 10: Percentage gain over TDMA in variable SNR

Figure 11: Data rate variation in AP transmissions

transmitted by an AP to multiple clients of the conference attendees. In infrastructure networks, stations only transmit to the access point and not to other stations; thus, stations can not use GRATIS. However, GRATIS can be used to merge downlink data packets. We have only chosen the packets transmitted from the AP in our trace analysis. Benefits for this protocol is maximized when there is a wide variety of SNR available among the clients, which facilitates packet combination at different rates. Although the traces did not provide the SNRs at which the clients are reachable from AP (this requires packet monitoring at each client) from figure 11, we notice that there is a wide variety of data rates used by the AP to transmit packets – as captured by 8 monitors on the first three days. On the fourth day, we noticed very low volume of data packets compared to other days, and most of the packets were transmitted in BPSK modulation. Instead of trying to analyze the reason for this inconsistency we focus on the first three days of trace to analyze possible gains in using GRATIS in 802.11 network.

We selected one random monitor on one of the three days, Monitor 4 on Aug-19, and computed the airtime gain achievable per min using our protocol. It would have been optimal to combine packets based on SNR at the clients. Since we do not have any SNR reported, we assume that if a packet is transmitted at a modulation rate, then the SNR is in between $\pm 2dB$ of the minimum SNR required for that modulation, and any GRATIS mode that operates within that range can be used to encode another packet decodable at higher SNR. This way, we are limited to using only Model, Mode2 and *Mode4* instead of possible 6 as mentioned in $\S3$. Using these modes, the base layer is reachable at BPSK, QPSK and 16QAM respectively, while the GRATIS layer is decodable at 64QAM, 16QAM and 64QAM respectively. We assume a lookahead buffer of just 10 queued packets to determine the possible combinations. The airtime requirement in 802.11 not only depends on the packet air-time, but also on the medium access time, which equals the DIFS time and a random backoff time. We used SIGCOMM 2004 dataset [19] to estimate an average medium access time per packet, which equals $2730\mu s$. Figure 12 shows the temporal variation in percentage gain in airtime per minute if GRATIS is used. The volume of data packets transmitted per min, and the normalized amount of packet transmitted per min in each modulation is also shown. Percentage airtime gain has been computed for two cases: considering only the DIFS time wait time of $34\mu s$ and including the idle time of $2730\mu s$ along with DIFS. Simple moving average with 10 data points, shows that the gain in airtime is more when we also consider the idle time along with DIFS. Significant amount of time is spent in medium access, which will be reduced if GRATIS can be used in 802.11 network. We notice that the volume of packets drop significantly at time ≈ 100 mins, which could probably be the lunch hour, and most of the packets were transmitted in BPSK. GRATIS did not find much opportunity to combine packets as there was limited variety in the modulation of the data packets.

This analysis shows that when the clients' SNR vary in a diverse range, GRATIS can be used to combine packets and gain airtime, which can essentially be used to transmit more packets and increase overall throughput of the network. To ensure that we receive similar gains in other days using the trace from other monitors as well, we computed average gain in each day for each monitor, as shown in figure 13. Results show consistent gains in other scenarios as well.

8 Discussions and Future Work

This paper presents GRATIS but also opens up a new set of open research problems. While GRATIS provides more flexibility by providing different intermediate rates and merging multiple packets to enhance the network throughput, the combinations cannot be chosen randomly while also ensuring good performance. Careful inspection has to be made before selecting any combination. We present two modes *Mode7* and *Mode8* shown in figure 14(a) and 14(b) respectively, chosen randomly from QPSK

Figure 12: Gain in airtime using GRATIS

Figure 13: Percentage gain over multiple days and monitors

Figure 14: Random selection of constellation points

and 16QAM constellations. The throughput and SNR requirement of these two modes are shown in figure 14(c). Clearly, the base layer of each of the modes provide BPSK data rate but requires a SNR as good as QPSK. Hence, these two modes cannot be used to yield improved aggregate throughput as the base layer has a sub-optimal data rate.

In our implementation and analysis discussed in this paper, we have used 1/2 rate convolution coding. Considering a 3/4 coding rate we can obtain a similar set of results but at presumably higher SNR, where the base layer and the GRATIS layer will have a 3/4 coding rate. By using all the basic rates and their GRATIS derivatives, we can have almost a linear relationship between SNR and achievable throughput, instead of the conventional step-wise discrete rate allocation. Therefore, from a MAC layer point of view, GRATIS inspires us to delve into the modalities of a novel rate adaptation algorithm for wireless networks and comprare with related SNR-based rate adaptation algorithms [26, 3]. Also in wireless mesh networks we can forward packets to multiple users from a common point (router/relay node) using GRATIS. This is particularly helpful when the involved node-pairs cannot over-hear each others transmission and hence cannot use network coding. We also intend to explore the possibilities of using GRATIS in improving protocols of higher layers like opportunistic routing algorithms.

9 Conclusion

GRATIS provides an efficient method of simultaneous packet transmission and reception. This increases the network throughput without compromising the throughput of one of the nodes while using widespread channel variability to simultaneously transmit an independent packet destined for another node. The GRATIS packet is indeed some extra free bits to the high SNR node, which it would have received after the completion of the first packet using a serialized medium access pattern as in 802.11. We have implemented the protocol in hardware and have shown the ease of implementation if the signal processing is done using a hybrid platform of software and hardware components. The experimental results show several possibilities of use of GRATIS giving unforeseen gains in throughput in wireless networks. Applying GRATIS on real-time packet trace analysis reveals that even with a few simple combinations, we can gain significant airtime, which can be further utilized to transmit more packets. Also through our analysis we show that GRATIS provides more flexibility with better error performance than other contemporary simultaneous packet transmission techniques, making it a suitable candidate for emerging wireless networks.

A Example BER calculation

In this section we present a worked out example of BER calculation for an overlapped modulated packet. In GRATISthe transmitted constellation is decoded using a different set of decision boundaries which yields a constellation of smaller size. For example, a transmitted constellation of 16QAM can be decoded as a QPSK as done in *Mode2* base layer. In such a case the BER for the base layer will be between QPSK and 16QAM. For a 16QAM constellation the possible constellation point sets are $\{\pm A, \pm A\}$, $\{\pm 3A, \pm A\}$, $\{\pm A, \pm 3A\}$ and $\{\pm 3A, \pm A\}$. A is a modulation dependent parameter given by $\sqrt{\frac{K_b E_b}{K_{mod}}}$, where K_b is the number of bits per symbol, E_b is the energy per bit and K_{mod} is a scaling factor so that all constellation points have unit energy: for 16QAM this scaling factor is $\sqrt{10}$ [1]. Thus $A = \sqrt{\frac{K_b E_b}{10}} = \sqrt{\frac{E_s}{10}}$, where E_s is the energy per symbol. Figure 3(a) shows a 16QAM constellation and the decision boundaries for a QPSK decoding. The constellation points are marked [0...15]. We recall from section 3, that for the base layer a jump from one signal point to another within a quadrant will not cause any bit error since all the points in one quandrant is mapped to one QPSK constellation

point. Therefore, we start by identifying the pair-wise squared Euclidean distances between transmitted constellation points that will cause a bit error.

$$\Delta_E(0,3) = 36A^2 \tag{4}$$

$$\Delta_E(0,15) = 36A^2 \tag{5}$$

$$\Delta_E(1,2) = 4A^2 = \Delta_E(7,8)$$
(6)

$$\Delta_E(1, 14) = 36A^2 = \Delta_E(7, 4) \tag{7}$$

$$\Delta_E(6,5) = 4A^2 \tag{8}$$

$$\Delta_E(6,9) = 4A^2 \tag{9}$$

Now, we can compute the conditional error probabilities of the points in the top-right quandrant viz. 0, 1, 6, 7 using eq. 1 as follows,

$$P_{WUB}(E|\vec{I}=0) = \frac{1}{2}\operatorname{erfc}\left(\sqrt{\frac{\Delta_E(0,3)}{4N_0}}\right) + \frac{1}{2}\operatorname{erfc}\left(\sqrt{\frac{\Delta_E(0,15)}{4N_0}}\right)$$

$$= \operatorname{erfc}\left(\sqrt{\frac{9A^2}{N_0}}\right)$$
(10)

$$P_{WUB}(E|\vec{I}=1) = \frac{1}{2} \operatorname{erfc}\left(\sqrt{\frac{\Delta_E(1,2)}{4N_0}}\right) + \frac{1}{2} \operatorname{erfc}\left(\sqrt{\frac{\Delta_E(1,14)}{4N_0}}\right)$$
(11)
$$= \frac{1}{2} \operatorname{erfc}\left(\sqrt{\frac{A^2}{N_0}}\right) + \frac{1}{2} \operatorname{erfc}\left(\sqrt{\frac{9A^2}{N_0}}\right)$$
$$P_{WUB}(E|\vec{I}=6) = \frac{1}{2} \operatorname{erfc}\left(\sqrt{\frac{\Delta_E(6,5)}{4N_0}}\right)$$
(11)

$$+\frac{1}{2}\operatorname{erfc}\left(\sqrt{\frac{\Delta_E(6,9)}{4N_0}}\right)$$

$$=\operatorname{erfc}\left(\sqrt{\frac{A^2}{N_0}}\right)$$
(12)

Combining eq. 10, 11 and 12 we get,

$$P_{WUB}(E) = \frac{4}{16} P_{WUB}(E|\vec{I}=0) + \frac{8}{16} P_{WUB}(E|\vec{I}=1) + \frac{4}{16} P_{WUB}(E|\vec{I}=6) = \frac{1}{2} \operatorname{erfc}\left(\sqrt{\frac{9A^2}{N_0}}\right) + \frac{1}{2} \operatorname{erfc}\left(\sqrt{\frac{A^2}{N_0}}\right)$$
(13)

Substituting the value of A,

$$P_{WUB}(E) = \frac{1}{2}\operatorname{erfc}\left(\sqrt{\frac{9E_s}{10N_0}}\right) + \frac{1}{2}\operatorname{erfc}\left(\sqrt{\frac{E_s}{10N_0}}\right)$$
(14)

Similarly, BER plots can be obtained for any constellation and any decision boundary. We plot the theoretical BER results with respect to varying symbol to noise ratio (E_s/N_0) in figure 2(a)

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