ADAPTIVE CODING AND TRANSMISSION PARADIGMS FOR WIRELESS CHANNELS

A Light-Hearted Overview

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ABSTRACT

Following a brief historical perspective on channel coding the concept of near-instantaneously adaptive wireless transceivers is introduced as a counter-measure of mitigating the channel-quality fluctuations experienced in wireless communications. It is argued that channel coded adaptive modulation schemes can be viewed as a lower complexity alternative of mitigating the channel quality fluctuations of wideband wireless channels in comparison to multiple-transmitter and multiple-receiver based space-time codes. However, provided that the complexity of the latter schemes employing multiple transmitters and receivers is affordable, the performance advantages of adaptive modulation and adaptive channel coding schemes erode, since the channel quality fluctuations of the wireless channel are effectively mitigated.

1. BRIEF HISTORY OF CHANNEL CODING

Following Shannon's ground-breaking predictions outlining the mathematical foundations of information theory [1], during the 1950s coding theory researchers embarked on contriving coding schemes that are capable of approaching Shannon's predictions. Fulfiling this ambition required about 45 years of intensive research by the entire channel coding community, which culminated in the invention of turbo codes by Claude Berrou and his colleagues [2, 3]. The history of channel coding research spanning over half a century is summarised in Figure 1 [4], showing particularly intensive activities during the 1990s, although the individual contributors are too numerous to name here.

In parallel with the impressive developments in the processing capabilities of the chips dedicated to channel encoding and decoding, over the years more and more complex solutions have been invented for various commercial and professional applications in both storage and transmission. In this light it became clear that the conception of channel coding and transmission schemes dedicated to a particular application is based on a complex set of contradictory design factors, which are summarised in Figure 2 [4].

Convolutional FEC codes date back to 1955 [5], which were discovered by Elias, while Wozencraft and Reiffen [6, 7], as well as Fano [8] and Massey [9] proposed various algorithms for their decoding. A major milestone in the history of convolutional error correction coding was the invention of a maximum likelihood sequence estimation algorithm by Viterbi [10] in 1967. A classic

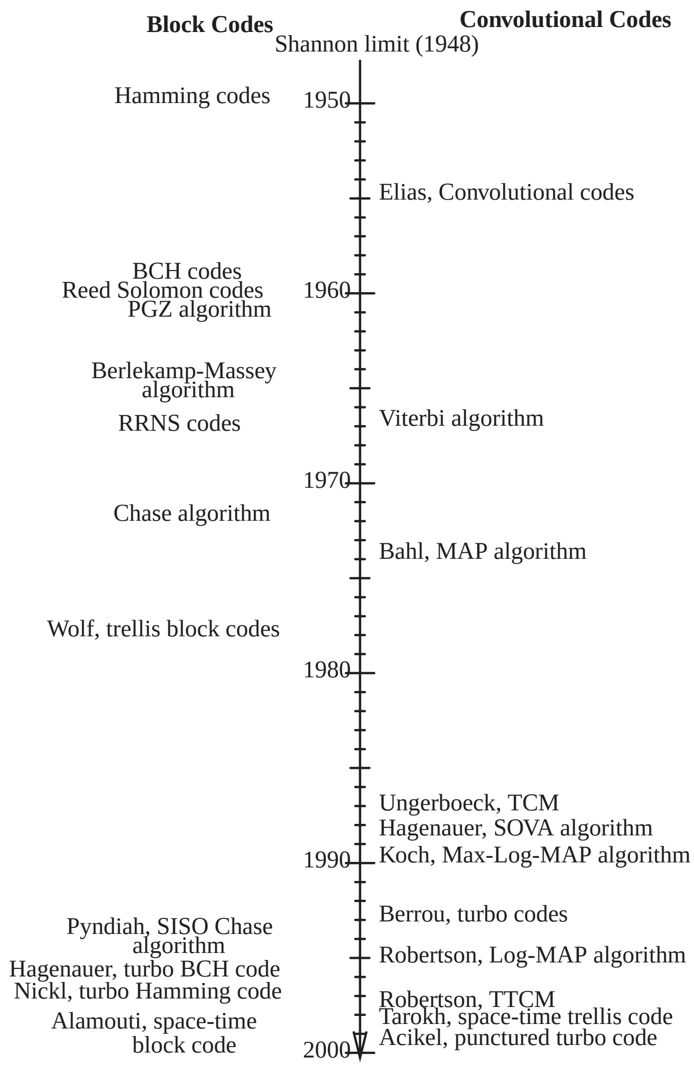

Figure 1: Brief history of channel coding research, ©John Wiley and IEEE Press, 2002, Hanzo, Liew, Yeap [4].

This overview is based on L. Hanzo, T.H. Liew, B.L. Yeap: Trubo coding, turbo equalisation and space-time coding, John Wiley - IEEE Press, 2002

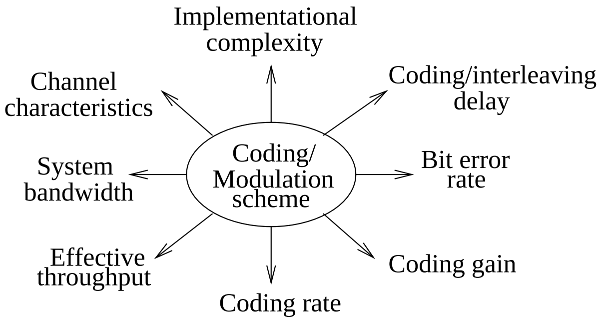

Figure 2: Factors affecting the design of channel coding and modulation schemes ©John Wiley and IEEE Press, 2002, Hanzo, Liew, Yeap [4].

interpretation of the Viterbi algorithm (VA) can be found, for example, in Forney's often-quoted paper [11]. One of the first practical applications of convolutional codes was proposed by Heller and Jacobs [12] during the seventies.

We note here that the VA does not result in minimum bit error rate (BER), it rather finds the most likely sequence of transmitted bits. However, it performs close to the minimum possible BER, which can be achieved only with the aid of the extremely complex full-search algorithm evaluating the probability of all possible 2^n binary strings of a $k-$bit message. The minimum BER decoding algorithm was proposed in 1974 by Bahl *et al.* [13], which was termed the Maximum A-Posteriori (MAP) algorithm. Although the MAP algorithm slightly outperforms the VA in BER terms, because of its significantly higher complexity it was rarely used in practice, until turbo codes were contrived by Berrou *et al.* in 1993 [2, 3].

Focusing our attention on block codes, the single-error correcting Hamming block code was too weak for practical applications. An important practical milestone was the discovery of the family of multiple error correcting Bose-Chaudhuri-Hocquenghem (BCH) binary block codes [14] in 1959 and in 1960 [15, 16]. In 1960, Peterson [17] recognised that these codes exhibit a cyclic structure, implying that all cyclically shifted versions of a legitimate codeword are also legitimate codewords. The first method of constructing trellises for linear block codes was proposed by Wolf [18] in 1978. Due to the associated high complexity, there was only limited research in trellis decoding of linear block codes [19, 20]. It was in 1988, when Forney [21] showed that some block codes have relatively simple trellis structures. Motivated by Forney's work, Honary, Markarian and Farrell *et al.* [19, 22, 23, 24, 25] as well as Lin and Kasami *et al.* [20, 26, 27] proposed various methods for reducing the associated complexity. The Chase algorithm [28] is one of the most popular techniques proposed for near maximum likelihood decoding of block codes.

Furthermore, in 1961 Gorenstein and Zierler [29] extended the binary coding theory to treat non-binary codes as well, where code symbols were constituted by a number of bits, and this led to the birth of burst-error correcting codes. They also contrived a combination of algorithms, which is referred to as the Peterson-Gorenstein-Zierler (PGZ) algorithm. In 1960 a prominent non-binary subset of BCH codes was discovered by Reed and Solomon [30]; they were named after their inventors Reed-Solomon (RS) codes. These codes exhibit certain optimality properties, since their codewords have the highest possible minimum distance between the legitimate codewords for a given code-rate. This, however, does not necessarily guarantee attaining the lowest possible BER. The PGZ decoder can also be invoked for decoding non-binary RS codes. A range of powerful decoding algorithms for RS codes was found by Berlekamp [31, 32] and Massey [33, 34]. Various soft decision decoding algorithms were proposed for soft decoding of RS codes by Sweeney [35, 36, 37] and Honary [19]. In recent years RS codes have found practical applications, for example, in Compact Disc (CD) players, in deep-space scenarios [38], and in the family of Digital Video Broadcasting (DVB) schemes [39], which were standardised by the European Telecommunications Standardisation Institute (ETSI).

Inspired by the ancient theory of Residue Number Systems (RNS) [40, 41, 42], which constitute a promising number system for supporting fast arithmetic operations [40, 41], a novel class of non-binary codes referred to as Redundant Residue Number System (RRNS) codes were introduced in 1967. An RRNS code is a maximum-minimum distance block code, exhibiting similar distance properties to Reed-Solomon (RS) codes. Watson and Hastings [42] as well as Krishna *et al.* [43, 44] exploited the properties of the RRNS for detecting or correcting a single error and also for detecting multiple errors. Recently, the soft decoding of RRNS codes was proposed in [4, 45].

During the early 1970s, FEC codes were incorporated in various deep-space and satellite communications systems, and in the 1980s they also became common in virtually all cellular mobile radio systems. However, for a long time FEC codes and modulation have been treated as distinct subjects in communication systems. By integrating FEC and modulation, in 1987 Ungerboeck [46, 47, 48] proposed Trellis Coded Modulation (TCM), which is capable of achieving significant coding gains over power and band-limited transmission media. A further historic breakthrough was the invention of turbo codes by Berrou, Glavieux, and Thitimajshima [2, 3] in 1993, which facilitate the operation of communications systems near the Shannonian limits. Turbo coding is based on a composite codec constituted by two parallel concatenated codecs. Since its recent invention turbo coding has evolved at an unprecedented rate and has reached a state of maturity within just a few years due to the intensive research efforts of the turbo coding community. As a result of this dramatic evolution, turbo coding has also found its way into standardised systems, such as for example the recently ratified third-generation (3G) mobile radio systems [49]. Even more impressive performance gains can be attained with the aid of turbo coding in the context of video broadcast systems, where the associated system delay is less critical, than in delay-sensitive interactive systems.

More specifically, in their proposed scheme Berrou *et al.* [2, 3] used a parallel concatenation of two Recursive Systematic Convolutional (RSC) codes, accommodating a so-called turbo interleaver between the two encoders [4]. At the decoder an iterative structure using a modified version of the classic minimum bit error rate Maximum A-Posteriori Algorithm (MAP) invented by Bahl *et al.* [13] was invoked by Berrou *et al.*, in order to decode these parallel concatenated codes. Again, since 1993 a large body of work has been carried out in the area, aiming for example for reducing the associated decoder complexity. Practical reduced-complexity decoders are for example the Max-Log-MAP algorithm proposed by Koch and Baier [50], as well as by Erfanian *et al.* [51], the Log-MAP algorithm suggested by Robertson, Villebrun and Hoeher [52], and the SOVA algorithm advocated by Hagenauer as well as Hoeher [53, 54]. Le Goff, Glavieux and Berrou [55], Wachsmann and Huber [56] as well as Robertson and Worz [57] suggested to use these codes in conjunction with bandwidth efficient modulation schemes. Further advances in understanding the excellent performance of the codes are due, for example to Benedetto and Montorsi [58, 59], Perez, Seghers and Costello [60]. During the mid-nineties Hagenauer, Offer and Papke [61], as well as Pyndiah [62] extended the turbo concept to parallel concatenated block codes as well. Nickl *et al.* show in [63] that Shannon's limit can be approached within 0.27 dB by employing a simple turbo Hamming code. In [64] Acikel and Ryan proposed an efficient procedure for designing the puncturing patterns for high-rate turbo convolutional codes. Jung and Nasshan [65, 66] characterised the achievable turbo coded performance under the constraints of short transmission frame lengths, which is characteristic of interactive speech systems. In collaboration with Blanz they also applied turbo codes to a CDMA system using joint detection and antenna diversity [67]. Barbulescu and Pietrobon addressed the issues of interleaver design [68]. The tutorial paper by Sklar [69] is also highly recommended as background reading.

Driven by the urge to support high data rates for a wide range of bearer services, Tarokh, Seshadri and Calderbank [70] proposed space-time trellis codes in 1998. By jointly designing the FEC, modulation, transmit diversity and optional receive diversity scheme, they increased the throughput of band-limited wireless channels. A few months later, Alamouti [71] invented a low-complexity space-time block code, which offers significantly lower complexity at the cost of a slight performance degradation. Alamouti's invention motivated Tarokh *et al.* [72, 73] to generalise Alamouti's scheme to an arbitrary number of transmitter antennas. Then, Tarokh *et al.*, Bauch *et al.* [74, 75], Agrawal [76], Li *et al.* [77, 78] and Naguib *et al.* [79] extended the research of space-time codes from considering narrow-band channels to dispersive channels [70, 80, 79, 71, 73].

A further important advance in the field, which was also proposed by Berrou and his colleagues, was the invention of turbo equalisation in 1995 [81], where instead of separate channel equalisation and channel decoding, these functions are jointly and iteratively performed. This technique is capable of attaining an improved performance after a number of consecutive iterations. Furthermore, it is capable of improving the channel estimation accuracy of wireless receivers and can also be efficiently exploited in the context of multi-user equaliser/detector assisted Code Division Multiple Access (CDMA) systems.

2. ADAPTIVE CODING AND TRANSMISSION FOR WIRELESS CHANNELS

The capacity of Gaussian channels is unambiguously characterised with the aid of the channel's bandwidth and its signal-to-noise ratio (SNR). By contrast, wireless channels [49] require a high number of parameters for their characterisation, such as the distribution of the fading envelope and phase, the Doppler frequency, the channel impulse response, etc. and hence the capacity of wireless channels is typically characterised for specific scenarios only [82]. The ultimate goal of wireless communications research is to devise transmission schemes, which are capable of communicating near the **capacity of the wireless channel**.

Since the instantaneous channel quality of wireless systems fluctuates over a wide range, as it is demonstrated in Figure 3, it is irrealistic to expect that conventional fixed-mode transceivers would maintain wireline-like transmission qualities. At the time of writing numerous existing wireless systems already exhibit some grade of adaptivity, as it was documented for example in the excellent overview article by Nanda *et al.* [83]. The third-generation (3G) systems UTRA/IMT200 [49] are also amenable to agile, Burst-by-Burst (BbB) reconfiguration [84, 85, 86] on the basis of 10ms transmission bursts with the advent of their sufficiently high-rate

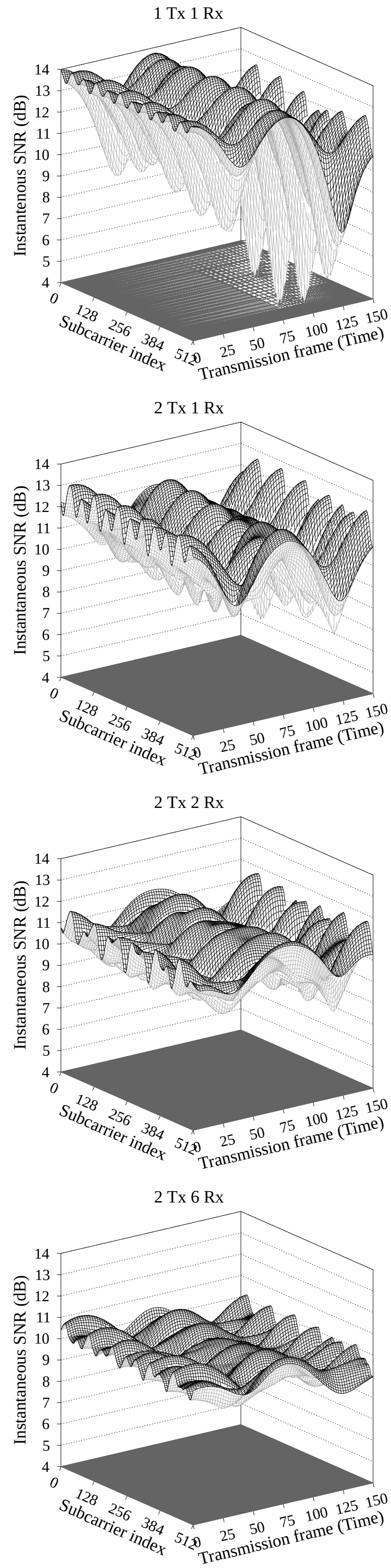

Figure 3: Instantaneous channel SNR versus time and frequency for a 512-subcarrier OFDM modem in the context of a single-transmitter single-receiver scheme as well as for the space-time block code $\mathbf{G}_2$ [71] using one, two and six receivers when communicating over an indoor wireless channel. The average channel SNR is 10 dB. ©IEEE, Hanzo, Liew and Yeap [4], 2002

and suitably low-delay control channels.

Albeit the well-known Orthogonal Variable Spreading Factor (OVSF) direct-sequence spreading codes [49, 87] have originally not been proposed for BbB-based reconfiguration, they can be potentially adapted on a near-instantaneous basis in an effort to counteract the near-instantaneous channel quality fluctuations of the wireless channel. Controlling the OVSF codes can also be potentially combined with near-instantaneous modulation mode control. The advantage of counteracting the near-instantaneous channel quality fluctuations in this way instead of the extensive employment of agile power control is that power control may inflict excessive co-channel interference in an effort to maintain the channel quality of specific users. By contrast, BbB-adaptive transmission constitutes a 'non-intrusive' way of mitigating the effects of transmission bursts, since only the user of interest adjusts its transmission mode appropriately.

The High PERformance Local Area Network standard known as HIPERLAN2 was also designed with a similar BbB-adaptive baseband modulation tool-box concept in mind [85], although again, the BbB employment of different transmission modes was not originally forseen. Nonetheless, the agile BbB-based application of adaptive source- and channel-coding as well as modulation [84, 86, 85] is emerging as a promising new system design paradigm in conjunction with the powerful channel prediction techniques proposed for example by Duel-Hallen, Hu and Hallen [88].

Over the past decades most channel coding schemes have been optimised for achieving the highest possible minimum distance amongst the legitimate codewords, which is an adequate design criterion for the Gaussian channels of wireline based or other quasi-stationary channels. However, the design of channel coding schemes for high error-rate, fading and dispersive wireless channels is still in its infancy [89]. Since the radio spectrum is a scarce resource, one of the most important objectives in the design of a digital cellular system is the efficient exploitation of the available spectrum in order to accommodate the ever-increasing traffic demands. Trellis Coded Modulation (TCM) [46, 4], which is based on combining the functions of coding and modulation, is a bandwidth efficient scheme that has been widely recognized as an excellent error control technique suitable for applications in mobile communications. Turbo Trellis Coded Modulation (TTCM) [90, 4] is a more recent channel coding scheme that has a structure similar to that of the family of power efficient binary turbo codes [2, 4], but employs TCM codes as component codes. TCM and TTCM schemes invoked Set-Partitioning (SP) based signal labeling, in order to achieve a higher Euclidean distance between the unprotected bits of the constellation. By contrast, parallel trellis transitions are associated with the unprotected data bits, which reduces the decoding complexity. In our TCM and TTCM schemes, random symbol interleavers were utilised for both the turbo interleaver and the channel interleaver.

Another powerful coded modulation scheme utilising bit-based channel interleaving in conjunction with Gray signal labeling, which is referred to as Bit-Interleaved Coded Modulation (BICM), was proposed in [91, 4]. It combines conventional convolutional codes with several independent bit interleavers, in order to increase the associated diversity order. With the aid of bit interleavers, the code's diversity order can be increased to the binary Hamming distance of a code, and the number of parallel bit-interleavers equals the number of coded bits in a symbol [91, 4]. The performance of BICM is better, than that of TCM over uncorrelated (or perfectly interleaved) fading channels, but worse than that of TCM in Gaussian channels due to the reduced Euclidean distance imposed by the associated "random modulation" inherent in a bit-interleaved scheme [91, 4]. Recently, iteratively decoded BICM using SP based signal labeling, referred to as BICM-ID has also been proposed [92, 93, 4]. The philosophy of BICM-ID is to increase the Euclidean distance of the BICM code and to exploit the full advantage of bit interleaving by a simple iterative decoding technique.

The question arises, whether - instead of designing channel codecs for wireless channels - it might be a better approach to attempt to render the wireless channel Gaussian-like and then invoke the classic channel codecs designed for Gaussian channels?

Adaptive transmission schemes are instrumental in achieving this objective. More specifically, adaptive modulation and transmission schemes [84, 86, 85] are attractive in terms of providing a high instantaneous throughput, when the instantaneous channel quality is high. By contrast, under low instantaneous channel quality conditions a lower-throughput but more robust transmission mode can be invoked. **The advantage of such adaptive transmission schemes is that the perceived channel quality becomes more Gaussian-like and hence the error statistics become less bursty. This in turn renders the channel codec more efficient and allows us to achieve potentially higher coding gains, than in conjunction with fixed-mode transceivers.**

3. APPLICATION EXAMPLE: ADAPTIVE MULTI-USER DETECTION AIDED TTCM/CDMA WIRELESS VIDEOPHONY[94]

As a powerful application example, in this section we characterise the achievable performance of a burst-by-burst adaptive coded modulation based wireless video telephone scheme supporting eight users employing the MultiUser Detection (MUD) aided CDMA system of Reference [94]. More specifically, the H.263 video codec [86] was used for transmitting 176x144-pixel Quarter Common Intermediate Format (QCIF) resolution video sequences scanned at 10 frames/s [86]. The burst-by-burst adaptive CDMA transceiver was capable of reconfiguring itself as a 2, 3, 4 or 6 bit/symbol TTCM scheme.

The achievable video quality was quantified in terms of the Peak-Signal-To-Noise-Ratio (PSNR) [86] and Figure 4 shows the video quality in terms of the PSNR versus the channel SNR for each of the modulation modes. As expected, the higher throughput bitrate of the 64QAM mode provides a better video quality. However, as the channel quality degrades, the video quality of the 64QAM mode is reduced and hence it becomes beneficial to switch from 64QAM to 16QAM, then to even lower throughput modulation modes, as the channel quality degrades. Although the inherently reduced bitrate of the more robust but lower-throughput BbB-adaptive TTCM modes resulted in a reduced video PSNR, this was less objectionable in subjective video quality terms, than the effects of channel errors would have been, had we not reconfigured the transceiver in a more robust TTCM mode.

4. MULTIPLE-INPUT, MULTIPLE-OUTPUT SYSTEMS VERSUS ADAPTIVE TRANSMISSION

It is a natural ambition to combine these adaptive transceivers with diversity aided Multiple Input, Multiple Output (MIMO) or space-time coded diversity systems [72, 4] in a further effort towards mitigating the effects of fading and hence rendering the channel more Gaussian-like. **A vital question in this context is, whether adaptive transceivers retain their performance advantages in conjunction with MIMOs?**

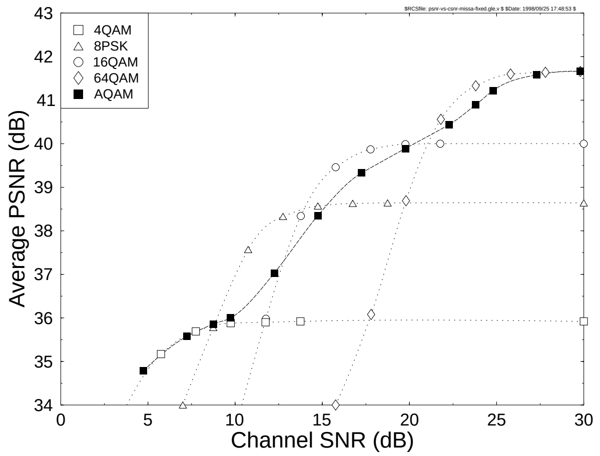

Figure 4: Average PSNR versus channel SNR for the four fixed TTCM modes of 4QAM, 8PSK, 16QAM as well as 64QAM and for the four-mode TTCM AQAM scheme using the H.263 encoded QCIF video sequence scanned at 10 frame/s, when communicating over the Universal Mobile Telecommunications System's Terrestrial Radio Access (UTRA) [87] scheme's vehicular channel A. The channel coded transmission rate varied near-instantaneously between 48 and 144 kbit/s ©IEEE VTC'2002 Spring, Chung, Ng, Kuan and Hanzo [94].

The inspection of Figure 3 allows us to intuitively answer this question, where the instantaneous channel SNR experienced by an Orthogonal Frequency Division Multiplex (OFDM) modem [4, 84, 85] versus both time as well frequency has been plotted. The OFDM modem has 512-subcarriers. Explicitly, in Figure 3 a conventional single-transmitter, single-receiver scheme as well as Alamouti's space-time block code $\mathbf{G}_2$ [71] using one, two and six receivers were employed for communicating over a low-dispersion indoor channel. The average channel SNR was 10 dB. We can see in Figure 3 that the variation of the instantaneous channel SNR for the single-transmitter and single-receiver scheme is severe and may become as low as 4 dB owing to deep fades of the channel. On the other hand, we can see that for the space-time block code $\mathbf{G}_2$ using two transmitters and one receiver the variation in the instantaneous channel SNR becomes less severe. Explicitly, by employing multiple transmit antennas - as shown in Figure 3 - the effect of the channels' deep fades may be significantly reduced.

As we increase the number of receivers, i.e. the diversity order, we observe that the variation of the channel becomes less dramatic. Effectively, by employing higher-order diversity, the fading channels have been converted to AWGN-like channels, as evidenced by the scenario employing the space-time block code $\mathbf{G}_2$ using six receivers. Since adaptive modulation only offers advantages over fading channels, we argue that using adaptive modulation might become unnecessary, as the diversity order is increased. Hence, adaptive modulation can be viewed as a lower-complexity alternative to space-time coding, since only a single transmitter and receiver is required. These informal conclusions are confirmed by the quantitative evidence seen in Figure 5 [4, 85], suggesting that no significant joint benefits accrue for BbB-adaptive systems employing high-order space-time coded transmit and receive diversity systems, since both of these regimes aim for mitigating the effects of fading and once the fading has been sufficiently mitigated for it to become near-constant, no further fading counter-measures are necessary.

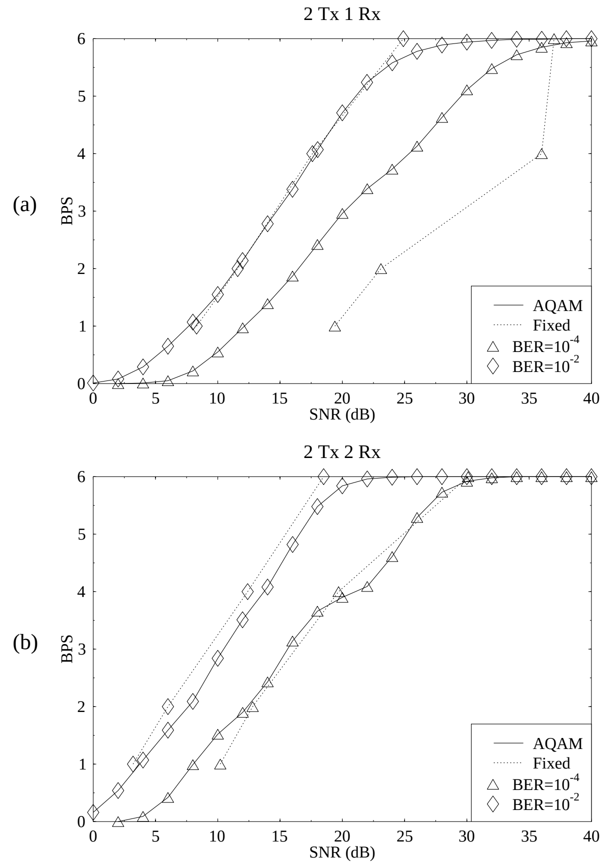

Figure 5: BPS throughput performance comparison between adaptive OFDM and fixed modulation based OFDM using Alamouti's space-time block code $\mathbf{G}_2$ employing (a) one receiver and (b) two receivers, when communicating over a low-dispersion indoor channel ©IEEE, Hanzo, Liew and Yeap [4], 2002. The target BER was set to either 1% or 0.01%.

5. CONCLUSIONS

In summary, we argue that channel coded adaptive modulation schemes can be viewed as a lower complexity alternative for mitigating the channel quality fluctuations of wideband wireless channels in comparison to multiple-transmitter and multiple-receiver based space-time codes. However, provided that the complexity of the latter schemes employing multiple transmitters and receivers is affordable, the performance advantages of adaptive modulation and adaptive channel coding schemes erode, since the channel quality fluctuations of the wireless channel are effectively mitigated.

The next challange ahead is that of finding low-complexity, low-latency coding and transmission schemes, achieving near--Shannonian performance for transmission over hostile, dispersive, fading wireless channels in the context of a variety of practical applications, some of which are exemplified in [4, 85, 84, 86, 87]. Finally, it is important to note in closing that the adaptive transceivers advocated in this overview have further network-layer benefits, since they allow the system to reconfigure the transceiver and hence to 'drop the transmission throughput, rather than dropping the call'. This is allows the system to approximately double the number of users supported [87].

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