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Editorial

Whilst the third generation (3G) mobile radio standards in Europe, in the U.S.A. and in Japan are approaching completion, researchers turned their attention towards the next generation of solutions. The research objectives became more ambitious, endeavouring to contrive appropriate system components capable of supporting a variety of multi-rate, multi-media services in various system environments. The mobile radio channel exhibits a time-variant channel quality, irrespective of whether indoor or outdoor scenarios are concerned. Additionally, there is a range of unpredictable propagation factors, such as the vehicular speed, the terrain, the local paraphernalia, etc. which influence the channel quality experienced by wireless transceivers. Although it is not particularly desirable a scenario, under these circumstances the instantaneous wireless channel capacity fluctuates and it can only be approached by "smart" transceivers, which are capable of varying their parameters in harmony with the channel quality variations.

A range of solutions have been advocated in the recent literature as to how to adapt the transceiver parameters. A well-known solution was proposed for example by Adachi and his team at NTT in Japan using the so-called Orthogonal Variable Spreading Factor (OVSF) codes in the context of the Japanese 3G Code Division Multiple Access (CDMA) proposal. However, a whole variety of other solutions have been analysed in the recent literature, such as for example employing multiple codes for high-rate users, which was favoured by Ottoson in this issue. A further technique is based upon varying the number of bits per symbol transmitted. Historically, this solution was advocated first by Steele and Webb at the University of Southampton in the U.K., leading to a spate of activities and stimulating further seminal research for example by Goldsmith and her colleagues at Caltech and at the University of Stanford in the U.S.A., who contributed the first article of this special issue.

These transceiver reconfiguration operations can be invoked on a static call-by-call basis in order to accommodate for example the different teletraffic and/or source signal representation quality as well as transmission integrity requirements of different users. In wireless environments, however, the channel's quality and the associated channel capacity is time-variant on a near-instantaneous basis. Hence an intelligent system may reconfigure itself during a call many times, in order to adapt to time-variant channel conditions. The benefits and the complexity implications of such intelligent multimode terminals are not widely documented. Hence the primary aim of this special issue is to stimulate further work in this area, whilst documenting some of the associated "cost/benefit" aspects of such intelligent multimode, multimedia terminals as well as to provide technical solutions for implementing some of the components.

In this context the first contribution by Alouini and Goldsmith quantifies the capacity of Nakagami multipath fading channels under a given average power constraint in conjunction with three different power- and rate-adaptation regimes. They provide closed-form solutions for the Nakagami-channel's capacity for each of the three considered power- and rate-adaptation regimes, demonstrating that rate adaptation is the key to increasing the link's spectral efficiency. These discussion then lead on to the performance analysis of practical constant-power, variable-rate M-QAM schemes over Nakagami channels, providing closed-form expressions for the outage probability, spectral efficiency and average bit-error-rate (BER) upon assuming perfect channel estimation and negligible time delay between the required channel quality estimation and modem mode adaptation. These elaborations are concluded with the analysis of the impact of time delay on the BER of adaptive M-QAM.

Naijoh, Sampei, Morinaga as well as Kamio at the University of Osaka and the Ministry of Post in Japan have also substantially advanced the state-of-the-art in adaptive modulation. Their contribution advocates a time division multiple access/time division duplex (TDMA/TDD)-based adaptive modulation scheme in conjunction with convolutional coding and a hybrid automatic repeat request (ARQ) protocol. The authors commence their discourse by considering the throughput limitation of conventional ARQ-assisted fixed-rate quaternary phase shift keying (QPSK), which suggests that an effective means of improving the throughput is the reduction of the ARQ re-transmission probability under low channel quality conditions and to increase the transmission bit rate per TDMA/TDD slot under favourable instantaneous channel conditions. Based on these strategies, this contribution proposes invoking adaptive modulation assisted by convolutionally coded type-II hybrid ARQ and evaluates the associated performance, demonstrating dramatic performance advantages.

Recently intensive research has been motivated by these adaptive single-carrier modulation principles also in multi-carrier Orthogonal Frequency Division Multiplex (OFDM) systems, amongst others by Rohling and his colleagues at the University of Hamburg-Harburg in Germany, whose contribution in this issue encompasses also the multiple access layer. Specifically, since in OFDM the total bandwidth is divided into a high number of subcarriers, where these subcarriers can be readily shared amongst all the users. Furthermore, the modulation modes can be individually allocated on a subcarrier by subcarrier basis. Hence low-quality subcarriers can even be potentially deactivated, and their transmit power can be more beneficially allocated to other high-quality subcarriers. In their contribution Gruenheid and Rohling present the combination of two different multiple access schemes, namely OFDM-FDMA and OFDM-TDMA, in conjunction with adaptive modulation. Different degrees of adaptivity are analysed and compared, in order to demonstrate the benefits of an “intelligent” multiple access and adaptive modulation strategy.

Another OFDM-based contribution of the issue is due to Chuang, which was cast in the context of dynamic packet assignment and interference suppression for wireless internet services. The anticipated high peak-bandwidth demands of the near-future are likely to require a tight frequency reuse. Dynamic packet assignment (DPA) has been previously proposed for OFDM-based wireless Internet access having a downlink transmission rate on the order of 1 Mb/s using a wide-area cellular infrastructure. The proposed DPA scheme is capable of allocating radio resources on a packet by packet basis and of reassigning them in about 100 msec, thereby gaining advantages in both statistical multiplexing terms as well as increasing the achievable area spectral efficiency due to the associated dynamic channel assignment (DCA) policies. Chuang’s paper contributes in the area of interference suppression cast in the context of the proposed OFDM-based internet service.

In recent years substantial advances accrued also in the field of intelligent forward error correction (FEC) coding, employing serial and parallel concatenated coding. The latter schemes are also often referred to as turbo codes, which employ iterative decoding techniques, passing soft-decision information back and forth between a pair of so-called soft-in soft-out type decoders and gradually improving the decoding performance, until a performance saturation is achieved within a fraction of a dB from the theoretical Shannonian limit. Turbo codecs require, however, a high turbo interleaving depth, and hence they are more amenable to non-interactive data transmission or to distributive information transmission, for example in broadcasting. Rate-compatible punctured turbo codes are also powerful in Automatic Repeat Request (ARQ) systems, which are often invoked in high integrity wireless data systems, where no transmission errors can be tolerated. A further novel research area closely related to turbo codes is the field of turbo equalisation, where channel decoding is jointly carried out with channel equalisation or interference cancellation. Joint channel decoding and soft multiple access interference cancellation was the topic of the paper by Ibrahim and Kaleh. Specifically, the authors used the channel decoder's output, in order to generate a soft-estimate of each user's signal. This estimate of the user's signal was then iteratively improved upon exchanging information between the channel decoder and interference canceller, before invoking a final hard-decision.

Annamalai and Bhargava have contributed to this issue on the topic of self-reconfigurable ARQ arrangements, commencing their discourse by pointing out that the effective throughput of ARQ-based systems falls rapidly under degrading channel conditions, when a high retransmissions rate has to be tolerated. In order to mitigate this problem, in their previous work the authors suggested to vary the FEC coding rate, compensating for the fluctuations in the channel conditions. In their contribution to this issue they considered four different algorithms for tracking the slowly varying channel quality fluctuations, which can be used for link adaptation in duplex radio communications. As a particular application the authors analysed the performance of stop-and-wait and selective-repeat ARQ schemes in conjunction with adaptive packet lengths.

An often encountered problem in multimode systems is the fluctuation of the transmitted signal envelope, which imposes a high linearity constraint on the power amplifiers employed. Only a good decade ago this problem was deemed so grave that it motivated the choice of a constant-envelope Gaussian Minimum Shift Keying (GMSK) based modem for the Global System of Mobile Communications known as GSM. This allowed the GSM system to employ class-C power amplification, which exhibits a high power-efficiency. Although in recent years further advances have been made in the field of wideband linear amplifiers, their power efficiency remains lower than for example that of linearised class-A/B amplifiers and dramatically lower than that of class-C amplifiers. These linear amplification problems are shared by multilevel Quadrature Amplitude Modulation (QAM), OFDM and CDMA. Fortunately the associated problems can be mitigated by a number of techniques, including precoding, which was advocated by Ottoson in this issue in the context of multicode CDMA, where the transmitted signal experiences large envelope variations as a result of transmitting the superposition of many independently spread signals. Ottoson notes that the envelope fluctuations of a multicode spread CDMA signal appear to increase as a function of the square root of the number of codes used, which limits the potential number of parallel codes. In order to mitigate this problem Ottoson proposes a precoding scheme, which is based on high-rate nonlinear block codes designed for the set of spreading codes used. The author's simulation results suggest that a precoded multicode system outperforms an uncoded multicode system in both single-user as well as in multiuser environments.

An important issue in spread spectrum systems is fast code acquisition, facilitating rapid user access and synchronisation with the system. The contribution by Spangenberg et al. describes a non-coherent technique for fast acquisition of direct sequence spread spectrum (DS/SS) signals in low earth orbit (LEO) satellite communication scenarios. The LEO environment is prone to high Doppler offsets, which are likely to cause major problems during the code acquisition phase due to the introduced frequency ambiguity. In their contribution the acclaimed authors from Edinburgh discuss the employment of a set of partial correlators for code phase acquisition, combined with the Fast Fourier Transform (FFT), demonstrating that the proposed architecture accelerates the synchronisation process compared to conventional techniques over a wide range of Doppler offsets.

Following the previous – mainly physical layer oriented contributions – the treatise by Wang and Hamdi proposes a novel medium access (MAC) protocol for future wireless multimedia personal communication systems, which the authors refer to as the hybrid and adaptive multiple access control (HAMAC) protocol. It is of paramount importance that the adaptive multimode physical layer does not impose detrimental constraints on the network-layer and that their synergy is maintained, despite the frequent transceiver reconfiguration operations imposed by the time-variant mobile channel. The protocol proposed by Wang and Hamdi integrates fixed assignment TDMA, reservation-based protocols and contention-based protocols, in order to support various classes of traffic such as constant-bit-rate (CBR), variable-bit-rate (VBR) and available-bit-rate (ABR) traffic. The main advantage of the proposed protocol is that the guaranteed service and simplified signaling features of TDMA are retained, which are complemented by the adaptive bandwidth allocation features and other advantages of packet reservation multiple access (PRMA)-like protocols.

The contribution by Sarker and Halme provides a study on slotted ALOHA, which is a well-established statistical multiplexing technique, exhibiting a range of advantageous properties. A popular variant of slotted ALOHA is the PRMA-like technique studied in the previous contribution by Wang and Hamdi. The general philosophy of slotted ALOHA-based statistical multiplexing techniques is that when an information source has new information to transmit, it contends for reserving some of the available resources offered by the multiple access channel, where the amount of reserved resources is ideally commensurate with the amount of information to be transmitted. For example, if these resources are constituted by a number of free time slots and no other sources contended for a reservation, then these time slots can be assigned to the source, which is becoming active. In case of contention collision, however, none of the contending sources is granted access and they continue contending, while using for example a random delay for preventing further collisions, until either they are granted the requested resources or until their information may become dropped, since it becomes obsolete. This scenario may be encountered for example in interactive speech systems, where “ageing” speech packets are discarded, in order not to jeopardize the transmission of newer packets. Again, a range of slotted ALOHA-related issues are analysed by Sarker and Halme in their contribution.

The last contribution by Efstathiou and Zvonar addresses a variety of issues related to the so-called enabling technologies of multi-standard “software radio” base stations. The basic concept of software radios is that most processing of the signal to be transmitted and/or received is carried out by a powerful baseband processor, which can be readily reconfigured under software control, in order to carry out a range of source- and channel coding, as well as modulation, transmission and reception functions. This treatise contributes in the field of quantifying the required specifications of the associated analogue-to-digital and digital-to-analogue converters as well as up- and down-converters.

Here we conclude scanning the issue, hoping that we raised your interest in probing further. We would like to close by thanking the numerous reviewers, who have genereously volunteered their valuable time, in order to improve the reader-appeal of the papers. Naturally, we would also like to thank the contributors and Kluwer for their valued assistance in bringing this issue to fruition. Our hope is that you will find the issue both thought-provoking and informative!

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(guest editors)