RZ 3497 (# 99399) 06/09/03
Electrical Engineering 29 pages

Research Report

Candidate Technologies for Next-Generation Wireless Access Systems

Pedro Coronel and Wolfgang Schott

IBM Research
Zurich Research Laboratory
8803 Rüschlikon
Switzerland

ABSTRACT

Because of the spectacular success that they have experienced in recent years, Wireless Local Area Networks have emerged as a key technology in the wireless evolution towards 4th-generation IP-based networks. Indeed, they are already competing with cellular networks to deliver data services in hotspots. Latest systems provide multimedia quality-of-service support at data rates of up to 54 Mbps, and are expected to play important roles in the enterprise and home networking environments. Nevertheless, the next generation of short-range wireless communication systems is called upon to fulfill very challenging requirements that far exceed what "state-of-the-art" systems can provide. In fact, they will have to compare positively with wired technologies in order to provide real broadband wireless services. This report aims at identifying the underlying requirements and the technologies that will have to be employed to meet these requirements. Because the requirements in data rate and link quality are virtually impossible to satisfy with single-antenna systems, we pay a special attention to multiple-antenna systems. We focus on the specific and significant design challenges that will have to be addressed at both the physical layer and the data link layer. Although this survey is not intended to be exhaustive, we try to point out important research activities that should be conducted in order to allow future systems to fully exploit the benefits that emerging transmission technologies provide.

I. INTRODUCTION

For several years now, Wireless Local Area Networks (WLANs) have been experiencing spectacular success. Primarily driven by the corporate need to enable flexible workplace mobility and the corresponding economic advantages, the use of these networks is also extending into the private consumer market. It is the tendency of private consumers to consider the wireless technology to be more elegant and a simpler solution to connect the ever growing number of home devices. In parallel to the development of private networks, the wireless public access has emerged owing to the interest of both small Wireless Internet Service Providers (WISP) and larger carriers. Although hotspots usage for leisure purposes is still not widespread, hot spots are actually proliferating in places where the business traveler can potentially be productive during traditionally "wasted" time (e.g. at airports, convention centers, train stations, etc).

Several organizations have played a crucial role in enabling the current popularity of WLANs. Prior to the ratification of the IEEE 802.11 standard in 1997, WLAN technologies were proprietary to the vendors. As a consequence, end-users could not customize their product by combining products from different vendors and, because only a few vendors existed, prices were substantial. Even with the 1997 standard, vendors did not show a common interest in guaranteeing compatibility for each of the Standard's three different physical layers. The 802.11b standard has drastically changed the WLAN landscape, opening the door for interoperability. The Wireless Ethernet Compatibility Alliance (WECA) has been decisive in reaching this goal by instigating vendors to test the interoperability of their 802.11x products. The successful completion of the tests allowed the vendors to use the "Wi-Fi" label in the advertising and packaging of their product. The marketing of "Wi-Fi" has been crucial because vendors across the world, and across industry segments, have joined WECA in order to ensure that they were keeping up with the development of wireless networking. Standardization bodies and industry alliances have established the conditions that allowed prices to spiral downward and volumes to increase. Indeed, in 1999, when the standard was ratified, the average price for an 802.11b chipset was in excess of $55. By mid-2002, the chipset cost was well under $20, and it is forecasted to be less than $7 by 2006 [1]. A first generation of 802.11a products was released in late 2001 and, following the same tendency, were commercialized at lower prices than expected. The satisfactory acceptance of 802.11a evidences the need for increasingly higher data rates. Although it presents disadvantages in terms of cost, range and power consumption, its peak data rate (up to five times that of 802.11b) and its ability to support a larger number of channels makes it a natural candidate for home and enterprise networks in the long term [2].

The forecasted growth for the WLAN hardware shipment is fairly steep and robust, reaching end-use revenue peak of $2,755 million by 2005 [1]. Meanwhile, the combination of 2.4/5 GHz products will also be introduced, allowing end-users to access whatever kind of technology is within reach. Nevertheless, as prices continue to fall dramatically, driving volume out the door, margins will decrease fast, with overall revenue growth forecasts being negative in 2006. As far as the public-area access is concerned, an increasing number of cellular operators are beginning to move into the market. Indeed, WLANs are not seen as a replacement of the existing networks but rather as complementary systems having the desirable advantage to work in an unlicensed spectrum. Forecasts predict again a tremendous growth in public-area access market-service revenue, from $3.9 million in 2001 to $224.7 million in 2005. Nevertheless, we note that after reaching its peak in 2002, the growth rate will slowly decrease during in subsequent years [3]. Even if the near future seems to be very sunny for the WLAN equipment and services markets, and moreover, that they likely will become mature in the midterm, the issue of identifying the required features of these systems over a longer term and the corresponding technologies to be developed naturally arises.

The demand for short-range wireless communications systems having increased capabilities is exploding. In particular, wireless broadband Internet creates the need for higher capacity and data rates than those achieved by current systems. Providing multimedia services (such as real-time conferencing or high-quality video streaming) together with traditional applications (Web browsing and email) in home, office or nomadic hotspot environments has become the targeted milestone for WLANs. In these contexts, they have to compare positively in terms of performance with competing technologies such as cable and digital subscriber line (xDSL). Although wireless systems offer significant advantages in areas such as deployment, scalability, maintenance and cost, a number of important issues still have to be addressed before they can enter the broadband market successfully.

This need has been recognized by several entities that are already working towards next-generation WLANs. For instance, within the IEEE 802.11 community, several task groups are addressing issues such as improved security (TGi), quality of service (TGe) and inter-access point protocols (TGf) among others. Additionally, it is worth noting the recent creation of the High Throughput Study Group (HTSG), whose goal is to "improve the 802.11 wireless LAN user experience by providing significantly higher throughput for current applications and to enable new applications and market segments" [4]. Consequently, their efforts currently focus on identifying the requirements and technologies, both at the physical and link layer, which will allow WLANs to enter the broadband market. Another entity that has been developing the concept of new-generation communication systems, including WLANs, is the industry-founded Wireless World Research Forum (WWRF). In its "Book of Visions 2001", several contributions by experts from academia and industry analyze the desired characteristics of future systems and draw up the corresponding research areas of interest [5]. According to this work, the general trend will be to integrate various types of systems (e.g. WLANs, Wireless Personal Area Networks (WPAN) and cellular systems) with a common IP-based backbone. Additionally, we point out the WWRF-initiated, but now independent, Wireless World Initiative (WWI) organization, which is collected concrete research proposals for next-generation wireless systems in the context of the 6th EU Framework Project (FP6). All the aspects that will form part of next-generation communication systems are intended to be considered, and, among these, attention is also being paid to short-range high-data-rate communication systems.

Finally, let us briefly mention additional standardization efforts in the broadband area that will share a number of emerging transmission technologies with future WLANs. First, the development of the base IEEE 802.16 standard is seeking to make Broadband Wireless Access (BWA) more readily available. The "last-mile/first-mile" connection in Wireless Metropolitan Area Networks (WMANs) is addressed by 802.16a. It applies to systems in licensed and unlicensed frequency bands that will provide data, video and voice services with a specified Quality of Service (QoS). In parallel, 802.16e is dealing with the combination of fixed and mobile BWAs, aiming at filling the mobility/throughput gap between WLANs and cellular networks. Furthermore, the recently created IEEE 802.20 has the mission to develop an efficient air interface for interoperable Mobile Broadband Wireless Access (MBWA). Ultimately, it should enable the transport of IP-based services targeted at business and residential end-user markets.

The purpose of this report is to describe the challenging issues that research needs to address towards next-generation communication systems, and in particular, future WLANs. Our survey will specifically deal with lower-layer considerations. The remaining part of this report is organized as follows. The next section will discuss the requirements that have to be met depending on different user scenarios in detail. Section III considers the improvements that can be achieved at the radio transceiver. In Section IV, we present the gains that can be achieved at system level and the corresponding techniques. Finally, we elaborate our conclusions in Section VI.

II. USER SCENARIOS AND REQUIREMENTS

Next-generation WLANs are expected to meet very challenging requirements. The exact nature of those as well as the extent to which they will have to be met are determined by the scenario in which the wireless access system will be used. We will therefore structure our discussion according to three general scenarios, and analyze their implications from a requirement perspective.

a. Home Environment

The home environment offers a large number of applications for which wireless systems are useful. Indeed, they enable communication among several devices, playing the role of an Ethernet connection. Depending on the application, the data rates necessary may vary from 100 kbps (e.g. sending a file to the printer) to 100 Mbps (e.g. video streaming). In general, highly sophisticated security protocols are not needed, but a security scheme that ensures controlled access, privacy and integrity should be provided. In order to ensure ubiquitous access, it might be necessary to install repeaters in addition to the central station. In that case, handover should be supported so that traffic can be redirected from one device to another without loosing the connection.

User-friendliness is a highly desirable attribute. The system should be of the type "plug and play", preventing the user from having to get involved in configuration or initialization tasks. Additionally, a major concern of most potential users will be to ensure that the system operates at low transmitting power levels. This is not only intended to limit human exposure to electromagnetic radiation, but also to reduce consumption. In this environment, the expenses incurred by the user should be kept at reasonable levels, and therefore, low device complexity is essential. From the point of view of the home end-user, backward compatibility with legacy WLAN or WPAN systems aims at reducing the migration costs and is consequently desirable.

The idea of using short-range communication systems as last-mile technology is also gaining momentum. This is evidenced by the creation of the IEEE 802.16 standard mentioned above. The objective would be to provide broadband Internet access and other services such as high-definition TV to residential customers at lower costs than with cable or xDSL. Again, the high bandwidth needed for these applications makes the development and use of new technologies indispensable. The introduction of mesh architectures would be a possible solution. They would serve as a series of wall-mounted routers that would configure themselves as signals were routed around the house. However, we emphasize that the broadband wireless access technology has to be cost-competitive, and therefore, has to provide high spectral efficiencies (e.g. for dense urban environments) and a high coverage range. Regarding the voice, data or video services provided by the Internet backbone network, QoS concepts should be used to make them available only if the necessary network provision is met.

Moreover, the home environment is also likely to use WLAN technology as a Wireless Local Loop (WLL) system. This application connects subscribers to the Public Switched Telephone Network (PSTN) using radio signals as a substitute for copper in all or part of the connection between subscriber and switch. The economic focus of WLL is not only aimed at deploying telephone services in emerging economies, but also at unlocking local loop competition in developed economies. Although voice services do not require high bandwidth, the optimization of the trade-off between coverage range, quality and number of APs is challenging in this application.

b. Enterprise Environment

The interest of enterprises in acquiring WLANs is to a large extent due to the simple way changes and moves can be dealt with. The resulting mobility is also seen as a contribution to productivity because mobile employees can make use of the tools they require without having to be at their desks. In the corporate environment, next-generation WLANs will have to measure up with 1 to 10 Gigabit Ethernet. As a complementary technology, wireless systems are intended to provide peak data rates of at least 100 Mbps in order to be attractive. Additional hurdles to be overcome are network management and security issues. While there is an abundance of solutions for wired networks, a considerable effort still has to be done to provide WLANs with strong management and security functionalities. The solutions must not only be robust for static scenarios, but also have to roam seamlessly among subnets with the end-user. Concerning applications, time-sensitive applications need to be supported, for example real-time conferencing. In fact, next-generation WLANs should provide native QoS support, and this is especially true in the corporate environment. The medium will consequently be shared in a way that improves the overall performance. Backward compatibility is a positive property that will justify a gradual migration to the next generation without requiring the entire investment to be concentrated at one point in time. Finally, more efficient power usage should allow increasing the coverage at ranges comparable with current ones. In this way, enhanced data rates will not be achieved at expenses of increasing the AP density, which will be positive from an infrastructure investment point of view.

c. Hotspot Environment

Although public-area access has recently become very popular owing to the appearance of wireless connectivity, wired technologies have already been used to provide it for a long time. A key driver of its development has been, and remains to be, the corporate requirement to increase workforce mobility and productivity. As a consequence, as far as hotspots are concerned, WLANs should be able to provide the applications that are demanded most by business travelers. A sine qua non condition to reach this goal is to enable high-speed Internet access, thus allowing access to the company's intranet. Moreover, the confidential nature of the information that will flow through the network necessitates strong security guarantees. The lack of security offered by current systems because of factors such as short key length, poor authentication mechanisms or vulnerable encryption, combined with the high number of potential threats existing in a public access environment, are slowing the development of this market. The expected adoption of Virtual Private Networks (VPN) protocols together with the work performed by 802.11 TGi will contribute to solve this problem.

Another aspect that has to be addressed involves the handoff capabilities of future WLANs. In order to become a viable service for the business traveler, wireless access has to be more consistent with respect to the mobility objective that has been driving its development. Therefore, the service has to become widespread, and handoff capabilities should be introduced. Wirelessly linking hotspots to the Internet, using for instance 802.16a, would contribute to keep the costs of interconnection reasonable and allow further hotspot expansion. Handoffs between different access points in a single network or between different technologies (e.g. GSM/802.11b), and roaming between networks belonging to different operators should be enabled. The effective coverage will consequently increase, and the user will remain aware of the available broadband-connection opportunities while traveling. Eventually, handoffs should be transparent to the user. Thus, appropriate mobility management should be implemented, based, for example, on existing concepts used in cellular networks, or on mobile capabilities of IPv6. Future network management should also allow users and/or applications to receive the requested amount of bandwidth. In this context, the introduction of QoS is essential because it ensures the availability of time- or bandwidth-critical services.

III. RADIO TRANSCEIVER TECHNOLOGIES

The wireless channel is highly challenging because it is impaired by path loss, shadowing and scattering, which not only cause the signal level to drop with distance, but also induce it to fade. Scattering engenders multipath propagation that spreads the signal in time, frequency (the motion of transmitter, receiver or scatterer is needed) and space. Additionally, the fact that the medium is generally shared among several users increases the level of interference that has to be dealt with. Several techniques can be used to offset these effects and even to convert some of them into benefits. In recent years, most of the attention has been focused on Multiple-Input Multiple-Output (MIMO) systems because of the spectacular improvements they can provide. For instance, MIMO technology is able to enhance the achievable throughput [6] and link reliability significantly with respect to traditional Single-Input Single-Output (SISO) systems. As we have seen, these are crucial requirements for next-generation communication systems, and. therefore, there is no doubt that MIMO systems will shape them. With a special attention to antenna arrays, we will elaborate next on candidate technologies for future communication systems and, in particular, future WLANs. Unless stated otherwise, the techniques considered are not restricted to MIMO systems.

a. Diversity

The principle of diversity it to provide the receiver with several independently faded replicas of the transmitted signal, such that, with high probability, at least one of them will not have experienced a deep fade. The higher the number of independent replicas, the higher this probability will be. Thus diversity leads to an improvement in link reliability or error rate. Although coding has the same purpose, the gains are of different nature. While the coding gain is evidenced by a shift to the left in the Symbol-Error Rate (SER) curve, the diversity gain manifests itself by increasing the magnitude of its slope. Therefore, the resulting SNR advantage due to diversity gain increases with SNR, whereas the one due to coding gain remains constant for increasing SNR. Diversity gain can be achieved in several ways. For example, if the signal time span is larger than the channel's coherence time, time diversity is achieved. Similarly, if the signal frequency span is larger than the channel's coherence bandwidth, frequency diversity is obtained. Unlike time and frequency diversity, *spatial diversity* requires the use of several antennas which have to be spaced apart by more than the coherence distance.

Receive and transmit diversity are special cases of spatial diversity, one of the key leverages provided by multi-antenna systems. Receive diversity is independent of the particular modulation and coding scheme used and only needs multiple antennas that experience independent fading to be effective. Transmit diversity requires the signal to be pre-processed or pre-coded before transmission. If the channel is known perfectly at the transmitter, techniques such as transmit-Maximum Ratio Combining (MRC) or eigenmode transmission can be used to achieve diversity. If no channel state information is available, transmit diversity can still be obtained by methods such as space-time or space-frequency coding. For two transmit antennas, a well-known example of such a technique is the Alamouti scheme, which orthogonalizes the channel irrespective of its realization and extracts full order diversity [7]. In general, for a MIMO system that employs M_R antennas at the receiver and M_T antennas at the transmitter, the maximum achievable spatial diversity order is given by $M_R M_T$. Depending on the channel conditions and the number of antennas used, it might be useless to assign all these degrees of freedom to achieve diversity. Indeed, some of them could be used to increase the throughput.

The concept of diversity gain as seen until now is well understood from the point of view of an average probability of error over the random channel. However, recall that diversity helps in stabilizing the fluctuations of the received signal strength and thus is related to reliability. Therefore, an outage analysis is better suited at capturing it because it evaluates the degree of performance that is guaranteed for a certain level of reliability. For that purpose, the authors of [8] use an outage-rate-dependent measure of diversity to quantify the loss or gain in diversity due to fading signal correlation, power/gain imbalance, and Ricean fading compared with classical i.i.d. Rayleigh fading. Another approach that characterizes the diversity order of the channel and that does not depend on the outage is to measure the fluctuations in received signal power. As the number of degrees of freedom offered by the channel increases, the variations become smaller and the achievable *effective diversity order* increases as well [8]. Although the relationship to the error probability exponent (i.e. slop of the SER curve) is not clear, it is consistent with the classical understanding that the fading channel approaches an AWGN link (i.e. no fluctuations at all) as the diversity order goes to infinity. The advantage of this concept is that it can be applied to MIMO channels with arbitrary statistics, thus providing insights into the influence of the propagation conditions on the performance over such channels.

b. **Spatial multiplexing**

Spatial multiplexing (SM), also known as BLAST (Bell-Labs Space-Time), is a technique that requires multiple antennas at both the transmit and receive sides, as well as a propagation environment that offers rich scattering [9-11]. The basic principle is to split the symbol stream to be transmitted into several streams and to transmit them simultaneously in the same frequency band. Under favorable scattering conditions, the receiver can separate the different co-channel signals and extract them using channel knowledge. This technique opens multiple data pipes that yield a linear increase in capacity (in min $\{M_R, M_T\}$). The cost of this significant improvement in spectral efficiency is only due to the additional hardware needed to use multiple antennas, but remarkably, no extra bandwidth or power is needed.

The performance of spatial multiplexing and consequently the enhancements in spectral efficiency that a MIMO system can achieve depend critically on the propagation conditions. In environments in which the channel exhibits a strong fixed component, i.e. Ricean component, large antenna spacings are required to enable the separation of the different streams at the receiver. A possible solution to ensure high multiplexing gains in a wide variety of propagations conditions is to have multiple central stations spatially multiplexing data to the same user. Nevertheless, this requires coordination among the central stations and thus introduces additional overhead. In order to keep this overhead at a reasonable level, it is necessary to better understand how the propagation environment influences the MIMO gains achievable through this strategy. Indeed, if the central stations providing better performance are identified, only a minimum number of them will have to be used for multiplexing, and the coordination overhead will be kept low.

c. **Array gain**

Array gain is achieved through coherent combination of the signals arriving at the receive antenna array. The resulting received SNR increases proportionally to the number of receive antennas. This translates into an improvement of the offered coverage. Array gain can be obtained by processing at either the receiver or the transmitter, but in any case, channel knowledge is a prerequisite. At the receiver side, Maximum Ratio Combining (MRC) is used to maximize the received SNR. At the transmitter, channel information is used to launch the signal from the different antennas such that it is received in phase. Nevertheless, retrieving the necessary channel information causes more practical problems at the transmitter than at the receiver.

d. **Co-channel interference cancellation**

Co-Channel Interference (CCI) arises in networks in which a small frequency reuse factor has been used for deployment. CCI mitigation is desirable in wireless networks because it enables better frequency reuse and thus improves spectral efficiency. Multiple antenna systems can be useful in further reducing the reuse factors used in traditional SISO systems by performing *interference cancellation* (IC). Because the transmitters or receivers are not co-located, there is no possibility of coordination between them — joint coding or decoding is not possible. Different receivers can exploit antenna arrays to cancel CCI, based on criteria such as maximum likelihood (ML), minimum mean square error (MMSE), or combinations thereof (e.g. MMSE-ML) together with training sequences. The amount of channel knowledge that is required depends on the criterion chosen as well as on performance needs. In general, it can be shown that the number of receive antennas required per interferer for CCI cancellation is given by the effective spatial rate (i.e. the number of independent symbols transmitted per symbol period over the M_T transmit antennas) of the signaling scheme employed by the interferer. Thus, if it uses spatial multiplexing, M_T antennas are needed at the receiver to cancel one interferer, whereas if Orthogonal Space Time Block Codes (OSTBC) is used, only one receive antenna is needed per interference source [12, Chap. 11]. From a mutual information point of view, if I denotes the dimensionality of the interferer, it can be shown that the effective number of receive antennas is reduced to $M_T - I$ [13].

e. Tradeoffs in multi-antenna systems

The use of multiple antennas at both ends of the wireless link offers significant leverages. As we have seen, link reliability is improved by spatial diversity, the spectral efficiency of the system is increased by spatial multiplexing, the system capacity in cellular networks is enhanced by using interference-cancellation techniques, and the improvement in SNR achieved by array gain extends the system coverage. These key benefits that are achievable in MIMO systems are well understood individually. Nevertheless, before optimum systems can be designed, the tradeoffs between those gains need to be understood in order to realize a maximum number of them *simultaneously*. Next, we examine some of the dependencies that have been analyzed in literature.

The error performance of a MIMO system depends on two factors: firstly, the achieved diversity order, which determines the slope of the SER curve, and secondly, the coding gain, which is approximately constant in the high-SNR regime. The concept of Space-Time Codes (STC) that perform coding across temporal and spatial dimensions was introduced in [14]. In addition, it was shown there that the error performance in frequency-nonselective channels can be controlled by a code design based on two criteria: The "rank criterion" ensures that the difference matrix between every pair of codewords is full rank in order to extract full spatial diversity. The "determinant criterion" aims at optimizing the coding gain by maximizing the determinant of the squared difference matrix. Using these two criteria, space-time trellis codes (STTC), which are an extension of trellis codes, can be designed to extract both spatial diversity and coding gain. However, in practical systems, the coding gain achieved is limited by the decoding complexity, which increases exponentially with the number of trellis states. In contrast, OSTBCs, to which the Alamouti scheme mentioned above belongs, exhibit a lower encoding/decoding complexity. Although they achieve full diversity if properly designed, they do not provide coding gain. Therefore, STTCs yield better performance than OSTBCs do, but at the cost of increased receiver complexity.

However, STBCs as well as STTCs have a spatial rate of one or less. We recall that one of the major benefits of MIMO systems is the possibility to open several data pipes to increase the capacity. SM can be thought of as a special STC with the spatial rate equal to M_T. For example, if we consider the case of an uncoded SM system over which symbols drawn from a signal set of size 2^q are transmitted, the resulting signaling rate is qM_T (i.e. the spatial rate is M_T) and the diversity gain reduces to M_R (recall that the maximum achievable diversity order is $M_T M_R$) [12]. Thus, the rate has been increased at the price of reducing the diversity advantage. Under good channel conditions, this reduction might be affordable from an error-performance point of view. We note that coding schemes that in terms of spatial rates lie in between the extremes of diversity coding and SM can be designed. For example, the linear dispersion framework proposed in [15] aims at maximizing the ergodic capacity (i.e. the capacity mean of an ergodic random channel) by using matrix modulation that spreads the symbols to be transmitted across space and time.

The tradeoff between diversity and multiplexing has been analyzed theoretically. In [16], new tighter closed-form bounds for the ergodic capacity and the variance of mutual information are reported. These results provide insights into the diversity-multiplexing tradeoff for Gaussian codebooks by relating the variance of the mutual information to the diversity order provided by the channel. For an uncorrelated channel, these results lead to the following concrete example illustrated in Figure 1. If the number of received antennas M_R is fixed, and more antenna are added to the transmitter while keeping $M_R > M_T$, the number of independent data streams that are spatially multiplexed is increased. In fact, the additional degrees of freedom are exploited to increase the throughput but not the diversity order. In contrast, because the number of spatial data pipes that can be opened is limited by the fixed number of receive antennas, when $M_T > M_R$, the additional antennas will be used to increase the diversity order of the system. The propagation parameters also have a strong impact on the gains that can be achieved. In the case of a spatial fading correlation, a loss in ergodic capacity was observed. For instance, assuming that $M_T = M_R$ and that only transmit correlation is present, an increase in correlation would decrease the number of effective antennas in the system, and consequently, the effective number of parallel data pipes that could be opened would also be decreased. However, this would lead to a higher *per-stream diversity order* (i.e. the achieved diversity is normalized by the number of independent streams opened). This information-theoretic framework is extended in [13] to take also the relation of these gains to interference and coding gains into account.

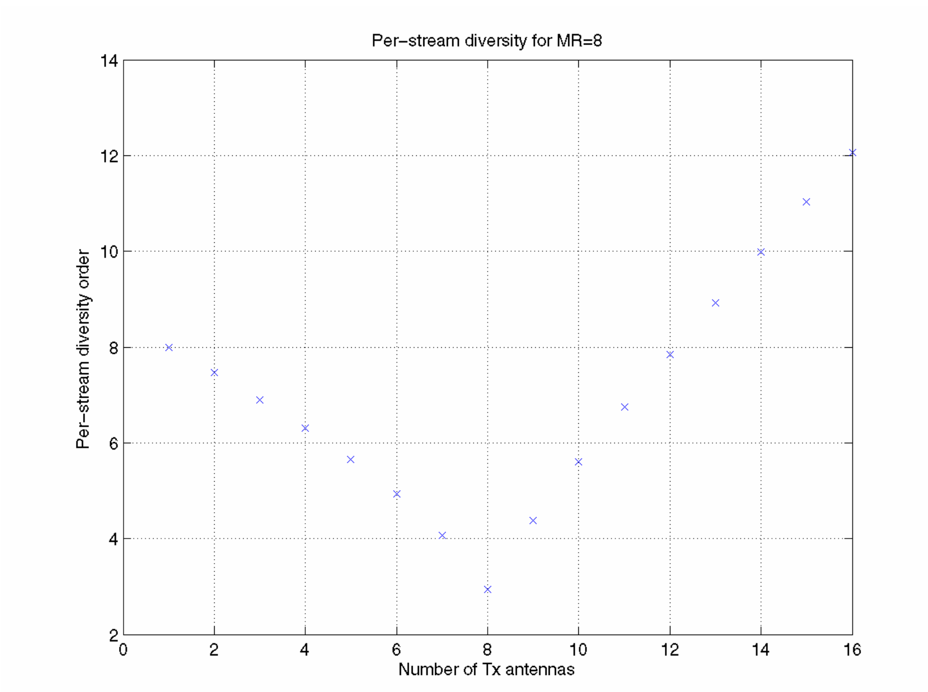

Fig. 1: Dependency of the per-stream diversity on M_T for $M_R = 8$ [13].

Whereas the above approach relates the gains obtained to the ergodic capacity, another framework based on the outage capacity is reported in [17]. (Recall that for non-ergodic random channels, the outage capacity $C_{OUT}(q)$ is defined as the capacity that is guaranteed for $(100 - q)$% of the channel realizations.) For i.i.d MIMO channels, [17] provides a characterization of the optimum diversity-multiplexing tradeoff curve valid for block lengths strictly bigger than the total number of antennas used in the system. This curve is given by a piecewise linear function that associates the corresponding diversity gain $d[k]$ ($d[k] = (M_T - k)(M_R - k)$) with each multiplexing gain $k = 0, \ldots, \min[M_T, M_R]$.

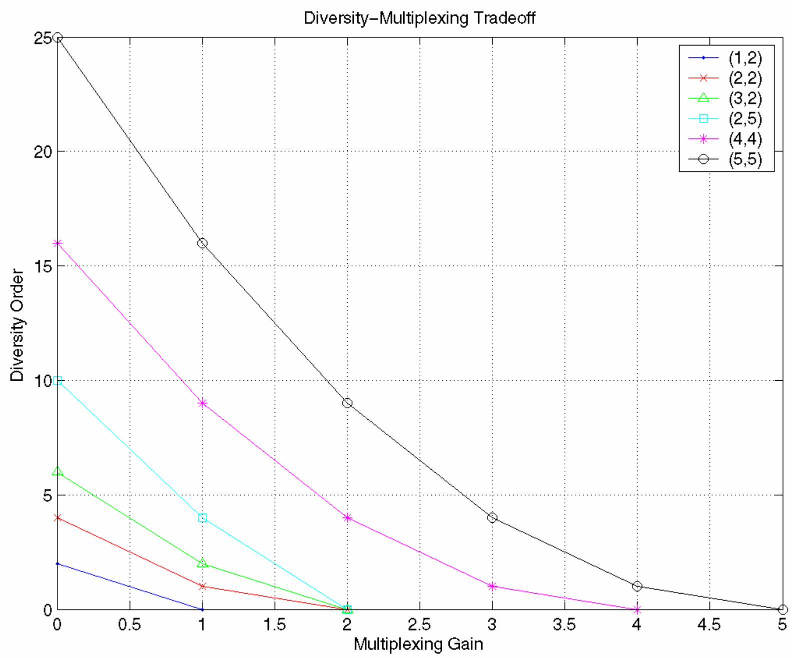

Fig. 2: Diversity-multiplexing tradeoff for several antenna configurations (M_T, M_R) [17].

Figure 2 shows the tradeoff curve for several antenna configurations. Clearly, the multiplexing gain is limited by the minimum number of antennas used in the system. Maximum gains for both multiplexing and diversity cannot be achieved simultaneously. This tradeoff is achievable by any scheme and is intended to be used to evaluate existing schemes or even to design new ones.

In summary, understanding the fundamental dependencies between the different MIMO gains is a crucial step in the design of optimum systems. Different information-theoretic frameworks have been introduced to grasp these dependencies. In particular, insights into the multiplexing-diversity tradeoff have been provided using both an ergodic and an outage analysis. More recently, results on how interference canceling and coding gains affect the multiplexing-diversity tradeoff have been reported. Nevertheless, further investigation efforts are needed to expand these insights and take advantage of them by developing codes that are flexible with respect to the tradeoffs. Finally, as we will see later, MIMO systems provide other gains that have to be included in the analysis.

f. Channel knowledge at the transmitter

So far we have considered schemes that do not assume any Channel State Information (CSI) at the transmitter. However, a transmission scheme that uses channel knowledge may be exploited to improve the link performance. There are various transmission methods that make use of CSI, e.g. pre-coding, adaptive rate/power control, beamforming, crosstalk cancellation, adaptive bit-loading, and so forth. In multi-carrier systems, CSI can be used to perform waterfilling over tones, to assign the transmit power optimally. The exact scheme to be used depends on a variety of factors, such as the nature of the available channel knowledge (full or partial), or the performance criterion to be optimized (average error rate or ergodic capacity). However, in terms of information-theoretic limits, there is little to be gained by exploiting CSI at the transmitter in a SISO system. In the multi-antenna case, however, the gain through even partial channel knowledge can be substantial (see [18] and references therein).

In the case of perfect channel knowledge at both ends of a MIMO channel, the Singular Value Decomposition (SVD) of the channel matrix can be used to process the transmitted and received signals linearly to obtain a set of parallel scalar channels. Then, the available transmit power may be optimally allocated to the individual channels. The case of incomplete channel knowledge, in which the channel mean or its covariance matrix is fed back to the transmitter, has also been analyzed in the literature [19]. Nevertheless, accurate estimates of the channel estimates are not always possible, particularly when dealing with a fast-fading channel. This fact has been taken into account in [20], where the proposed error performance criterion depends on the estimation quality. Side information is utilized for improving a predetermined OSTBC by means of a linear transformation. When the quality of the side information degrades, the optimum transformation is found to be a scaled unitary matrix and therefore, the codewords are transmitted without modification. On the other hand, when the estimate quality is high, beamforming in the direction of the strongest eigenmode of the channel is performed. Transmitter strategies in the presence of CSI, and in particular for the broadband MIMO channel, still are areas of current research.

g. Frequency-Selective Channels and OFDM

Broadband channels offer frequency diversity due to delay spread. However, the complexity of ML detection and even of sub-optimum equalizers needed for single carrier modulation grows exponentially with the product of bandwidth and delay spread. In this context, Orthogonal Frequency Division Multiplexing (OFDM) significantly reduces the receiver complexity. Using a cyclic prefix, it avoids the need for temporal equalization at the price of a small penalty in spectral efficiency. If N denotes the number of subcarriers used, OFDM decomposes the frequency-selective channel of bandwidth B into N orthogonal flat-fading channels with bandwidth B/N. Moreover, because tones spaced by more than the channel coherence bandwidth will experience independent fading, OFDM can exploit frequency diversity by coding and interleaving across tones (coded OFDM).

The complexity reduction achieved in the SISO case is even more important in a MIMO system. Indeed, if the modulation and demodulation parameters are appropriately synchronized across all transmit and receive antennas, every OFDM subcarrier can be treated as an independent narrowband frequency flat MIMO channel. The optimum transmitter energy allocation is to split it evenly across antennas and tones when the channel is not know at the transmitter, but if the channel is known, waterpouring across space and frequency may be used to maximize capacity. The work reported in [21, 22] investigates the use of space-time coding in OFDM-based broadband systems in which both spatial and frequency diversity are available. If L is used to denote the number of taps in the channel, the maximum achievable diversity order of the system is given by LM_TM_R. Thus, a frequency selective channel potentially offers L times more diversity than a flat-fading channel does. A noticeable fact is that a space-time code achieving full spatial diversity in a narrowband environment will in general not achieve full diversity in the OFDM broadband case. In fact, the diversity and coding gains achieved depend on the rank and eigenvalues of the transmit and receive correlation matrices. Therefore, modified criteria for code design that take into account these differences are proposed. Using these criteria, [23] presents a systematic method for designing codes that allows a variable multiplexing-diversity tradeoff. In addition to the advantage in diversity that a frequency-selective MIMO channel offers, it will in general also provide a higher multiplexing gain than a MIMO flat-fading channel does. This is due to the fact that frequency selectivity improves not only the outage capacity but also the ergodic capacity [24].

The coding scheme considered above assumes that the receiver has channel knowledge. However, acquiring knowledge of the fading coefficients is very challenging in the flat-fading case and even more so in the frequency-selective case as the presence of multiple paths increases the number of coefficients to be estimated. Whereas the coherent case is representative for fixed or low-mobility wireless systems, the fast-fading channels, in which future mobile broadband access systems will operate, motivates the analysis of non-coherent systems. The maximum achievable diversity order for non-coherent MIMO-OFDM has been shown to be the same as in the coherent case [25]. Furthermore, using unitary constellations, [25] derives the code design criteria and provides non-coherent code constructions that achieve full space-frequency diversity.

Before ending this section, let us mention the spectral efficiency issue that arises when using a cyclic prefix OFDM system. We have seen that this method converts the frequency-selective channel into a set of independent flat-faded subchannels for every subcarrier. The fades are removed by dividing each subchannel output by the channel transfer function at the corresponding subcarrier provided they are not zero. If CSI is available at the transmitter, power loading is a way to bypass channel fades. Nevertheless, gaining channel knowledge might be too costly, and usually, error-control coding is used to mitigate the loss due to channel nulls on or close to the transmitted subcarriers. In order to avoid the bandwidth overexpansion caused by coded OFDM, the use of Zero-Padding (ZP) OFDM has recently been proposed [26]. If the number of zeros equals the length of the cyclic prefix, the two schemes yield the same spectral efficiency. Nevertheless, without bandwidth-consuming channel coding, ZP-OFDM guarantees symbol recovery and FIR equalization regardless of the location of the channel's nulls. This is achieved at the cost of some increase in complexity. For ZP-OFDM to be potentially considered for future multicarrier communication systems, efficient time and frequency synchronization methods have to be developed.

h. Multiuser systems

The three traditional access techniques employ time, frequency or code dimensions to differentiate them (TDMA, FDMA and CDMA, respectively). Other multiple-access techniques as well as hybrid schemes are considered for future systems. Let us first mention Orthogonal Frequency Division Multiple-Access (OFDMA), which consists in allocating a subset of OFDM tones on a per-user basis. Because OFDM converts each tone into an orthogonal flat-fading channel, Multiple Access Interference (MAI) could be avoided with an adequate cyclic prefix and proper synchronization. Each subcarrier can then be modulated and coded with data rates that are adaptively determined to maximize the overall error performance. Higher-level modulations can be used for good subcarriers, whereas subcarriers that are in deep fades will not be used at all. On the other hand, an extension to OFDMA is the so-called Multi-Carrier CDMA (MC-CDMA), in which a spreading sequence is applied in the frequency domain [27]. An MC-CDMA system does not experience significant intersymbol interference (ISI) because it is also composed of narrowband signals having a symbol duration that is much larger than the delay spread. In these systems, multiple access is achieved with different users transmitting at the same set of subcarriers by using orthogonal spreading codes. Thus, MAI is directly related to the correlation properties of the spreading sequences. As in the case of OFDMA, the data rates can be adapted on a per-tone basis. The optimum receiver for MC-CDMA in a delay-spread channel has been found to be of the RAKE type.

Moreover, multicarrier systems usually span a wider frequency range than the coherence bandwidth, thereby achieving frequency diversity.

Multiple antennas can also be used at the central station to support multiple users with one or more antennas. This technique, known as Spatial Division Multiple-Access (SDMA) [28], exploits the differences in the users' spatial signatures at the central station to communicate with them simultaneously in the same frequency band. The main benefit of SDMA is that it allows channel reuse within a cell and consequently, improves spectral efficiency. While a single-user MIMO system is a point-to-point link, the multiuser system is a *broadcast channel* in the downlink and a *multiple access channel* in the uplink (see Figure 3). From an information-theoretic perspective, the link capacity defined for the first case translates into a capacity region in the second. The relationship between the multiple-access and broadcast-capacity regions is still a subject of research. Another open research problem concerns the optimum user-collision strategy for multiuser MIMO systems. In the case of a single-antenna multipath fading channel without CSI at the transmitter, it was shown in [29] that collision among users in frequency is desired to maximize the sum capacity. Therefore, the question is to know how much collision, either in time or frequency, is required in the MIMO case to achieve capacity gains. In terms of transceiver complexity, it is clearly desirable to have as little collisions as possible. This consideration underlines the necessity to better understand the performance-complexity tradeoff from a collision perspective to design efficient transceivers.

A problem that arises in the multiuser context is the unfairness due to the assignment of sub-channels to users. While in the single user case only the performance of the overall link really matters, in the multiuser case each user link has a targeted performance and rate, and they might not always be achievable. Moreover, because the distance to the central station of the different users is different, two users might have strong differences in Signal-to-Interference-plus-Noise Ratio (SINR). If not properly compensated, these differences hurt the user having the weaker received signal power. Therefore, the statistics of all users to which simultaneous transmission occurs have to be taken into account to use a fair strategy. In order to understand how performance gains can be realized in the multiuser MIMO context, the ultimate performance limits of the coding and signal-processing schemes used have to be assessed.

The design of space-time or space-frequency codes for several users is particularly challenging because a cooperative encoding across all the antennas as in the single-user case is not always possible (coding/decoding cooperation can only be done at the central station). In the broadcast case, although cooperation is possible between the transmit antennas, the multiple users at the receiver side are not able to coordinate their decoding. Owing to the fact that independent decoding recovers the signal of interest by treating the others as interference noise, it is outperformed by joint decoding which exploits the structure in other users' signal. The coding problem at the central station for the random fading multiple-access channel still is an open problem, particularly concerning the power coordination. In general, it remains to investigate efficient code-design criteria to achieve diversity and coding gains depending on the propagation conditions for mixed encoding/decoding (i.e. joint and independent) across antennas.

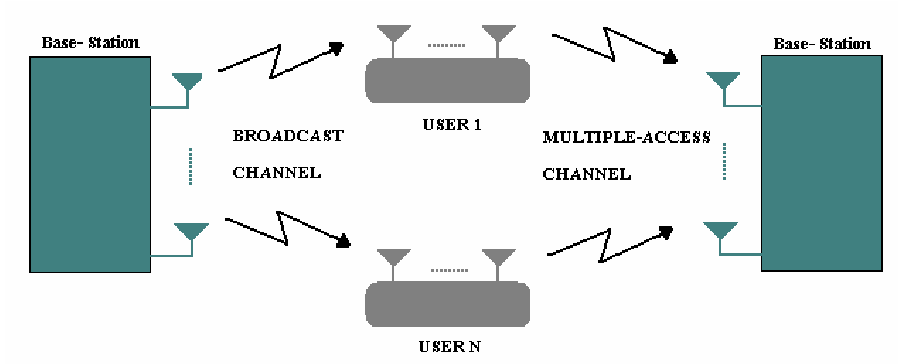

Fig. 3: The MIMO broadcast and multiple-access channels

A possible solution to decouple the MIMO multiuser channel into independent single-user channels is to employ code-division multiple access (CDMA). Again, only perfectly orthogonal codes guarantee full decoupling and therefore, the correlation properties of the codes will determine the resulting MAI. In a context in which CDMA is used to allow several users to share a common channel, space-time or space-frequency coding can also be employed to realize spatial and frequency gains. As pointed out, [29] demonstrates that collision-based multiple-access schemes in single-antenna systems are required in order to achieve ergodic sum capacity in a broadband channel. Provided that this result is extendable to the MIMO case, this would mean that multiple users should occupy a set of joint signal-space dimensions. On the other hand, the lower the dimensions of the transmit signal space, the higher the achievable diversity gain (redundancy is introduced). These considerations show the necessity of jointly designing the spreading sequences and space-time (or. frequency) codes to perform optimum signaling in a MIMO-CDMA system.

i. Time- vs. Frequency-Division Duplexing

There are several tradeoffs between Frequency-Division Duplexing (FDD) and Time-Division Duplexing (TDD) approaches. FDD is geared towards radio communication systems that provide individual radio frequencies for each user, and thus is ideal for point-to-point symmetric and predictable traffic (e.g. voice traffic). Because each transceiver simultaneously transmits and receives radio signals, the frequency allocation used for the downlink and uplink must be sufficiently isolated. Furthermore, the frequency separation must be coordinated to permit the use of inexpensive RF technology. TDD enables each transceiver to operate as either a transmitter or receiver on the same frequency, and eliminates the need for separate up- and downlink frequency bands. By allocating both links in an adequately time-shared manner, TDD can handle asymmetric and unpredictable traffic better. Recalling that Internet traffic is bursty, and thus that the bandwidth needed by the up- and downlink may vary with time, we see that TDD is the best candidate duplexing method for next-generation wireless systems. However, there is a time latency due to the fact that communications are not fully duplexed. The use of a guard interval that is at least equal to the round-trip time is necessary to avoid interference between different time slots. This necessary guard interval increases with the mobile/central station distance, but timing advance can be used to reduce it.

j. Iterative methods

The “Turbo principle” should be included among the techniques to be used in next-generation wireless systems [30]. In fact, its underlying idea can be applied to many detection/decoding problems such as channel estimation, equalization, coded modulation, multiuser detection, MIMO detection, joint source and channel decoding, and so forth. It has been used for the first time in the decoding of error-correcting codes and has been shown to perform very close to the Shannon capacity. The basic idea is that the receiver has two soft detectors/decoders that iteratively update two probability densities. The more iterations are performed, the higher the reliability of the decoder output). Indeed, turbo schemes for multi-user and multi-signal stream detection can achieve maximum likelihood performance at a fraction of the computational cost. Additional research is needed to investigate iterative methods based on the Turbo principle that exploit the benefits of emerging techniques.

k. Channel estimation

Depending on the application, different channel parameters need to be estimated with varying accuracy. There are several ways to retrieve this information. In pilot-aided methods, training or pilot sequences are transmitted in order to enable the estimation. As the number of channel parameters increases, additional training has to be provided to identify the additional parameters. In contrast, blind identification techniques do not use training signals. A number of such techniques have been developed based on specific properties, e.g. the channel's cyclo-stationarity. Finally, semi-blind methods mix both training- and blind-identification-based techniques. Whereas the receiver can employ these estimation schemes to characterize the channel through which the signal has traveled, this is not directly possible at the transmitter. Channel estimates can be obtained at the transmitter either by estimating the channel at the receiver and using feedback, or by using the reciprocity principle in duplex transmission. The accuracy of the first approach depends on how the feedback delay compares with the channel coherence time, e.g. the delay has to be small compared with the coherence time to ensure that the estimate is not outdated. The reciprocity-principle approach, on the other hand, states that at a given time, frequency and location, the up- and downlink channels are equal. However, duplexing schemes support both links simultaneously and need to isolate these links to prevent interference. As a consequence, some lack of reciprocity is enforced, leading to estimation errors. Nevertheless, if the separation either in time, frequency or space is small compared with the channel coherence in that dimension, the estimate will be accurate. This implies that the reciprocity principle can in general not be used in FDD systems owing to the physical limits on duplexing filters that isolate the up- and downlinks. In fact, the frequency separation that they achieve is usually 5% of the operating frequency, leading to a separation that is larger than the channel coherence bandwidth [12]. Therefore, the estimation techniques depend very much on the specific system in which they are to be used. For example, by taking into account a system parameter such as the length of the cyclic prefix in an OFDM system, [31] derives a mean square error estimator for a Single-Input Multiple-Output (SIMO) OFDM system that is robust for any kind of channel provided that the excess delay does not exceed the cyclic prefix. This method reduces the estimation complexity because it does not need *a priori* information on the channel statistics, and the filter does not have to be adapted to changing statistics. Trends for future channel-estimation algorithms include, among others, the optimization of the tradeoff between the training overhead and the estimation quality, the characterization and optimization of the feedback channel bandwidth when CSI is to be used at the transmitter, and the improvement of the robustness of the estimation scheme against channel variations.

The investigation of efficient channel-estimation schemes for MIMO systems is also a research objective. MIMO systems ask for more training effort because the number of parameters to be estimated is proportional to the number of transmit antennas. Interesting results on channel estimation for MIMO have been provided in the context of the SM system developed at Bell Labs, i.e. BLAST [32]. It is reported that for a SM system, the required transmission interval grows approximately linearly with the number of transmit antennas. This requirement implies that for the purpose of maximizing the effective transmission rate (i.e. the number of message bits), half of the transmission-time interval has to be used for training. Ideally, the goal would be to achieve the high transmission rates offered by SM while avoiding training for channel estimation. That is what the blind estimation scheme proposed in [33] for OFDM-based systems in frequency-selective channels does. The only *a priori* information needed by that algorithm is an upper bound on the channel order (i.e. excess delay). The fundamental idea of the estimation scheme lies in pre-multiplying the individual streams of each antenna by a periodic sequence. This introduces a different signature in the cyclostationary domain, and cancels out the influence of all but one antenna at a time, thereby allowing the channel matrix identification. Although this approach does not need training, it presents a high complexity. We also point out the analysis that is performed from an information-theoretic standpoint of the optimum training amount in [34] and which is extended to frequency-selective channels in [35]. These results are a valuable basis to further investigate efficient MIMO estimation schemes for future wireless communication systems. In summary, such schemes should be robust and provide a good estimation quality while keeping the introduced overhead and the implementation complexity reasonable.

IV. SYSTEM-LEVEL ISSUES

The requirements imposed on future wireless communications systems in terms of data rates and QoS ask for significant improvements over current systems. As we have seen, MIMO systems have been recognized as a key enabling technology for meeting these requirements with considerably higher spectral efficiency and link reliability. Additionally, the use of multicarrier transmission in conjunction with multiple antennas offers important advantages in broadband wireless environments. In order to understand accurately how these technologies should be integrated in future wireless systems, research cannot ignore system-level considerations. Indeed, adequate attention has to be paid to how the capabilities that new transmission schemes offer should be exploited by the upper-layer protocols so as to deliver the best possible QoS in terms of throughput, delay and jitter to the end-users. The system efficiency would further improve if different layers were allowed to interact in a controlled way. This calls for a cross-layer design approach in which the transmission schemes and protocols would no longer be designed independently. Hereafter, we will discuss several ideas to be investigated at the system level that aim in this direction.

a. MAC design

Medium Access Control (MAC) protocols define rules to allow efficient and fair access to the shared wireless medium, and thereby play a crucial role in determining the network performance. Next-generation communication systems require a MAC protocol supporting native QoS, implying that it should be able to associate appropriate service attributes with each application (bandwidth, delay and delay jitter) and ensure that the network supports them. Clearly, in cases where the necessary provisions are not available, no access to the service should be allowed (i.e. admission control). In order to meet this goal, the MAC protocol has to prioritize different transmissions according to their QoS class on both the up- and the downlink. As we shall see later in this section, scheduling is a very useful technique in this respect since it allows the support of multiple service flows on a link.

Usually, a MAC protocol provides either connection-oriented or connectionless services. Traditionally connection-oriented services are provided for voice transmission. In the wireless environment, voice channels are dedicated for the duration of a call (i.e. circuit switching). In order to obtain a dedicated channel, an initiation sequence is required to connect the called and calling parties. On the other hand, connectionless services are targeted to cases where dedicated resources are not required. Wireless data networks are not well supported by circuit switching, as their traffic is often short, bursty, and has periods of inactivity. In particular, packet switching is a common technique used for connectionless services, in which all users access the channel randomly without requiring a setup procedure. Packet switching breaks each message into smaller units for transmission. This helps to reduce the packet-error rate and the latency. Nevertheless, when fragmentation is done, a certain amount of control information has to be added to each packet to provide source and destination identification as well as error-recovery capabilities (acknowledgement, Automatic Repeat Requests (ARQ), sequencing, error detection…). Because the channel is utilized only when bursts of information are being received or sent, packet switching provides excellent channel efficiency for bursty transmissions [36]. Additionally, MAC protocols can be specifically designed to carry both voice and data traffic by enabling connection and connectionless services [37]. In the example given, a TDMA mechanism is used to provide an isochronous voice service, while Carrier Sensing Multiple-Access with Collision Avoidance (CSMA/CA) is employed for asynchronous data services. The superframes used by the MAC incorporates both services by alternating Contention-Free Periods (CFP), in which a control point manages the access to the medium, and Contention Periods (CP), in which the stations contend to access the medium. In addition, the CFPs can be omitted if they are not needed. The main benefit in combining both is that real-time services can be supported while still ensuring good throughput for asynchronous data, a desirable property in future-generation communication systems.

The basic MAC protocol used in the 802.11 WLAN standard is the Distributed Coordination Function (DCF) that works as a "listen-before-talk" scheme based on CSMA/CA. In order to support isochronous services as well, the standard defines the Point Coordination Function (PCF), which lets the stations have priority access to the wireless medium, coordinated by a station called Point Coordinator (PC). There is no contention among the stations: either the PC polls them, asking for a pending frame, or it transmit the frames it has pending for a given station. With PCF, a CFP and a CP alternate over time. During the CFP, the PCF is used for accessing the medium, whereas the DCF is used during the CP [38]. There are problems with the PCF that led to the current activities to enhance the protocol with QoS-support capabilities. For instance, the transmission time of polled stations is not under the control of the PC and therefore it is not possible to guarantee QoS to other stations that are polled during the remainder of the CFP. To support QoS, 802.11 Task Group E defines enhancements to the above MAC protocol [39]. There may still be the two phases of operation, i.e. a CP and a CFP, which alternate continuously over time. The Enhanced DCF (EDCF) is used in the CP only, whereas a Hybrid Coordination Function (HCF) is used in both phases. The EDCF introduces Traffic Categories (TCs) which allow MAC packets to be delivered through multiple instances within one station, each one having its TC-specific parameters. A single station can therefore implement up to eight transmission queues realized as virtual stations inside it, with different QoS parameters determining their corresponding priorities. Furthermore, EDCF is complemented by the introduction of the HCF, which may initiate delivery of MAC packets to the stations whenever it wants because it has higher priority. During either CP or CFP, the HCF can poll stations and grant them transmission opportunities. Finally, the controlled contention mechanism allows stations to request the allocation of transmission opportunities by making the HCF aware of their individual transmission needs. Therefore, the HCF provides the means for delivering time-bounded traffic at the cost of a significant increase in protocol complexity. A comprehensive overview of the new QoS-supporting features of 802.11e compared with the legacy standard can be found in [40].

Although this modified protocol support QoS, its main drawback lies in the throughput limitations of the 802.11 MAC itself. In fact, all 802.11 physical layers, which provide data rates ranging from 2 up to 54 Mbps, use the same MAC protocol. As the data rate increases, the overhead of the protocol also increases. This statement has been proved in a theoretical study of the DCF reported in [41], where the overhead is shown to limit the maximum throughput as the data rate provided by the physical layer goes to infinity. It is worth noting that this upper limit is found for the best-case scenario: the channel is noise-free and only one station has a packet to send at a given time. Therefore, real conditions such as a noisy channel and collisions will further degrade the achievable throughput. For instance, if the size of the payload in 802.11a is 1000 bytes, the maximum throughput that can be achieved is 24.7 Mbps when the speed provided by the physical layer is 54 Mbps. Furthermore, if the data rate was allowed to go to infinity the maximum throughput obtained would not exceed 51 Mbps for the same payload size. The only way to have a throughput that increases with the data rate would be to allow the payload sizes to grow as well. As we have seen, a packet-switched service such as CFP performs better for small packets. Therefore, regardless of the speed future transmission strategies will reach, the current 802.11 MAC protocol will not be able to exploit it fully. These results force the conclusion that a new MAC protocol that will take into account the specificities of the physical layer it interacts with has to be designed for future WLANs systems. We note that this makes the requirement of backward compatibility (i.e. with legacy protocols) impossible to fulfill, at least if full compatibility is intended. The work reported in [42] is in contrast with these considerations because it shows the ability of 802.11e MAC to provide QoS for demanding applications in terms of delay constraints and throughput.

Several ideas concerning the characteristics of the new MAC protocol have already been formulated. For example, an adaptive protocol that would be reconfigurable depending on the traffic conditions has been proposed: for high-load traffic, a TDMA-based protocol would be used, whereas for lower loads, random access would be preferred. Interference-aware MAC protocols have also been proposed in order to support a dynamic change of transmission scheme depending on the level of multiuser interference. Nevertheless, further research is still necessary to investigate these and other schemes.

b. Radio-Link Protocol

A Radio-Link Protocol (RLP) is a recognized mechanism designed for the wireless environment to deal specifically with the types of impairments found on the radio link between the mobile and the central station. The RLP is positioned at the link layer of the protocol stack and is responsible for asynchronously transporting data between unspecified upper-layer entities. The detailed mechanisms employed by an RLP are usually specific to a particular air interface and tailored to it. For instance, it can provide mechanisms to deal with errors on the link. Error detection can be performed alone or in conjunction with correction or retransmission. It may also help to control the delay encountered in transmissions where the exact scheme can be designed to provide bounded delays. Different mechanisms that recover from all or part of the transmission errors are employed to provide delivery guarantees. Finally, the so-called bandwidth conservation schemes aim at compressing headers or payloads, for example on an application-specific basis. Although some of these concepts have already been employed in current standards, research still has to determine the most efficient way to integrate them in future systems. Next, we will discuss two RLP schemes that are very promising candidates to be included into next-generation communication systems.

Link Adaptation – Link-Adaptation (LA) algorithms are a powerful tool to increase the data rate and spectral efficiency of wireless networks. The basic idea behind these algorithms is to adapt the transmission parameters dynamically to the changing environmental and interference conditions. The fundamental set of parameters to be adapted includes modulation and coding, but it can also be extended to parameters such as the power level, the spreading factor, the signaling bandwidth and others. In practical systems, the possible values that the adjusted parameters will take on are grouped together into a set of modes. The purpose of LA is to ensure that the most efficient mode is always used over the varying channel conditions. In systems in which only one mode is available, LA has generally been designed to maintain acceptable performance for poor channel conditions. Consequently, when the channel is good, the system is unable to take advantage of this and does not exploit the full channel capacity. In contrast, if several modes are available, the system can provide increased spectral efficiency under favorable conditions and reduce it to ensure robustness under bad conditions. Regarding the adaptation rate of the LA algorithm, it clearly has to be larger than the channel's coherence time to derive the maximum benefit. Because the wireless channel exhibits not only small-scale but also large-scale fading, adaptation gains can still be realized at lower rates.

A sensitive issue in LA algorithms is the definition of a channel quality indicator based on which the adaptation will be done. There are various metrics that can be employed to compute the switching threshold between different modes [43]. Often the metric chosen is the average signal-to-noise ratio (SNR) measured at the receiver. For a given error performance target, the minimum SNR that achieves it in each mode is computed and retained as a switching threshold. Then, the measured average SNR is used to select the optimum mode according to the computed thresholds and is fed back to the transmitter. This method poses a number of problems under real-world conditions. Indeed, the delay introduced by the feedback operation and other implementation limitations will not allow an instantaneous adaptation, and if this delay is bigger than the channel's coherence time, the new mode will not be optimum. Furthermore, when the channel exhibits fading within the SNR estimation window, it becomes more complicated to derive the switching thresholds for a given error performance because the pdf of the SNR is a function involving many parameters (fading statistics over time and frequency, number of antennas used in the system, transmission and reception schemes…). In that case, higher-order statistics of the SNR have to be considered to obtain statistical information about the SNR distribution. This simplifies the threshold determination as it can then be done based on the moments of the SNR distribution and rather than on the exact distribution. Nevertheless, the order and the number of moments to be computed have to remain low in order to keep the complexity reasonable. The challenge consists in balancing the estimation accuracy with the computation efficiency.

Another metric that can be used to perform LA relies on tracking errors in the reception of packets for each candidate mode. The observed Packet-Error Rate (PER) provides explicit channel information for each trained mode. However, the estimation accuracy depends on the number of packets observed within a window, and therefore, this method is traffic-dependent. If the user does not send or receive packets, one loses track of the channel quality. Consequently, the reaction time of the algorithm is difficult to control. In general, measuring the ability of the scheme to perform fast adaptation under realistic traffic and without requiring additional bandwidth (e.g. due to regular scheduling of training packets) is an open research problem. In summary, the SNR-based method is able to adapt quickly, but an accurate computation of the switching thresholds is difficult to obtain. On the other hand, the error-tracking method provides accurate statistics only after observing a substantial amount of traffic, making adaptation slow. A current research topic would be to find a robust scheme by combining the two approaches.

The motivation for using metrics that do not require feedback from the receiver lies not only in the practical problems caused by providing the transmitter with CSI, but also in the fact that, in general, current systems (e.g. 802.11) do not support any protocol means for the receiver to inform the transmitter about the actual channel quality. In this context, the algorithm investigated in [44] uses only information that is available at the transmitter to determine the link quality and perform the corresponding adaptation. The idea is to use the error-recovery procedure defined in the 802.11 MAC. The transmitter maintains two counters for each destination MAC address, one for successful transmissions, the other for failed transmissions. Successful transmissions indicate good channel conditions and failed transmissions, on the other hand, evidence a deterioration of the propagation environment. In the case of failure, the corresponding counter is incremented, and the success counter is set to zero (and vice versa in the success case). Each counter has an associated threshold that determines how many failure or success events have to be observed before switching to a new operational mode. The main concern is to optimize the switching thresholds to ensure good performance. The simulations reported [44] show that with appropriate threshold optimization, this adaptation scheme results in a substantial throughput increase compared with fixed data rates and that it performs very close to a system in which a "genie", having perfect knowledge of the propagation conditions, would choose the optimum transmission parameters. Furthermore, this method provides a significant enhancement of the delay performance with respect to any fixed rate scheme.

Concerning future wireless communication systems, current LA schemes should be extended to take into account new dimensions such as frequency and space, in which different transmissions schemes may also be adapted. As we have seen, the use of MIMO systems calls for intelligent ways to map the signal to be transmitted to the various antennas. This mapping must be done according to the channel characteristics, and this is where LA has to come into play. Indeed, the tradeoff between multiplexing and diversity should be determined as a function of the channel-matrix characteristics. For instance, signals that are sent independently over a rank-deficient MIMO channel cannot be recovered. In that case, a space-time coding approach, in which the transmitted signal is redundant, should be used. In general, MIMO channels with high rank are able to provide higher multiplexing gains, whereas channels with high Frobenius[1] norm are better suited for space-time coding. Experimental studies have shown that adapting the MIMO transmission mode to the spatial channel statistics can result in performance improvement of one order of magnitude in terms of bit-error rate [45]. Therefore, the antenna-mapping strategy should be treated as an additional parameter to be adapted by the LA algorithm. In the same way, multicarrier systems call for a scheme to map the information bits efficiently over the different tones. The LA algorithm would associate an independent mode (modulation and coding) to each tone. The idea would be to avoid transmitting over faded subcarriers and use high-level modulation on tones offering good channel conditions. Although this technique leads to high capacity gains, its implementation is impractical because of the substantial amount of channel knowledge required at the transmitter. An alternative solution that is still beneficial is to perform this adaptation on a subband basis, where several tones are grouped to be assigned the same mode (i.e. aggregation).

The challenges encountered in designing efficient LA algorithms can be summarized as follows. The adaptation thresholds, obtained either by simulation or measurement, can be expressed in terms of average SNR or post-processing SNR (used in MIMO systems for transparency of the antenna setup parameters). In order to keep the computational complexity of the LA algorithm at a reasonable level, the goal is to trade off the amount of statistical information computed and the resulting accuracy of the adaptation. The adaptation rate is also an issue of big concern because to obtain larger capacity gains, it has to be faster than the channel variations. Additionally, practical limitations include the signaling overhead for mode-change messages, which consumes bandwidth and time resources. In multicarrier systems, aggregation can be used to reduce the computational complexity but it also reduces the adaptation benefits in highly frequency-selective channels. Finally, when the adaptation algorithm uses feedback to provide the transmitter with the necessary information, attention should be paid to the information that is being fed back. For example, instead of acquiring CSI and providing it to the transmitter, the receiver could directly process this information to identify the appropriate mode to be employed and communicate this mode to the transmitter. This would reduce the amount of information to be fed back because the number of modes is typically small. Further reductions of the feedback bandwidth can be achieved by ensuring that feedback is only performed when a change of mode is required.

Automatic Repeat Request – In order to reduce the Frame-Error Rate (FER) seen by upper layers, the RLP typically uses an Automatic Repeat Request (ARQ) error-recovery mechanism, which is considered a key tool in dealing with bursty errors occurring in wireless channels. The principle of ARQ is to implement a low-latency acknowledgement and retransmission mechanism. Note that this technique is complementary to coding, as it introduces redundancy during the fraction of time when data gets corrupted. Clearly, ARQ introduces time diversity into the system because of its capability

[1] The Frobenius norm of a matrix is the sum of squares of all its elements.

to recover from noise, interference, and fades. There are two main approaches to flow and error control provided by ARQ. In the *stop-and-wait* mechanism, retransmission takes place if a frame is not correctly acknowledged before a timeout occurs at the sender. This method does not perform well in a packet environment, where a large number of small frames are transmitted. An alternative is the *sliding-window* protocol, where a single ACK message can be used to acknowledge several frames. To handle errors, the receiver sends a special reject control frame to indicate a missing or corrupted frame. In the main form of the sliding-window protocol, i.e. *Go-back-N*, upon reception of this message, the sender resends the damaged frame and all others until the end of the window. Consequently, the transmitter must retain all un-ACKed frames. Another method that allows the sender to resend only the missing frames is *selective-reject* ARQ. However, this method requires more sophistication at both link ends as the frames have to be handled out of sequence.

In addition to these traditional retransmission methods, Hybrid ARQ (HARQ) schemes have also been developed. The term HARQ is used to describe any combined coding and ARQ scheme in which unsuccessful transmission attempts are used in decoding instead of being discarded. The simplest form of Hybrid ARQ was proposed by Chase [46], where the basic idea was to send a number of repeats of each coded data packet and allow the decoder to combine multiple received copies of the coded packet weighted by the SNR prior to decoding. Another form of Hybrid ARQ is Incremental Redundancy (IR), which aims at incrementally transmitting additional redundant information as long as the decoding of a packet fails. The additional redundant information is combined with the information received previously, resulting in enhanced coding. IR offers the potential for better performance with high initial code rates at the cost of additional memory and decoding complexity. Performance can be further enhanced by exploiting the ability to handle changes in available resources: the system can intentionally change the amount of redundancy sent on a retransmission.

The benefits of IR have been recognized and integrated into standardized systems. In 3G CDMA standards bodies, the consensus is to explicitly define and allow IR, while still allowing Chase combining as a subset of IR. Several studies have been made in that context to assess the performance of the different HARQ variants. For instance, we cite [47], where the performance improvements of HARQ are demonstrated by means of simulation. HARQ is shown to enable fast LA by making the selection of the initial mode tolerant to errors. Whereas Chase combining works well at low speeds, IR outperforms it at higher speeds and when several users have to be supported. Regarding the integration of such a method into next-generation systems, further research efforts have to be undertaken. The performance of several schemes should be assessed and compared, taking into account other future transmission methods, e.g. MIMO and multicarrier systems. Furthermore, the efficient combination of HARQ and LA is a logical objective to be investigated as the need for a retransmission denotes a bad channel quality, offering the possibility of doing a mode-switch. This fact is confirmed by the afore-mentioned LA scheme reported in [44].

c. Scheduling

Multiuser diversity is a form of diversity inherent in wireless networks. Using the same independence argument as for the diversity concepts considered previously, it is made available by the independent time-varying channels that various users experience. It is exploited by scheduling the different users when their instantaneous channel quality is near its peak.

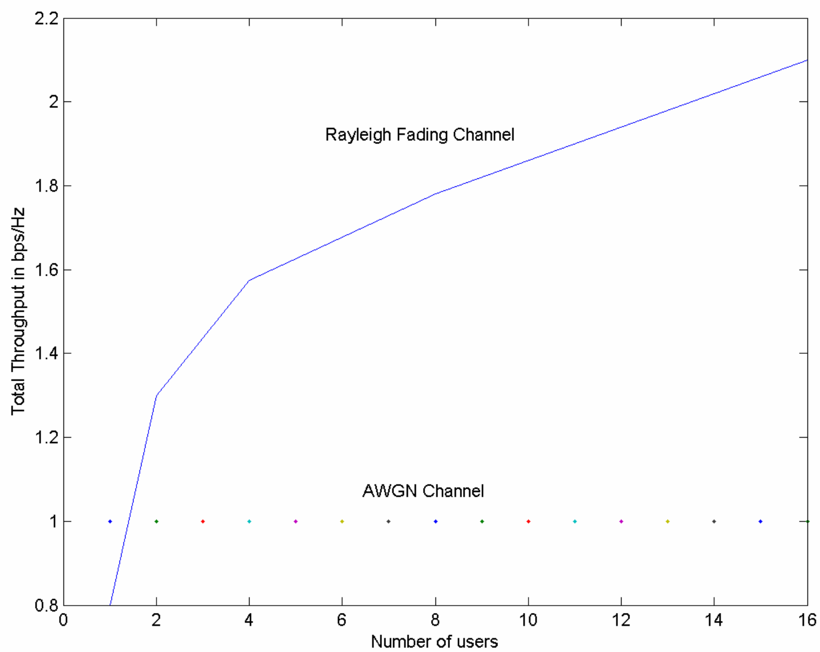

Fig. 4: Sum capacity of the AWGN and Rayleigh fading channels with average SNR = 0 dB [49].

In order to be able to perform such a scheduling, the channel fluctuations of the different users need to be tracked. The benefits of multiuser diversity are evidenced by the information-theoretic results reported in [48]. In a single-cell environment, the authors focused on the uplink with multiple users communicating through time-varying fading channels with the central station. In order to maximize the total information-theoretic capacity, they showed that the optimum strategy is to schedule at any one time the user with the best channel. In a system in which numerous users experience independent time-varying channels, the probability at any time to have one user with an instantaneous SNR that is much higher than the average is high. Therefore, if the best user is chosen at all times, the spectral efficiency of the system can be made higher than that of a channel without fades having the same average SNR. From this point of view, fading appears like a source of randomness that can be exploited. The resulting sum-capacity of a system that schedules the user with the best channel in each time slot is illustrated in Fig. 4 for an increasing number of users.

A factor limiting the gains obtained by multiuser diversity is the dynamic range of the channel fluctuations. Indeed, the wider the fluctuations, the peakier the users' channels and consequently, the higher the multiuser diversity gain. In practical systems, channels can show small variations due to a LOS component and poor scattering. Slowly-varying channel also cause problems in applications with demanding delay constraints because the transmissions cannot wait for the peak of a given user. In that case, multiple antennas at the transmitter can be used to induce artificial channel fluctuations by modulating the signal from each antenna with a pseudo-random sequence of complex gains [49]. These results suggest that the channel variations should be made as large as possible so that each user can be scheduled when that user has a very strong channel. This is accomplished by varying the strengths of both the signal and interference a user receives. Therefore, multiple antennas, which are traditionally used to improve the reliability of point-to-point links, are used in that approach to induce high fluctuations in the Multi-User (MU) MIMO channel such that high scheduling gains are obtained.

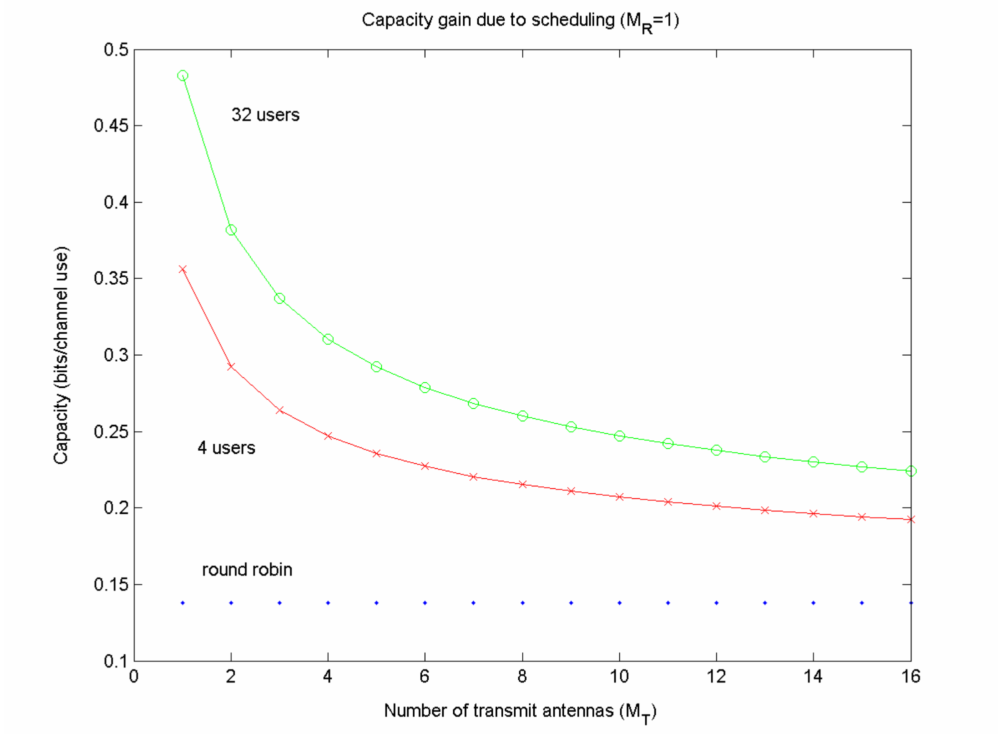

Fig. 5: Scheduling capacity gain w.r.t. a round-robin scheduling [50].

In order to characterize the amount of fluctuation provided by the channel, it is useful to analyze it from an information-theoretic point of view. As we have seen, the ergodic capacity growth in a MIMO system is known to be proportional to the minimum number of antennas used. In contrast, it turns out that the variance of the instantaneous capacity can grow very slowly and even shrink as the number of antennas increases. In fact, as more antennas are being added to the system in order to achieve MIMO benefits, the channel "hardens", thereby limiting the gains provided by scheduling. This channel hardening as well as its implications for the scheduling gains expected in a multiuser MIMO environment and the amount of overhead needed for rate control with feedback were studied in [50]. Figure 5 illustrates the capacity gains that these authors obtained compared with a round-robin approach for 4 and 32 users. As it can be seen, the higher the number of transmit antennas M_T, the smaller the gain in capacity (note that M_R is fixed). These results are valid in a purely Rayleigh fading environment, but they also showed that by adding shadow fading, both MIMO and scheduling gains could be obtained simultaneously.

In single-antenna systems, typically the user with the best instantaneous SNR is scheduled for transmission. As far as multi-antenna systems are concerned, it is unclear which measure should be used to schedule the different transmissions. We recall that MIMO channels with high rank are better suited to SM and that space-time coding fits MIMO channels with a large Frobenius norm better. This consideration shows that scheduling in a MIMO system requires a careful adaptation of the signaling strategy. Therefore, the question of optimum allocation of the spatial degrees of freedom between space-time/frequency coding, SM, and inducing channel fluctuations arises. Moreover, in broadband channels, the presence of frequency diversity would be another dimension to be exploited in the design of optimum scheduling algorithms. Further research in this area should assess the impact of varying channel conditions on the optimum scheduling strategies to be used and derive scheduling algorithms that would be able to exploit knowledge of the MIMO channel statistics.

VI. CONCLUSION

Wireless LANs have emerged as a key technology in the wireless evolution towards next-generation communication systems. Integrating emerging transmission technologies such as multiple antennas into these systems promises significant improvements in terms of spectral efficiency, error performance, and QoS provision. Indeed, future WLANs will be able to compete with wired technologies in the home networking and corporate environments, and enhance the services provided in hotspots. Nevertheless, this integration still requires very challenging research to be conducted.

The use of MIMO techniques in wireless systems offers spatial multiplexing gain, diversity gain, interference-canceling gain and array gain. While these leverages are well understood individually, the underlying dependencies are not yet clear. Their investigation is a crucial step if the degrees of freedom provided by multiple antennas are to be shared optimally depending on the propagation conditions. Moreover, in contrast to SISO systems, exploiting channel knowledge at the transmitter can yield significant performance enhancements. Nevertheless, the transmitter strategies required are still an issue necessitating additional research. In order to utilize channel knowledge at either the transmitter or the receiver, efficient channel-estimation schemes that balance estimator accuracy, implementation complexity, and the amount of training introduced have to be determined. Additionally, multicarrier systems have been shown to exploit the frequency diversity present in the broadband channel efficiently. In particular, OFDM shows a reduction in implementation complexity, which is especially interesting when used in conjunction with multiple antennas. As far as the multiuser capabilities of MIMO systems are concerned, there are still numerous open research problems. For instance, the ultimate performance limits of the multiple access and broadcast channels have to be assessed. In an environment in which encoding/decoding cooperation across the antennas is not always possible, efficient coding schemes for those channels need to be designed. Furthermore, the necessity of jointly designing the coding and multiple-access schemes has been pointed out. In order to enable an efficient design of transmitter and receiver architectures for future wireless systems, challenging research activities leading to appropriate combinations of the transmission techniques considered have to be carried out.

At the data-link layer, we discussed the enhancements that have to be done at the MAC and RLP sub-layers. In particular, current MAC protocols for WLANs are not very efficient because they introduce significant overhead, thereby limiting the quality of service they provide to upper layers. In fact, they are not able to exploit the capabilities of the physical layer fully. A joint design with the physical layer is envisioned as the solution to improve MAC schemes for wireless systems in general. Link-adaptation protocols that match the transmission parameters to the channel conditions should be developed in order to use the channel resources optimally. The dimensions introduced by MIMO and multicarrier systems should be integrated, not only in the adaptation modes, but also in ARQ techniques or variants thereof. Scheduling is considered to enhance the spectral efficiency of the system significantly by exploiting multiuser diversity. Optimum scheduling schemes need to be determined for MIMO systems. For that purpose, a new measure on which the scheduling decision should be based has to be derived. Furthermore, the investigation of the dependencies between multi-user diversity gains and other MIMO gains is an aspect of high importance because it will allow the optimum allocation of the degrees of freedom provided by the channel.

This report aims at identifying candidate technologies for next-generation wireless systems. Although the discussion is far from exhaustive, an accurate picture of the challenges that research is facing has been provided. Future systems, led by WLANs, will be able to meet the expectations placed on them if the leverages provided by the use of multiple antennas in conjunction with more traditional techniques are fully exploited at the physical and data link layers.

ACKNOWLEDGMENTS

The authors thank Professor Helmut Bölcskei for his valuable input to this survey as well as the members of the Wireless Communications Group at IBM's Zurich Research Laboratory for helpful discussions.

REFERENCES

[1] G. Paulo, "Wherever, Whenever: WLANs Make Business Networking Nomadic," Report No. IN020176LN, In-Stat / MDR, July 2002.

[2] N. Theodosopoulos and J. Susman, Global Telebits, UBS Warburg, October 14, 2002.

[3] A. Cravens, "How Hot are Hotspots: The Developing Market for Public Area Access," Report No. MU0110SG, Cahners In-Stat Group, December 2001.

[4] J. Rosdahl, "Draft Project Authorization Request HT-SG," Doc. No. 02/798, November 2002, available at http://grouper.ieee.org/groups/802/11/Documents

[5] "The Book of Visions 2001-Visions of a Wireless World," Wireless World Research Forum, December 2001, draft available at http://www.wireless-world-research.org

[6] E. Telatar, "Capacity of Multi-Antenna Gaussian Channels," Technical Report No. BL0112170-950615-07TM, AT&T Bell Laboratories, 1995.

[7] S. M. Alamouti, "A Simple Transmit Diversity Technique for Wireless Communications," *IEEE J. Sel. Areas Commun.*, vol. 16, pp. 1451-1458, Oct. 1998.

[8] R. Nabar, H. Bölcskei and A. Paulraj, "Outage Performance of Space-Time Block Codes for Generalized MIMO Channels," preprint, submitted to *IEEE Trans. Inform. Theory*, March 2002.

[9] G. Foschini, "Layered Space-Time Architecture for Wireless Communications in a Fading Environment when using Multi-Element Antennas," *Bell Labs Tech. J.*, pp. 41-59, 1996.

[10] A. Paulraj and T. Kailath, "Increasing Capacity in Wireless Broadcast Systems using Distributed Transmission/Directional Antennas," US patent, (5,345,599), 1994.

[11] E. Telatar, "Capacity of multi-antenna Gaussian channels," *European Trans. Tel.*, vol. 10, no. 6, pp. 585-595, Nov./Dec. 1999.

[12] A. Paulraj, R. Nabar and D. Gore, *Introduction to Space-Time Wireless Communications*, Cambridge Univ. Press, 2003.

[13] H. Bölcskei, "Fundamental Performance Tradeoffs in Coherent MIMO Signaling," submitted to Globecom 2003.

[14] V. Tarokh, N. Seshadri and A. R. Calderbank, "Space-Time Codes for High Data Rate Wireless Communication: Performance Criterion and Code Construction," *IEEE Trans. Inform. Theory*, vol. 44, no. 2, pp. 744-765, March 1998.

[15] B. Hassibi and B. M. Hochwald, "High-Rate Codes that are Linear in Space and Time," *IEEE Trans. Inform. Theory*, vol. 48, no. 7, pp. 1804-1824, July 2002.

[16] O. Oyman, R. Nabar, H. Bölcskei and A. Paulraj, "Characterizing the Statistical Properties of Mutual Information in MIMO Channels: Insights into Diversity-Multiplexing Tradeoff", preprint, submitted to *IEEE Trans. Signal Processing*, Dec. 2002.

[17] L. Zheng and D. Tse, "Diversity and Multiplexing: A Fundamental Tradeoff in Multiple Antenna Channels," *IEEE Trans. Inform. Theory*, to appear.

[18] G. Caire and S. Shamai (Shitz), "On the Capacity of Some Channels with Channel State Information," *IEEE Trans. Inform. Theory*, vol. 45, no. 6, pp. 2007-2019, Sept. 1999.

[19] E. Visotsky and U. Madhow, "Space-Time Transmit Precoding With Imperfect Feedback," *IEEE Trans. Inform. Theory*, vol. 47, no. 6, pp. 2632-2639, Sept. 2001.

[20] G. Jongren, M. Skoglund and B. Ottersten, "Combining Beamforming with Orthogonal Space-Time Block Coding," *IEEE Trans. Inform. Theory*, vol. 48, no. 3, pp. 611-627, March 2002.

[21] H. Bölcskei and A. Paulraj, "Space-Frequency Coded Broadband OFDM Systems," Invited presentation at *IEEE WCNC 2000,* Chicago (IL), Sept. 2000.

[22] H. Bölcskei, M. Borgmann and A. Paulraj, "Impact of the Propagation Environment on the Performance of Space-Frequency Coded MIMO-OFDM," *IEEE J. Sel. Areas Commun.* (special issue on "MIMO Systems and Applications"), Apr. 2003, to appear.

[23] ---, "Space-Frequency Coded MIMO-OFDM with Variable Multiplexing-Diversity Tradeoff", *IEEE ICC 2003, Anchorage, Alaska*, May 2003, to appear.

[24] H. Bölcskei, D. Gesbert and A. Paulraj, "On the Capacity of OFDM-Based Spatial Multiplexing Systems", *IEEE Trans. Commun.*, vol. 50, no. 2, pp. 225-234, Feb. 2002.

[25] H. Bölcskei and M. Borgmann, "Code Design for Non-Coherent MIMO-OFDM Systems," *Allerton Conf. on Communication, Control, and Computing*, Monticello, IL, Oct. 2002 (invited paper).

[26] B. Muquet *et al.*, "Cyclic Prefix or Zero Padding for Wireless Multicarrier Transmissions?" *IEEE Trans. Commun.*, vol. 50, no. 12, pp. 2136-2148, Dec. 2002.

[27] G. Fettweis *et al.*, "On Multi-carrier Code Division Multiple Access (MC-CDMA) Modem Design," *Proc. IEEE 44th VTC*, vol. 3, pp. 1670-1674, June 1994.

[28] B. Ottersten, "Array Processing for Wireless Communications," *Proc. 8th IEEE Signal Processing Workshop on Statistical Signal and Array Processing*, pp. 466-473, Corfu, Greece, June 1996.

[29] R. G. Gallager, "An Inequality on the Capacity Region of Multi-access Multipath Channels," in *Communications and Cryptography – Two Sides of One Tapestry*, pp. 129-139, Kluwer, Dordrecht, 1994.

[30] J. Hagenauer, "The Turbo Principle in Communications," *International Symposium on Information Theory and its Applications*, Xi'an, PRC, Oct. 2002.

[31] S. Furrer and D. Dahlhaus, "Mean Bit-Error Rates for OFDM Transmission with Robust Channel Estimation and Space Diversity Reception," in *Proc. International Zurich Seminar on Broadband Communications*, pp. 47-1–47-6, Zurich, Switzerland, Feb. 2002.

[32] T. L. Marzetta, "BLAST Training: Estimating Channel Characteristics for High Capacity Space-Time Wireless," *Proc. Allerton Conf. on Communication, Control and Computing*, Monticello (IL), Sept. 1999.

[33] H. Bölcskei *et al.*, "Blind Channel Identification and Equalization in OFDM-based Multi-Antenna Systems," *IEEE Trans. Signal Processing*, vol. 50, no. 1, pp. 96-109, Jan. 2002.

[34] B. Hassibi and B. Hochwald, "Optimal Training in Space-Time Systems," *Proc. 44th Asilomar Conf. on Signals, Systems and Computers*, vol. 1, pp. 743-747, 2000.

[35] H. Vikalo et al., "Optimal Training for Frequency-Selective Fading Channels," *Proc. IEEE International Conf. on Acoustics, Speech and Signal Processing*, vol. 4, pp. 2105-2108, 2001.

[36] T. S. Rappaport, *Wireless Communications: Principles and Practice*, Prentice Hall PTR, 1996.

[37] J. Lansford, "HomeRF™/SWAP: A Wireless Voice and Data System for the Home," *IEEE Proceedings of ICASSP*, vol. 6, 2000.

[38] IEEE 802.11 WG, "Reference number ISO/IEC 8802-11:1999 (E) IEEE STD 802.11, 1999 edition. International Standard [for] Information Technology-Telecommunications and information exchange between systems - LAN/MAN Specific Requirements- Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications," 1999.

[39] IEEE 802.11 WG, "Draft Supplement to STANDARD FOR Telecommunications and Information Exchange between Systems - LAN/MAN Specific Requirements – Part 11: Wireless Medium Access Control (MAC) and physical layer (PHY) specifications: MAC Enhancements for Quality of Service (QoS)," IEEE 802.11e/D2.0, Nov. 2001.

[40] S. Mangold *et al.*, "IEEE 802.11e Wireless LAN for Quality of Service," *Proc. European Wireless*, vol. 1, pp. 32-39, Florence, Italy, Feb. 2002.

[41] Y. Xiao and J. Rosdahl, "Throughput and Delay Limits of IEEE 802.11," *IEEE Commun. Lett.*, vol. 6, no. 8, pp. 355-357, August 2002.

[42] H. L. Truong and G. Vannuccini, "Performance Evaluation of the QoS Enhanced IEEE 802.11e MAC Layer," *57th IEEE Semiannual Vehicular Technology Conference*, Jeju Korea, April 2003.

[43] S. Catreux, D. Gesbert and R. W. Heath, Jr., "Adaptive Modulation and MIMO Coding for Broadband Wireless Data Networks," *IEEE Commun. Mag.*, pp. 108-115, June 2002.

[44] P. Chevillat, J. Jelitto, A. Noll Barreto and H. L. Truong, "A Dynamic Link Adaptation Algorithm for IEEE 802.11a Wireless LANs" *IEEE Int. Conf. on Communications (ICC),* Anchorage, Alaska, May 2003.

[45] R. U. Nabar *et al.*, "Performance of Multi-antenna Signaling Strategies using Dual-Polarized Antennas," in *Intl. Symp. on Wireless Personal Multimedia Communications (WPMC)*, Aalborg, Denmark, pp. 175-180, Sept. 2001.

[46] D. Chase, "Code Combining: A Maximum-Likelihood Decoding Approach for Combining an Arbitrary Number of Noisy Packets," *IEEE Trans. Commun.*, vol. 33, pp. 593-607, May 1985.

[47] R. Love *et al.*, "Incremental Redundancy for Evolutions of 3G CDMA Systems," *IEEE 55th Vehicular Technology Conference*, vol. 1, pp. 454-458, 2002.

[48] R. Knopp and P. Humblet, "Information Capacity and Power Control in Single Cell Multiuser Communications," in *Proc. IEEE Int. Computer Conf. (ICC'95)*, Seattle, WA, June 1995.

[49] P. Viswanath, R. Laroia and D. Tse, "Opportunistic Beamforming using Dumb Antennas," *IEEE Trans. Inform. Theory*, vol. 48, pp. 1277-1294, June 2002.

[50] B. M. Hochwald, T. L. Marzetta and V. Tarokh, "Multi-antenna Channel Hardening and its Implications for Rate Feedback and Scheduling," preprint, submitted to *IEEE Trans. Inf. Theory*, 2002.