Adaptive Modulation and MIMO Coding for Broadband Wireless Data Networks

Severine Catreux and Vinko Erceg, Iospan Wireless Inc.
David Gesbert, University of Oslo
Robert W. Heath, Jr., The University of Texas at Austin

ABSTRACT

Link adaptation techniques, where the modulation, coding rate, and/or other signal transmission parameters are dynamically adapted to the changing channel conditions, have recently emerged as powerful tools for increasing the data rate and spectral efficiency of wireless data-centric networks. While there has been significant progress on understanding the theoretical aspects of time adaptation in LA protocols, new challenges surface when dynamic transmission techniques are employed in broadband wireless networks with multiple signaling dimensions. Those additional dimensions are mainly frequency, especially in multicarrier systems, and space in multiple-antenna systems, particularly multiarray multiple-input multiple-output communication systems. In this article we give an overview of the challenges and promises of link adaptation in future broadband wireless networks. We suggest guidelines to help in the design of robust, complexity/cost-effective algorithms for these future wireless networks.

INTRODUCTION

The wireless communications industry is gaining momentum in both fixed and mobile applications. The continued increase in demand for all types of wireless services (voice, data, and multimedia) is fueling the need for higher capacity and data rates. Although improved compression technologies have cut the bandwidth needed for voice calls, data traffic will demand much more bandwidth as new services come online. In this context, emerging technologies that improve wireless systems spectrum efficiency are becoming a necessity, especially in broadband applications. Some popular examples include smart antennas, in particular multiple-input multiple-output (MIMO) technology, coded multicarrier modulation, link-level retransmission, and adaptive modulation and coding techniques [1].

Popularized by cellular wireless standards such as Enhanced Data GSM Evolution (EDGE), adaptive modulation and coding techniques that can track time-varying characteristics of wireless channels carry the promise of significantly increasing data rates, reliability, and spectrum efficiency of future wireless data-centric networks. The set of algorithms and protocols governing adaptive modulation and coding is often referred to as *link adaptation* (LA).

While substantial progress has been accomplished in this area to understand the theoretical aspects of time adaptation in LA protocols, more challenges surface as dynamic transmission techniques must take into account the additional signaling dimensions explored in future broadband wireless networks. More specifically, the growing popularity of both multiple transmit antennas systems — MIMO and multi-input single-output (MISO) — and multicarrier systems such as orthogonal frequency-division multiplexing (OFDM) creates the need for LA solutions that integrate temporal, spatial, and spectral components together. The key issue is the design of robust low-complexity and cost-effective solutions for these future wireless networks.

The article is organized as follows. First we introduce traditional LA techniques. We then discuss other emerging approaches for increasing the spectral efficiency in wireless access systems with an emphasis on interactions with the LA layer design. We focus on smart antenna techniques and coded multicarrier modulations. We continue with a short overview of space-time broadband wireless propagation characteristics. We then explore various ways of capturing channel information and provide some guidelines for the design of sensible solutions for LA. Finally, we emphasize the practical limitations involved in the application of LA algorithms and give examples of practical performance.

 0163-6804/02/$17.00

LINK ADAPTATION FUNDAMENTALS

The basic idea behind LA techniques is to adapt the transmission parameters to take advantage of prevailing channel conditions. The fundamental parameters to be adapted include modulation and coding levels, but other quantities can be adjusted for the benefit of the systems such as power levels (as in power control), spreading factors, signaling bandwidth, and more. LA is now widely recognized as a key solution to increase the spectral efficiency of wireless systems. An important indication of the popularity of such techniques is the current proposals for third-generation wireless packet data services, such as code-division multiple access (CDMA) shemes like cdma2000 and wideband CDMA (WCDMA) and General Packet Radio System (GPRS, GPRS-136) including LA as a means to provide a higher data rate [2].

The principle of LA is simple. It aims to exploit the variations of the wireless channel (over time, frequency, and/or space) by dynamically adjusting certain key transmission parameters to the changing environmental and interference conditions observed between the base station and the subscriber. In practical implementations, the values for the transmission parameters are quantized and grouped together in what we refer to as a set of *modes*. An example of such a set of modes, where each mode is limited to a pair of modulation level and coding rate, is illustrated in Table 1 (extracted from [2]). Since each mode has a different data rate (expressed in bits per second) and robustness level (minimum signal-to-noise ratio (SNR) needed to activate the mode), they are optimal for use in different channel/link quality regions. The goal of an LA algorithm is to ensure that the most efficient mode is always used, over varying channel conditions, based on a mode selection criterion (maximum data rate, minimum transmit power, etc). Making modes available that enable communication even in poor channel conditions renders the system robust. Under good channel conditions, spectrally efficient modes are alternatively used to increase throughput. In contrast, systems with no LA protocol are constrained to use a single mode that is often designed to maintain acceptable performance when the channel quality is poor to get maximum coverage. In other words, these systems are effectively designed for the worst-case channel conditions, resulting in insufficient utilization of the full channel capacity.

The capacity improvement offered by LA over nonadaptive systems can be remarkable, as illustrated by Fig. 1. In this figure, we represent the link-level spectral efficiency (SE) performance (b/s/Hz) vs. the short-term average SNR γ in dB, for four different uncoded modulation levels referred to as binary phase shift keying (BPSK), quaternary PSK (QPSK), 16-quadrature amplitude modulation (QAM), and 64-QAM. The SE was obtained for each modulation by taking into account the corresponding maximum data rate and packet error rate (PER), which is a function of the short-term average SNR. We highlighted the SE curve of two systems in red and green. The first system is nonadaptive and constrained to use the BPSK modulation only. Its corresponding SE vs. SNR is represented by the red curve. The second system uses adaptive modulation. Its corresponding SE is given by the envelope of the individual curves and is represented in green. It is seen that each modulation is optimal for use in different quality regions, and LA selects the modulation with the highest SE for each link. The performance of the two systems is equal for SNR up to 10 dB. However, in the range of higher SNRs, the SE of the adaptive system is up to six times that of the nonadaptive system. When averaging the SE over the SNR range for a typical power-limited cellular scenario, the adaptive system is seen to provide a close to threefold gain over the nonadaptive system.

The example in Fig. 1 is ideal since it assumes that the modulation level is perfectly adapted to the short-term average SNR, and that the proba-

Scheme	Modulation	Maximum Rate [kb/s]	Code Rate
MCS-9	8 PSK	59.2	1
MCS-8		54.5	0.92
MCS-7		44.8	0.76
MCS-6		29.6	0.49
MCS-5		22.4	0.37
MCS-4	GMSK	17.6	1
MCS-3		14.8	0.80
MCS-2		11.2	0.66
MCS-1		8.8	0.53

Table 1. *EGPRS modulation and coding schemes and peak data rates.*

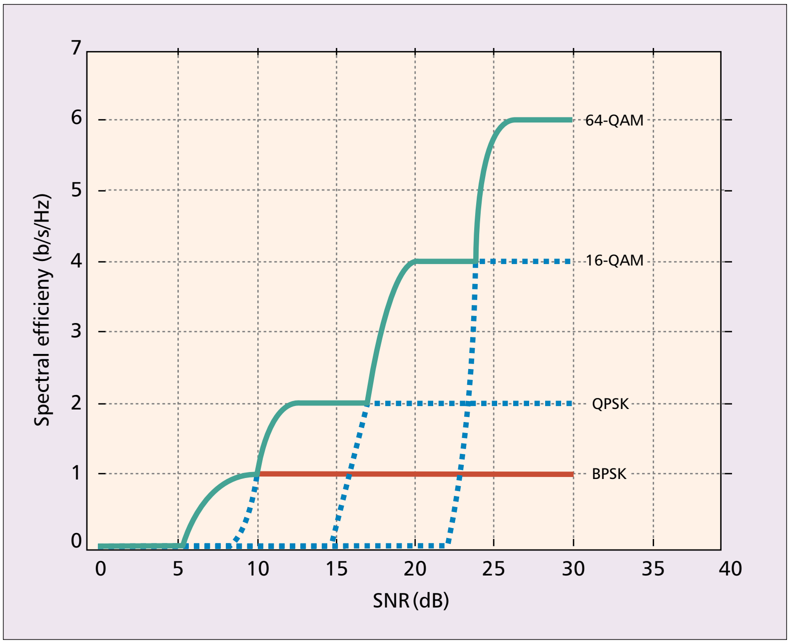
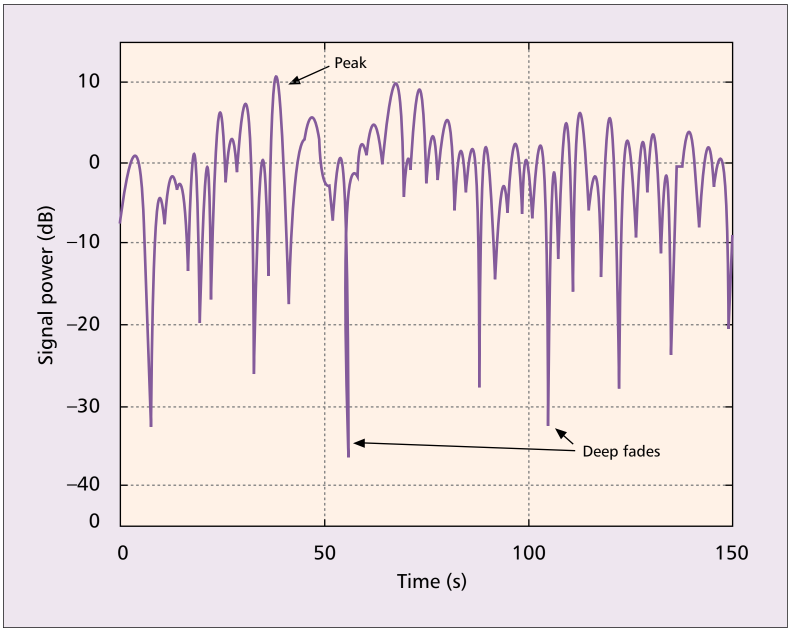


Figure 1. *Spectral efficiency for various modulation levels as a function of short-term average SNR.*



■ **Figure 2.** *Signal fading over time.*

bility of error as a function of the SNR is exactly known; for example, here we considered an additive white Gaussian noise (AWGN) channel, which corresponds to an instantaneous channel measurement. That assumption is only true for instantaneous feedback and is not practical due to delays in the feedback path. When there is delay, as explained later, the first and higher-order statistics of the fading channel should be incorporated to improve the adaptation. Furthermore, other dimensions such as frequency and space (where different transmission schemes may be adapted) may yield further gains simply by providing additional degrees of freedom exploitable by LA.

Expanding the Dimensions of Link Adaptation

Smart Antennas

"Smart antenna" technology is widely recognized as a promising technique to increase the spectrum efficiency of wireless networks [3]. Systems that exploit "smart antennas" usually have an array of multiple antennas only at one end of the communication link (e.g., at the transmit side, such as MISO systems; or at the receive side, such as single-input multiple-output, (SIMO, systems). A more recent idea, however, is multi-array or MIMO communication [4, 5] where an antenna array is used at both the transmitter and receiver. The potential of MIMO systems goes far beyond that of conventional smart antennas and can lead to dramatic increases in the capacity of certain wireless links [4]. In the so-called BLAST scenario [5], each antenna transmits an independently modulated signal simultaneously and on the same carrier frequency. Alternatively, the level of redundancy between the transmitting antennas can be increased to improve robustness by using so-called space-time codes [6].

Multi-Antenna-Element Systems and LA — In MIMO and MISO systems, the presence of multiple transmit antenna elements calls for an efficient way of mapping the bits of the message(s) to be sent to the various signals of individual antenna elements. The mapping can and must be done in different ways as a function of both the channel characteristics and the benefit desired from the smart antennas. For instance, in the MIMO case, the mapping in a multiplexing/BLAST scheme [5] tends to minimize the redundancy between the various antenna signals in order to favor maximum data rate. In contrast, a typical space-time coding approach [6] will introduce a lot of redundancy in an effort to maximize the diversity gain and achieve a minimum bit error rate (BER). The properties of the instantaneous or averaged space-time channel vector/matrix (e.g., the rank and condition number) play a critical role in the final selection of mapping strategy, just as the SNR does when picking a modulation or coding scheme for transmission. This is because independent signals transmitted over a rank-deficient MIMO channel cannot be recovered. In this respect it is clearly understood that the antenna mapping strategy must be treated as one component in the joint optimization of the signaling by the LA layer. Practical examples of this are considered in the last section of this article describing performance simulation results.

Multicarrier Systems

Broadband transmission over multipath channels introduces frequency selective fading. Mechanisms that spread information bits over the entire signal band take advantage of frequency selectivity to improve reliability and spectrum efficiency. An example of such a multipath-friendly mechanism is frequency-coded multicarrier (e.g., OFDM) modulation [7]. Transmission over multiple carriers calls for a scheme to map the information bits efficiently over the various carriers. Ideally, the mapping associates an independent coding and modulation scheme (or mode) with each new carrier. The idea is to exclude (e.g., avoid transmitting over) faded subcarriers, while using high-level modulation on subcarriers offering good channel conditions. While this technique leads to high theoretical capacity gain, it is highly impractical since it requires significant knowledge about the channel at the transmitter, thereby relying on large signaling overhead and heavy computation load. Alternate solutions based on adapting the modes on a per-subband basis (as opposed to per-subcarrier basis) are less demanding in overhead and select a unique mode for the entire subband, while still profiting from the frequency selectivity (see next section).

Adaptive Space-Time-Frequency Signaling

Before presenting the possible approaches to designing LA in broadband multi-antenna systems, we give a brief overview of wireless propagation channel modeling aspects in space, time, and frequency dimensions relevant in LA design.

The Space-Time-Frequency Wireless Channel

An ideal link adaptation algorithm adjusts various signal transmission parameters according to current channel conditions in all of its relevant dimensions. It is well known that unlike wired channels, radio channels are extremely random and the corresponding statistical models are very specific to the environment. Propagation models are usually categorized according to the scale of the variation behavior they describe:

- *Large-scale variations* include, for instance, the path loss (defined as the mean loss of signal strength for an arbitrary distance between transmitter and receiver) and its variance around mean, captured as a log-normal variable referred to as *shadow fading* and caused by large obstructions.
- *Small-scale variations* characterize the rapid fluctuations of the received signal strength over very short travel distances or short time durations due to multipath propagation. For broadband signals, these rapid variations result in fading channels that are frequency selective.

A typical realization of a fading signal over time is represented in Fig. 2. The superposition of component waves (or multipath) leads to either constructive (peaks) or destructive interference (deep fades). Time-selective fading is characterized by the so-called channel coherence time, defined as the time separation during which the channel impulse responses remain strongly correlated. It is inversely proportional to the Doppler spread and is a measure of how fast the channel changes: the larger the coherence time, the slower the change in the channel. Clearly it is important for the update rate of LA to be less than the coherence time if one wishes to track small-scale variations. However, adaptation gains can still be realized at much lower rates thanks to the large-scale variations.

In a multipath propagation environment, several time-shifted and scaled versions of the transmitted signal arrive at the receiver. When all delayed components arrive within a small fraction of the symbol duration, the fading channel is frequency nonselective, or flat. In wideband transmission, the multipath delay is often nonnegligible relative to the symbol interval, and frequency-selective fading results. Fig. 3 shows the time-varying frequency response of a channel model taken from [8] over 2 MHz bandwidth. In this type of channel, the variation of the signal quality may be exploited in both the frequency and time domains.

The third dimension LA may exploit is fading selectivity over space, which will be observed in a system that employs multiple antennas. Space selectivity occurs when the received signal amplitude depends on the spatial location of the antenna, and is a function of the spread of angles of departure of the multipaths from the transmitter, and the spread of angles of arrival of the multipaths at the receiver.

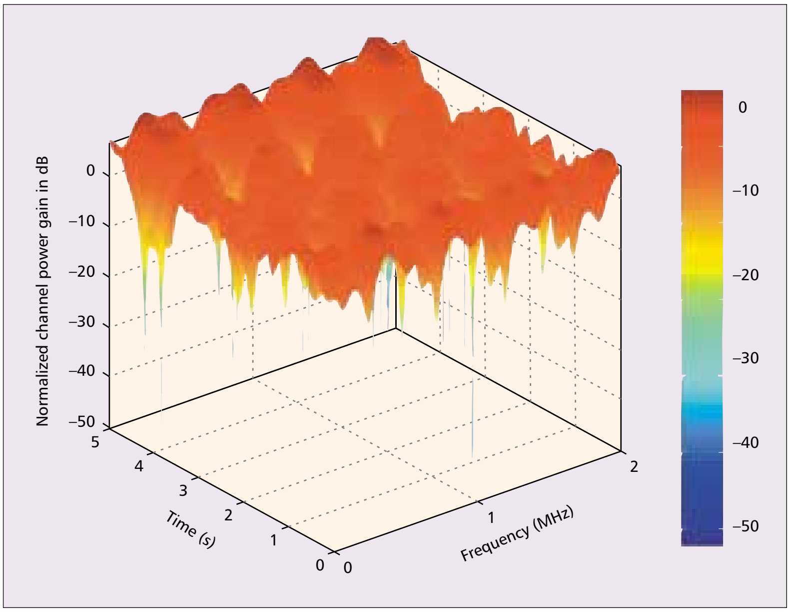


Figure 3. *Frequency response vs. time for a multipath channel.*

Adaptation Based on Channel State Information

The general principle of LA is to:

- Define a channel quality indicator, or so-called channel state information (CSI), that provides some knowledge on the channel
- Adjust a number of signal transmission parameters to the variations of that indicator over the signaling dimensions explored (time, frequency, space, or combination thereof)

There are various metrics that may be used as CSI. Typically, SNR or signal-to-noise-plus-interference ratio (SINR) may be available from the physical layer (e.g., by exploiting power measurements in slots without intended transmit data). At the link layer, packet error rates (PERs) are normally extracted from the cyclic redundancy check (CRC) information. BERs are sometimes available. In this section we review the respective use of this type of CSI in the design of the LA protocol, with emphasis on time adaptation and for an error-rate-constrained system. We first consider the traditional example of LA using SNR measurement with the perfect instantaneous feedback introduced earlier. We show the limitations of this scheme, and move on to more sophisticated types of adaptation.

Adaptation Based on Mean SNR — To implement adaptive transmission, the CSI must be available at either the transmitter or receiver. Often, such information consists of the SNR measured at the receiver [2]. In this case, a possible solution for LA is as follows:

1. Measure the SNR at the receiver.
2. Convert the SNR information into BER information for each mode candidate.
3. Based on a target BER, select for each SNR measurement the mode that yields the largest throughput while remaining within the BER target bounds.
4. Feed back the selected mode to the transmitter.

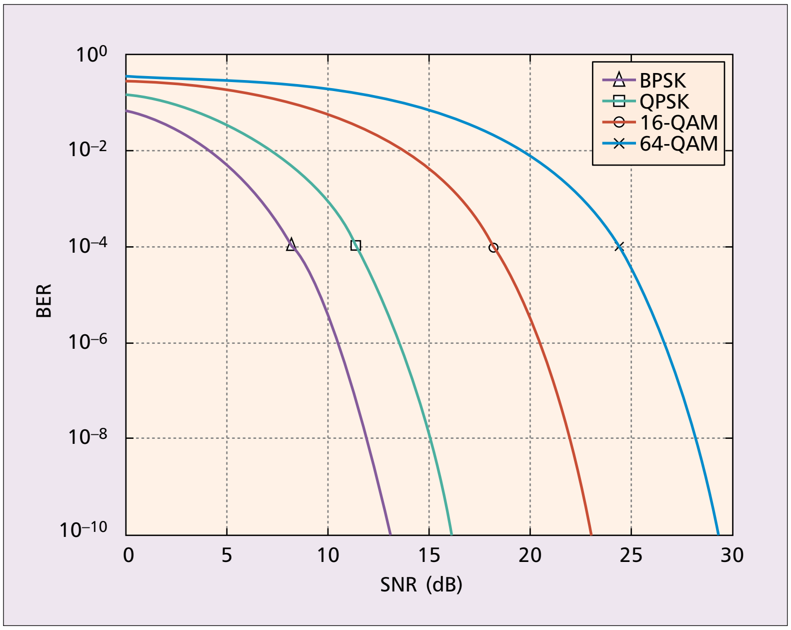


Figure 4. *BER for various modulations level as a function of short-term average SNR.*

Step 1 corresponds to the assessment of the CSI. Step 2 refers to the computation of the adaptation (or switching) thresholds. In this case, a threshold is defined as the minimum required SNR for a given mode to operate at a given target BER. Step 3 refers to the selection of the optimal mode, based on a set of thresholds and SNR measurement. Step 4 is concerned with the feedback of information to the transmitter. Under ideal assumptions, the implementation of these steps is straightforward. For example, let us assume a channel that is fading over time only (we leave aside the two other dimensions for simplicity). The conversion from mean SNR to BER can be made only if the mean SNR is measured in a very short time window, so each window effectively sees a constant nonfading channel. Let us therefore assume further that the SNR can be measured instantaneously, and the LA algorithm aims at adapting a family of uncoded M-QAM modulations to each instantaneous realization of the SNR. Established closed-form expressions for the AWGN channel may be used to express the BER as a function of the SNR γ assuming ideal coherent detection. Figure 4 represents a set of these BER curves for modulations BPSK, QPSK, 16-QAM and 64-QAM. The set of adaptation/switching thresholds is then obtained by reading the SNR points corresponding to a target BER. For example, if the target BER is 10^{-4}, the thresholds are 8.4, 11.4, 18.2, and 24.3 dB, respectively (as indicated by the markers on the figure).

Of course the scenario presented above relies on ideal assumptions. In practice, the feedback delay and other implementation limitations will not allow for mode adaptation on an instantaneous basis, and the effective update rate may be much slower than the coherence time. In that case the conversion of the SNR into BER information is not as simple as the formulation available for the AWGN model, because the real channel may exhibit some fading within the SNR averaging window. This calls for the use of second and higher order statistics of the SNR instead of just the mean.

Adaptation Based on Multiple Statistics of the Received SNR — We assume here that the CSI is measured over an arbitrary time window (flat fading case) set by the system-level constraints of the LA protocol. If multicarrier modulation is used, a two-dimensional time-frequency window may be used. The mapping between the SNR and the average BER is determined using the probability density function (pdf) of the SNR over that window. Unfortunately, in real channels this pdf cannot be obtained via simple analysis because it is a function of many parameters. It depends on:

- The channel fading statistics over frequency and time (which often have different distributions).
- The relationship between the length of the window in time and the channel coherence time.
- The relationship between the length of the window in frequency and the channel coherence bandwidth.
- In the case of multi-antenna-based systems, the SNR is determined post-antenna combining; therefore, the pdf also depends on such system parameters as the number of antennas used on the transmit and receive sides, antenna separation, antenna polarization, and transmission and reception schemes.

Instead of trying to estimate the full pdf of the SNR over the adaptation window, one can simplify the problem by estimating limited statistical information from the pdf, such as the k-order moment over the adaptation window, in addition to the pure mean (first order moment). These statistics provide only an approximation of the pdf of the received SNR. They are useful, however, when k can be kept low and yet yield sufficient information for a reasonably accurate mapping of the SNR into BER information. The first moment of the SNR captures how much power is measured at the receiver on average. The second moment of the SNR over the time (cf. frequency) dimension captures some information on the time (cf. frequency) selectivity of the channel within the adaptation window. Higher-order moments give further information on the pdf. However, they are also more computationally demanding, so there is a trade-off between accuracy and computation efficiency.

With moment-based CSI, the adaptation thresholds are a function of multiple statistics of the received SNR. This introduces simplicity and flexibility to the LA algorithm, since the adaptation thresholds no longer rely on any particular channel conditions. They remain valid for any Doppler spread, delay spread, and Ricean K-factor, and in the case of multi-antenna systems, they do not depend on any assumption made on the number of transmit and receive antenna, antenna polarization, and so on, since the effect of all these factors is captured by the low order moments of the SNR ($k > 1$) and, to a large extent, by the first and second order moments alone.

Incorporating Packet or Bit Error Rate Information — In some cases, the CSI relies on tracking errors in the reception of packets of data for each mode candidate [9]. The data reception history is updated and maintained until all modes have been trained, and there is a correspondence between mode and PER. This mode of adaptation provides explicit information on the observed link quality for each possible mode candidate instead of relying on theoretical BER curves. However, the limitation comes with the number of packets observed in any window. The method relies on the estimation of PER statistics, which can require up to several thousands of packets to be transmitted for a given mode in order to obtain a reliable estimate, thereby possibly making the adaptation loop slow. Unless one decides to use a training packet at regular frequent times, most probably only large-scale channel variations may be exploited by such adaptive transmission methods. Furthermore, these methods are traffic-dependent, making the algorithm reaction time harder to control (than, e.g., SNR-based methods) and offering no possibility of monitoring the channel quality when the user does not send or receive traffic. In particular, in the absence of traffic, one loses track of channel quality and may have to conservatively reinitialize the LA.

In general, how effective these methods can be in realistic traffic and bandwidth constraints is an open research problem. In particular, it is critical to measure the ability of the scheme to lend itself to a very fast adaptation scenario without significant bandwidth loss.

Note that packet error information can be exploited differently. The CSI can be monitored in an iterative manner, such as in the concept of incremental redundancy (IR), proposed for standards IS-136 and EDGE in [10]. The idea behind IR is to incrementally transmit additional redundant information as long as decoding of a packet fails. The additional redundant information is combined with the previously received information, resulting in enhanced coding.

CSI Based on a Combination of SNR and Error Statistics Information — The types of CSI described above all come with their pros and cons. SNR-based CSI offers the flexibility to adapt the modes on a very fast basis; however, it relies on the computation of adaptation/switching thresholds that may be inaccurate. The accuracy of the threshold mechanism increases by taking into account higher order statistics of SNR than just the mean. Error-based CSI captures accurate performance of the modes; however, this accuracy is reached only after a substantial amount of traffic is observed, especially in the low error rate region, making adaptation slow. An important topic of current research is to combine all types of CSI together to yield both accuracy and robustness over a wide range of channels, adaptation rates, and traffic conditions.

Space-Time-Frequency Adaptation

In a system with multiple antennas at the receiver and/or transmitter, the SNR not only varies over time and frequency but also depends on a number of parameters including the way the transmitting signals are mapped and weighed onto the transmit antennas, the processing technique used at the receiver, and the antenna polarization and propagation related parameters such as the pairwise antenna correlation. In a space-time-frequency adaptation scheme it is desirable that the adaptation algorithm be able to select the best way of combining antennas at all times (e.g., choosing between a space-time coding approach, a beamforming approach, and a BLAST approach in a continuous way). Furthermore, it should do so in a systematic way that is transparent to the antenna setup itself. For example, ideally, the same version of the LA software is loaded in the modem regardless of the number of antennas of this particular device or their polarization. One possible solution to capture the effects of all these parameters in a transparent manner is to express the channel quality (and therefore the adaptation thresholds) of multi-antenna-based systems in terms of *post-processing* SNR as opposed to simple preprocessing SNR levels measured at the antennas. In this case, the variation of the post-processing SNR is monitored over time, frequency, and space, thus enabling the LA algorithm to exploit all three dimensions. For CSI purely based on error statistics, the channel quality of multi-antenna-based systems is directly expressed post-processing, since errors are detected at the end of the communication chain.

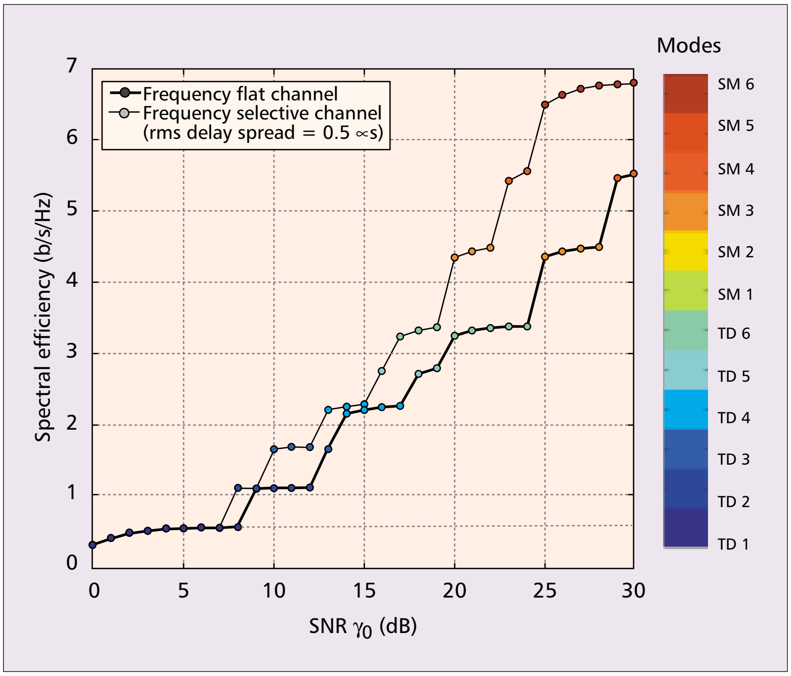
Practical Limitation and Implementation Issues

We identify certain LA algorithm design challenges as follows:

Determination of Adaptation Thresholds. The thresholds can be obtained by either simulation or measurements. They are expressed in terms of time-frequency average receive SNR (SISO systems) or post-processing SNR (for multi-antenna systems) and are indexed by the higher-order statistical averages computed. It is often a challenge to compute them accurately because the required sample size grows as the order of the statistical average increases. Therefore, the trade-off consists of picking the least amount of statistical channel information to be computed while still describing the essence of the channel behavior.

Adaptation Rate. The adaptation rate gives information on what kind of channel variations the LA algorithm is tracking. SNR-based algorithms can be used for either very fast or very slow adaptation. In slow mode, the algorithm tracks large-scale variations influenced by user location within the cell, road traffic, seasons, wind, and cell deployment. In fast mode, it tracks small-scale variations influenced by frequency- and time-selective fading and local changes in SNR. It is easy to understand that faster adaptation leads to larger capacity gain, since the channel variations are exploited in a more accurate manner. However, fast adaptation has practical limitations, in both time and frequency. Fast adaptation increases the number of mode-change messages that must be sent over the air, consuming bandwidth and time resources. There is a trade-off between performance gain and amount

In general, how effective these methods can be in realistic traffic and bandwidth constraints is an open research problem. In particular, it is critical to measure the ability of the scheme to lend itself to a very fast adaptation scenario without significant bandwidth loss.

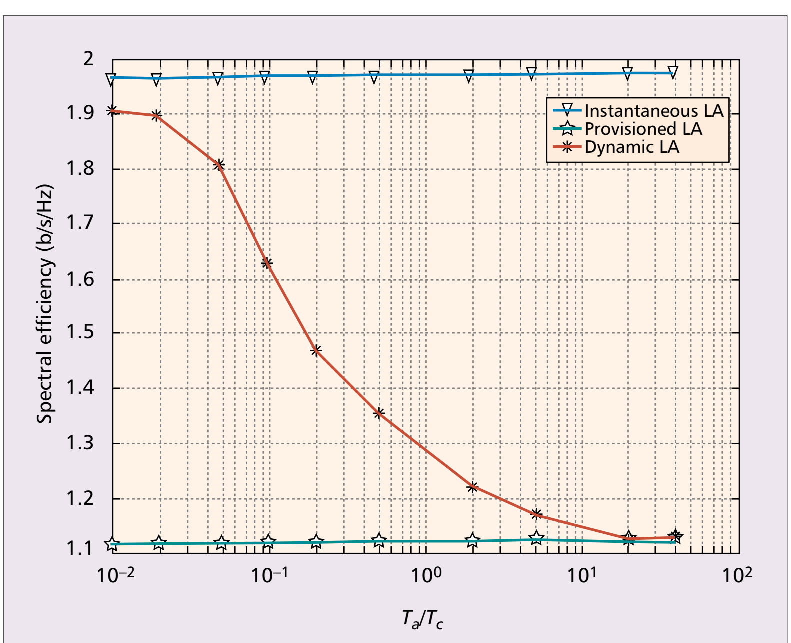


■ **Figure 5.** *Spectral efficiency vs. long-term average SNR with and without adaptive signaling with and without frequency selectivity.*

of resources allocated to control messages, as seen in the simulation section.

In the frequency domain, an adaptive scheme for multicarrier systems was proposed in [7] where the mode is adapted on a per-subcarrier basis. This results in large signaling overhead and heavy computation. An alternative is to aggregate subcarriers into subbands and instead adapt the modes on a per-subband basis. This is motivated by the observation that depending on the subcarrier bandwidth, the channel is roughly constant for adjacent groups of subcarriers. The required overhead diminishes as more subcarriers are aggregated together; however, the benefits of aggregation decrease for highly frequency selective channels.

Feedback. The feedback messages inform the transmitter of the CSI or mode predicted by the receiver. The feedback load should be minimized since it consumes resources that would be otherwise used for data. If the CSI is fed back, the transmitter uses that information to make the mode decision. Otherwise, this decision is made at the receiver, and only the mode is fed back to the transmitter. This latter option is usually more efficient since there are typically a small number of modes, thereby limiting the amount of feedback. Furthermore, it is easily understood that the mode is not fed back at each prediction, but rather only if the new predicted mode differs from the mode the transmitter is currently using.



■ **Figure 6.** *Spectral efficiency vs. normalized adaptation window, for various adaptation rates at fixed long-term average SNR.*

Performance Evaluation

In this section we illustrate the performance attainable by an LA algorithm (where the adaptation is based on a combination of SNR and PER statistics) in a simulated scenario. We use as an example a broadband wireless MIMO-OFDM-based system that provides data access to stationary Internet users. Note that despite the users' stationarity, the environment is still time-varying and some gain may be obtained by tracking these variations. The adjustable transmission parameters are the modulation level, coding rate, and transmission signaling scheme. One possible transmission signaling scheme is to demultiplex the user signal among the several transmit antennas (what we refer to as *spatial multiplexing*, SM); the other is to send the same copy of the signal out of each transmit antenna with a proprietary coding technique based on the concept of delay diversity. We refer to this latter scheme as *transmit diversity* (TD). A particular combination of modulation level, coding rate, and transmission signaling scheme is referred to as a *mode*. The system may use six different combinations of modulation level and coding rate, and two different transmission signaling schemes, resulting in a total of 12 candidate modes, indexed as modes SM i and modes TD i, where $i = 1, \ldots, 6$.

Figure 5 presents the system spectral efficiency (SE) performance in b/s/Hz vs. the long-term average SNR γ_0 for a frequency-flat and a frequency-selective (rms delay spread is 0.5 µs) environment. The results highlight the great capacity gain achievable by LA over a system using a single modulation. The dashed line in the lower part of the figure shows the SE vs. SNR of a system using mode TD 1 only. It is seen that the SE remains at 0.5 b/s/Hz, regardless of the SNR. In contrast, the SE of the adaptive system increases with SNR as different modes are used in different SNR regions. It is also shown that exploiting the extra dimension (over frequency) provides additional gain in a frequency selective channel, mostly for higher SNR where the higher mode levels (with larger SE) may be used.

Figure 6 shows the system spectral efficiency (SE) performance in b/s/Hz vs. the normalized adaptation window defined as T_a/T_c, where T_a and T_c denote the adaptation window and channel coherence time, respectively. The channel is considered frequency flat. Thus, the results are independent of the channel Doppler spread. Three curves are plotted for a fixed long-term average SNR of 10 dB. The upper blue curve represents the SE of instantaneous LA, that is, the mode is adapted for each instantaneous realization of the SNR. This scenario may not achievable in practice due to practical limitations, but we use it here as an upper bound on the performance obtainable with a practical adaptation rate. For an average SNR at 10 dB, the upper bound is almost 2 b/s/Hz. The lower green curve represents the SE of provisioned LA, that is, the mode is adapted based on long-term SNR statistics. The SE value is taken from Fig. 5 at SNR of 10 dB, where it is read to be equal to 1.1 b/s/Hz. Since provisioned LA is the slowest way of adapting to the time-varying channel components, the corresponding SE is used as a lower bound on the performance obtainable with a practical adaptation rate. Finally, the red curve in between shows the variation of the SE as a function of the normalized adaptation window. When the ratio T_a/T_c is small, the adaptation is fast and the performance approaches the theoretical upper bound. When the ratio T_a/T_c is large, the adaptation is slow and the performance converges toward the lower bound. In general, it is seen that a twofold capacity gain may be achieved between the slowest and fastest adaptation rates.

Conclusion

This article reviews the fundamentals of adaptive modulation and coding techniques for MIMO broadband systems and illustrates their potential to provide significant capacity gains under ideal assumptions. Other emerging techniques for increasing spectral efficiency in wireless broadband access systems are presented; smart antenna techniques and coded multicarrier modulations, and their interactions with the LA layer design were emphasized. Following a short overview of space-time broadband wireless propagation characteristics, we explore various ways to capture channel information and provide some guidelines on the design of sensible solutions for LA. Implementing optimum LA is challenging due to practical limitations, but simulated performance of a realistic broadband wireless MIMO-OFDM-based system using LA is very encouraging.

References

[1] D. Gesbert *et al.*, "Technologies and Performance for Non Line-of-Sight Broadband Wireless Access Networks," *IEEE Commun. Mag.*, vol. 40, no. 4, Apr. 2002., pp. 86–95.

[2] S. Nanda, K. Balachandran, and S. Kumar, "Adaptation Techniques in Wireless Packet Data Services," *IEEE Commun. Mag.*, Jan. 2000, pp. 54–64.

[3] *IEEE Pers. Commun.*, Special Issue, "Smart Antennas," vol. 5, no. 1, Feb. 1998.

[4] G. J. Foschini and M. J. Gans, "On Limits of Wireless Communications in a Fading Environment When Using Multiple Antennas," *WL. Pers. Commun.*, vol. 6, no. 3, Mar. 1998, pp. 311–35.

[5] G. J. Foschini *et al.*, "Simplified Processing for Wireless Communication at High Spectral Efficiency," *IEEE JSAC — WL. Commun. Series*, vol. 17, no. 11, Nov. 1999, pp. 1841–52.

[6] A. F. Naguib, N. Seshadri and A. R. Calderbank, "Increasing Data Rate over Wireless Channels," *IEEE Sig. Proc.*, May 2000, pp. 76–92.

[7] T. Keller and L. Hanzo, "Adaptive Multicarrier Modulation: A Convenient Framework for Time-Frequency Processing in Wireless Communications," *Proc. IEEE*, vol. 88, no. 5, May 2000, pp. 611–40.

[8] V. Erceg *et al.*, "Channel Models for Fixed Wireless Applications," IEEE 802.16 Broadband Wireless Access Working Group. IEEE802.16.3c-01/29r4 (http://www.ieee802.org/16/tg3/contrib/802163c-01_29r4.pdf).

[9] S. S. Gilbert *et al.*, "Communication of Data Reception History Information," U.S. Patent # 5,559,810.

[10] R. Van Nobelen *et al.*, "An Adaptive Radio Link Protocol with Enhanced Data Rates for GSM Evolution," *IEEE Pers. Commun.*, Feb. 1999, pp. 54–63.

Implementing optimum LA is challenging due to practical limitations but simulated performance of a realistic broadband wireless MIMO-OFDM- based system, using LA, is very encouraging.

Biographies

SEVERINE CATREUX (scatreux@iospanwireless.com) received her M.Sc. degree in electrical engineering from National Institute of Applied Sciences (INSA), Rennes, France, in 1996, and her Ph.D. degree from INSA in 2000. Between 1996 and 1999 she was at the University of Victoria, British Columbia, Canada, where she studied array signal processing techniques for digital radio communications systems. From April 1999 to early 2000 she completed her doctoral studies in the Wireless Communications Research Department of AT&T Labs Research, with emphasis on the data throughputs attainable by MIMO systems. In April 2000 she joined Iospan Wireless Inc., San Jose, California, a start-up company developing a high-speed broadband fixed wireless system based on MIMO-OFDM technology. Her research interests are in the areas of adaptive array signal processing for digital communications, wireless MIMO systems, and adaptive transmission techniques.

DAVID GESBERT (gesbert@ifi.uio.no) holds a Ph.D. degree from Ecole Nationale Superieure des Telecommunications (ENST), Paris, France, 1997. From 1993 to 1997 he was with France Telecom Research (CNET), Radio Systems Department, Paris. From April 1997 to October 1998 he was a postdoctoral fellow in the Information Systems Laboratory, Stanford University. In October 1998 he took part in the founding team of Iospan Wireless Inc., formerly Gigabit Wireless Inc., a startup company promoting high-speed wireless Internet access networks using adaptive MIMO, OFDM, and other state-of-the-art applied wireless research. In 2001 he became an independent consultant and joined in parallel the Signal Processing Group, Department of Informatics, at the University of Oslo, Norway, as an adjunct associate professor. He is the author of over 40 papers and 15 patents, granted or pending, in the area of wireless systems. His research interests are in the areas of high-speed wireless data/IP networks, smart antennas and MIMO, link layer, and system optimization.

VINKO ERCEG (verceg@iospanwireless.com) received his B.S.E.E. in 1988 and his Ph.D.E.E. in 1992, both from the City University of New York. From 1990 to 1992 he was a lecturer at the Electrical Engineering Department at City College of New York, concurrently working with SCS Mobilecom, Port Washington, New York, on spread-spectrum systems for mobile communications. In 1992 he joined AT&T Bell Laboratories and became a principal member of technical staff in the Wireless Communications Research Department of AT&T Labs-Research in 1996. He participated in various aspects of wireless research, including signal propagation modeling, systems engineering, and performance analysis. He joined Iospan Wireless Inc. in February 2000, where he now serves as director and principal engineer in channel modeling and systems validation. In 2001 he chaired the IEEE 802.16 working group on broadband wireless access standards in developing NLOS channel models. His research interests include radio propagation and channel characterization, capacity and coverage prediction of cellular systems, and MIMO wireless systems.

ROBERT W. HEATH JR. (rheath@ece.utexas.edu) received his B.S.E.E. (1996) and M.S.E.E. (1997) degrees from the University of Virginia, Charlottesville, and his Ph.D.E.E. (2002) degree from Stanford University, California. From 1998 to 1999 he was with Iospan Wireless Inc.. From 1999 to 2001 he served as a senior consultant for Iospan Wireless Inc. In January 2002 he joined The University of Texas at Austin where he serves as an assistant professor. His research group, the Wireless Systems Innovations Laboratory, focuses on the theory, design, and practical implementation of wireless systems. His current research interests are coding, modulation, equalization, and resource allocation for MIMO wireless communication systems.