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Chapter 8: Advanced modulation and coding for future-ready wireless access technologies

8.1. Introduction

Novel wireless access technologies are key enablers towards realizing future 6G networks. These technologies must support the future use cases and vertical industries envisioned for 6G, and be capable of delivering ultra-massive access, along with ultra-high reliability, ultra-low latency, and extremely high data rates. Towards this direction, future wireless networks are expected to rely on advanced physical-layer capabilities. Advanced modulation and coding schemes for future wireless systems are indeed key enablers towards realizing future 6G networks. The ongoing research activities in the field of advanced modulation and coding are indeed of paramount importance not only for achieving the challenging requirements of future wireless systems from a capacity point of view, but also for being capable of delivering reliable wireless services (Awathankar et al., 2024; Chamola et al., 2025; Fatima & Kondamuri, 2025).

Of course, the above stringent requirements refer to very high spectral efficiency. Such high rates, together with the demand to serve a huge number of users, make the task for the physical layer challenging. In addition, advanced techniques, such as non-terrestrial networks, intelligent reflecting surfaces, integrated satellite and terrestrial networks, leveraging hybrid beamforming, massive machine-type communications with large-scale antenna arrays at the base station, and mobile edge computing, are expected to be utilized in the next generation wireless systems. All these operations require a robust wireless infrastructure, which can also assist users in tracking down the narrow beams formed by the advanced technologies mentioned above. In addition, enhanced modulation and coding techniques are required to guarantee the confidentiality and secrecy of the information transmitted for transport applications (Filo et al., 2021; Kumar et al., 2025; Singh et al., 2025).

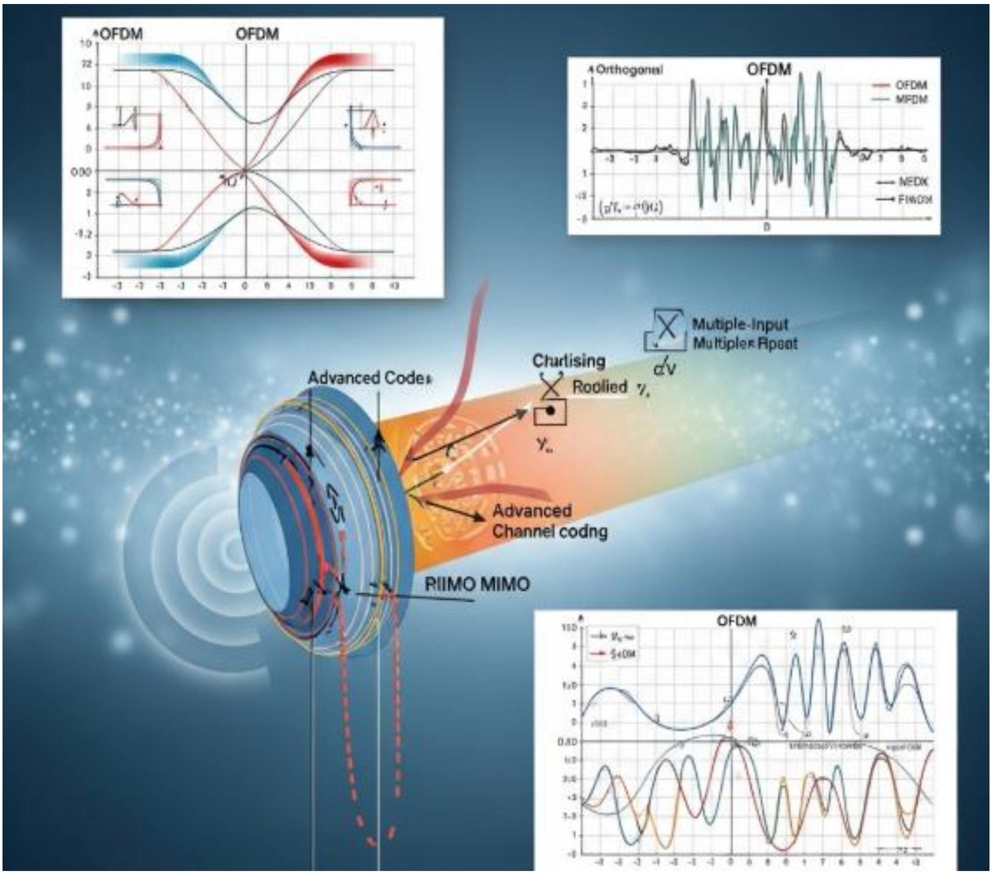

Fig 8.1: Advanced Modulation and Coding

The first generation mobile communication system pioneered the D-AMPS and TACS systems. In these systems, analog modulations were used to provide circuit-switched voice services utilizing the FDMA and TDMA methods. From the second generation systems, digital techniques have been applied mainly to the mobile communication systems. The second generation mobile system provided a digital speech service by using codecs and a TDMA technique. Here, the modulation schemes are GMSK and DQPSK, which are spectrally efficient compared with D-AMPS. In these systems, a three-dimensional Gaussian-shaped raised cosine criterion is used for the system specification, and consideration is given to the multi-user detection by the channel equalizer. From the mobile communication system, stable, high-quality speech service with low bit error rates on the order of 10–3 can be provided by digital adaptive equalization and channel coding with a coding gain lower than 5 dB.

8.2. Overview of Wireless Access Technologies

With the development of Digital Circuit Technology and Radio Device Technology, wireless transmission shows advantages such as low cost, wide service area, and convenient mobility. Whether terrestrial or space-wireless transmission, it is increasingly being used for high-speed information networks. Various wireless access technologies have been constructed and commercialized. In this section, the primary features and access technology issues for each stage of wireless access technology are described from a historical viewpoint. In accordance with this historical method, the described wireless techniques, service category, channel access method, and applied modulation are described.

8.3. Modulation Techniques

To transmit information over a wireless channel, the modulation of a signal as a function of a carrier allows it to accumulate energy into a frequency band, to translate the signal into radio frequencies, and to provide more robustness to the wireless communication. Several different modulation methods exist, in broad categories, to modulate either a signal in the time domain, or a signal in the floating domain which is a changing magnetic field in order to induce an electromagnetic wave. Knowing the performance of modulation techniques is critical for the design of wireless transceivers, because any limitation or specialization at any sub-block level must be correctly taken into account to have the desired performances of the wireless communication. For wireless communications, three main categories of modulation exist: analog modulation, digital modulation, and hybrid modulation, which are described in the subsequent sections in order of decreasing signal dimensionality. As a general overview, analog modulation allows modulating an infinite number of values on a signal (thus reproducing the input signal), and is for example used in electrical TV or FM radio transmissions. Digital modulation allows to convert a finite number of values into a signal, for example transmitting 1 as high voltage, and 0 as low voltage. One digital modulation method would be Pulse Amplitude Modulation, which allows sending finite intensities of a signal, as used in most optical fiber communications. Finally, hybrid modulation is used to map a discrete set of symbols into a signal for transmission. Hybrid modulation can for example be Pulse Amplitude Modulation with Phase Shift Keying or Quadrature Amplitude Modulation, which are widely used in mobile wireless applications with phase-coherent demodulation and without frequency-selective fading.

8.3.1. Analog Modulation

When wireless technology demand increased to support mass service delivery, growth of available bandwidth demanded more spectrum-efficient technologies in order to increase the available capacity. In doing so, reliable and efficient techniques from information theory became inevitable. The true evolution stage of the wireless data economy began with the introduction of digital modulation techniques. Before that, however, the main modulation techniques were analog in nature. While few doubt the great advantages of employing digital modulation techniques in cellular technology today, the fundamental modulation techniques did not go on without merit. In order to appreciate how far we have come in exploiting the potential of the communications channel, we recap briefly the major analog modulation techniques.

In amplitude modulation, the amplitude of the carrier is varied in proportion to that of the modulating signal. Traditionally, this modulation technique is implemented to communicate speech signals between terminals sharing the same subscriber line. The greatest disadvantage of analog modulation techniques is that their error performance relies heavily on the characteristics of the transmitted signal. Therefore, transmitted signals with large dynamic range will produce a poor error performance when modulation is performed using wideband resources. As a matter of fact, this technique operates in the non-coherent mode and consequently, a large amount of the channel's potential is wasted here. In addition, what severely limits the number of users simultaneously accommodated with this technique is the sensitivity to noise, namely that the equalizer does not recover well the original transmitted signal in the presence of a noise component. If the power of the noise component is large compared to the power of the original signal, no amount of processing is going to help to improve the formula used to decide which of the different transmitted signals is the one that will better recover the transmitted signal.

8.3.2. Digital Modulation

Digital modulation employs disjointed, countable symbols representing information bits. The symbols are typically achieved through pulse amplitude modulation with a fixed pulse shape, guiding the envelope of a bandpass-modulated first derivative of a rectangular pulse. Generally, modulation of the same carrier frequency and antenna is used for the adjacent symbols, but in GMSK modulation, this is supplemented by a frequency shift of half a symbol duration. As the symbol period of digital modulation is long compared with the equivalent energy to bandwidth product, the envelope has to compensate for the non-continuous phase of the modulated carrier, as it does for the raised cosine pulse shape of DMT, traditional QAM, and other digital modulations. Furthermore, common QAM modulations can be modeled as numerous coherent subcarriers which apply phase rotations. These practical QAMs thus have a wide variation of distance between adjacent signals, trashing ISI-free single carrier modulation from their remote-symbol spacing. Programmable gold codes may be used for matrix mapping to avoid large variation in distances, but this is rarely considered. MIMO techniques can also be used to compress the signal distances and enlarge the signal size without the multipath nulling effects of these techniques.

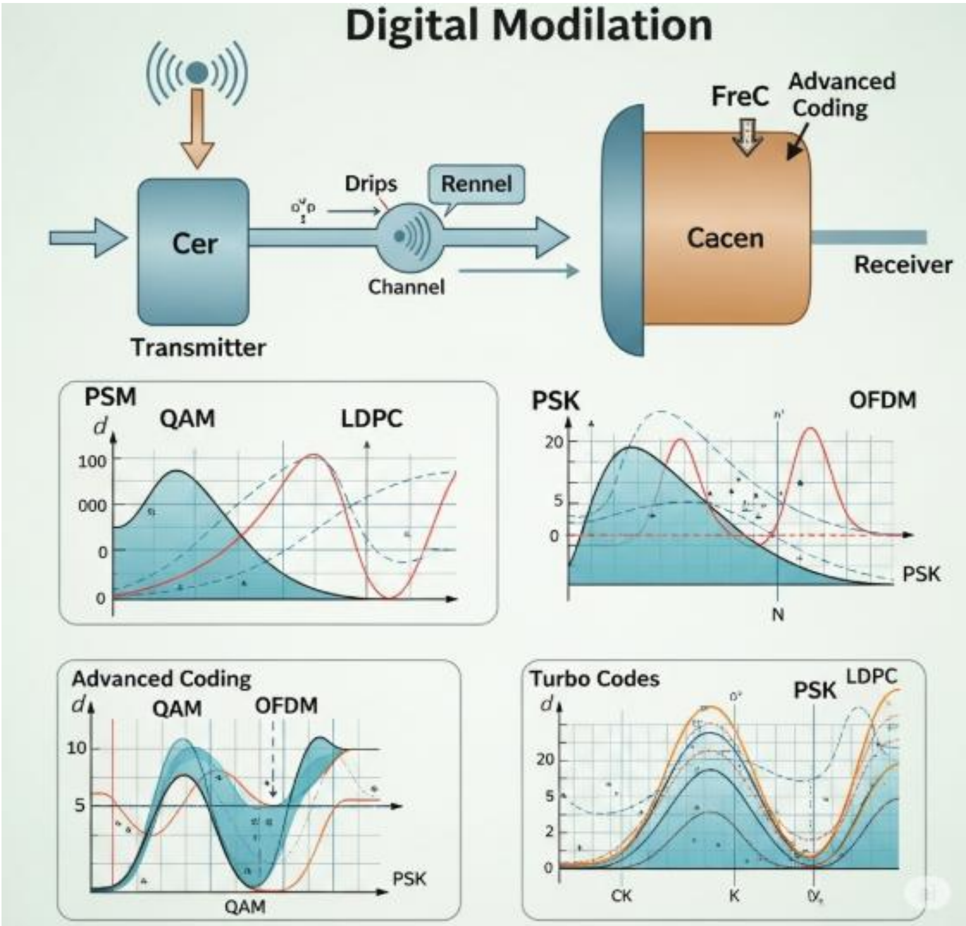

Fig 8.2: Digital Modulation of Advanced Modulation

Digital modulation is applied in DMT, Constant Modulus modulation, traditional QAM, MSK, CSK, and their variants. The opposite of modulation is AC coupling to pass high frequencies between devices. Despite digital modulation's straightforward definition, there are essentially no known systems that cannot be modeled as analog modulation with a precision-enforceable bandwidth, hand-created chunking of mappings, and an exact, amplifiable probability density function. All implementations of digital modulation have been effectively shown to introduce very high distortion effects so that the modulated signal is not representative of the original digital signal. Error correction codes must be enforced, usually with the desire to operate with as little frequency bandwidth as possible, to ensure functionality at reasonable distances.

These previous solutions are mainly dedicated to high-capacity long-haul fiber transmission systems or to Wireless Systems that work in the mmWave band. In first case, the advantages of Hybrid Modulation rely on lowering the fiber-imposed restrictions such as Nonlinear Shannon Limit and FEC Threshold Q factor, thus overcoming the performance limitations due to the high number of subcarriers or the high baudrate implemented in the DMT-modulated system. In the second case, Hybrid Modulation is used to transmit very high capacity data streams and is implemented to reduce the electronic-digital signal processing complexity related to Nyquist Pulse Shaping which arises when high baudrate PAM signals are used.

8.3.3. Hybrid Modulation

Hybrid Modulation combines the advantages of both Analog and Digital Modulation schemes. The first proposed Multi-Level Hybrid Modulation system was HPM/DMT which utilizes the HPM PNU QPSK to transmit the phase and amplitude information of the transmitted DMT signal. After that, several other Multi-Level Hybrid Modulation solutions have been presented including: Hybrid HPM/DMT, which employs HP QPSK modulation to transmit amplitude and phase information of the DMT signal; PM-APSK-DMT, which proposes the use of the six-space phase modulated irregularly shaped amplitude and phase modulation to carry digital serial data and is very suitable for high speed long-haul transmission; TCM/HPM-DMT, employing Trellis Coded Modulation Pm QPSK to transmit the phase and amplitude information of the DMT signal; and TCM/PM-APSK-DMT, developed to transmit the digital information encoded with TCM on the 6P-APSK modulation constellation.

8.4. Coding Strategies

Coding and modulation require the input to have a structure that is known to the transmitter and receiver, which makes detection and extraction feasible. Coding strategies and modulation work differently when compared. Coding means adding redundancy without any level-shift. Code modulation creates huge patterns of symbols, with different levels induced, thus introducing capital innovations without any history.

The goal of source coding, error correction coding and channel coding is to add redundancy to a stream of bits for reliable communication over a bandwidth limited channel. The redundancy incurs a loss to the transmission rate, or code rate. The channel introduces a source of uncertainty in the form of noise, interference, dispersive distortion, fading, etc.

Error correction codes protect transmitted data solutions from being rendered useless. They are added to the information data but do not need to be compressed. Adding parity bits or check-sums is the most commonly known method. This is done at a programming or software level after the compression has been applied and is independent of the compression format.

Channel coding uses the concept of breaking your compressed data stream into packets and then applying redundancy to each packet. The most efficient method is with Forward Error Correction. In FEC, error detecting codes called parity check codes are added to the packet stream before applying channel encoding. Coding is performed at the codec level since it is a function of packet size and coding methods. FEC is used in conjunction with fragmentation methods. Thus, coding strategies are highly dependent on the format and method of source encoding applied. This is the basis of a Coding Strategies Table.

The tradeoff space of the overall coding strategy is complex, involving many interacting factors: Method, Name, Packet Size, Type, Source Degree, Source Code Rate, Mode, MCS, Code Rate x, Code Rate, Bit Errors, Block Errors, Service.

8.4.1. Error Correction Codes

Error correction codes were invented by Claude Shannon in 1948. If we denote the data bits that we want to send and the codeword that we will send through the channels, the codebook is defined by the error correction code C if it relates $u \rightarrow w$. This transformation is not necessarily injective because we might just want to protect a limited number of data bits. For example, we might have just 1 bit of data, but we send 3 bits of codeword that would protect the information against one error. For this reason, C is normally a "large" transformation where $|C| >> 1$, being $|C|$ the size of the codebook. Shannon proved that it is possible to find a code such that for very large data lengths, we can approach the channel capacity, that is the Shannon limit, with an arbitrary small probability of uncorrected error.

Error correction codes are forcing the mapping from the set of information symbols into a set of codewords. If we denote U as the alphabet used for the source symbols, the number of states should limit the number of source symbols and should be "small." It is this information that error correction algorithms transform from the source into the corresponding error correction code. There are two possible approaches to accomplish error correction codes. The first one consists of selecting among all the possible source code values those ones that are less probable of being transmitted. These less probable source values map evenly (or as close as possible) to the error code with lower probability distribution.

8.4.2. Channel Coding

Engineers and researchers with military, space, or telecom experience are funneled toward the task of finding the best compromise among various defined qualities. This is speed, loss probability, spectral efficiency, coding delay, achievable distances, diversity, complexity, delay, latency, error detection, bandwidth expansion, decoding delay, line rate, overhead, state, finite size, encoder speed, erasure codes, distance, diversity, physical layer security, computation speed, probing code, decision feedback, receiving or decision, depth, error floor, symbol probability, identity, iterative decoding, conjugate, intelligent, joint, conditions, Markov model, hybrid codes, generative scores, and many more. Channel coding involves the modification of transmitted data such that it can be reliably decoded, despite being subject to degrading effects in the noisy channel. These effects could range from noise or interference that distorts the coding signal, to channel fading, to decrypted eavesdroppers.

8.4.3. Source Coding

The most important feature of source coding is the removal of redundancy to convey a signal as concisely as possible. This process occurs before the signal is converted into its own representation of bits. The most widely used source coder in commercial applications is the Transform Video Coding, which transforms pixel difference signals (after motion prediction) into a combination of cosine basis functions over an adjustable window length. A Uniform Scalar Quantizer uniformly quantized large portions of the cosine coefficients to the same value generating a highly redundant signal over longer intervals. This redundancy is removed using variable-length codes to convert the input variable bit-width sample into the shortest codeword bits based on a predetermined distribution of the quantized coefficients.

This coding defines a mapping between the quantized source signals and a reduced number of codewords or symbols, each having lengths given by denoted as L(s). This is similar to an approach to coding theory for source codes which states that redundancy can be removed if the average rate is less than the source entropy. The idea is that source redundant symbols are assigned small lengths whereas source distribution symbols that seldom ever occur are assigned increasingly larger lengths. It naturally follows that source code mapping into codewords must be bijections as no loss of information can be tolerated. Thus, this mapping can either take the form of codebook entries or apply some integer algorithmic mapping used widely in practical applications.

8.5. Advanced Modulation Schemes

In order to meet the increasing demand for higher data rates in wireless communication, well-controlled spectrum allocation is no longer an adequate solution. New advanced modulation schemes, such as Orthogonal Frequency Division Multiplexing, massive MIMO technologies, and higher-order modulation, are essential tools to perform the data transfer more efficiently through the wireless channel. They utilize the physical layer in a much more efficient manner compared to other methods. By dividing one regular bit stream into several parallel streams, all the parallel streams can jointly use a much wider bandwidth with very high data rates. Due to the nature of multiplexing, such advanced modulation schemes require accurate synchronization and channel estimation equipment at the receivers, while suffering from large PAPR, spectral leakage, and narrowband interference problems.

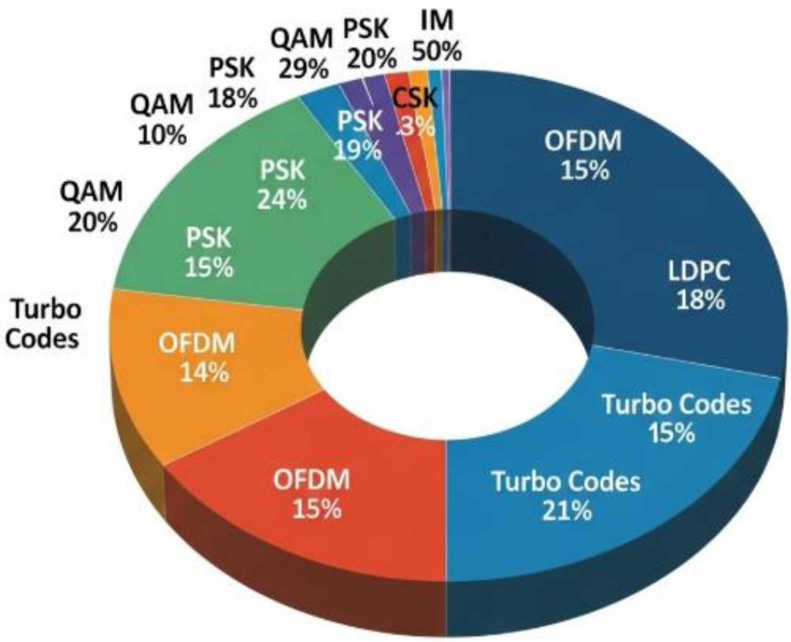

Fig 8.3: Advanced Modulation and Coding for Future-Ready Wireless Access

It is known that a linear modulation scheme such as phase-shift keying can be implemented by arithmetic rotation of the message signal. Due to the nature of geometry, the number of symbols that can be arbitrarily close to the Nyquist pulse is much bigger than the number of levels of the modulation signal. By hierarchically or combinatorially rotating the PSK points using both frequency and amplitude techniques, the non-linear M-scaled modulation signals can be implemented. Furthermore, by increasing the signal complexity, higher-order modulation can convey hundreds to thousands of bits per symbol, approaching the Shannon capacity for practical networks. Other advanced modulation schemes, such as index-coded modulation, generalized low-density parity-check modulation, constellation shaping, lattice-coded modulation, spatial modulation, and generalized spatial modulation, have also been developing rapidly. With careful designing such as signal shaping to avoid singularities and memory-assisted modulation, these advanced modulations achieve near capacity performance while also being easy and efficient at the receiver's operations, thereby creating exciting new possibilities for the future of wireless communications.

In particular, a better understanding of the DMT performance limits, and of DMT optimization techniques, is needed in the direction of rich-scattering multipath environments because they achieve a full-resolution in time and also, combined with space diversity from multiple-input multiple output antennas, or with space–time coding.

8.5.1. Orthogonal Frequency Division Multiplexing (OFDM)

The original DMT modulation was introduced to the digital subscriber line to cope with the severe multipath fading typical of this channel condition. The main idea behind this discrete multi-tone is to encode symbols on different orthogonal carriers and transmit them in parallel. In communication theory, it is known that the spectral efficiency of a band-limited channel can be maximized if the signal is designed to match the channel. In this case, the task of transmitting symbols encoded at high rate over a baseband band-limited is converted into that of transmitting a broad-band white-noise process into a band-pass channel. In general, multiple carrier modulations consider uniform and non-uniform partitions of the frequency axis. In the case of DMT, the total bandwidth is partitioned into N equal-width subbands. The N subbands are then modulated with N independent streams of symbols chosen from a discrete alphabet, which are transmitted through N parallel modulated discrete time systems.

The DMT takes advantage of the frequency selectiveness of the wireless channel by using the channel frequency response to remove the intersymbol interference at the receiver. In particular, the DMT is not only one of the most important advanced modulation techniques adopted in the current standards, but more importantly, it is expected to play a fundamental role also in future radio access protocols. Thus, it is of paramount importance to make pioneering research efforts to better understand the fundamental limits of the DMT, and identify areas in which there is still room for improvement. In addition, a careful study of the performance limitations imposed by the system's discrete nature is also needed, considering techniques like turbo codes, low density parity check codes or lift and map joint source-channel coding on the DMT.

8.5.2. Massive MIMO

Massive multiple-input and multiple-output (MIMO), a wireless technology, uses large numbers of antennas to simultaneously serve many users with high reliability and energy efficiency in a relatively small region. Massive MIMO eliminates multi-user interference and estimates the channel state information by spatially multiplexing the users, with the base station employing hundreds of antennas, possibly tens, to serve many tens of mobile terminals in a small area. It provides order-of-magnitude improvements in energy efficiency through the use of simple modulation or non-modulation techniques. Studies show improvements in spectral efficiency of 10-100 times, reduction in the cost for wideband and distributed wireless access networks by 70-95%, and reduction in the power expenditure for wireless terminals by many times. Massively large MIMO enjoy greater error exponents, asymptotically guaranteed multiuser diversity gain, and tend to provide a service outage-less quality for some time in a given ad-hoc area, such as at a super bowl stadium while catering to the mobile terminals that need data download services.

Massive MIMO, operating on large bandwidths, can serve quantized and analog data by employing selection-based or time division multiplexing (TDM)-based user selection techniques while ensuring that the resource reuse interval is much larger than the inverse of the channel estimation bandwidth. Antenna selection offers a low-complexity solution for implementing large MIMO systems and time division multiple access (TDMA) reduces the transmission power coefficient, enhancing the error performance. -bit exponent allows the access points to be placed away from cell boundaries.

8.5.3. Higher-Order Modulation

The concept of higher-order modulation refers to the notion of pairing more than one bit to the same constellation point in a digital modulation mapping. This is referred to as dense modulation. All coded modulation schemes assume that bits are paired together according to some rules. However, most schemes do not pair to the maximum. This leads to transmitters wherein only a small fraction of the modulation alphabet is actually used, while the other points remain empty. However, received symbols that are empty are weakly detected against noise, at least compared to used and thus filled points. The weakness of the detection of empty points does not mean that those points do not cause interference, though. On the decoding side, by construction the likelihoods of visited symbols at a receiver tend toward zero as the distance from the point tends to infinity. This means that the large set covers more of the area associated with a symbol than the small point does.

Higher-order modulation can be viewed as transmitting symbols that carry more than one bit, in the sense that symbols are seen as denoting a string of bits rather than a bit value. These types of modulation generally allow to increase system throughput in a region where conventional, discrete modulation is limited to relatively low values. Channel versions of classical codes permit to pool fixed-alphabet probabilities by sharing modulation points, though less efficiently down to the domain-specific modulated family just beyond constellation shaping, and do also allow to use those less efficient modulated designs even when some part of the draw have probabilities near zero. Such codes employ larger constellations than discrete or shared designs and cause values that drop with increasing modulation order for the same segmental throughput, but allow transmission rates that only vary according to the small modulation order law.

8.6. Coding Techniques in Modern Systems

The introduction of error-correcting redundancy into wireless systems has undergone significant development since the pioneering works in 1948. For the first generation of digital wireless access systems, convolutional error-correcting codes were the allied actor. Such codes, generally combined with Viterbi decoding, were adopted in the second generation of wireless systems. They have been used in the third generation, associated with Reed-Solomon block codes, and in the fourth generation LTE wireless access technological solution since they offer trade-off good performance-low complexity. The remarkable advances in VLSI technologies have allowed the use of increasingly complex decoders. This has triggered the great research activity, design, and implementation of powerful channels with subtractive joint decoding schemes.

While the ideal codes are unbounded and achieve the Shannon limit, perfect codes of finite length cannot exist. They, with good distance approaching but not achieving the Shannon limit, are defined as “good” codes. The parity-check value must be very low for a code to be considered “good”. For large message lengths, the rates of the “good” codes are asymptotically high. The use of good codes is, however, necessary. In the VLSI technology era, codes such as turbo codes and low-density parity-check codes appeared that jointly exploited the potential of parallelizable soft-in-soft-out decoders. In the last decade, polar codes recently received renewed interest since they turned out to be capacity achieving under particular conditions. We will provide below a brief overview of the origins and aspects of tame interest of LDPC codes, turbo codes, and polar codes.

8.6.1. Low-Density Parity-Check Codes

In nearly all practical high-speed wireless channels, we are faced with the presence of a fairly high level of noise. The result is an effective SNR that is often less than 0 dB and sometimes as low as −1 or −2 dB. At the same time, the information rates over these channels must be high. Clearly, the limits tell us that FEC codes must now approach their maximum performance, which essentially mandates that we use codes that come within a fraction of 1 dB of the maximum likelihood decoding performance of the given channel. Classical block codes whose size grows as the rate −R work well near capacity for blocs in the FEC sense. But the speed of these codes imposes severe processing constraints, especially under QAM constellation size growth requirements. The advent of efficient near-capacity decoding has triggered a revolution in modern FEC codes.

First adopted in run-length-limited constrained coding applications in the 1970s, these codes are now finding their way into concatenated turbo code families that have provided product-like near-capacity performance results with simple iterative decoding algorithms. As delay lines and memories with associated processing requirements have become ubiquitous, these codes have been rediscovered. They are now receiving a revitalization and are being product coded in a variety of distinct and complementary families. Constructed as part of punctured convolutional codes, interleaved trellis codes, turbo block codes, convolutional turbo codes and decodable rate-1 uncodable trellis codes, LDPC codes in either non-binary or binary form are again at the forefront of near-capacity practical coding techniques for large information message sizes and the complex modulations currently being utilized in high-rate wireless and satellite communication products.

8.6.2. Turbo Codes

In order to achieve maximum capacities close to error probabilities close to the Shannon theoretical limit, some additional components are required to the channel decoder by computationally approaching infinite code length. Therefore, turbo codes use two Convolutional codes decoders. When turbo codes were proposed, the SOVA algorithm could be used to provide the required soft values for external iteration. Turbo codes showed that the performance of these codes was close to the Shannon limits for short and medium message lengths. The main advantage of these codes is that they are able to achieve additional coding gains by increasing the number of practical iterations at the decoder.

The fundamental advance recently made with these codes was the implementation and practical performance of turbo codes. However, such codes need a careful design to get the best performance in practical wireless systems, for example the size of the interleaver and the code rates. These codes have become very popular because of their practical application in wireless systems and became increasingly adopted at the beginning of the third generation digital mobile radio telecommunications systems. However, if a small-size block is to be coded, then the performance of these error correcting codes will be very poor and more powerful codes are required to mitigate such decrease in the performance. This can be done by newly formed Convolutional codes formed into a Trellis structure.

8.6.3. Polar Codes

Polar codes represent a noteworthy advancement in the domain of error-correcting codes by demonstrating Shannon's channel coding theorem for symmetric binary discrete memoryless channels. This was a pivotal moment in coding theory, inspiring innovative sequence design approaches in other coding forms, particularly in the design of LDPC codes. Polar codes possess unique and desirable attributes for practical applications, including relatively straightforward encoding and decoding algorithms, performance close to capacity with practical block lengths, and the adaptation of channel polarization methods to several areas beyond channel coding. These factors have led to the selection of polar codes for several upcoming standards such as 5G New Radio and Wi-Fi 7.

This chapter will present a concise overview of polar codes, covering the basic principles upon which polar codes are based, their encoding and decoding algorithms, and code performance versus block length dependencies.

Polar Codes

Polar coding is a low-complexity coding technique that permits reliable communication at near capacity for a broad class of channels with discrete and continuous outputs with input-output dependencies that are "strong enough." Polar codes employ a hierarchical decomposition of the channel, wherein the transfer characteristics of each subchannel are associated with a binary input-output dependent function called the Bhattacharyya parameter. Depending on the nature of the input-output dependency, the Bhattacharyya function approaches zero for progressively more inputs when the parameters are specified to belong to a functionally continuous family of parameters. When this is the case, the Bhattacharyya function predicts the approach to capacity.

8.7. Conclusion

The quest to encounter mobile services in an increasing number of frequency bands has issued several speed requirements. To fulfill these needs for next-generation wireless access, a series of advanced techniques aiming to increase the spectral efficiency beyond state-of-the-art solutions have been proposed. However, this increase may not be retained for the increased exploitation of higher carrier frequencies, demanding relaxations on transmitter, channel, and receiver designs. Usage of higher carrier frequencies brings additional challenges, such as path loss and multipath fading, ease of blocking, feasibility of affordable and effective linearizing techniques for power amplifiers, and increased susceptibilities to ambient noise, thermal noise, phase noise, timing jitter, flicker noise, and modulation synthesis noise, as well as the added difficulty of RF instrumentations.

No solution will prevail for all scenarios. The benefits of higher carrier frequencies, such as smaller spatial form factors, potentially increased secrecy, and lower-power consumption, need to be traded-off against the challenges posed on terminal devices and required relay and backbone support architectures. Accordingly, this chapter addressed a number of modern and modernization techniques that may help satisfy current and future demands on wireless communications. Regardless of carrier frequency, better modulation and coding will, in all instances, assist with reaching the limits imposed by Shannon's theorems. The applications of higher-order modulations, non-rectangular constellations, coded modulations, distributed source coding, and coding of multiple signals will help decisionwired devices provide the best performance for required resources. More importantly, the consolidation of these advanced techniques will be necessary in order to provide a service discriminatory transmission for services requiring reduced latency and tolerance to errors, as well as services demanding reduced physical resources due to temporal restrictions.

8.7.1. Future Trends

Advanced wireless communication technologies such as 5G and beyond may require the use of a sizable amount of regulated spectrum at millimeter-wave bands. However, propagation at these bands is not only heavily attenuated in free-space, but also suffers from very intense attenuation due to oxygen absorption. In normal cellular deployments using frequency division duplexing, signals at a typical mobile station show large disparity between forward transmission and that for reverse transmission. Making matters worse, the nature of scattering at mmWave and 300 GHz bands leads to large channel delay spread making them very dispersive even at small cells. Design of practical wireless communication systems becomes a challenging problem due to no known efficient tools available for design of coding and modulation that can deal with the diverse adverse phenomena. The outcome of ATM systems may be well-shortened mobile battery life, necessitating more base station deployments, leading to an overall poor user experience and huge cost for operators.

In this chapter, we discuss the current and future trends in relation to modulation, coding, and synchronization for mobile wireless access technologies. The chapter will also cover a tutorial on a few of the latest advances in the area set to serve as enabling technologies for future 5G and beyond setups. First, we discuss the spectral efficiency and energy efficiency of any wireless access system, the recently observed paradigm shift towards greater energy efficiency in the sixth mobile generation, and the likely requirements for future wireless access technologies in terms of modulation, coding, and synchronization. This observation is of huge importance as spectral efficiency is determined at the physical layer, is the main metric any designer should seek to optimize using the resources available, while that for energy efficiency involves other area non-predictable such as the cost of base stations, user devices, and behalf as whole.

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