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BANDWIDTH-EFFICIENT WIRELESS MULTIMEDIA COMMUNICATIONS: LIMITATIONS, SOLUTIONS AND CHALLENGES INVITED PAPER

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ABSTRACT

Commencing with the brief history of mobile communications and the portrayal of the basic concept of wireless multimedia communications, the implications of Shannon's theorems regarding joint source and channel coding for wireless communications are addressed. Following a brief introduction to speech, video and graphical source coding as well as the cellular concept, a rudimentary overview of flexible, reconfigurable mobile radio schemes is provided. We then summarise the fundamental concepts of modulation, introduce an adaptive modem scheme and argue that third-generation transceivers might become adaptively re-configurable under network control in order to meet backwards compatibility requirements with existing systems and to achieve best compromise amongst a range of conflicting system requirements in terms of communications quality, bandwidth requirements, complexity and power consumption, robustness against channel errors, etc.

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1 THE WIRELESS COMMUNICATIONS SCENE

Since the end of the last century, when Marconi and Hertz demonstrated the feasibility of radio transmissions, mankind has endeavoured to fulfil the dream of wireless multimedia personal communications, enabling people to communicate with anyone, anywhere, at any time, using a range of multimedia services. The evolution of wireless systems and their subsystems has been well documented in a range of monographs by Jaykes [1], Lee [2], Parsons and Gardiner [3], Feher [4] and others. Glisic and Vucetic [5] as well as Prasad [6] concentrated on various aspects of Code Division Multiple Access (CDMA) in their monographs, while the compilation of excellent overviews edited by Glisic and Leppanen [7] treated both Time Division Multiple Access (TDMA) and CDMA along with a range of other associated aspects, such as smart antennae [53, 55, 56], trellis coding as well as emerging topics, referred to as 'time-space' processing [55], 'per-surviver' processing [57] etc.

Meyer et al [8] focused on various modern receiver techniques in their monograph. Steele [9] compiled a monograph, which considers most physical layer aspects of modern TDMA systems, including speech and channel coding, modulation, frequency hopping, and so on, amalgamating them in the last chapter in the context of the Global System of Mobile Communications known as GSM. Further important references are for example those by Rappaport [10], Garg and Wilkes [11] or the compilation edited by Gibbson [12]. These developments are also portrayed in magazine special issues [13]-[18] and excellent reviews by McDonald [19], Steele [20]-[22], Cox [30], Li and Qiu [32], Kucar [31] etc. This treatise attempts to provide an update on some of the subsystems and trends in the broad field of wireless multimedia communications. Let us commence our discourse with a glimpse of history.

The first mobile radio systems were introduced by the military, police and other emergency services, most of which were limited to voice only communications. During the pre-VLSI era the realisable signal processing complexity was severely limited and hence the handsets provided typically poor voice quality at a high cost. This was due to the phenomenon of multi-path wave propagation, where the different multi-path components arriving at the receiver's antenna suffer different attenuation and phase rotation and hence they sometimes add constructively, sometimes destructively. This situation is further aggravated by the so-called delay-spread, when the various propagation paths have rather different path-lengths and consequently exhibit different delays, spilling intersymbol interference (ISI) into the adjacent signalling or symbol intervals. These phenomena can today often be combated by sophisticated signal processing methods at the cost of added implementational complexity, which was not possible in the pre-VLSI era. Hence, until quite recently, the quality and variety of wireless services has been inferior to conventional tethered communications.

The first public cellular radio system, known as the Advanced Mobile Phone Service (AMPS) was introduced in 1979 in the United States, shortly followed by the Nordic Mobile Telephone (NMT) system in Scandinavia in 1981. The first British system was the Total Access Communications System (TACS) operated by Cellnet and Vodafone, while the Japanese introduced the Nippon Advanced Mobile Telephone System (NAMTS). All of these so-called **first-generation** national systems were based on analogue frequency modulation (FM) but used digital network control. However they did not support international roaming.

In 1982 CEPT (Conference Europeene des Postes et Telecommunication), the main governing body of the European PTT's, created the Groupe Speciale Mobile (GSM) Committee and tasked it with standardising a digital cellular Pan-European public mobile communication system to operate in the 900 MHz band. This was followed by the launch of experimental programmes of different types of digital cellular radio systems in a number of European countries. By the middle of 1986 nine proposals were received for the future Pan-European system, and GSM organised a trial in Paris to identify the one having the best performance. The technical details of the candidate systems are described in references [33], [34], [35], [36] and [37], while a short summary of their salient features was given in reference [39]. A detailed description of the standardised GSM system's main features can be found in reference [40]. This scheme constitutes the first so-called **second-generation** public land mobile radio (PLMR) system, which was designed for the worst-case propagation scenario of high-elevation antennae providing radio coverage for large rural cells. The corresponding channel conditions and techniques for mitigating their effects will be highlighted during our further discourse.

Following GSM, in 1989 the American second generation scheme known as the Digital Advanced Mobile Phone (DAMPS) system [41] had also been standardised, with the advantage of being able to accommodate three higher quality digital channels in a conventional 30 kHz analogue AMPS channel slot. Its unique feature is that similarly to the Japanese second generation scheme referred to as the Public Digital Cellular (PDC) system [42] it uses a 2 bits/symbol non-binary modem, which implicitly assumes a more benign propagation environment than that of the GSM PLMR system. The improved wave propagation conditions are a consequence of employing so-called micro-cells, where, in contrast to hostile PLMR system, the high antenna elevation is reduced to below the urban sky-line. Hence there is typically a strong line-of-sight (LOS) path between the base station (BS) and mobile station (MS), reducing the fading depth and mitigating the effect of delay-spread induced ISI. These issues will be re-visited in more depth at a later stage.

With respect to the improved propagation conditions the multi-level IS-54 and PDC systems provide a seamless transition towards the so-called cordless telecommunications (CT) system concept contrived mainly for friendly indoors office and domestic propagation environments. Hence CT products are designed to have a low transmitted power and small coverage area, where typically there is a dominant line-of-sight (LOS) propagation path between the Fixed Station (FS) and Portable Station (PS). The low transmitted power and small transmission range facilitate a low-complexity, low-cost, light-weight construction. The standardisation and development of CT products was hallmarked by the British CT2 system, the Digital European Cordless Telephone (DECT) and the Japanese Handyphone (PHP) systems. A further important milestone was the standardisation of the British DCS-1800 system, which is essentially an up-converted GSM system implemented at 1.8 GHz. The definition of the so-called half-rate GSM system supporting twice as many subscribers within the 200 kHz channel bandwidth, as the full-rate system was also an important development in the field. These second generation systems and CT schemes were described in dedicated chapters of reference [12].

Currently there exist a range of initiatives world wide, which attempt to define the **third generation** personal communications network (PCN), which is referred to as a personal communications system (PCS) in North America. The European Community's Research in Advanced Communications Equipment (RACE) programme [13, 12] and the consecutive framework referred to as Advanced Communications Technologies and Services (ACTS) programme [44, 45] spearheaded these initiatives. In the RACE programme there were two dedi-

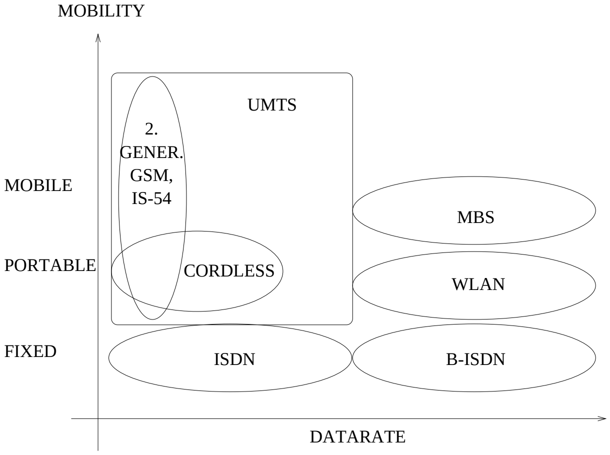


Figure 1. Stylised mobility versus bitrate plane classification of existing and future wireless systems

cated projects, endeavouring to resolve the on-going debate as regards to the most appropriate multiple access scheme, studying Time Division Multiple Access (TDMA) [13, 40, 9, 41, 12] and Code Division Multiple Access (CDMA) [13, 9, 43, 12].

European third generation research is conducted under the umbrella of the so-called Universal Mobile Telecommunications System (UMTS) [13] initiative and sofar the following proposals have been submitted to ETSI [54]: wideband CDMA [46, 47, 48], Adaptive TDMA [49] (ATDMA), hybrid TDMA/CDMA [50], Orthogonal Frequency Division Multiplex (OFDM) [51, 68] and Opportunity Driven Multiple Access (ODMA). We note that the Nokia testbed portrayed in [48] was designed with video transmission capabilities up to 128 kbps in mind. Similarly, cognizance was given to the aspects of less bandwidth-constrained, ie higher-rate video communications by the Japanese wideband CDMA proposal [52] for the Intelligent Mobile Terminal IMT 2000 emerging from NTT DoCoMo. These standardisation activities are portrayed in more depth in [54].

In the ACTS workplan [44] there are a number of projects dealing with multimedia source- and channel coding, modulation and multiple access techniques for both cellular and wireless local area networks (WLANs). These studies will design the architecture and produce demonstration models of the universal mobile telecommunications system (UMTS), which the Europeans intend to accomplish before the turn of the century. Somewhere along the line UMTS is expected to merge with the CCIR study on the future public land mobile telecommunications system (FPLMTS). These systems are characterised by the help of Figure 1 in terms of their expected grade of mobility and bitrate. These fundamental features predetermine the range of potential applications.

Specifically, the fixed networks are evolving from the basic 2.048 Mbit/s Integrated Services Digital Network (ISDN) towards higher-rate broad-band ISDN or B-ISDN. A higher grade of mobility, which we refer to here as portability, is a feature of cordless telephones, such as the DECT, CT2, PHP etc systems, although their transmission rate is more limited. The DECT systems is the most flexible one amongst them, allowing the multiplexing of 23 single-user channels in one direction, which provides rates up to 23×32 kbps = 736 kbps for advanced services. Wireless local area networks (WLAN) can support bitrates up to 155 Mbits/s in order to extend existing Asynchroneous Transfer Mode (ATM) links to portable terminals, but they usually do not support full mobility functions, such as location update or handover from one BS to another. A rapidly evolving field gaining also considerable commercial interest is associated with the research and development of High-Performance LANs (HIPERLAN) [66, 67] for 'customer premises' type communications. Contemporary second generation PLMR systems, such as GSM and IS-54 cannot support high bitrate services, since they typically have to communicate over lower quality channels, but they exhibit the highest grade of mobility, including high-speed international roaming capabilities.

The third generation UMTS is expected to have the highest grade of fexibility both in terms of its service bitrate range and in terms of mobility. In its design cognizance is given to the second generation systems. Indeed, we may anticipate that some of the subsystems of GSM and DECT may find their way into UMTS, either as a primary sub-system, or as a component to achieve backward compatibility with systems in the field. This approach may result in hand-held transceivers that are intelligent multimode terminals, able to communicate with existing networks, while having more advanced and adaptive features that we would expect to see in the next generation of wireless multimedia personal communication networks. Following the above brief overview of the wireless communications scene let us now briefly speculate on the practical embodiment of the multimedia communicator of the near future.

2 OUTLINE

Following the above brief historical overview in the rest of this treatise we concentrate mainly on bandwidth-efficient low-rate systems, although many of the proposed techniques are suitable for high-rate systems as well. Following some introductory conceptual notes as regards to a possible manifestation of the future wireless multimedia communicator in Section 3, we analyse the ramifications of Shannon's message for wireless systems in Section 4. This is followed by three Sections on speech, video and graphical source coding, before we focus our attention on transmission aspects. Section 8.1 highlights the basic cellular concept, while Section 8.2 introduces a few multiple access concepts, leading on to introducing the concept of 'software radios' or adaptive intelligent transceivers in

Section 9. We then make a short excursion to the field of modulation schemes in Section 11 and forward error correction (FEC) coding, before concluding with the portrayal of the expected system performance figures characterising such an intelligent multimode speech system in Section 13 and the characterisation of a videophone transceiver in Section 14.

The paper addresses the so-called physical-layer functions of wireless systems in more depth, but attempts also to devote some attention to higher-layer aspects, such as multiple access, dynamic channel allocation, handover, etc. Given the wide scope of this treatise, it is inevitable that some important trends and seminal contributions by highly acclaimed authors remain beyond its coverage, although with the number of references provided there is sufficient scope for the interested reader to probe further in certain deeper subject areas.

3 WIRELESS MULTIMEDIA COMMUNICATOR

A possible manifestation of the multimedia PS is portrayed in Figure 2, which is equipped with a bird-eye camera, microphone, liquid-crystal screen, serving both as video-telephone screen as well as a computer screen. The conventional keybord is likely to be replaced by a pressure-sensitive writing tablet, facilitating optical handwriting recognition [208]-[213], signature verification etc.

The pivotal implementational point of such a multimedia PS is that of finding the best compromise amongst a number of contradicting design factors, such as low power consumption, high robustness against transmission errors amongst various channel condition, high spectral efficiency, good audio/video quality, low-delay, high-capacity networking and so forth. In this contribution we will address a few of these issues in the context of the proposed PS depicted in Figure 2. The time-variant optimisation criteria of a flexible multi-media system can only be met by an adaptive scheme, comprising the firmware of a suite of system components and invoking that combination of speech codecs, video codecs, embedded channel codecs, voice activity detector (VAD) and modems, which fulfills the currently prevalent requirement [68].

These requirements lead to the concept of arbitrarily programmable, flexible so-called software radios [16], which is virtually synonymous to the so-called tool-box concept invoked for example in the forthcoming Motion Pictures Expert Group (MPEG) 4 video codec proposed for wireless video communications [70]. This concept appears attractive also for UMTS-type transceivers. A few examples of such optimisation criteria are maximising the teletraffic carried or the robustness against channel errors, while in other cases minimisation of the bandwidth occupancy, the blocking probability or the power consumption is of prime concern.

The corresponding network architecture is shown in

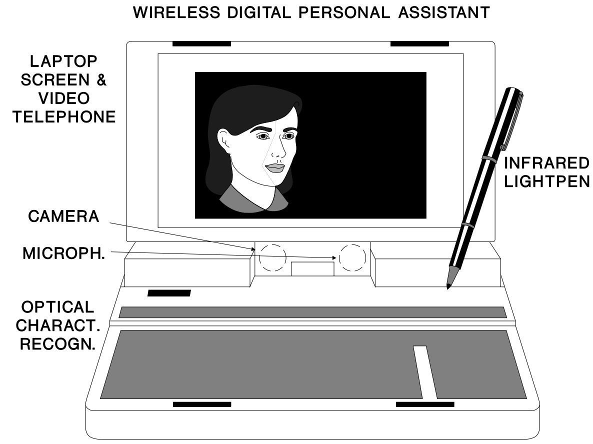


Figure 2. Wireless Multimedia Communicator

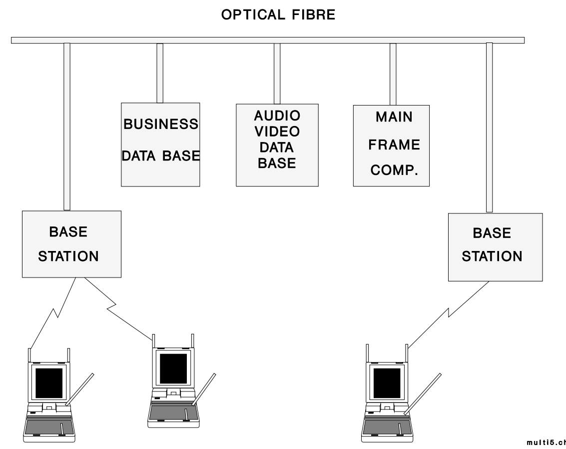


Figure 3. Wireless Multimedia Network

Figure 3. The multimedia PSs communicate with the so-called BSs in their vicinity, which are interconnected either directly using optical fibre with eachother, or in more complex systems via the so-called Mobile Switching Centres (MSC). The PSs can access through BS a range of services, including business databases, multimedia databases, main-frame computers, etc. Let us now turn our attention to some of the information theoretical aspects of wireless communications, in order to be able to understand the underlying systems technical ramifications.

4 SHANNON'S MESSAGE AND ITS IMPLICATIONS FOR WIRELESS CHANNELS

In mobile multimedia communications it is always of prime concern to maintain an optimum compromise in terms of the contradictory requirements of low bit rate, high robustness against channel errors, low delay and low complexity. The minimum bit rate at which the condition of distortionless communications is possible is determined by the entropy of the multimedia source message. Note however that in practical terms the minimum information transmission rate required for the lossless representation of the source signal, which is referred to as the source entropy is only asymptotically achievable, as the encoding memory length or delay tends to infinity. Any further compression is associated with information loss or coding distortion. Note that the optimum source encoder generates a perfectly uncorrelated source coded stream, where all the source redundancy has been removed, therefore the encoded symbols are independent and each one has the same significance. Having the same significance implies that the corruption of any of the source encoded symbols results in identical reconstructed signal distortion over imperfect channels.

Under these conditions, according to Shannon's fundamental work [72, 73, 75], best protection against transmission errors is achieved, if source and channel coding are treated as separate entities. When using a block code of length N channel coded symbols in order to encode K source symbols with a coding rate of $R = K/N$, the symbol error rate can be rendered arbitrarily low if N tends to infinity and the coding rate to zero. This condition also implies an infinite coding delay. Based on the above considerations and on the assumption of Additive White Gaussian Noise (AWGN) channels, source and channel coding have historically been separately optimised.

Mobile radio channels are typically subjected to multipath propagation and hence constitute a more hostile transmission medium than AWGN channels, exhibiting pathloss, lognormal slow fading and Rayleigh fast fading [217, 216]. Furthermore, if the signalling rate used is higher than the channel's so-called coherence bandwidth [217, 216], additional impairments are inflicted by dispersion, which is associated with frequency domain linear distortions. Under these circumstances the channel's error distribution versus time becomes bursty and an infinite-memory symbol interleaver is required in order to disperse the bursty errors and render the errors as independent, as possible, such as over AWGN channels. Clearly, for mobile channels many of the above mentioned, asymptotically valid ramifications of Shannon's theorem have a limited applicability.

A range of practical limitations must be observed, when designing wireless multimedia links. Although it is often possible to reduce the required bit rate of state-of-art multimedia source codecs while maintaining a certain reconstructed signal quality, in practical terms this is only possible at a concomittant increase of the implementational complexity and encoding delay. A good example of these limitations is the half-rate GSM speech codec, which was required to approximately halve the encoding rate of the 13 kbps full-rate codec, while maintaining less than quadrupled complexity, similar robustness against channel errors and less than doubled encoding delay. Naturally, the increased algorithmic complexity is typically associated with higher power consumption, while the reduced number of bits used to represent a certain speech segment intuitively implies that each bit will have an increased relative significance. Accordingly, their corruption may inflict increasingly objectionable speech degradations, unless special attention is devoted to this problem. It is worth noting that despite its quadruple complexity the half-rate GSM speech codec maintains a lower power consumption due to low-power 3V-technology than the first launched full rate codec had.

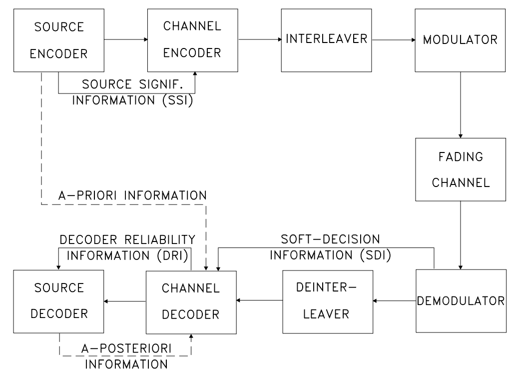


Figure 4. Intelligent transceiver schematic

In a somewhat simplistic approach one could argue that due to the reduced source rate we could accommodate an increased number of parity symbols using a more powerful, implementationally more complex and lower rate channel codec, while maintainig the same transmission bandwidth. However, the complexity, quality, robustness trade-off of such a scheme would not be very attractive.

A more intelligent approach will be required in order to design better wireless multimedia transceivers [73, 74] for bursty mobile radio channels. The simplified schematic of such an intelligent transceiver is portrayed in Figure 4. Perfect source encoders operating close to the information-theoretical limits of Shannon's predictions can only be designed for stationary source signals, a condition not satisfied by most multimedia source signals. Further previously mentioned limitations are the encoding complexity and delay. As a consequence of these limitations the source-coded stream will inherently contain residual redundancy and the correlated source symbols will exhibit un-equal error sensitivity, requiring un-equal error protection. Following Hagenauer [73, 74] we will refer to the additional knowledge as regards to the different importance or vulnerability of various source coded bits as source significance information (SSI), whereas to the confidence associated with the channel decoder's decisions as decoder reliability information (DRI).

These additional links between the source- and channel codecs are also indicated in Figure 4. Further potential performance gains are possible, when exploiting the a-posteriori information accruing from decoding a received message. This is possible for example, when the difference between consecutive decoded symbols violates some threshold condition and thereby facilitates the detection of a channel decoding error. Then the channel decoder can attempt a second tentative decoding by passing the second most likely corrected message to the source decoder, which in turn subjects this again to the previously failed threshold test, etc. A variety of such techniques have succesfully been used in robust source-matched source- and channel coding [73, 74, 82, 83]. Another practical manifestation of the time-variant source statistics of speech signals is the fact that during silent speech spurts some speech codecs do not surrender their reserved physical link, they reduce their output bitrate instead, which can reduce the interference inflicted to other users in so-called Code Division Multiple Access (CDMA) systems, such as the American IS-95 system [43]. Video codecs, such as the variable-rate MPEG 1 [80] and MPEG 2 [81] codecs even more explicitly rely on the fluctuation of the source statistics. For example, when a new object is introduced in the scope of the camera, which cannot be predicted on the basis of already known previous video frames, then the bitrate is typically increased.

The role of the Interleaver and Deinterleaver [79] seen in Figure 4 is to rearrange the channel coded bits before transmission. The mobile radio channel typically inflicts bursts of errors during deep channel fades, which often overload the channel decoder's error correction capability in certain source signal segments, while other segments are not benefiting from the channel codec at all, since they may have been transmitted between fades and hence are error-free even without channel coding. This problem can be circumvented by dispersing the bursts of errors more randomly between fades so that the channel codec is faced always with an 'average-quality' channel, rather than the bi-modal faded/non-faded condition, although only at the cost of increased system delay, which may become an impediment in interactive multimedia communications. In other words, channel codecs are most efficient, if the channel errors are near-uniformly dispersed over consecutive received segments.

In its simplest manifestation an interleaver is a memory matrix that is filled with channel coded symbols on a row-by-row basis, which are then passed on to the modulator on a column-by-column basis. If the transmitted sequence is corrupted by a burst of errors, the deinterleaver maps the received symbols back to their original positions, thereby dispersing the bursty channel errors. An infinite memory channel interleaver is required in order to perfectly randomise the bursty errors and therefore to transform the Rayleigh-fading channel's error statistics into that of an AWGN channel, for which Shannon's information theoretical predictions apply. Since in interactive multimedia communications the tolerable delay is strictly limited, the interleaver's memory length and efficiency is also limited. For further details on the effects of various interleavers on the error correction codec's efficiency the interested reader is referred to Reference [79].

A specific deficiency of the above mentioned rectangular interleavers is that in case of a constant vehicular speed the Rayleigh-fading mobile channel typically produces periodic fades [217, 216] and error bursts at travelled distances of $\lambda/2$, where λ is the carrier's wavelength, which may be mapped by the rectangular interleaver into another set of periodic bursts of errors. Again, a range of more random re-arrangement or interleaving algorithms exhibiting a higher performance than rectangular interleavers have been proposed for mobile channels in Reference [79], where also a variety of practical channel coding schemes have been portrayed. Section 5 gives a brief overview of the recent activities in speech source coding, Section 6 provides a rudimentary introduction to video source coding, while Section 7 highlights the principles of graphical source coding. For a full review of speech source coding schemes for mobile systems the interested reader is referred to references [84]- [91], joint source and channel coding was the subject of [92], whereas modulation and transmission arrangements for wireless channels have been studied in [4, 6, 69, 9, 68].

Returning to Figure 4, soft decision information (SDI) is passed by the demodulator to the FEC decoder, indicating that the demodulator refrained from making a hard-decision concerning the received bit. Instead, it passes the estimated reliability of the received information to the FEC decoder, thereby improving its efficiency. The channel state information (CSI), which is in simple terms representative of the current fade depth, can be used to weight the the SDI in the detection process. This weighted reliability information is then often used by the channel decoder in order to invoke maximum likelihood sequence estimation (MLSE) based on the Viterbi algorithm [311, 79] in order to improve the system's performance with respect to conventional hard decision decoding. Following the above rudimentary review of Shannon's infromation theory, the rest of this treatise is devoted to practical issues of wireless multimedia communications. Let us initially consider briefly the recent advances in speech source coding.

5 SPEECH SOURCE CODING

5.1 A historical perspective on speech codecs

Following the 64 kbits/s Pulse Code Modulation (PCM) and 32 kbps Adaptive PCM (ADPCM) G.721 Recommendations standardised by the International Telecommunications Union (ITU), in 1986 the 13 kbits/s Regular Pulse Excitation (RPE) [105, 106] codec was selected for the Pan-European mobile system known as GSM, and more recently Vector Sum Excited Linear Prediction (VSELP) [107, 108] codecs operating at 8 and 6.7 kbits/s were favoured in the American IS-54 and the Japanese PDC wireless networks. These developments were followed by the 4.8 kbits/s American Department of Defence (DoD) codec [112]. The state-of-art was documented in a range of excellent monographs by O'Shaughnessy [87], Furui [88], Anderson and Seshadri [92], Kondoz [89], Kleijn and Paliwal [90] and in a tutorial review by Gersho [78]. More recently the 5.6 kbits/s half-rate GSM quadruple-mode Vector Sum Excited Linear Predictive (VSELP) speech codec standard developed by Gerson et al [109] was approved, while in Japan the 3.45 kbits/s half-rate PDC speech codec invented by Ohya, Suda and Miki [113] using the so-called Pitch Synchronous Innovation (PSI) CELP principle was standardised. Other currently investigated schemes are the Prototype Waveform Interpolation (PWI) proposed by Kleijn [114], Multi-Band Excitation (MBE) suggested by Griffin et al [115] and Interpolated Zinc Function Prototype Excitation (IZFPE) codecs advocated by Hiotakakos and Xydeas [116]. In the low-delay, but more error sensitive backward adaptive class the 16 kbps ITU G.728 codec [117] developed by Chen et al from the AT&T speech team hallmarks a significant step. This was followed by the equally significant development of the more robust, forward-adaptive 15 ms delay G.729 ACELP arrangement proposed by the University of Sherbrooke team [122, 123], AT&T and NTT [118]. Lastly, the standardisation of the 2.4 kbps DoD codec led to intensive research in this very low-rate range and the Mixed Excitation Linear Predicitve (MELP) codec by Texas Instrument was identified [119] in 1996 as the best overall candidate scheme.

Before concluding our discourse on speech codecs let us briefly highlight the problems associated with 7kHz bandwidth -so-called commentatory quality speech coding.

5.2 Wideband speech codecs

For the sake of completeness we note briefly that 7kHz bandwidth speech codecs offer more transparent speech quality than their narrowband counterparts at typically higher bitrate and algorithmic complexity.

One of the problems associated with full-band coding of wideband speech is the codec's inability to treat the less predictable high-frequency, low-energy speech band, which was tackled by the ITU G.722 codec using split-band or sub-band coding. Although the upper subband is important for maintaining an improved intelligibility and naturalness, it only contains a small fraction of the speech energy, which is on the order of 1% and therefore its bitrate contribution has to be limited appropriately. The ITU G.722 codec [131] uses two equal-width subbands, whose signals are encoded employing ADPCM techniques and has the ability of transmitting speech at 64, 56 or 48 kbps, while allocating 0, 8 or 16 kbps capacity for data transmission.

Quackenbush [132] suggested a transform-coded approach in order to allow for a higher flexibility in terms allocating the bits available, which was proposed originally by Johnston [133] for 30kHz-sampled high-fidelity audio signals and reduced the bitrate required according to the lower sampling rate of 16kHz. Ordentlich and Shoham proposed a low-delay Celp-based 32 kbps wideband codec [134], which achieved a similar speech quality to the G.722 64 kbps codec at a concomitant higher complexity. The backward-adaptive LPC filter used had an order of 32, which was significantly lower than the filter order of 50 used in the G.728 codec [117]. The G.728 filter-order of 50 was able to cater for long-term periodicities of upto 6.25 ms, corresponding to pitch frequencies down to 160 Hz at a sampling rate of 8kHz without a LTP, allowing better reconstruction for female speakers. The filter order of 32 at a sampling frequency of 16 kHz cannot cater for long-term periodicities. Nonetheless, the authors opted for using no LTP. In contrast to the G.728 codebook of 128 entries here 1024 entries were used to model the 5-sample excitations.

In a contribution by Black, Kondoz and Evans [135] the backward- adaptive principle was retained for the sake of low delay, but it was combined with a split-band approach. The low-band was encoded by a backward-adaptive CELP codec using a 10-th order LPC filter updated over 14 8 kHz-sampled samples or 1.75 ms and the authors argued that it was necessary to incorporate a forward adaptive LTP in order to counteract the potentially damaging error feedback effect of the backward-adaptive LPC analysis. The upper-band typically contains a less structured, noise-like signal, which has a slowly varying dynamic range. Black et al here proposed to use a 6th order forward-adaptive predictor updated over a 56-sample interval, which is quadrupled in comparison to the low-band. Backward- adaptive prediction would be unsuitable for this less accurately quantised band, which would precipitate the effect of quantization errors in future segments.

The prestigous speech coding group at Sherbrooke University [136, 137, 138] proposed a range of ACELP-based codecs, since Laflamme, Adoul Salami et al argued that ACELP codecs are amenable to wideband coding, when employing vast codebooks in conjunction with a reduced-complexity focused codebook search strategy using a number of encapsulated search loops. This technique facilitates searching only a fraction of a large codebook, while achieving a similar performance to that of a full-search. Suffice to say here that this technique was proposed by the authors also for the ITU G.729 8 kbps low-delay codec using a 15-bit ACELP codebook and five encapsulated loops [121, 122].

Here we conclude our discussion of speech source codecs and briefly classify a range of video codecs suitable for wireless videophony and other wireless visual communications services, before focusing our attention on wireless transmission aspects.

6 VIDEO SOURCE CODING

6.1 Motivation and Background

Motivated by the proliferation of wireless multimedia services [139, 140], a plethora of video codec schemes have been proposed for various applications [141]-[156], but the perhaps most significant advances in the field are hallmarked by the MPEG4 initiative [70]. The design of videophone schemes centres around the best compromise amongst a number of inherently contradictory specifications, such as video quality, bit rate, implementational complexity, robustness against channel errors, coding delay, bitrate fluctuation and the associated buffer length requirement, etc. Many of these aspects have been treated in a number of established monographs by Netravali and Haskell [143], Jain [191], Jayant and Noll [85] as well as Gersho and Gray [149]. A plethora of video codecs have been proposed in the excellent special issues edited by Tzou, Mussmann and Aigawa [157], by Hubing [158] and Girod et al [159] for a range of bitrates and applications, but the individual contributions by a number of renowned authors are too numerous to review. Khansari, Jalali, Dubois and Mermelstein [166] as well as Mann Pelz [180] reported promising results on adopting the H.261 codec for wireless applications by invoking powerful signal processing and error-control techniques in order to remedy the inherent source coding problems due to stretching its application domain to hostile wireless environments. Färber, Steinbach and Girod [167]-[170] also contributed substantially towards advancing the state of art in the context of the H.263 codec as well as in motion compensation [168, 169], as did Eryurtlu, A.H. Sadka, A.M. Kondoz [174, 175]. Further important contributions in the field were due to Chen et al [181], Illgner and Lappe [182] Zhang [183], Ibaraki, Fujimoto and Nakano [184], Watanabe et al [185] etc, the MPEG4 consortium's endeavours [71], the efforts of the mobile audio-video terminal (MAVT) consortium. Vector quantisation based schemes were advocated by Ramamurthy and Gersho [149] as well as by Torres and Huguet [150]. A major feature topic of the European Community's Fourth Framework Programme [44, 45] on Advanced Communications Technologies and Services (ACTS), is video communications over a range of wireless and fixed links.

In this Section initially we focused our attention on the design and performance evaluation of wireless video telephone systems, suitable for the robust transmission of Quarter Common Intermediate Format (QCIF) sequences over conventional mobile radio links, such as the Pan-European GSM system [40], the American IS-54 [41] and IS-95 [43] systems as well as the Japanese PDC system [42]. In contrast to existing standard codecs, such as the ITU H.261 scheme and the MPEG1 [80], MPEG2 [81] and MPEG4 [70] arrangements, our proposed video codec's fixed, but arbitrarily programable bitrate facilitates its employment also in future intelligent systems, which are likely to vary their bitrate in response to various propagation and teletraffic conditions. We will conclude the Section with a brief overview of the ITU H.263 standard video codec, which is a flexible scheme, suitable for a range of multimedia visual applications at various bitrates and video resolutions.

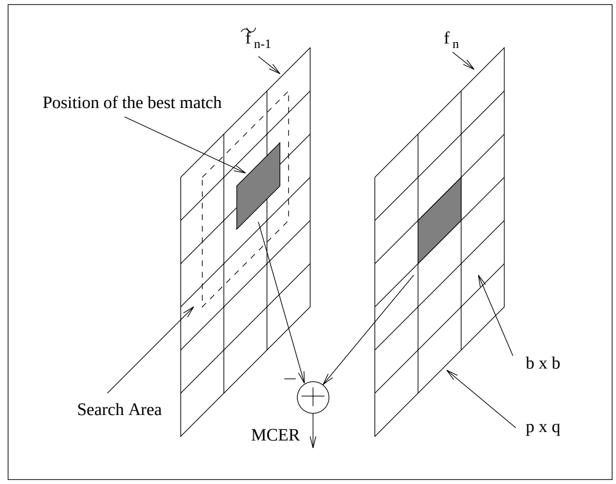


Figure 5. Simplified schematic of motion compensation ©J. Streit [186], 1996

6.2 Motion Compensation

The ultimate goal of low-rate image coding is to remove redundancy in both spatial and temporal domains and thereby reduce the required transmission bit rate. The temporal correlation between successive image frames is typically removed using block-based motion compensation, where each block to be encoded is assumed to be a motion-translated version of the previous locally decoded frame.

The vector of motion translation or motion vector (MV) is typically found by the help of correlation techniques, as seen in Figure 5. Specifically, a legitimate motion translation region or search scope is stipulated within the previous locally decoded frame, the block to be encoded is slid over this region according to a certain algorithm and the location of highest correlation is deemed to be the destination of the motion translation. Motion compensation (MC) is then carried out by subtracting the appropriately motion translated previous decoded block from the one to be encoded in order to generate the so-called motion compensated error residual (MCER). Clearly, the image is decomposed in motion translation and MCER, and both components have to be encoded and transmitted to the decoder for image reconstruction. The motion compensation removes some of the temporal redundancy and the variance of the MCER becomes much lower than that of the original image, which ensures bit rate economy.

The MCER frame can then be represented using a range of techniques [190], including subband coding [144,

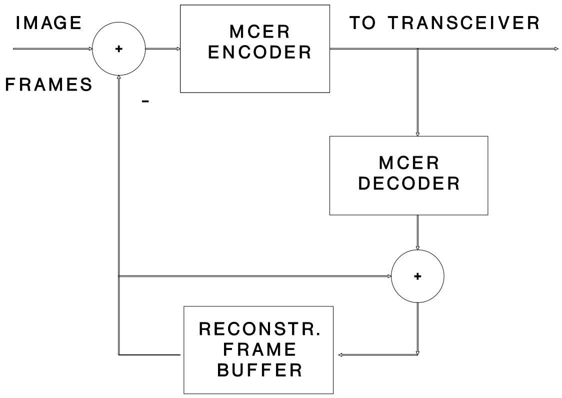


Figure 6. Simple video codec schematic

145] (SBC), wavelet coding [146], Discrete Cosine Transformation [191, 80, 81, 188] (DCT), vector quantisation (VQ) [149]-[151] or Quad-tree [147, 148, 155, 189] (QT) coding. Some of these techniques will be highlighted in the forthcoming Subsections.

When a low codec complexity and low bit rate are required, the motion compensation technique described above can be replaced by simple frame-differencing. In frame-differencing the whole of the previous locally decoded image frame is subtracted from the one to be encoded without the need for the above correlation-based motion prediction, which may become very computationally intensive for high-resolution, high-quality video portraying high-dynamic scenes. Such a simple video codec schematic based on simple frame-differencing is shown in Figure 6. Although the MCER residual variance remains somewhat higher for frame-differencing than in case of full motion compensation, there is no pattern-matching search, which reduces the complexity and no MVs have to be encoded, which may reduce the overall bit rate. Observe in Figure 6 that after frame-differencing the encoded MCER is conveyed to the transceiver and also locally decoded. This is necessary to be able to generate the locally reconstructed video signal, which is invoked by the encoder in subsequent MC steps. The encoder uses the locally reconstructed, rather than the original input video frames, since these are not available at the decoder, which would result in mis-alignment between the encoder and decoder. This local reconstruction operation is carried out by the adder in the Figure, superimposing the decoded MCER on the previous locally decoded video frame. The operations are similar, if full MC is used. Practical codecs, such as for example the ITU H.263 scheme, often combine the so-called inter-frame and intra-frame coding techniques on a block-by-block basis, where MC is employed only if it was deemed advantageous in MCER reduction terms.

In case of highly correlated consecutive video-frames the MCER typically exhibits 'line-drawing' characteristics, where large sections of the frame difference signal are 'flat', characterised by low pixel magnitude values, while the motion contours, where the frame differencing has failed to predict the current pixels on the basis of the previous locally decoded frame are represented by larger values, as seen in at the centre of Figure 9. Consequently, efficient MCER residual coding algorithms must be able to represent such textured MCER patterns adequately, a topic to be addressed in the forthcoming subsections. Let us initially consider a bandwidth-efficient cost-gain quantised DCT-based codec [188].

6.3 DCT-based Video Codec

Our DCT-based video codec's outline is depicted in Figure 7. The DCT [191] has been popular in video compression standards [80, 81], since it exhibits a so-called energy compaction property, implying that upon transforming a correlated or predictable signal to the spatial frequency domain most of its energy will be compacted to a few high-energy, low-frequency coefficients. This is a consequency of the Wiener-Khnitsin theorem, stating that the power spectral density (PSD) and the autocorrelation function (ACF) are Fourier transform pairs. Hence the flat ACF of a predictable, slowly-varying signal implies a compact low-pass type PSD, which is amenable to compression, since in the spatial frequency domain a lower number of coefficients has to be transmitted than in the temporal domain. It is important to note that the MC often removes most of the redundancy from the correlated temporal domain video frame and hence the DCT of the MCER may even result in an expanded spatial frequency domain representation, which can be conteracted for example by adaptive bit allocation schemes. Strobach [147] proposed quad-tree coding in order encode the MCER and mitigate this problem. Alternative frequency domain solutions include subband coding [144, 145] (SBC) or wavelet coding [146], which facilitate a flexible control over the allocation of bits in the spatial frequency domain. The MPEG standard codecs [80, 81] and the H.261, H.263 codecs scan and entropy code the DCT coefficients and also allow direct encoding of the more correlated video signal on a block-by-block basis. Vector quantisation (VQ) [149]-[151] can be carried out both in the frequency and the time domains, but a persistant deficiency is their difficulty to handle sharp edges adequately.

Returning to the DCT principle, our proposed DCT-based codec was designed to achieve a time-invariant compression ratio associated with a fixed but programmable encoded video rate of 5-13 kbps [1]. The codec's operation is initialised in the intra-frame mode, but once it switched to the inter-frame mode, any further mode switches are optional and only required if a drastic scene change occurs.

[1]The MA sequence encoded at various bitrates can be viewed under the WWW address http://www-mobile.ecs.soton.ac.uk

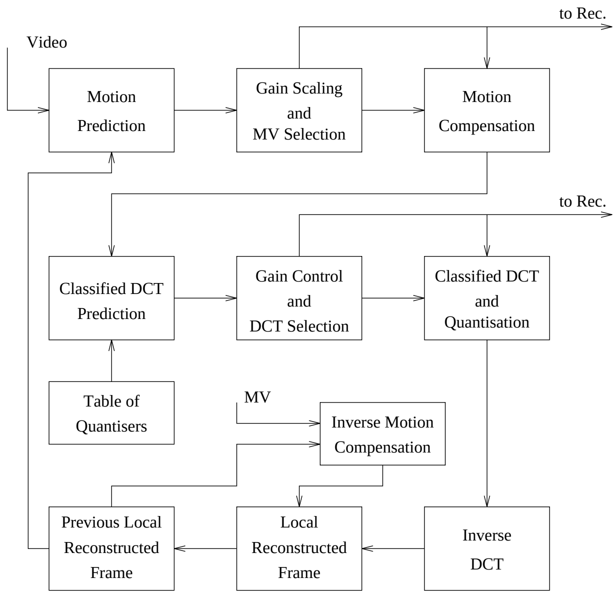


Figure 7. DCT-codec schematic ©IEEE, Hanzo & Streit [188], 1995

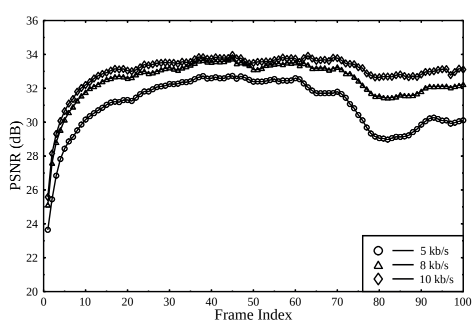


Figure 8. PSNR versus frame index performance at various bitrates for the 'Miss America' sequence ©IEEE, Hanzo & Streit [188] 1995

In the intra-frame mode the encoder transmits the coarsely quantised block averages for the current frame, which provides a low-resolution initial frame required for the operation of the inter-frame codec at both the commencement and during later stages of communications in order to prevent encoder/decoder misalignment. For 176×144 pixel ITU standard Quarter Common Intermediate Format (QCIF) images in a specific scenario [188] we limited the number of video encoding bits per frame to 1136, corresponding to a bitrate of 11.36 kbps at 10 frames/s.

In the motion-compensation 8×8 blocks are used. At the commencement of the encoding procedure the motion compensation (MC) scheme determines a motion vector (MV) for each of the 8×8 blocks using full-search. The MC search window is fixed to 4 × 4 pels around the center of each block and hence a total of 4 bits are required for the encoding of 16 possible positions for each MV. Before the actual motion compensation takes place, the codec tentatively determines the potential benefit of the compensation in terms of motion compensated error energy reduction. Then the codec selects those blocks as 'motion-active' whose gain exceeds a certain threshold. This method of classifying the blocks as motion-active and motion-passive results in an active/passive table, which consists of a one bit flag for each block, marking it as passive or active.

Pursuing a similar approach, gain control is also applied to the Discrete Cosine Transform (DCT) based compression. Every block is DCT transformed and quantised. In order to take account of the non-stationary nature of the motion compensated error residual (MCER) and its time-variant frequency-domain distribution, four different sets of DCT quantisers were designed. The quantisation distortion associated with each quantiser is computed in order to be able to choose the best one. Ten bits are allocated for each quantiser, each of which are trained Max-Lloyd quantisers catering for a specific frequency-domain energy distribution class. All DCT blocks whose coding gain exceeds a certain threshold are marked as DCT-active resulting in a similar active/passive table as for the motion vectors. For this second table we apply the same run length compression technique, as above. Again, if the number of bits required for the encoding of the DCT-active blocks exceeds half of the maximum allowable number, blocks around the fringes of the image are considered DCT-passive, rather than those in the central eye and lip sections. If, however, the active DCT coefficient and activity-table do not fill up the fixed-length transmission burst, the thresholds for active DCT blocks is lowered and all tables are recomputed.

The bit allocation scheme was designed to deliver 1136 bits per frame, which is summarised in Table 1. The encoded bitstream begins with a 22 bit frame alignment word (FAW). This is necessary to assist the video decoder's operation in order resume synchronous operation after loss of frame synchronisation over hostile fading channels. The partial intra-frame update refreshes only 22 out of 396 blocks every frame. Therefore every 18 frames or 1.8 seconds the update refreshes the same blocks. This periodicity is signalled to the decoder by transmitting the inverted FAW. A MV is stored using 13 bits, where 9 bits are required to identify one of the 396 the block indexes using the enumerative method and 4 bits for encoding the 16 possible combinations of the X and Y displacements. The 8×8 DCT-compressed blocks use a total of 21 bits, again 9 for the block index, 10 for the DCT coefficient quantisers, and 2 bits to indicate which of the four quantiser has been applied. The total number of bits becomes $30 \cdot (13{+}21){+}22 \cdot 4 + 22 + 6{=}1136$, where six dummy bits were added in order to obtain a total of 1136 bits suitable in terms of bit packing requirements for the specific forward error correction

FAW	PFU	MV Index	MV	DCT Index	DCT	Padding	Total
22	22×4	30×9	30×4	30×9	30×12	6	1136

Table 1. Bit Allocation Table per QCIF Video Frame for the Fixed-rate DCT Codec ©IEEE, Hanzo & Streit [188] 199

block codec used.

The encoded parameters are transmitted to the decoder and also locally decoded in order to be used in future motion predictions. The video codec's Peak SNR (PSNR) versus frame index performance is shown in Figure 8, where the PSNR is defined, as the conventional Signal-to-Noise Ratio (SNR), except that instead of the actual video signal power a video pixel value of 255 is assumed, yielding a pixel power of 255^2 for all pixel positions across the video frame. Since 255 is the highest possible value for an 8-bit pixel representation, the PSNR is typically higher, than the conventional SNR. The codec proposed was subjected to bit-sensitivity analysis and a Quadrature Amplitude Modulation [68] (QAM) based source-sensitivity matched transceiver was designed in order to transmit the video stream over wireless channels. The interested reader is referred to reference [188] for further details. Having described the principles of DCT-based video coding let us now consider QT coding of the MCER [189].

6.4 Quadtree structured coding

The proposed QT-codec shares the structure of the previous DCT-based scheme portrayed in Figure 7, but employs QT-coding of the MCER. Quad-trees (QT) represent a sub-class of the so-called region growing techniques, where the image, in our case the MCER generated by the MC scheme is described by the help of variable size sectors characterized by similar features, in this case, similar grey levels. Explicitly, the MCER is described in terms of two sets of parameters, the structure of similar regions and their grey levels. Note that the information characteristic of the QT structure is potentially much more sensitive to bit errors than the grey level coding bits.

Before QT decomposition takes place, the frame difference signal is divided in 16×16-pixel blocks perfectly tiling the original difference frame. Creating the QT regions is a recursive operation. Considering each individual pixel, two or more neighbours are merged together if a certain merging criterion is satisfied. This criterion may be, for instance, a similar grey level. This merging procedure is repeated until no more regions satisfy the merging criterion, hence no more merging is possible. Similarly, the QT regions can be obtained in a top-down approach, dividing the MCER in a number of sections, if the sections do not satisfy the similarity criterion, and continue until the pixel level is reached and no further splits are possible.

The quad-tree approach is one possible implementation of the socalled region growing techniques. This process can be observed in Figure 9. For a rectangular region an algorithmically attractive implementation is, when commencing at the pixel level, four quadrants of a square are merged together, if the matching criterion is met. The grey levels of the quadrants of a square are represented by $m1 \ldots m4$ and their mean is computed according to $m = (m_1 + m_2 + m_3 + m_4)/4$. If the absolute difference of all four pixels and the mean grey level is less than the system parameter σ, then these pixels satisfy the merging criterion. Explicitly, a simple merging criterion can be formulated as follows:

$$(|m - m_1| < \sigma) \cap (|m - m_2| < \sigma) \cap (|m - m_3| < \sigma) \cap (|m - m_4| < \sigma) = \text{True}, \quad (1)$$

where $\cap$ represents the logical AND operation.

It is expected that if the system parameter σ is reduced, the matching criterion becomes more stringent and hence less merging takes place, which is likely to increase the required encoding rate at a concommittant improvement of the MCER's representation quality. In contrast, an increased σ value is expected to allow more merging to take place and hence reduce the bit rate, as we will show in our results Section.

If the merging criterion is satisfied, the mean grey level m becomes the grey level of the merged quadrant in the next generation, and so on. At this stage it is important to note that the quality control threshold σ does not need to be known to the QT decoder. Therefore the image representation quality can be rendered position-dependent within the frame being processed, which allows weighting to be applied to important image sections, such as the eyes and lips without increasing the complexity of the decoder or the transmission rate.

Pursuing the top-down QT decomposition approach, the frame difference signal constitutes a so-called node in the QT. After splitting this node gives rise to four further nodes, which are classified on the basis of the 'similarity criterion'. Specifically, if all the pixels at this level of the QT differ from the mean m by less than the threshold σ, then they are considered to be a so-called 'leaf node' in the QT. Hence they do not have to be subjected to further 'similarity tests', they can be represented simply by the mean value m.

If, however, the pixels constituting the current node to be classifed differ by more than the threshold σ, the pixels forming the node cannot be adequately represented

Figure 10. Enhanced sample codebook with 128 8×8 vectors ©Streit and Hanzo[190], 1997

by their mean m and thus they must be further split, until the threshold condition is met. This repetitive splitting process is continued, until there are no more nodes to split, since all the leaf nodes satisfy the threshold criterion, as shown in Figure 9. Consequently the QT structure describes the contours of similar grey levels in the frame difference signal.

In order to be able to reproduce the encoded image, not only the grey levels of the leaf nodes, but also the QT structure must be efficiently encoded and communicated to the decoder. Fortunately, the QT structure can be efficiently described by the help of a variable-length code. At the commencement of image communications a low-resolution version of the first image frame is encoded and transmitted to the decoder in order to assist in its operation. Then the MCER signal is computed, which is subjected QT coding (QTC) before transmission to the decoder. Again, the schematic of the QT codec obeys the structure of Figure 6. The QT coded MCER is locally decoded and added to the previous locally decoded frame and stored in the frame buffer for the duration of one frame in order to generate the next block estimates for the MC operation. A range of techniques related to the optimum QT splitting and bit allocation techniques were suggested in [189], where also details of the source-matched video transceiver can be found. Adequate video quality was achieved for the Miss America sequence for a bitrate of 11.36 kbps, when using 10 frames/s scanned QCIF images.

Without aiming for an indepth treatment we briefly allude to the concept of vector quantised video codecs, where the MCER of an 8×8 pixel block is represented by the best matching entry of the two-dimensional codebook shown in Figure 10. This principle also allowed us to contrive an 11.36 kbps QCIF codec for wireless video telephony, the details of which were presented in Reference [187], where the full transceiver performance over fading channels is also characterised. The above-mentioned range of fixed-rate video codecs is compared in terms of error resilience and video quality in Reference [190]. In References [187]-[189] a range of flexible reconfigurable multi-level transceivers were designed for the transmission of the VQ-, DCT- and QT-coded video streams by allocating an addition physical speech channel for video telephony. However, for reasons of space economy these results were not included here. Similar video PSNR versus channel SNR results are here provided using the ITU H.263 video codec and a reconfigurable transceiver in order to characterise the expected video performance in Figures 27 and 28.[2]

In closing we note again that the literature of video compression is very rich [139]-[156] and recent developments led to the definition of the MPEG1, MPEG2, H.261 and H.263 standards. Although these codecs rely on vulnerable variable-length coding techniques, work is also under way towards contriving more robust coding algorithms, such as those to be incorporated in the forthcoming MPEG4 scheme [71, 70]. In the next Subsection we briefly highlight the features of the standard H.263 scheme, which is an error-sensitive, variable-rate scheme, but achieves a very high compression ratio and hence to date it is the best existing standardised video codec. We will also propose appropriate transmission techniques to support its operation in a wireless videophone scheme.

6.5 The H.263 ITU Codec

The H.263 codec was detailed in References [193, 194], while a number of transmission schemes designed for accommodating its rather error-sensitive bit-stream were proposed in [195]-[177]. As an illustrative example, in Table 2 we summarised the various video resolutions supported by the H.261 and H.263 ITU codecs, in order to demonstrate their flexibility [192]. Their uncompressed bitrates at frame scanning rates of both at 10 and 30 frames/sec for both grey and colour video are also listed. The mature H.261 standard defined two different picture resolutions, namely QCIF and CIF, while the H.263 codec has the ability to support five different resolutions. All H.263 decoders must be able to operate in sun-QCIF (SQCIF) and QCIF modes and optionally support CIF, $4 \times$CIF and $16 \times$CIF formats.

The H.261 and H.263 codecs share the simplified schematic of Figure 11, which operate under the instructions of the coding control block, selecting the required inter/intra frame mode, the quantisation and bitallocation scheme etc. DCT is invoked to compress either the original or the MCER blocks and the encoded video signal is also locally decoded and stored in the frame memory

[2] The exposition of this Section can be augmented by studying the corresponding real-time video quality under http://www-mobile.ecs.soton.ac.uk

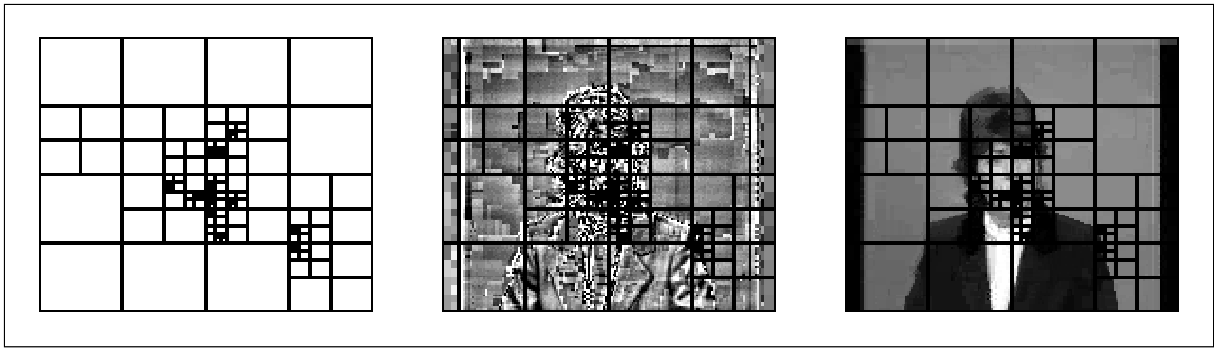


Figure 9. Quad-tree segmentation example with and without overlaid MCER and original video frame ©Streit & Hanzo, [189], 1996

Video Format	Luminance dimensions	No. of Pels per frame	Uncompressed bitrate (Mbit/s)			
			10 frame/s		30 frame/s	
			Grey	Colour	Grey	Colour
SQCIF	128 x 96	12 288	0.983	1.47	2.95	4.42
QCIF	176 x 144	25 344	2.03	3.04	6.09	9.12
CIF	352 x 288	101 376	8.1	12.2	24.3	36.5
4CIF	704 x 576	405 504	32.4	48.7	97.3	146.0
16CIF	1408 x 1152	1 622 016	129.8	194.6	389.3	583.9
CCIR 601	720 x 480	345 600	27.65	41.472	82.944	124.416
HDTV 1440	1440 x 960	1 382 400	110.592	165.888	331.776	497.664
HDTV	1920 x 1080	2 073 600	165.9	248.832	497.664	746.496
SQCIF: Sub-Quarter Common Intermediate Format QCIF: Quarter Common Intermediate Format CIF: Common Intermediate Format HDTV: High Definition Television						

Table 2. Various video formats and their uncompressed bitrate. Upon using compression 10-100 times lower average bit rates are realistic.

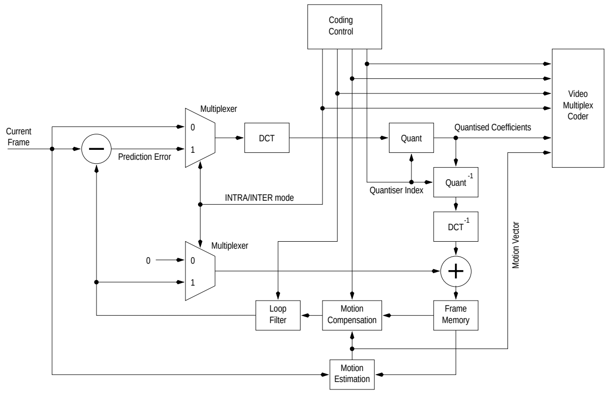


Figure 11. Simplified H.261/H.263 schematic©Cherriman [192], 1995

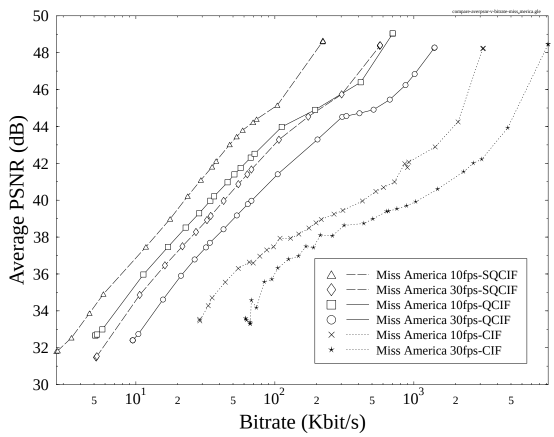


Figure 12. Image quality (PSNR) versus coded bitrate, for H.263 "Miss America" simulations at 10 and 30 frames/s using SQCIF, QCIF and CIF sequences ©Cherriman [192], 1995

in order to be used in future MC steps. All encoded information is multiplexed for transmission by the video multiplex coder. The codec's PSNR versus encoded bitrate performance is portrayed in Figure 12 for "Miss America" simulations at 10 and 30 frames/s using SQCIF, QCIF and CIF sequences [192]. Observe in the Figure the codec guarantees near-linear rate-scalability over a wide operating range, which is partly explained by the extensive employment of entropy coding schemes. The performance of a complete adaptive videophone system will be portrayed after considering the associated wireless transmission aspects.

Here we curtail our discussions on video codecs and provide some notes on another aspect of multimedia communications, namely on graphical correspondence.

7 GRAPHICAL SOURCE CODING

7.1 Background

Telewriting is a multimedia telecommunication service enabling the bandwidth-efficient transmission of handwritten text and line graphics through fixed and wireless communication networks [196]-[201]. Differential chain coding (DCC) has been successfully used for graphical communications over E-mail networks [196] or teletext systems [199], where bit rate economy is achieved by exploiting the correlation between successive vectors. References [197] and [202] addressed also some of the associated communications aspects. A plethora of further excellent treatises were contributed to the literature of chain coding by R. Prasad and his colleagues from Delft University [203]-[205].

7.2 Fixed length differential chain coding[206]

In chain coding (CC) a square-shaped coding ring is slid along the graphical trace from the current pixel, which is the origin of the legitimate motion vectors, in steps represented by the vectors portrayed in Figure 13. The bold dots in the Figure represent the next legitimate pixels during the graphical trace's evolution. In principle the graphical trace can evolve to any of the surrounding eight pixels and hence a three-bit codeword is required for lossless coding. Differential chain coding [203] (DCC) exploits that the most likely direction of stylus movement is a straight extension, corresponding to vector 0 and with a gradually reducing probabilty of sharp turns corresponding vectors having higher indeces. Explicitly, we have found the while vector 0 typically has a probability of around 0.5 for a range of graphical source signals, including English and Chinese handwriting, a Map and a technical Drawing, the relative frequency of vectors ± 1 is around 0.2, while vectors $\pm 2, \pm 3$ have probabilities around 0.05. This suggests that the coding efficiency can be improved using the principle of entropy coding by allocating shorter codewords to more likely transitions and longer ones to less likely transitions.

In reference [206] we embarked on exploring the potential of a graphical coding scheme dispensing with variable length coding, which we refer to as fixed length differential chain coding (FL-DCC). FL-DCC was contrived in order to comply with the time-variant resolution- and/or bit rate constraints of intelligent adaptive multimode terminals, which can be re-configured under network control to satisfy the momentarily prevailing tele-traffic, robustness, quality, etc system requirements. In order to maintain lossless graphics quality under lightly loaded traffic conditions, the FL-DCC codec can operate at a rate of $b = 3$ bits/vector, although it has a higher bit rate than DCC. However, since in voice and video coding typically perceptually unimpaired lossy quantisation is used, we embarked on exploring the potential of the reconfigurable FL-DCC codec under $b < 3$ low-rate, lossy

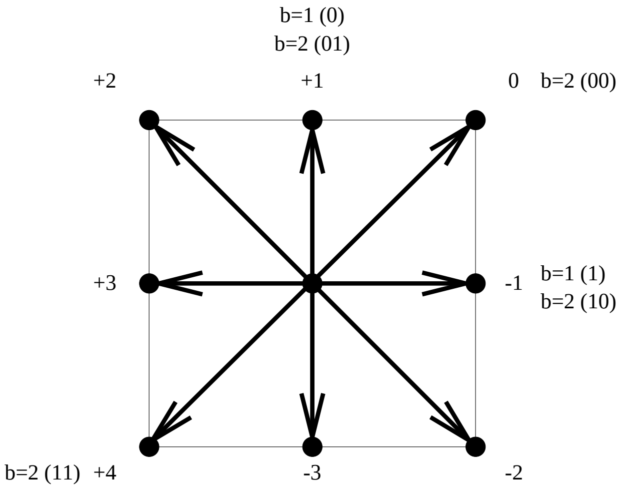


Figure 13. Coding ring

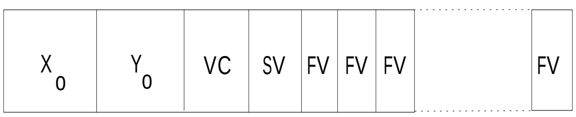


Figure 14. FL-DCC Coding Syntax ©IEEE, Yuen and Hanzo [206], 1995

conditions.

Based on our findings as regards to the relative frequencies of the various differential vectors, we decided to evaluate the performance of the FL-DCC codec using the $b = 1$ and $b = 2$ bit/vector lossy schemes. As demonstrated by Figure 13, in the $b = 2$-bit mode the transitions to pixels -2, -3, +2, +3 are illegitimate, while vectors 0, +1, -1 and +4 are legitimate. In order to minimise the effects of transmission errors the Gray codes seen in Figure 13 were assigned. It will be demonstrated that, due to the low probability of occurance of the illegitimate vectors, the associated subjective coding impairment is minor. Under degrading channel conditions or higher tele-traffic load the FL-DCC coding rate has to be reduced to $b = 1$, in order to be able to invoke a less bandwidth efficient, but more robust modulation scheme or to generate less packets contending for transmission. In this case only vectors +1 and -1 of Figure 13 are legitimate. The subjective effects of the associated zig-zag trace will be removed by the decoder, which can detect these characteristic patterns and replace them by a fitted straight line.

In general terms the size of the coding ring is given by $2n\tau$, where $n = 1, 2, 3\ldots$ is referred to as the order of the ring and τ is a scaling parameter, characteristic of the pixel separation distance. Hence the ring shown in Figure 13 is a first order one. The number of nodes in the ring is $M = 8n$.

The data syntax of the FL-DCC scheme is displayed in Figure 14. The beginning of a trace can be marked by a typically 8 bit long pen-down (PD) code, while the end of trace by a pen-up (PU) code. In order to ensure that these codes are not emulated by the remaining data, if this would be incurred, bit stuffing must be invoked. We found that in complexity and robustness terms using a 'vector counter' (VC) constituted a more attractive alternative for our system. The starting coordinates X_0, Y_0 of a trace are directly encoded using for example 10 and 9 bits in case of a video grahics array (VGA) resolution of 640×480 pixels.

The first vector displacement along the trace is encoded by the best fitting vector defined by the coding ring as the starting vector (SV). The coding ring is then translated along this starting vector to determine the next vector. A differential approach is used for the encoding of all the following vectors along the trace, in that the differences in direction between the present vector and its predecessor are calculated and these vector differences are mapped into a set of 2^b fixed length b-bit codewords, which we refer to as 'fixed vectors' (FV).

We designed a wireless 4QAM-based [68] transceiver for the transmission of FL-DCC encoded graphical source signals and evaluated the system's robustness over Rayleigh-fading channels with second-order switched-diversity, using automatic repeat requests limited to a maximum of three transmission attempts (TX3) [207]. Here we refrain from providing PSNR versus channel SNR curves, for these the interested user is referred to [207]. However, the corresponding subjective graphical quality and the associated PSNR values are summarised in Figure 15 for the channel SNR range of 5-12 dB, respectively. Due to its low channel capacity requirement the FL-DCC coded signal is readily accommodated by the voice signal during passive speech spurts, when using a voice activity detector (VAD) [40]. Finally, it is noteworthy that the ITU standardised two different chain coding schemes in the T.150 Recommendation for use over conventional low-BER fixed telephone lines. However, for wireless channels the proposed FL-DCC scheme is preferable due to its higher robustness and programmable-rate operation.

We note that an associated multimedia signal manipulation relying on writing tablets is the field of handwriting recognition for both on-line and off-line applications [208]-[213]. Many of the techniques used are based on hidden Markov models (HMMs), which are widely employed in the field of speech recognition. Previous research has shown that HMMs are applicable to both off-line [212] and on-line [208, 210, 213] handwriting recognition problems. The advantage of such statiscal methods is that they can handle variability in the writing process of an individual, but they are also capable of identifying and capturing the individual features of the handwritten characters by taking into account dynamic, pressure dependent features. In many applications the handwritten data is described by the directional writing angle as a

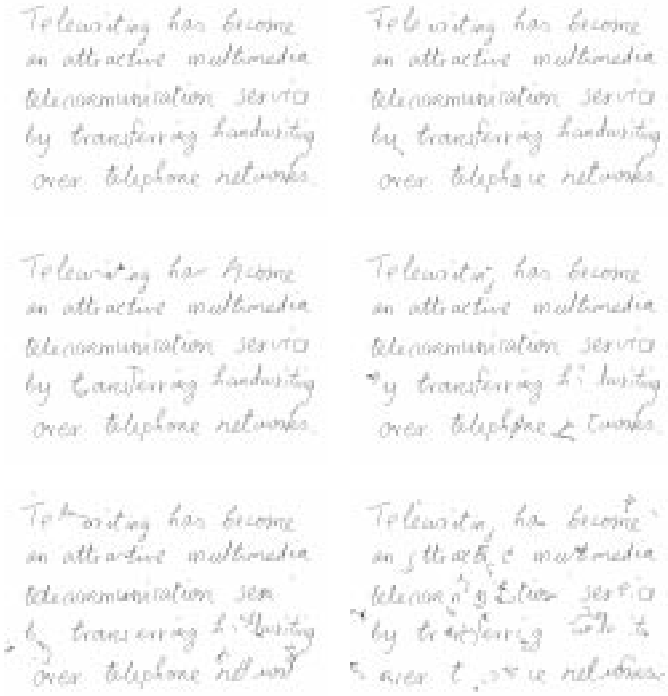


Figure 15. Subjective effects of transmission errors for the $b = 1$ 16QAM, RD, TX3 scheme for PSNR values of (left to right, top to bottom) 49.47, 42.57, 37.42, 32.01, 27.58 and 21.74 dB, respectively. ©IEEE, Yuen and Hanzo [206], 1995

function of the distance along the writing trajectory.

Following the above brief excursion to graphical source compression and signal processing, here we turn our attention to wireless communications aspects, commencing with a review of the frequency re-use concept of cellular systems.

8 CELLUAR COMMUNICATIONS BASICS

8.1 The Cellular Concept

A common feature of the previously mentioned mobile radio systems is that communications take place between a stationary base station (BS) and a number of roaming mobile stations (MSs) or portable stations (PSs) [1]-[9]. The BS's and the MS's transmitter is expected to provide a sufficiently high received signal level for the far-end receivers in order to maintain the required communications integrity. This is usually ensured by power control. The geographical area in which this condition is satisfied is termed as a traffic cell, which typically has an irregular shape, depending on the prevailing propagation environment determined by terrain and architectural features as well as the local paraphernalia. In theoretical sudies often a simple hexagonal cell structure is favoured for its simplicity, where the BSs are located at the centres of the cells.

In an ideal situation the total bandwidth available to a specific mobile radio system could be allocated within

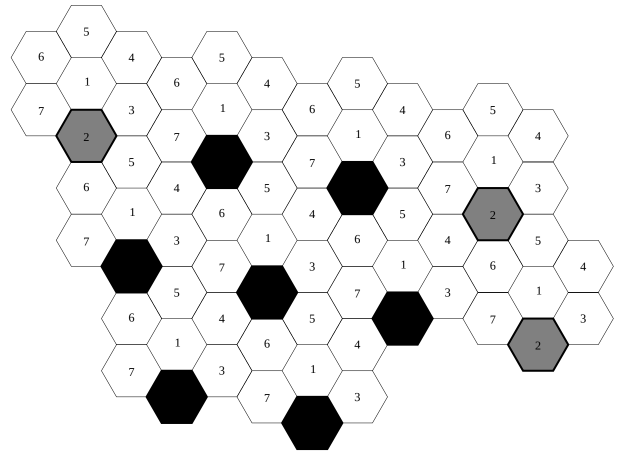


Figure 16. Hexagonal cells and seven-cell clusters

each cell, assuming that there is no energy spilt in the adjacent cell's coverage area. However, since wave propagation cannot be shielded at the cell boundary, PSs near the cell edge would experience approximately the same signal energy within their channel bandwidth from at least two BSs. This phenomenon is called **co-channel interference**. A remedy to this problem is to devide the total bandwidth B_{total} in frequency slots of $B_{cell} = B_{total}/N$, and assign a mutually exclusive reduced bandwidth of B_{cell} to each traffic cell within a so-called **cluster** of N cells, as demonstrated in Figure 16 for $N = 7$. The seven-cell clusters are then tesselated in order to provide contiguous radio coverage. Observe from the figure that the phenomenon of **co-channel interference** between the black co-channel cells having an identical frequency set is not eliminated, but due the the increased co-channel BS distance or **frequency re-use distance** the interference is significantly reduced. Note also that in analytical and simulation-based interference studies the 'second tier' of interfering cells, which are hatched, is typically neglected.

A consequence of the above cellular concept is however that the total number of MSs that can be supported simultaneously over a unit area is now reduced by a factor of N. This is because assuming a simple frequency division multiple access (FDMA) scheme, where each MS is assigned a radio frequency carrier and a user bandwidth of B_{user} now only $M = B_{cell}/B_{user}$ number of MS can be serviced, rather than $N \cdot M = B_{total}/B_{user}$. This problem can be circumvented by making the clusters as small as the original cells, which is achieved by reducing the transmitted power. In fact, further reduction of the cell-size has the advantage of serving more and more users, while requiring a reduced transmitted signal power and hence light-weight batteries. A further favourable effect is that the smaller the traffic cell, the more benign the propagation environment due to the presence of a dominant line-of-sight (LOS) propagation path and the mitigated effects of the multi-path propagation. These arguments lead to

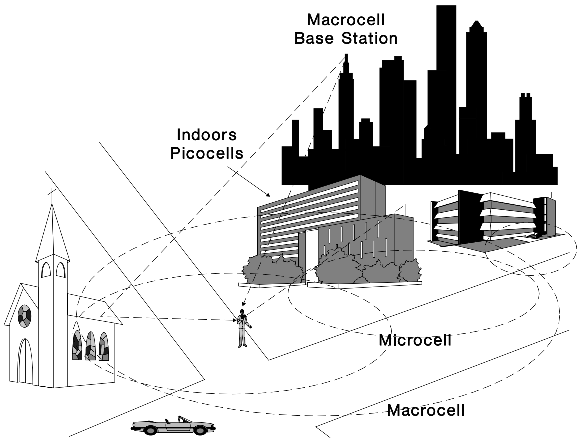


Figure 17. Various traffic cells

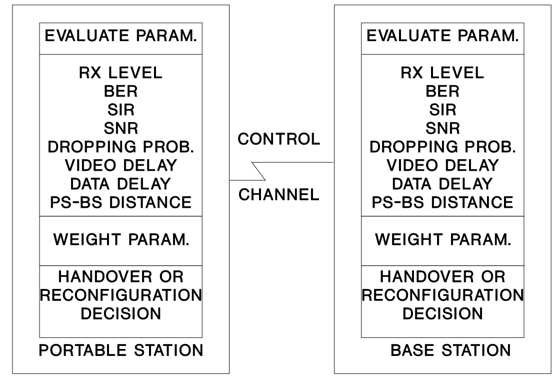


Figure 18. Handover Control Parameters

the concept of **micro- and pico-cells** [20]-[22], which are often confined to the size of a railway station or airport terminal and an office, respectively. Cluster sizes smaller than $N = 7$ are often used in practice in order to enhance the system's spectral efficiency, but such schemes require modulation arrangements that are resilient against the increased co-channel interference. These different cells will have to co-exist in practical systems, where for example an 'over-sailing' macrocell can provide an emergency handover capability for the microcells, when the MS roams in a propagation blind-spot but cannot hand over to another microcells, since in the target microcell no traffic channels are available.

The various cell scenarios are exemplified by Figure 17, where the conventional macro-cell BS is allocated to a high tower, illuminating a large area but providing a rather hostile propagation cell, since often there is no line-of-sight (LOS) path between the BS and PS. Hence the communications are more prone to fading than in the stylised microcell illuminated by the antenna at the top of the lower buildings. The smaller microcells typically use lower transmit power and channel most of the energy in the street canyon, which mitigates the signal's variability and hence has more benign fading. Furthermore, microcells also reduce the signal's dispersion due to pathloss differences. Lastly, indoors picocells provide typically even better channels and tend to mitigate co-channel interferences due to partitions and ceilings.

The previously mentioned hand-over process is crucial as regards to the perceived grade of service (GOS) and a wide range of different complexity techniques have been proposed for example by Tekinay, Jabbari and Pollini in References [23, 24] for the various existing and future systems, some of which are summarised in Figure 18. Explicitly, the BS and PS keep compiling the statistics of a range of communications quality parameters shown in the Figure and weight them according to the prevalent optimisation criterion, before a handover or mode of operation reconfiguration command is issued. Although the PS plays an active role in monitoring the various parameters, these are typically reported to the BS, which carries out the required decisions.

A further important issue associated with the cellular concept and cellular planning is **dynamic channel allocation** (DCA), where a variety of algorithms can be invoked to support the system's operation [25]-[29] in order to mitigate the effects of co-channel interference (CCI) and to maximise the number of traffic channels supported. These techniques are typically invoked in cordeless telephone systems, such as DECT and CT2 [12]. The basic concept of DCA is fairly plausible, since the MS scans the physical channels in order to identify the specific channel exhibiting the lowest signal level, before camping on the one, which was deemed to inflict the lowest level of co-channel interference. Chuang et al [25]-[27] documented the performance of a variety of DCA algorithms, arguing that these techniques under certain conditions can converge to a local minimum of the total interference averaged over the network. In closing we note that there exists a different dynamic channel allocation philosophy, which was proposed by Bernhardt [28, 29], advocating the employment of the worst possible channel that satisfies a minimum signal-to-intereference (SIR) condition. This is equivalent to invoking the worst 'just tolerable' physical channel for communications, which intuitively allows a more compact frequency re-use pattern to be employed, but naturally requires a robust modulation scheme. The overall benefit is an expected higher number of accommodated users.

The family of DCA algorithms is closely related to a range of multiple access schemes, hence we consider multiple access next.

8.2 Multiple Access

The physical channel, which the MS and BS use for their communications can be manifested by a given frequency slot, assigned to the MS for the entire durations of a call. This was the case in most first-generation Frequency Division Multiple Access (FDMA) mobile radio systems. In the second-generation systems, such as the Pan-European GSM [330], the American IS-54 [41] and the Japanese PDC system [42], Time Division Multiple Access (TDMA) was proposed, assigning the whole bandwidth of a TDMA carrier to a MS for a fraction of the time, ie for the duration of a time-slot. The American IS-95 CDMA system [43] uses all the system bandwidth all the time for all users, communicating with orthogonal so-called signature codes. However, these systems employ so-called contentionless bandwidth allocation, where the physical channel is not exploited to its full capacity due to being assigned to users also during their passive speech spurts, when they are listening or thinking, etc.

By contrast, **statistical multiplexing** schemes surrender the physical channel during passive speech spurts, when the channel is not actively used by the MS. This often leads to substantially increased user numbers being supported by the system. A range of multiple access (MAC) protocols have been advocated in the literature [218]-[228], most of wich were featured in a recent excellent overview by Li and Qiu in Reference [32].

Packet reservation multiple access (PRMA) is a statistical multiplexing method for conveying speech signals via time division multiple access (TDMA) systems, which was proposed by Goodman[218] and Wei [220]. A range of various PRMA-assisted CT systems were proposed in [229]-[233]. The operation of PRMA is based on the voice activity detector (VAD) being able to reliably detect inactive speech segments [330]. Inactive users' TDMA time slots are allocated to other users, who become active. The users, who are just becoming active, have to contend for the available time slots with a certain permission probability P_p, which is an important PRMA parameter to be augmented at a later stage.

Previously colliding users contend for the next available time-slot with a less than unity permission probability P_p, in order to prevent them from consistently colliding in their further attempts to attain reservation. If more than one user is contending for a free slot, neither of them will be granted it. If, however, only one user requires the time slot, he can reserve it for future use, until he becomes inactive. Under heavily loaded network conditions, when many users are contending for a reservation, a speech packet might have to contend for a number of consecutive slots. When the contention delay exceeds a latency of about 30 ms, the contending speech packet of typically 20 ms duration must be dropped. The probability of packet dropping must be kept below 1%, a value inflicting minimal degradation in terms of perceived speech quality.

Suffice to say here that in order to find the optimum permission probability P_p, the number of users supported at less than 1% packet dropping must be determined for various P_d values and the curve's maximum has to be identified. The rule of thumb is that when a system can support twice the number of slots in comparison to another, the corresponding P_p value must be halved, in order to maintain the desirable contention rate for each slot. When P_p is high, too vigorous contentions are encouraged, resulting in unacceptably high collision rates. By contrast, too low a P_p value does not exploit the system's full teletraffic capacity due to a modest statistical multiplexing gain.

The perfomance potential of PRMA was analysed using the so-called equilibrium point analysis technique by Nanda, Goodman and Timor[221]. The underlying assumption of the equilibrium point analysis is that the PRMA system can be characterised by a Markov model and the number of users entering a given Markov state is identical to the number of users leaving it. The above PRMA technique was refined by Dunlop et al. [222], where the authors have restricted contentions to so-called contention mini-slots, thereby mitigating the effects of packet collisions. his scheme was termed PRMA++. PRMA was also suggested by Eastwood et al [232] for multiplexing multimedia users' transmission packets for transmission to the BS.

A technique referred to as Dynamic Time Division Multiple Access (D-TDMA) was advocated for integrated voice and data communication also by Dunlop et al [223], where so-called 'request minislots' are employed for channel acquisition and the 'information slots' are assigned in a second phase. Another efficient MAC protocol was suggested by Amitay and Nanda, which the authors referred to as Resource Auction Multiple Access (RAMA) [224], where only one user is granted access to the system at any instant, hence preventing collisions. Dynamic Reservation Multiple Access (DRMA) was introduced by Li and Qiu [219], while Brecht et al suggested the employment of Statistical Packet Assignment Multiple Access (SPAMA) [228] for supporting variable-rate multi-media traffic. The statistical nature of the proposed centralized slot assignment scheme facilitated an accurate matching of bitrate requirements for different multimedia services with a minimal amount of signalling, while maintaining a throughput of up to 93% at the cost of a low MAC delay.

A further alternative to support similar multi-rate multimedia users was also proposed by Brecht et al, which was termed Multi-frame PRMA (MF-PRMA) [226, 329]. Here we emphasize that most of the above statistical multiplexer schemes function also as multimedia packet multiplexers, supporting the delivery of multirate, multimedia traffic on a demand basis, giving cognizance to the different so-called stability constraints of speech video and data sources. Speech and interactive video are delay-sensitive, while data and distributive video is not. However, data and run-length coded variable-rate video are extremely error-sensitive, hence requiring higher integrity than speech and fixed-rate non-runlength coded video. Some of these aspects were also addressed in References [232, 228, 226, 227].

The first-generation PLMR systems were designed for low traffic density and the typical cell-radius was often of the order of tens of miles. Even the second-generation GSM system [40] was contrived to be able to cope with the hostile large-cell environment of 35 km radius rural cells. Hence it incorporated sophisticated and power-hungry signal processing, in order to be able to combat a wide range of channel impairments, associated with the hostile large-cell PLMR environment. The less robust DAMPS [41] and the second-generation Japanese digital mobile radio (JDMR) systems [42] reflect the more recent trend of moving towards small cells, exhibiting benign propagation characteristics, a tendency also adopted by the CT systems CT2 and DECT [12]. These propagation aspects are well-understood in the wireless communications community, but the above cell-size dependent propagation factors can be augmented by referring to the relevant literature [1, 2, 3, 216, 217]. Hence here we refrain from detailing deeper aspects of the wireless propagation environment.

Having covered some of the wireless communications basics we are now equipped to consider the flexible system architecture of a mobile multi-media communicator of the next generation.

9 FLEXIBLE MULTIMEDIA SYSTEM SCHEMATIC

The schematic of a flexible, toolbox-based multimedia PS is portrayed in Figure 19 [152]. The pivotal implementational point of such a multi-media PS is that of finding the best compromise amongst a number of contradicting design factors, such as power consumption, robustness against transmission errors, spectral efficiency, audio/video quality and so forth [229]. In this contribution we will address a few of these issues, mainly concentrating on the modulation and systems aspects of the proposed PS depicted in Figure 19. The time-variant optimisation criteria of a flexible multi-media system can only be met by an adaptive scheme, comprising the firmware of a suite of system components and loading that combination of speech codecs, video codecs, embedded channel codecs, voice activity detector (VAD) and modems, which fulfills the prevalent one [68]. A few examples are maximising the teletraffic carried or the robustness against channel errors, while in other cases minimisation of the bandwidth occupancy, the blocking probability or the power consumption is of prime concern.

Focussing our attention on the speech and video links displayed in Figure 19, the voice activity detector (VAD) [40] is deployed to control the packet reservation multiple access (PRMA) slot allocator [68, 229]. A further task of the 'PRMA Slot Allocator' is to multiplex digital source data from facsimile and other data terminals with the speech as well as graphics and other video signals to be transmitted.

Again, PRMA is a relative of slotted ALOHA contrived for conveying speech signals on a flexible de-

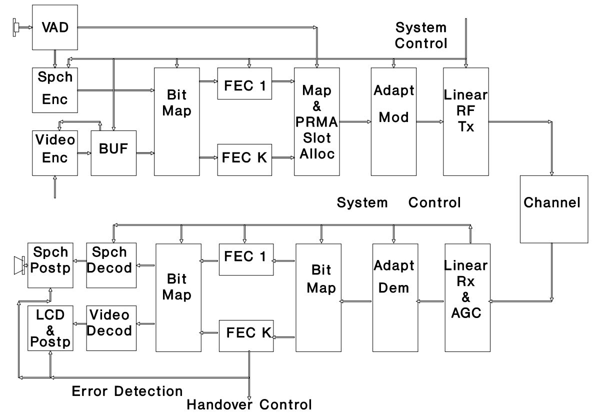


Figure 19. Multimedia UMTS Communicator Schematic

mand basis via time division multiple access (TDMA) systems. PRMA was documented in a series of excellent treatises by Goodman et.al [218]-[221], while various PRMA-assisted transceiver schemes were proposed in references [229]-[233].

The voice activity detector (VAD) [40, 330, 231] queues the active speech spurts to contend for an uplink TDMA time-slot for transmission to the BS. Inactive users' TDMA time slots are offered by the BS to other users, who become active and are allowed to contend for the un-used time slots with a less than unity permission probability. This measure prevents previously colliding users from consistently keep colliding in their further attempts to attain a time-slot reservation. If several users contend for an available slot, neither of them will be granted it, while if only one user requires the time slot, he can reserve it for its future communications. When many users are contending for a reservation, the collision probability is increased and hence a speech packet might have to contend for a number of consecutive slots, until its maximum contention delay of typically 32 ms expires. In this case the speech packet must be dropped, but the packet dropping probability must be kept below 1%, a value inflicting minimal degradation in perceivable speech quality in contemporary speech codecs.

Control traffic and system information is carried by packet headers added to the composite signal by the 'Bit Mapper' before K-class source sensitivity-matched forward error correction coding (FEC) takes place. Observe that the 'Video Encoder' supplies its bits to an adaptive buffer (BUF) having a feed-back loop. If the PRMA video packet delay becomes too high or the buffer fullness exceeds a certain threshold, the video encoder is instructed to lower its bit rate, implying a concomitant dropping of the image quality.

The Bit Mapper assigns the most significant source coded bits (MSB) to the input of the strongest FEC codec, FEC K, while the least significant bits (LSB) are protected by the weakest one, FEC 1. K-class FEC coding is used after mapping the speech and video bits to their appropriate bit protection classes, which ensures source sensitivity-matched transmission. 'Adaptive Modulation' is deployed [68, 229], with the number of modulation levels, the FEC coding power and the speech/video source coding algorithm adjusted by the 'System Control' according to the dominant propagation conditions, bandwidth and power efficiency requirements, channel blocking probability or PRMA packet dropping probability. If the communications quality or the prevalent system optimisation criterion cannot be improved by adaptive transceiver re-configuration, the serving BS will hand the PS over to another BS providing a better grade of service.

One of the most important and reliable parameters used to control these algorithms is the 'Error Detection' flag of the FEC decoder of the most significant bit (MSB) class of speech and video bits, namely FEC K. This flag can also be invoked to control the speech and video 'Postprocessing' algorithms. The adaptive modulator transmits the user bursts from the PS to the BS using the specific PRMA slot allocated by the BS for the PS's speech, data or video information via the linear radio frequency (RF) transmitter (Tx). Although the linear RF transmitter has a low power efficiency, its power consumption is less critical due to the low transmitted power requirement of the multi-media PCN than that of the digital signal processing (DSP) hardware.

The receiver structure essentially follows that of the transmitter. After linear class-A amplification and automatic gain control (AGC) the 'System Control' information characterising the type of modulation and the number of modulation levels must be extracted from the received signal, before demodulation can take place. This information also controls the various internal bit mapping algorithms and invokes the appropriate speech and video decoding as well as FEC decoding procedures. After 'Adaptive Demodulation' at the BS the source bits are mapped back to their original bit protection classes and FEC decoded. As mentioned, the error detection flag of the strongest FEC decoder, FEC K, is used to control handovers or speech and video postprocessing. The FEC decoded speech and video bits are finally source decoded and the recovered speech arrives at the earpiece, while the video information is displayed on a flat liquid crystal display (LCD).

The system control algorithms of the re-configurable mobile multimedia communicator will dynamically evolve over the years. PSs of widely varying complexity will coexist, with newer ones providing backward compatibility with existing ones, while offering more intelligent new services and more convenient features. After this rudimentary system-level introduction let us now focus our attention on a range of modern transceiver techniques.

10 MODERN TRANSCEIVER TECHNIQUES

In recent years a powerful architecture referred to as software radio [16] was advocated by many researchers, essentially employing a flexible baseband signal processing 'toolbox' of speech-, video- and channel codecs, modulation and 'user signature' functions plus a sufficiently wideband, linear radio frequency stage, which is interfaced to the baseband section using a high-speed digital to analogue converter. The software radio system architecture is capable of supporting intelligent multimode and multistandard operation, although many of the research issues are in their infancy at the time of writing. Smart antennae and co-channel interference reduction adaptive beam-forming methods inherited from radar and sonar researchers were characterised for example by Litva and Lo [53], Baier, Blanz and Schmalenberger [56] and were also advocated by Kohno [55] in the context of intelligent, so-called 'space-time processing' receivers.

Furthermore, the sophisticated 'per-survivor processing' detection algorithms proposed by Polydoros and Chugg [57] and the receiver techniques portrayed in the monograph by Meyer, Moeneclaey and Fechtel [8] are expected to improve the performance of the next generation of receivers. Verdu [58] contrived the so-called optimum multiuser detector for CDMA transceivers, albeit its complexity is exponentially proportional to the number of users detected by the scheme, which may become excessive. These multiuser detection techniques exploit the apriori information concerning the user signature or spreading sequences and the channel estimates derived, in order to remove the CDMA-specific multiuser interference and hence to approach the 'single-user' Shannonian performance. The more practical sub-optimum multiuser receivers are often classified as interference cancellation (IC) or suppression and joint detection (JD) arrangements [5].

Despite accurate power control some user signals arrive at the receiver at a higher power and hence IC attempts initially to detect the highest power user's signal from the superpostion of all users' signal. Upon error-free detection the strongest user's signal can be re-modulated and deducted from the received multiuser signal, a procedure that can be invoked for the next strongest users in turn, until all users are detected. The philosophy of multi-user detection is slightly different, since it is based on recognizing that the multi-user interference (MUI) is essentially similar to inter-symbol interference (ISI) and hence the classic equalization techniques [68] developed for conventional ISI cancellation in dispersive channels can be readily modified for this application. Explicitly, the MUI of CDMA, which is generated by a number of users, can be interpreted as conventional ISI inflicted by a multipath channel, having the same number of paths, as the number of interfering CDMA users. Joint detection research was spearheaded in recent years by Jung and Blanz [59], by A.Klein, G.K.Kaleh and P.W.Baier [60] and a number of other researchers at the University of Kaiserslautern in Germany, but the impressive individual contributions in this field are too numerous to mention [61]-[65]. Let us now consider the issues affecting the choice of the appropriate modulation scheme.

11 MODULATION ISSUES

11.1 Choice of Modulation

In some of the European mobile systems, such as the Pan-European GSM system [40, 330, 214] or the Digital European Cordless Telecommunications (DECT) scheme constant envelope partial response Gaussian Minimum Shift Keying (GMSK) [9, 69] is employed. Its main advantage is that it ignores any fading-induced amplitude fluctuation present in the received signal and hence facilitates the utilisation of power-efficient non-linear class-C amplification. In third generation personal communication systems however benign pico- and micro-cells will be employed, where low transmitted power and low signal dispersion are characteristic. Hence the employment of more bandwidth efficient multilevel modulation schemes becomes realistic. In fact the American and Japanese second generation digital systems, namely IS-54 and PDC, have already opted for 2 bits/symbol multi-level modulation [68].

The basic schematic of a modem is shown in Figure 20. If an analogue source signal must be transmitted, the signal is first low-pass filtered and analogue-to-digital converted. The generated digital bitstream is then mapped to complex modulation symbols, such as those seen in Figure 21, which are suitable for transmission over the bandlimited channel. In Figure 21 there are 16 such complex points, represented by 2 distinct amplitude and 8 different phase values, which can be described by 4 bits. This mapping operation is carried out by the MAP block of Figure 20, assigning the appropriate so-called in-phase (I) and quadrature-phase (Q) components to 4 incoming bits. In order to ensure that I and Q components do not change abruptly, which would require an infinite bandwith, the square-root N block carries out the so-called Nyquist-filtering operation in the base-band, before the signal is up-converted to the intermediate frequency (IF) band by the help of two carrier waves, which are in 90 degrees phase-shift, ie they are orthogonal. This orthogonality allows us to transmit the independent I and Q components within the same bandwidth, whithout interfering with eachother. Such so-called multi-level constellations are more prone to channel impairments than binary schemes due to their comparatively low distance between constellation. Hence they are employed typically over benign channels, where the guaranteed higher Shannonian channel capacity can be exploited this way to provide higher bit rates.

Multi-level modulation schemes have been considered in depth in reference [68] and in Chapters 17 and 18 we have shown that the bandwidth efficiency and minimum required signal-to-noise ratio (SNR) and Signal-to-Interference Ratio (SIR) of a modulation scheme in a given frequency re-use structure is dependent on the bit error ratio (BER) targeted. The required Bit Error Rate (BER) in turn is dependent on the robustness of the source codecs used. Furthermore, in indoors scenarios the partitioning walls and floors mitigate the co-channel interference and this facilitates the employment of 16-level Quadrature Amplitude Modulation (16QAM).

The so-called 'maximum minimum distance' square-shaped QAM constellation [234, 68] is optimum for transmissions over Additive White Gaussian (AWGN) channels, simply because the minimum distance amongst the modulation constellation points is as high, as possible. In other words, this constellation has the highest possible average distance amongst its constellation points under the constraint of a given average power, yielding the highest 'noise protection distances', when contaminated by AWGN. Until quite recently QAM developments were focussed on the benign AWGN telephone line and on point-to-point radio applications [235], which led to the definition of the CCITT telephone circuit modem standards V.29-V.33 based on various QAM constellations ranging from uncoded 16-QAM to trellis coded (TC) 128-QAM. In recent years QAM research for hostile fading mobile channels has been motivated by the ever-increasing bandwidth efficiency demand for mobile telephony [236]-[247], although it requires power-inefficient class A or AB linear amplification [248]-[251]. However, the power consumption of the low-efficiency class-A amplifier [250], [251] is less critical than that of the digital speech, image and channel codecs. Out-of-band emissions due to class AB amplifier non-linearities generating adjacent channel interferences can be reduced by some 15-20 dB using the adaptive predistorter proposed by Stapleton et. al. [252]-[254].

When using the square-shaped 16-QAM constellation it is essential to be able to separate the information modulated on to the in-phase (I) and quadrature-phase (Q) carriers by the help of coherent demodulation, invoking the Transparent-tone-in-band (TTIB) principle invented by McGeehan and Bateman [256], [257], [258] or invoking Pilot Symbol Assisted Modulation (PSAM) [255]. Although these methods can eliminate the residual BER, they are significantly more complex to implement than their non-coherently detected differentially coded counterparts. Based on the above arguments and constrained by the high bandwidth efficiency requirement in this treatise we have opted for non-coherently detected 16-QAM.

11.2 Non-Coherent Star 16-QAM

The pivotal point of differentially coded non-coherent QAM demodulation is that of finding a rotationally symmetric QAM constellation, where all constellation points are rotated by the same amount. Such a rotationally sym-

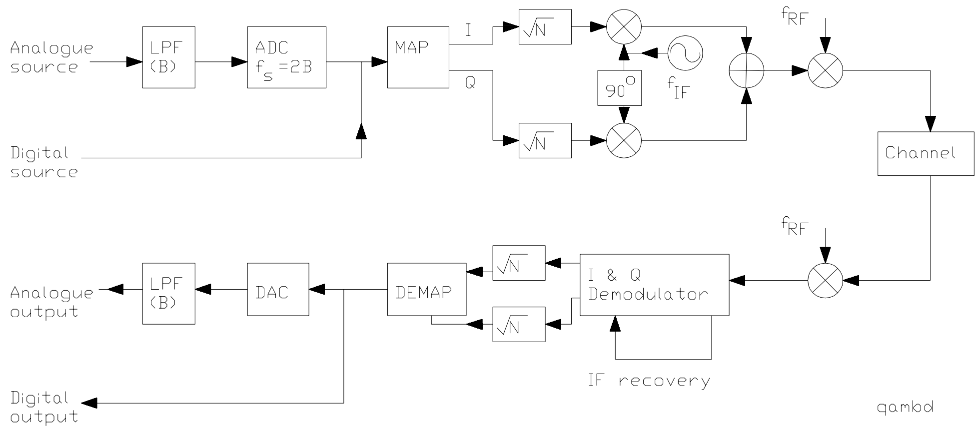


Figure 20. Basic modem schematic ©Webb, Hanzo, 1994, [68]

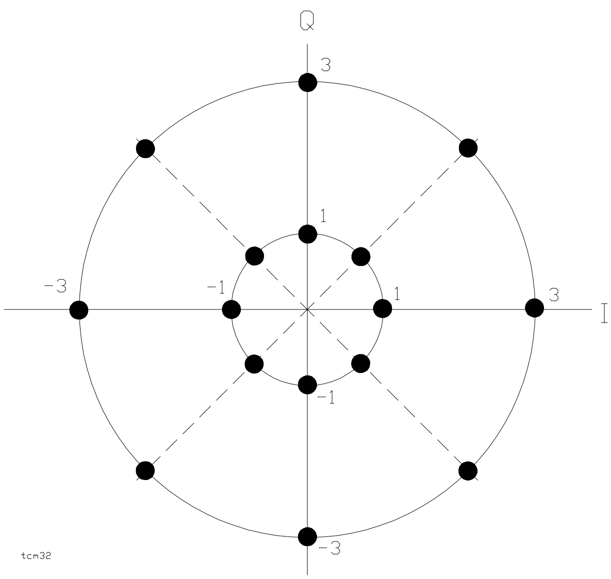


Figure 21. Star QAM constellation

metric 'star-constellation' was proposed in [245], which is shown in Figure 21. A disadvantage of the proposed star 16-QAM (16-StQAM) constellation is its lower average energy. While square 16-QAM had an average phasor energy of $10d^2$, the 16-StQAM halves this value to $5d^2$, where $2d$ is the phasor spacing of the I and Q components. This implies a 3 dB disadvantage over Gaussian channels, but via Rayleigh fading channels this SNR penalty becomes less.

Our differential encoder obeys the following rules. The first bit $b1$ of a four-bit symbol is differentially encoded onto the phasor magnitude, yielding a ring-swap for an input logical one and maintaining the current magnitude, i.e., ring for $b1 = 0$. Bits $(b2, b3, b4)$ are then differentially Gray-coded onto the phasors of the particular ring pin-pointed by $b1$. Accordingly, $(b2, b3, b4) = (0, 0, 0)$ implies no phase change, $(0, 0, 1)$ a change of 45^o, $(0, 1, 1)$ a change of 90^o, etc.

The corresponding non-coherent differential 16-StQAM demodulation is equally straightforward, having decision boundaries at a concentric ring of radius 2 and at phase rotations of $(22.5^o + n.45^o)$ $n = 0 \ldots 7$. Assuming received phasors of P_t and P_{t+1} at consecutive sampling instants of t and $t+1$, respectively, bit $b1$ is inferred by evaluating the condition:

$$|\frac{P_{t+1}}{P_t}| \geq 2. \tag{2}$$

If this condition is met, $b1 = 1$ is assigned, otherwise $b1 = 0$ is demodulated. Bits $(b2, b3, b4)$ are then recovered by computing the phase difference

$$\Delta\Theta = (\Theta_{t+1} - \Theta_t) \pmod{2\pi} \tag{3}$$

and comparing it against the decision boundaries $(22.5^o + n.45^o)$ $n = 0 \ldots 7$. Having decided which rotation interval the received phase difference $\Delta\Theta$ belongs to, Gray-decoding delivers the bits $(b2, b3, b4)$.

From our previous discourse it is plausible that the less dramatic the fading envelope and phase trajectory fluctuation between adjacent signalling instants, the better this differential scheme works. This implies that lower vehicular speeds are preferred by this arrangement, if the signalling rate is fixed. Therefore the modem's performance improves for low pedestrian speeds, when compared to typical vehicular scenarios. Alternatively, for a fixed vehicular speed higher signalling rates are favourable, since the relative amplitude and phase changes introduced by the fading channel between adjacent information symbols are less drastic. For a full treatise on QAM the interested reader is referred to [68]. Let us now highlight the current research aspects of adaptive modems, which can adjust the number of bits per transmitted symbol, in order to adapt to time-variant channel conditions.

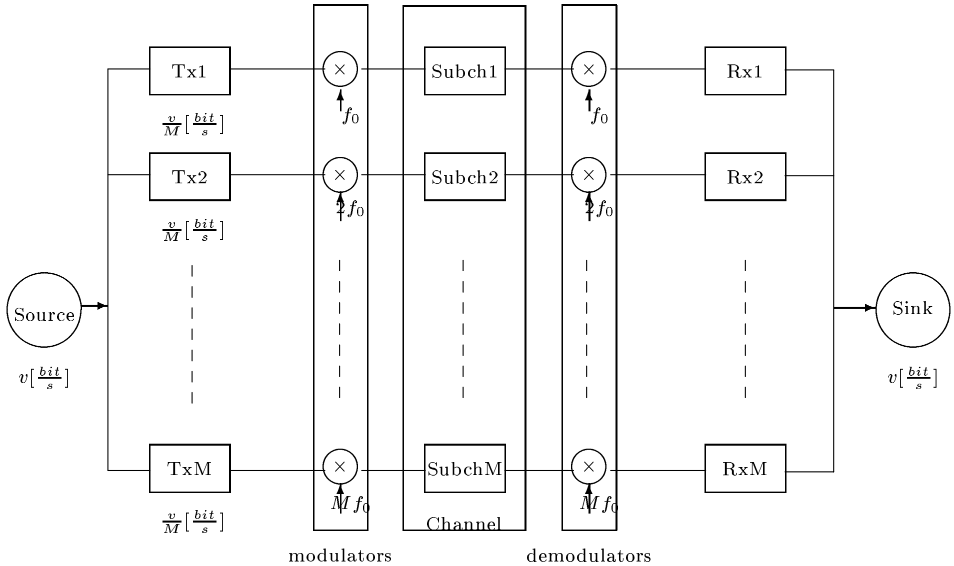


Figure 22. Simplified blockdiagram of the orthogonal parallel modem

11.3 Burst-by-burst adaptive modems

Burst-by-burst adaptive multi-level modulation was first suggested by Steele and Webb in References [259, 260, 68] for slowly-fading wireless pedestrian channels, inspiring intensive further research in recent years [262]-[271], in particular by Kamio, Sampei, Sasaoka, Morinaga, Morimoto, Harada, Okada, Komaki and Otsuki at Osaka University and the Ministry of Post in Japan, [261]-[264], as well as by Goldsmith and Chua [265] at CalTech in the USA or by Pearce, Burr and Tozer [266] in the UK. The proposed schemes provide a means of realising some of the time-variant channel capacity potential of the fading wireless channel [2, 274], invoking a more robust Transmission Scheme (TS) on a burts-by-burst basis, when the channel is of low quality and vice-versa, while maintaining a certain target bit error rate (BER) performance. The most appropriate TS is dependent upon the time-variant instantaneous Signal-to-noise Ratio (SNR) and Signal-to-interference Ratio (SIR). The TS can be chosen according to the following regime [267]:

$$\mathrm{TS} = \begin{cases} \text{No Transmission (Notx)} & \text{if} \quad l_1 > s^2/N \\ \text{BPSK} & \text{if} \quad l_1 \le s^2/N < l_2 \\ \text{QPSK} & \text{if} \quad l_2 \le s^2/N < l_3 \\ \text{Square 16 Point QAM} & \text{if} \quad l_3 \le s^2/N < l_4 \\ \text{Square 64 Point QAM} & \text{if} \quad s^2/N \ge l_4, \end{cases} \qquad (4)$$

where s is the instantaneous signal level, N is the average noise power, and l_1, l_2, l_3 and l_4, are the BER-dependent optimised switching levels. Time Division Duplex (TDD) was proposed, in order to exploit the reciprocity of the channel under high SIR-conditions, which allowed us to estimate the prevalent SNR on a burst-by-burst basis [268]. The reciprocity of the up- and down-link channel conditions in the TDD frame is best approximated, if the corresponding TDD slots are adjacent.

Switching levels(dB)	l_1	l_2	l_3	l_4
Mean-Speech (1%)	3.31	6.48	11.61	17.64
Mean-BER Data (0.01%)	7.98	10.42	16.76	26.33

Table 3. Switching levels for speech and computer data systems through a Rayleigh channel, shown in instantaneous channel SNR (dB) to achieve Mean BERs of 1×10^{-2} and 1×10^{-4}, respectively

In Reference [267] the analytical upper-bound performance of such a scheme was characterised over slow Rayleigh-fading channels, while in [270] an un-equal protection phasor constellation for signalling the current TS was proposed. The problem of appropriate power assignment was discussed for example in [263, 265].

In Reference [269] a combined BER- and Bits per Symbol (BPS) based optimisation cost-function was defined and minimised, in order to find the required TS switching levels for maintaining average target BERs of 1×10^{-2} and 1×10^{-4}, irrespective of the instantaneous channel SNR. These BER values can then be further mitigated by forward error correction coding and in case of the lower BER scheme can be rendered virtually error-free. The former scheme was referred to as the speech TS, while to the latter as the adaptive data TS. The optimised TS switching levels l_1, l_2, l_3 and l_4 are summarised in Table 3 [267]. The average BPS performance B of this adaptive modem was derived for a Rayleigh fading channel in Reference [267], which can be written as:

$$
\begin{aligned}
B &= 1 \cdot \int_{l_1}^{l_2} F(s,S)\,ds + 2 \cdot \int_{l_2}^{l_3} F(s,S)\,ds \\
&+ 4 \cdot \int_{l_3}^{l_4} F(s,S)\,ds + 6 \cdot \int_{l_4}^{\infty} F(s,S)\,ds,
\end{aligned} \tag{5}
$$

where $F(s,S)$ is the PDF of the Rayleigh channel, S is the average power and the integrals characterise the received signal level domains, where the 1, 2, 4 and 6 bits/symb TSs of Equation 4 are used. In References [271, 272] the latency performance of these schemes was quantified and frequency hopping as well as statistical multiplexing were proposed to mitigate its latency and buffer requirements.

11.4 Equalisation Techniques

The performance of wideband wireless channel equalizers was studied by a large cohort of researchers, such as Narayanan and Cimini [275], Wu and Aghvami [276] as well as by Gu and Le-Ngoc [277]. In order to achieve fast equalizer coefficient convergence, these contributions typically invoked the Kalman algorithm [68] and its diverse incarnations, such as the Square Root Kalman scheme, although Clark and Harun [278] argued that there were only marginal performance differences between the Kalman Algorithm and the Least Mean Squared [68] (LMS) algorithm in typical practical situations. Maximum likelihood sequence estimator (MLSE) type receivers typically outperform Decision Feedback Equalizers (DFE) at the cost of higher complexity. A range of hybrid compromise-schemes were proposed for example by Wu and Aghvami [276] as well as by Gu and Le-Ngoc [277].

Let us now turn our attention to a transmission scheme, which is particularly suitable for high-rate transmission over frequency selective fading channels, although it refrains from employing the above equalisation techniques.

11.5 Orthogonal Frequency Division Modulation

In this Section we briefly introduce Frequency Division Multiplexing (FDM), also referred to as Orthogonal Multiplexing (OMPX), as a means of dealing with the problems of frequency selective fading encountered, when transmitting over a high-rate wideband radio channel. The fundamental principle of orthogonal multiplexing originates from Chang [279], and over the years a number of researchers have investigated this technique [280]–[291]. Despite its conceptual elegance, until recently its employment has been mostly limited to military applications due to implementational difficulties. However, it has recently been adopted as the new European digital audio broadcasting (DAB) standard, it is also a strong candidate for digital terrestrial television broadcast (DTTB) and for a range of other high-rate applications, such as 155 Mbps wireless Asynchronous Transfer Mode (ATM) local area networks. These wide-ranging applications underline its significance as an alternative technique to conventional channel equalisation in order to combat signal dispersion [292]–[296].

In the FDM scheme of Figure 22 the serial data stream of a traffic channel is passed through a serial-to-parallel convertor, which splits the data into a number of parallel sub-channels. The data in each sub-channel are applied to a modulator, such that for M channels there are M modulators whose carrier frequencies are f_0, f_1, $...f_M$. The difference between adjacent channels is Δf and the overall bandwidth W of the N modulated carriers is $M\Delta f$.

These M modulated carriers are then combined to give an FDM signal. We may view the serial-to-parallel convertor, as applying every Mth symbol to a modulator. This has the effect of interleaving the symbols into each modulator, hence symbols S_0, S_M, S_{2M},.... are applied to the modulator whose carrier frequency is f_1. At the receiver the received FDM signal is demultiplexed into M frequency bands, and the M modulated signals are demodulated. The baseband signals are then recombined using a parallel-to-serial convertor.

The main advantage of the above FDM concept is that because the symbol period has been increased, the channel delay spread is a significantly shorter fraction of a symbol period than in the serial system, potentially rendering the system less sensitive to ISI than the conventional serial system. In other words, in the low-rate subchannels the signal is no longer subject to frequency-selective fading, hence no channel equalisation is necessary.

A disadvantage of the FDM approach shown in Figure 22 is the increased complexity over the conventional system caused by employing M modulators and filters at the transmitter and M demodulators and filters at the receiver. It can be shown that this complexity can be reduced by the use of the discrete Fourier transform (DFT), typically implemented as a Fast Fourier Transform (FFT) [68]. The subchannel modems can use almost any modulation scheme, and 4- or 16-level QAM is an attractive choice in many situations.

The FFT-based QAM/FDM modem's schematic is portrayed in Figure 23. The bits provided by the source are serial/parallel converted in order to form the n-level Gray coded symbols, M of which are collected in TX buffer 1, while the contents of TX buffer 2 are being transformed by the IFFT in order to form the time-domain modulated signal. The digital-to-analogue (D-A) converted, low-pass filtered modulated signal is then transmitted via the channel and its received samples are collected in RX buffer 1, while the contents of RX buffer 2 are being transformed to derive the demodulated signal. The twin buffers are alternately filled with data to allow for the finite FFT demodulation time. Before the data is Gray coded and passed to the data sink, it can be equalised by a low complexity method, if there some dispersion within the narrow subbands. For a deeper tutorial exposure the interested reader is referred to Chapter 15 of reference [68].

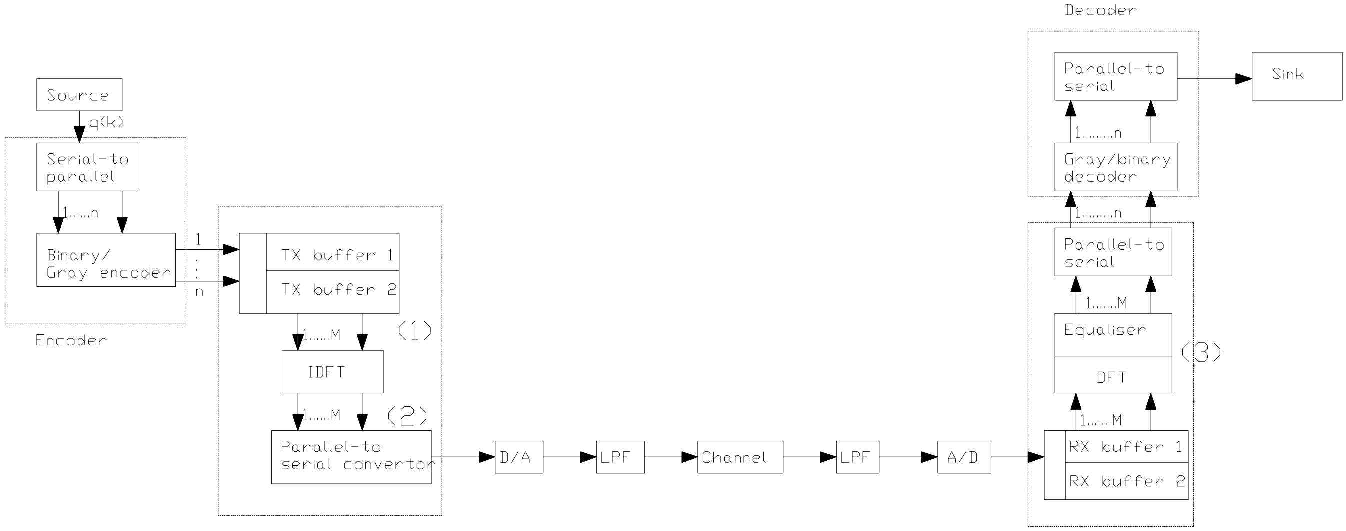


Figure 23. FFT-based OFDM modem schematic ©Webb, Hanzo, 1994, [68]

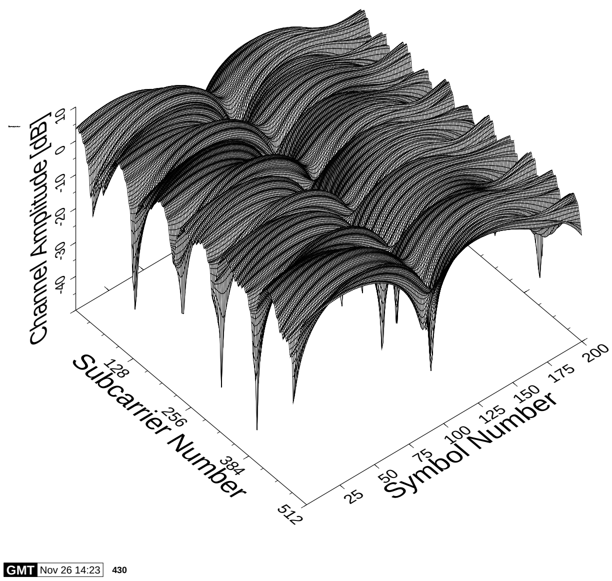


Figure 24. Frequency response in the Bandwidth of $M \times f_0$ for the 512-channel OFDM system at 155 Mbps ©IEEE, Cherriman, Keller, Hanzo [344]

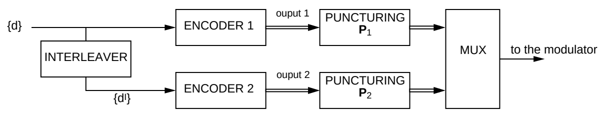
Before concluding this Section, we describe a typical channel characteristic of wireless local area networks (WLAN) transmitting at a rate of 155 Mbps, where the above OFDM scheme can be advantageously employed. The 155 Mbps rate is used in Asynchronous Transfer Mode (ATM) systems. We assumed an indoors airport terminal or warehouse environment of dimensions 100m $\times$ 100m and a 7-path channel, corresponding to the four walls, ceiling and floor plus the line-of-sight (LOS) path. The LOS path and the two reflections from floor and ceiling were combined into one single path in the impulse response. The worst-case impulse response associated with the highest path length and delay spread is experienced in the farthest corners of the hall, which was determined using inverse second power law attenuation and the speed of light for the computation of the path delays. The corresponding frequency response was plotted in Figure 24 for a 512-channel system, as a function of both OFDM frame index and suchannel index. Observe the very hostile frequency selective fading in the Figure, which is efficiently combated by the OFDM modem, since for the narrow subchannels the channel can be considered more or less flat fading. Again, the residual fading can be equalised using a simple pilot-assisted equaliser. In closing we note that a variety of OFDM-related aspects were investigated in References [179]-[301].

After this discussion on modulation techniques let us briefly consider ways of reducing the BER using FEC techniques.

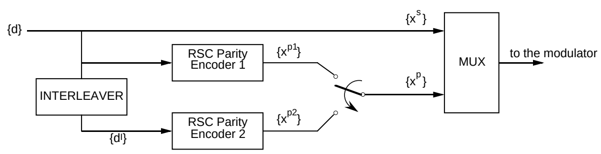
12 CHANNEL CODING

The highest coding gain over AWGN channels is achieved using trellis-coded modulation (TCM) [305], rather than consecutive FEC coding and modulation. Recently similarly attractive TCM schemes have been proposed for fading mobile channels [306], [307]. In order to provide TCM schemes having un-equal source-sensitivity matched error protection similarly to our approach in references [246]-[247], [145] Wei [309] suggested a range of non-uniformly spaced phasor constellations. Another method proposed by Wei [309] was to deploy a number of independent TCM schemes having different grade of protection and multiplex the sequences for transmission.

Both convolutional and block codes have been successfully used to combat the bursty channel errors [79]. Cox, Hagenauer et al. [302] proposed rate compatible punctured convolutional (RCPC) codecs [303], in order to provide bit sensitivity matched FEC protection for a subband speech codec using a R=1/2 rate so-called mother code, where some of the encoded output bits can be obliterated or punctured from the bit-stream. This then allows the designer to create a variety of different-rate bit-protection classes, while using the same decoder and protecting the more error-sensitive bits by a stronger low-rate code and the more robust source-coded bits by a higher-rate, less powerful FEC code.

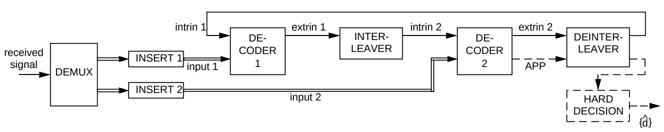


(a) General Turbo Encoder

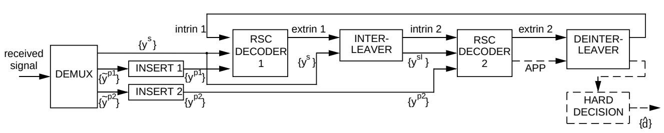


(b) Original RSC Turbo Encoder

Figure 25. General (a) and originally proposed rate 1/2 RSC (b) turbo code encoder structures©[327] Didascalou, 1996



(a) General Turbo Decoder



(b) RSC Turbo Decoder

Figure 26. General (a) and RSC (b) turbo code decoder structures©[327] Didascalou, 1996

12.1 Turbo Coding

In recent years significant advances have been made towards the Shannonian performance predictions with the introduction of the so-called **turbo codes** [310] and using so-called iterative decoding techniques, which will be briefly highlighted below. Turbo coding was proposed by Berrou, Glavieux and Thitimajshima [310], where the information sequence is encoded twice, using so-called Recursive Systematic Convolutional (RSC) encoders. As seen in Figure 25, the second encoding takes place after a pseudo-random interleaving of the original information sequence, in order to render the two encoded data sequences approximately statistically independent of each other.

In most turbo coding schemes a pair of half rate Recursive Systematic Convolutional (RSC) encoders are used, where each RSC encoder generates a so-called systematically encoded output stream, which is equivalent to the original information sequence, as well as a stream of parity information. The two parity sequences are then punctured before being transmitted along with the original information sequence to the decoder. This puncturing of the parity information allows a wide range of coding rates to be realised, and often half the parity information from each encoder is sent. Along with the original data sequence this results in an overall coding rate of 1/2.

As portrayed in Figure 26, at the receiver two RSC decoders are used. Special decoding algorithms must be invoked, which accept so-called soft inputs and give soft outputs for the decoded sequence, rather than binary bits. These soft inputs and outputs provide not only an indication of whether a particular bit was a 0 or a 1, but also a so-called likelihood ratio, which quantifies the probability that the bit has been correctly decoded. The turbo decoder operates iteratively. In the first iteration the first RSC decoder provides a soft output giving an estimation of the original data sequence based on the soft channel inputs alone. It also provides a so-called *extrinsic* output. The extrinsic output for a given bit is based not on the channel input for that bit, but on the information carried by surrounding bits and the constraints imposed by the code being used. This extrinsic output from the first decoder is used by the second RSC decoder as *apriori information*, and this information together with the channel inputs is used by the second RSC decoder to give its soft output and extrinsic information. In the second iteration the extrinsic information from the second decoder in the first iteration is used as the apriori information for the first decoder, and using this apriori information the decoder has an increased probability of decoding more bits correctly than it did in the first iteration. This cycle continues, where at each iteration both RSC decoders producing a soft output and extrinsic information based on the channel inputs and apriori information obtained from the extrinsic information provided by the previous decoder. After each iteration the Bit Error Rate (BER) in the decoded sequence drops, but the improvements obtained with each iteration falls as the number iterations increases so that for complexity reasons usually only 8 or 16 iterations are used.

If a bit appears in a rather corrupted section of the first encoded sequence, due to the independent parity information inherent in the second encoded sequence the second decoder may be able to assist the first one to correctly decode the original information. The independent information delivered by the first decoder is likely to improve the performance of the second decoder, the output of which can now be again invoked by the first decoder. This iterative decoding process can be continued, alternating between decoding in the trellis of the first decoder and that of the second one in order to achieve near-Shannonian performance.

Hagenauer and Hoeher proposed to use the so-called soft-output Viterbi [311] algorithm for the decoding of Turbo codes in Reference [312] , while Hagenauer, Offer and Papke [313] investigated also the feasibility of employing block codes as constituent codes, although most research is carried out in the context of convolutional codes. Robertson et al [315] and Jung [317] investigated various turbo decoders, while Jung et al [318] and Barbulescu et al [319] studied various interleaving aspects of turbo coding. A range of other associated turbo-coding issues were considered in References [320]-[323], while Breiling et al recently proposed an optimum non-iterative turbo decoder [324, 325, 326, 328], finding the maximum likelihood decoded information in single non-iterative decision step.

Following our rudimentary description of some of the components of wireless multimedia systems, in the next Section we consider the performance of a multimode transceiver, accommodating the firmware of a range of system components and hence ensuring a high grade of flexibility in terms of complexity, speech quality, robustness against channels errors, user capacity etc.

13 MULTIMODE SPEECH SYSTEM PERFORMANCE

13.1 Fixed Signalling-rate Scenario[229, 68]

In the comparative study [229] we presented simulation results giving BER, bandwidth occupancy and an estimate of complexity for 4 bit/symbol 16-Star QAM modems in order to characterise the potential of a UMTS-like system, 2 bit/symbol $\frac{\pi}{4}$-shifted differential quadrature phase shift keying ($\frac{\pi}{4}$-DQPSK) modems, since they are used in the IS-54 and PDC systems as well as binary Gaussian minimum shift keying (GMSK) modems. We used Packet reservation multiple access (PRMA), since it provided substantial improvements over TDMA in terms of the number of users supported.

Specifically, in our simulations we used the GMSK, $\frac{\pi}{4}$-DQPSK and 16-Star QAM modems combined with both the unprotected low-complexity 32 kbps ADPCM codec (as in DECT and CT2) and the 13 kbps RPE-LTP GSM codec with its twin-class FEC. Each modem had the option of either a low or a high complexity demodulator. The high complexity demodulator for the GMSK modem was a maximum likelihood sequence estimator based on the Viterbi algorithm [311], while the low complexity one was a frequency discriminator. For the two multilevel modems either low complexity non-coherent differential detection or a maximum likelihood correlation receiver (MLH-CR) was invoked. Synchronous transmissions and perfect channel estimation were used in evaluating the relative performances of the systems listed in Table 4. Our results represent performance upper bounds, allowing relative performance comparisons under identical circumstances.

The system performances applied to microcellular conditions. The carrier frequency was 2GHz, the data rate 400 kBd, and the mobile speed 15m/s. At 400 kBd in microcells the fading is flat and usually Rician. The best and worst Rician channels are the Gaussian and Rayleigh fading channels, respectively, and we performed our simulations for these channels to obtain upper and lower bound performances. Our conditions of 2GHz, 400 kBd and 15m/s are arbitrary. They correspond to a fading pattern that can be obtained for a variety of different conditions, for example, at 900 MHz, 271 kBd and 23m/s. We compared the performances of the systems defined in Table 4 when operating according to our standard conditions.

Returning to Table 4, the first column shows the system classification letter, the next the modulation used, the third the demodulation scheme employed, the fourth the FEC scheme and the fifth the speech codec employed. The sixth column gives the estimated relative order of the complexity of the schemes, where the most complex one having a complexity parameter of 12 is the 16-Star QAM, MLH-CR, BCH, RPE-LTP arrangement. All the BCH-coded RPE-LTP schemes have complexity parameters larger than six, while the unprotected ADPCM systems are characterised by values of one to six, depending on the complexity of the modem used. The speech Baud rate and the TDMA user bandwidth are given next.

An arbitrary signalling rate of 400 kBd was chosen for all our experiments, irrespective of the number of modulation levels, to provide a fair comparison for all the systems, under identical propagation conditions. Again, these propagation conditions can be readily converted to arbitrary Bd-rates upon scaling the vehicular speed appropriately. The 400 kBd systems have a total bandwidth of $400/1.35 = 296$ kHz, $2 \cdot 400/1.62 = 494$ kHz and $4 \cdot 400/2.4 = 667$ kHz, respectively. When computing the user bandwidth requirements, we took account of the different bandwidth constraints of GMSK, $\frac{\pi}{4}$-DQPSK and 16-QAM, assuming an identical Baud rate.

In order to establish the speech performance of systems A-L we evaluated the Segmental Signal-to-noise Ratio (SEGSNR) versus channel SNR, and Cepstral Distance (CD) [68] versus channel SNR characteristics of these schemes. These experiments yielded 24 curves for AWGN, and 24 curves for Rayleigh fading channels, constituting the best and worst case channels, respectively. Then for the twelve different systems and two different channels we derived the minimum required channel SNR value for near-unimpaired speech quality in terms of both CD and SEGSNR. These values are listed in columns 13 and 14 of Table 4. A range of interesting system design issues can be inferred from the table, but due to lack of space here we refrain from a deeper discussion and refer the interested reader to Reference [229] for more detail.

We note, that the bandwidth efficiency gains tabulated are reduced in SIR-limited scenarios due to the less dense frequency reuse of multilevel modems. Nevertheless, multilevel modulation schemes result in higher PRMA gains than their lower level counterparts. Following the above fixed Baud-rate scenario, let us now consider a fixed bandwidth situation in the next Section.

13.2 Fixed Bandwidth Scenario[339]

13.2.1 Background and Motivation

In another study [339] Packet Reservation Multiple Access (PRMA) assisted adaptive modulation using 1, 2 and 4 bit/symbol transmissions was proposed as an alternative to Dynamic Channel Allocation (DCA) in order to maximise the number of users supported in a traffic cell. The cell was divided in three concentric rings and in the central high Signal-to-noise ratio (SNR) region 16-level Star Quadrature Amplitude Modulation (16-StQAM) was used, in the first ring Differential Quaternary Phase Shift Keying (DQPSK) was invoked, while in the outer ring Differential Phase Shift Keying (DPSK) was utilised. In our diversity-assisted modems a channel SNR of about 7, 10 and 20 dB, respectively, was required in order to maintain a bit error ratio (BER) of about 1 %, which can then be rendered error-free by the binary BCH error correction codes used. A 4.7 kbps, 30 ms frame-length Algebraic Code Excited Linear Predictive (ACELP) speech codec [91] was employed, protected by a quad-class source-sensitivity matched BCH coding scheme [79], yielding a total bit rate of 8.4 kbps. A GSM-like Voice Activity Detector (VAD) controls the PRMA-assisted adaptive system. The achievable capacity improvement due to PRMA will be discussed at a later stage.

Dynamic channel allocation [341] (DCA) and packet reservation multiple access [68] (PRMA) are techniques which potentially allow large increases in capacity over a fixed channel allocation (FCA) time division multiple access (TDMA) system. Although both DCA and PRMA can offer a significant system capacity improvement, their capacity advantages typically cannot be jointly exploited, since the rapid variation of slot occupancy resulting from the employment of PRMA limits the validity of interference measurements, which are essential for the reliable operation of the DCA algorithm. One alternative to tackle this problem is to have mixed fixed and dynamic frequency re-use patterns, but this has the disadvantage of reducing the number of slots per carrier for the PRMA scheme, thus decreasing its efficiency.

In this study we proposed diversity-assisted adaptive modulation as an alternative to DCA. The cells must be frequency planned as in a FCA system using a bi-

1 Syst.	2 Modulator	3 Detector	4 FEC	5 Speech Codec	6 Complexity Order	7 Baud Rate (KBd)	8 TDMA User Bandw. (kHz)	9 No of TDMA Users per Carrier	10 No of PRMA Users per Carrier	11 No of PRMA Users per slot	12 PRMA User Bandw. (kHz)	13 Min SNR (dB) AWGN	14 Min SNR (dB) Rayleigh
A	GMSK	Viterbi	No	ADPCM	2	32	23.7	11	18	1.64	14.5	7	∞
B	GMSK	Freq. Discr.	No	ADPCM	1	32	23.7	11	18	1.64	14.5	21	31
C	$\frac{\pi}{4}$-DQPSK	MLH-CR	No	ADPCM	4	16	19.8	22	42	1.91	10.4	10	28
D	$\frac{\pi}{4}$-DQPSK	Differential	No	ADPCM	3	16	19.8	22	42	1.91	10.4	10	28
E	16-StQAM	MLH-CR	No	ADPCM	6	8	13.3	44	87	1.98	6.7	20	∞
F	16-StQAM	Differential	No	ADPCM	5	8	13.3	44	87	1.98	6.7	21	31
G	GMSK	Viterbi	BCH	RPE-LTP	8	24.8	18.4	12	22	1.83	10.1	1	15
H	GMSK	Freq. Discr.	BCH	RPE-LTP	7	24.8	18.4	12	22	1.83	10.1	8	18
I	$\frac{\pi}{4}$-DQPSK	MLH-CR	BCH	RPE-LTP	10	12.4	15.3	24	46	1.92	8	5	20
J	$\frac{\pi}{4}$-DQPSK	Differential	BCH	RPE-LTP	9	12.4	15.3	24	46	1.92	8	6	18
K	16-StQAM	MLH-CR	BCH	RPE-LTP	12	6.2	10.3	48	96	2.18	4.7	13	25
L	16-StQAM	Differential	BCH	RPE-LTP	11	6.2	10.3	48	96	2.18	4.7	16	24

GMSK: Gaussian MinimumShift Keying
$\frac{\pi}{4}$-DQPSK: Differential Phase Shift Keying
16-StQAM: 16-level Quadrarure Amplitude Modulation
MLH-CR: Maximum Likelihood Correlation Receiver
BCH: Bose Chaudhuri Hocquenghem FEC coding
ADPCM: Adaptive Differential Pulse Code Modulation
RPE-LTP: Regular Pulse Excited speech codec with Long Term Prediction
TDMA: Time Division Multiple Access
PRMA: Packet Reservation Multiple Access

Table 4. System parameters [229], ©IEEE, 1994, Williams, Hanzo et al

Parameter	GSM	DECT	Unit
Channel Bandwidth	200	1728	kHz
Symbol Rate	133	1152	kBd
Bursts per Frame	48	416	

Table 5. Parameters of the GSM like and DECT like adaptive modulation PRMA systems

nary modulation scheme. When adaptive modulation is deployed, the throughput is increased by permitting high level modulation schemes to be used by the mobiles roaming near to the centre of the cell, which therefore will require a lower number of PRMA slots to deliver a fixed number of channel encoded speech bits to the base station (BS). In contrast, mobile stations (MS) near the fringes of the cell will have to use binary modulation in order to cope with the prevailing lower signal-to-noise ratio (SNR) and hence will occupy more PRMA slots for the same number of speech bits. Specifically, our adaptive system uses three modulation schemes, namely binary differential phase shift keying (DPSK) ie one bit per symbol at the cell boundary, quaternary differential phase shift keying (DQPSK), ie two bits per symbol at medium distances from the BS, and 16 level star quadrature amplitude modulation [68] (16-StQAM), which carries four bits per symbol close to the centre of the cell.

13.2.2 PRMA-assisted Adaptive Modulation

PRMA schemes have been documented for example in reference [68]. However, in the proposed PRMA-assisted adaptive modulation scheme mobile stations (MS) can reserve more than one slot in order to deliver up to four bursts per speech frame, when DPSK is invoked towards the cell edges. When a free slot appears in the frame, each mobile that requires a new reservation, contends for it based on a permission probability, P_p. If the slot is granted to a 16-StQAM user, that slot is reserved in the normal way. If the slot is granted to a DQPSK user, then the next available free slot is also reserved for that user. Lastly, if the slot is granted to a DPSK user, then the next three free slots must also be reserved for this particular user. In this way, users that require more than one slot are not disadvantaged by forcing them to contend for each slot individually. If, however, there are less than three slots available, DQPSK or 16-StQAM users still may be able to exploit the remaining slots.

Again, we found that the difference in signal to noise ratio (SNR) required for the different diversity-assisted modulation schemes in order to maintain similar bit error ratios (BER) was approximately 3 dB between DPSK and DQPSK, and 12dB between DPSK and StQAM, when transmitting over Rayleigh-fading channels in our GSM- and DECT-type systems.

Thus, using an inverse fourth power pathloss law, DPSK was invoked between radii $0.84R$ and the cell boundary, R, which is one quarter of the cell area. StQAM was used between the cell centre and $0.5R$, which is a further quarter of the cell area and DQPSK in the remaining area, which constitutes half of the total cell area. Accordingly, considering the number of slots needed by the various modulation schemes invoked and assuming a uniform traffic density, we can calculate the expected number of required slots per call as

$$E(n) = \frac{1}{4}.4 + \frac{1}{2}.2 + \frac{1}{4}.1 = 2.25 \text{ slots.}$$

Since a binary user would require 4 slots, this implies a capacity improvement of a factor of $4/2.25 \approx 1.78$.

13.2.3 Adaptive GSM-like Schemes

The basic systems features are summarised in Table 5, where all modulation schemes assumed an excess bandwidth of 50%, resulting in a symbol rate, which is 2/3 of the total bandwidth. If we consider the case of the 16-StQAM modem which uses four bits per symbol, the 316 channel coded bits to be transmitted may be encoded as 79 4-bit symbols. An additional pair of symbols was used to encode the modulation type, which was repeted three times in order to facilitate majority logic decisions. A further pair of dummy symbols was allocated for so-called power ramping [68], yielding a total of 83 symbols - including all overheads - per 30ms speech frame. This corresponds to a single-user signalling rate of 83/30 ms $\approx$ 2.77 kBd, allowing us to create INT{133.33 kBd / 2.77 kBd} $\approx$ 48 time slots, where the INT{ } function represents the integer part of the bracketed expression. When the DQPSK mode of operation is selected in areas of somewhat lower signal strength, we have to use two traffic bursts in order to convey the 316 bits of information, and when the binary DBPSK mode is selected, four bursts are required. Accordingly, we select the appropriate modulation type within the traffic cell considered, as a function of the received signal strength.

Explicitly, the 8.4 kbps channel-coded rate, after accommodating the packet header carrying the required control information, allowed us to create 48 or 416 slots per 30 ms frame in the GSM-like and DECT-like systems respectively, as shown in the Table. Specifically, when using the 133.33 kBd GSM-like adaptive PRMA schemes, we can create 48 slots per 30 ms speech frame, which is equivalent to 12 slots for a binary only BPSK system, since four slots are required for the transmission of a 30 ms speech packet. When the quaternary system is used, 24 pairs of slots can be created. Note that when fixed channel allocation is used, the adaptive scheme and the binary only scheme can use the same cluster size. A quaternary only system requires a 3dB greater SIR than the binary scheme. According to Lee [342],

$$\frac{D}{R} = \sqrt{3K} \tag{6}$$

System	Slots	Permission Probability	Simultaneous Calls	Normalised by cluster size K	Improvement over binary with PRMA	Improvement over binary without PRMA
DBPSK	12	0.5	19	19	-	58%
DQPSK	24	0.4	44	31.1	64%	159%
Adaptive	48	0.5	36	36	89%	200%

Table 6. Improvements in capacity possible with adaptive modulation PRMA with 48 slots ©IEE, Williams, Hanzo, Steele, 1995, [339]

System	Slots	Permission Probability	Simultaneous Calls	Normalised by cluster size K	Improvement over binary with PRMA	Improvement over binary without PRMA
DBPSK	104	0.1	220	200	-	112%
DQPSK	208	0.1	470	332	51%	219%
Adaptive	416	0.1	400	400	82%	285%

Table 7. Achievable capacity improvements for the adaptive modulation PRMA with 416 slots ©IEE, Williams, Hanzo, Steele, 1995, [339]

where D is the distance to the closest interferer, R is the cell radius, and K is the cluster size. The prevailing Signal-to-Interference Ratio (SIR) can be expressed as

$$\mathrm{SIR} \approx \left(\frac{D}{R}\right)^{\gamma} \tag{7}$$

where γ is the path loss exponent and hence

$$K = \frac{1}{3}(\mathrm{SIR})^{\frac{2}{\gamma}}. \tag{8}$$

In this treatise we have used a pathloss exponent of $\gamma = 4$, and therefore increasing the SIR by 3dB requires that the cluster size be increased by a factor of $\sqrt{2}$. The packet dropping versus number of users performance of all the different schemes were evaluated for their respective optimum P_p values, which were different for the different schemes supporting different numbers of time slots.

We found that a maximum of 19 simultaneous calls can be supported at a packet dropping probability of 1%, when using the binary scheme with a PRMA permission probability of 0.5. For the sake of comparison, this system can support only 12 TDMA users, each requiring four slots per frame. By contrast, the quaternary scheme can support 44 simultaneous calls, when using the optimum permission probability of 0.4, assigning two slots per frame for each of them. The corresponding TDMA scheme could only support 24 such users. Lastly, our 48-slot adaptive scheme can accommodate 36 simultaneous calls, while using the optimum permission probability of 0.5. The capacity improvements attainable by the proposed GSM-like scheme are presented in Table 6.

13.2.4 Adaptive DECT-like Schemes

In our DECT-like schemes we have INT{1152 kBd/2.77 kBd}=416 slots per frame for the adaptive PRMA system. This is equivalent to 104 slots for a binary only system and 208 slots for a quaternary only system. Again, a quaternary only system requires a 3dB greater SIR than the binary scheme and so the cluster size should be increased by a factor of $\sqrt{2}$.

We found that the binary scheme can support up to 220 simultaneous calls at a packet dropping probability of 1%. When opting for the 208 slot quaternary scheme, the packet dropping versus number of users performance curve reveals that this system can accommodate 470 simultaneous calls with a permission probability of 0.1. Finally, the packet dropping performance of the 416 slot adaptive scheme suggests that the number of supported simultaneous conversations is about 400, when opting for a permission probability of 0.1. The achievable capacity improvements for our DECT-like system are displayed in Table 7.

In conclusion, adaptive modulation with PRMA gives the expected three- to four-fold capacity increase over the binary scheme without PRMA. Generally the greater the number of slots, the greater the advantage of PRMA over non-PRMA systems, since the statistical multiplexing gain approaches the reciprocal of the speech activity ratio. Furthermore, PRMA-assisted adaptive modulation achieves an additional 80% capacity increase over PRMA-assisted binary modulation [340]. The speech performance of our adaptive system evaluated in terms of Segmental Signal-to-noise Ratio (SEGSNR) and Cepstral Distance (CD) is unimpaired by channel effects for SNR values in excess of about 8, 10 and 20 dB, when using diversity-assisted DPSK, DQPSK and 16-StQAM, respectively, although in dispersive environments a reduced performance is expected.

Let us now consider the expected performance of a similar intelligent multimode video system.

14 VIDEOPHONE SYSTEMS

Below we follow the approach of Cherriman et al [176] and as an example let us consider transmitting QCIF images, where the video codecs were programmed to generate 3560, 2352 and 1176 bits per frame. At scanning rate of 10 frames/s these coding modes resulted in video bitrates of 35.6 kbps, 23.52 kbps and 11.76 kbps, respectively.

In our earlier work we have shown that in Quadrature Amplitude Modulation (QAM) schemes the bits assigned to the transmitted non-binary modulation symbols exhibit different bit error rates, defining a number of different integrity classes [68]. The number of integrity classes depends on the number of modulation-levels, and in 4-QAM there is only one such integrity class, in 16-QAM there are 2, while in 64-QAM there are 3 classes, often also referred to as sub-channels. By using different strength FEC codes on each QAM sub-channel it is possible to equalise the probability of errors on the sub-channels. This means that all sub-channels FEC codes should break down at approximately the same channel SNR. This is desirable if all bits to be transmitted are equally important. Since our datastreams are variable length coded, one error can cause a loss of synchronisation. Therefore in this case most bits are equally important, and so equalisation of QAM sub-channels BER is desirable. The FEC codes used in our system are summarised in Table 8, where a $BCH(n,k,t)$ code represents a binary Bose-Chaudhuri-Hocquenghen code encoding k bits to n bits and capable of correcting t errors per codeword [79]. After including a control header, pilot and ramp symbols [68], as suggested in Reference [176], the FEC-coded single-user signalling rates became 11.84 kBd in all three modem operating modes. In case of a Nyquist excess bandwidth of 50% this implies a single-user bandwidth requirement of 17.74 kHz. For example in the 200 kHz bandwidth of the Pan-European GSM system 8 voice-only users or supported, which corresponds to a 25 kHz user bandwidth. Consequently, our videophone stream can replace a speech stream, making wireless videophony realistic, if the cost of an additional timeslot is acceptable to the users.

The video system performance was evaluated under the propagation conditions of a vehicular speed of 30 mph, signalling rate of 11.84 kBd and propagation frequency of 1.9 GHz. The corresponding single-user bandwidth requirement is about 16 kHz, when using a modulation excess bandwidth of 35%, or a Nyquist roll-off factor of 0.35. In the various operating modes investigated the PSNR versus channel SNR curves of Figures 27 and 28 were obtained for Additive White Gaussian Noise (AWGN) and Rayleigh channels, respectively. Since both the H.261 and H.263 source codecs have had similar robustness against channel errors, and their transceivers were identical, the associated 'corner SNR' values, where unimpaired communications broke down were virtually identical for both systems over both AWGN and Rayleigh channels. However, as expected, the H263 codec again exhibited always higher video quality at the same bitrate or system bandwidth.

Modulation scheme	FEC codes used
4 QAM	BCH(255,147,14)
16 QAM	Class 1: BCH(255,179,10) Class 2: BCH(255,115,21)
64 QAM	Class 1: BCH(255,199,7) Class 2: BCH(255,155,13) Class 3: BCH(255,91,25)

Table 8. FEC codes used for 4, 16 and 64 QAM©Cherriman, Hanzo[176]

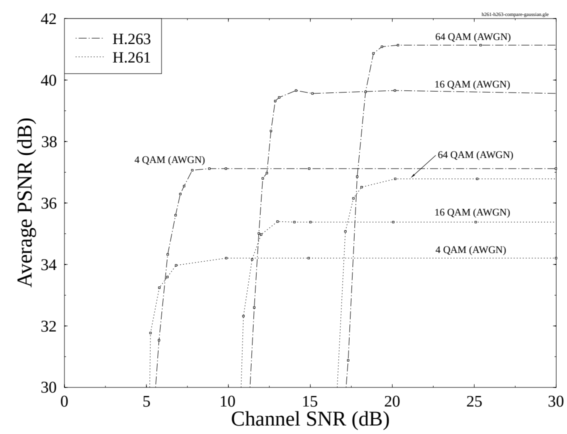


Figure 27. Performance comparison of the proposed adaptive H.261 and H.263 transceivers over AWGN channels©Cherriman, Hanzo[176]

Our current endeavours are focussed on exploring the quality versus bitrate performance of both systems for various image resolutions, in order to be able to provide the required video quality, bit rate, frame rate, image size and resolution on a demand basis in intelligent adaptive multimode transceivers.

15 SUMMARY AND CONCLUSIONS

This overview attempted to highlight some of the basic issues in bandwidth-efficient tetherless multimedia communications. Due to its wide scope it was impossible to offer a full exposure of any of the topics, although we endeavoured to provide references for the motivated reader

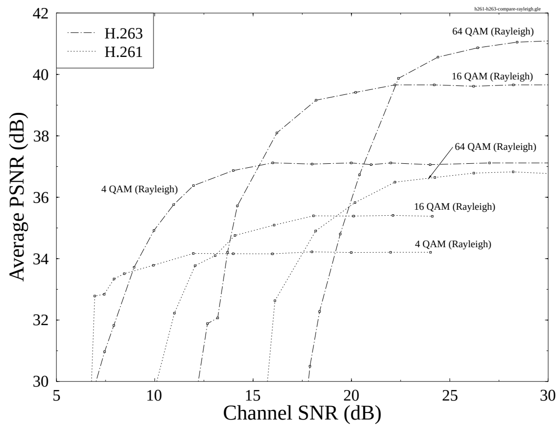


Figure 28. Performance comparison of the proposed adaptive H.261 and H.263 transceivers over Rayleigh channels©Cherriman, Hanzo[176]

to probe further in most topics of interest. In the source coding area substantial advances have been made over the past few years. A range of high-quality, error resilient speech codecs have been developed and with the advent of the 8 kbps G.729 ITU speech codec this field reached a remarkable state of maturity. This is the first ITU codec, which was designed with the high prevalant BER of wireless systems in mind. Currently research is under way towards the definition of the 4 kbps ITU standard speech codec. In order to provide narrowband wireless multimedia services, we have proposed a variety of fixed-rate video codecs, which can generate a bit stream that can be accommodated by allocating an additional physical channel for video transmissions. We also suggested and H.263-based fixed-rate multimode transceiver for video telephony. Furthermore, handwriting or graphical information can also be robustly encoded using FL-DCC and multiplexed for example using the previously described PRMA or SPAMA schemes with speech and video signals. Multiplexing variable-rate multimedia sources was detailed for example in References [232, 228, 227]. When the increased complexity of turbo codecs becomes acceptable due to the advances in low-voltage VLSI, the performance of wireless personal communicators can approach the Shannonian performance limits more closely. Indeed, over AWGN channels a BER of 10^{-5} was reported by Pietrobon [345] using a 1/3-rate hardware turbo codec operating at 356 kbit/s and employing an interleaver size of 65 536 bits at an E_b/N_0 value of 0.32 dB, where E_b and N_0 are the bit energy and the noise power spectral density, respectively. In closing we note that the intelligent wireless multimode, multimedia concept advocated in this contribution is likely to provide substantial benefits for the potential users and its implementation is realistic at the current state-of-the-art.

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REFERENCES

[1] W.C.Jakes: "Microwave mobile communications", *John Wiley & Sons*, New York, 1974.

[2] W.Y.C.Lee: "Mobile cellular communications", *McGraw Hill*, New York, 1989.

[3] J.D.Parsons and J.G.Gardiner: "Mobile communication systems", *Blackie*, London 1989.

[4] K. Feher: Wireless digital communications: Modulation and spread spectrum applications, Prentice Hall, 1995

[5] S. Glisic, B. Vucetic: Spread spectrum CDMA systems for wireless communications, Artech house publishers, 1997, ISBN 0-89006-858-5

[6] R. Prasad: CDMA for wireless personal communications, Artech House, 1996

[7] S. G. Glisic, P.A. Leppanen (Ed.): Wireless communications - TDMA versus CDMA, Kluwer academic publishers, 1997, ISBN 0-7923-8005-3

[8] H. Meyer, M. Moeneclaey, S. Fechtel: Digital Communications receivers, John Wiley, 1997 ISBN 0-471-50275-8

[9] R. Steele (Ed.): *Mobile Radio Communications*, IEEE Press-Pentech Press, 1992

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

[11] V. K. Garg and J. E. Wilkes, *Wireless and Personal Communications Systems.* Prentice Hall, 1996.

[12] J. D. Gibson: The mobile communications handbook, IEEE Press-CRC Press, 1996

[13] IEEE Personal Communications: The magazine of nomadic communications and computing, Feb. 1995, Vol. 2, No. 1 - Special Issue: The European Path Toward UMTS

[14] IEEE Personal Communications: The magazine of nomadic communications and computing, Apr. 1995, Vol. 2, No. 2 - Feature Topic: Wireless Personal Communications

[15] IEEE Communications Magazine, Nov. 1995, Vol. 33, No. 11 - Feature Topic: Universal Telecommunications at the Beginning of the 21st Century

[16] IEEE Communications Magazine, May 1995 Vol. 33 No. 5 - Feature Topic: Software Radios

[17] IEEE Communications Magazine, Jan. 1995, Vol. 33, No. 1 - Feature Topic: Wireless Personal Communications

[18] IEEE Communications Magazine, Feb. 1996, Vol. 34, No. 2 - Feature Topic: European Research in Mobile Communications, pp 60-106

[19] V.H.MacDonald, "The cellular concept", *Bell System Tech. J.*, Vol 58, No.1, pp.15-41, January 1979.

[20] R.Steele, "Towards a high capacity digital cellular mobile radio system", *Proc. of the IEE*, PtF, No.5, pp.405-415, August 1985.

[21] R.Steele and V.K.Prabhu, "Mobile radio cellular structures for high user density and large data rates", *Proc. of the IEE*, Pt F, No.5, pp.396-404, August 1985.

[22] R.Steele, "The cellular environment of lightweight hand-held portables", *IEEE Communications Magazine*, pp.20-29, July 1989.

[23] S. Tekinay, B. Jabbari: A measurement-based prioritisation scheme for handovers in mobile cellular networks, IEEE JSAC, 1992, Vol. 10, No. 8, pp 1343-1350

[24] G.P. Pollini: Trends in handover design, IEEE Comms. Mag., March 1996, pp 82-90

[25] J. C.-I Chuang: Performance issues and algorithms for dynamic channel assignment, IEEE JSAC Vol. 11, No. 6, Aug. 1993, pp 955-963

[26] J.C.-I. Chuang, N. Sollenberger: Performance of autonomous dynamic channel assignment and power control for TDMA/FDMA wireless access, IEEE JSAC, Vol 12, No. 8, Oct. 1994, pp 1314-1323

[27] M.M.-L. Cheng, J.C-I. Chuang: Performance evaluation of distributed measurement-based dynamic channel assignment in local wireless communications, IEEE JSAC, Vol. 14, No. 4, May 1996, pp 698-710

[28] R.C. Bernhardt: Timeslot management in digital portable radio systems, IEEE Tr. on Veh. Techn., Feb. 1991, pp 261-272

[29] R.C. Bernhardt: Timeslot re-assignment in a frequency reuse TDMA portable radio system, IEEE Tr. on Veh. Techn., Aug. 1992, pp 296-304

[30] D.C. Cox: Wireless personal communications: A perspective, pp 209-241 in J.D. Gibbson (Ed.): The mobile communications handbook, IEEE Press-CRC Press, 1996

[31] A.D. Kucar: Mobile radio: An overview, pp 242-262 in J.D. Gibbson (Ed.): The mobile communications handbook, IEEE Press-CRC Press, 1996

[32] V.O.K. Li, X. Qiu: Personal Communications Systems, Proc. of the IEEE, Vol. 83, No. 9, Sept. 1995, pp 1210-1243

[33] *Proceedings of the Nordic Seminar on Digital Land Mobile Radio Communication (DMR), Espoo, Finland,* February 1985.

[34] *Proceedings of the Second Nordic Seminar on Digital Land Mobile Radio Communication (DMRII), Stockholm, Sweden,* October 1986.

[35] *Proceedings of the International Conference on Digital Land Mobile Radio Communication (ICDMC) Venice, Italy,* June/July 1987.

[36] *Proceedings of Digital Cellular Radio Conference, Hagen, FRG,* October 12-14, 1988.

[37] GSM Recommendation, 1988, (Group Speciale Mobile or Global System of Mobile Communication) European Telecommunications Standardisation Institue (ETSI)

[38] A. Moloberti: "Definition of the radio subsystem for the GSM Pan-European digital mobile communication system" *Proc. of ICDMC, Venice, Italy,* pp. 37-46, June/July 1987.

[39] A.W.D. Watson. "Comparison of the contending multiple access methods for the pan-European mobile radio systems". *IEE Colloquium, Digest No:1986/95,* pp. 2/1-2/6, 7 October 1986.

[40] L. Hanzo, J. Stefanov: The Pan-European Digital Cellular Mobile Radio System-known as GSM, pp 677-773, Chapter 8 in R. Steele, Mobile Radio Communications, Pentech Press, 1992

[41] Dual-mode subscriber equipment - Network equipment compatibility specification, Interim Standard IS-54, Telcomm. Industry Association (TIA), Washington, DC, 1989, TIA

[42] Public Digital Cellular (PDC) Standard, RCR STD-27, Research and Development Centre for Radio Systems, Japan, PDC

[43] Mobile station - Base station compatibility standard for dual-mode wideband spread spectrum cellular system, EIA/TIA Interim Standard IS-95, Telcomm. Industry Association (TIA), Washington, DC, 1993, TIA

[44] Advanced Communications Technologies and Services (ACTS), Workplan, European Commission, DGXIII-B-RA946043-WP, Aug. 1994

[45] J. Schwartz da Silva, B. Arroyo-Fernandez, B. Barani, J. Pereira, D. Ikonomou: Mobile and personal Communications: ACTS and Beyond; ibid, pp 379-415

[46] K. Pehkonen et al: A performance analysis of TDMA and CDMA based air interface solutions for UMTS high bit rate services, Proc. of PIMRC'97, Helsinki, Sept. 1997, pp 22-26

[47] T. Ojanpera et al: Analysis of CDMA and TDMA for 3rd generation mobile radio systems, Proc. of VTC'97, Vol. 2, pp 840-844, Phoenix, USA, May 1997

[48] K. Pajukoski et al: Wideband CDMA test system, Proc. of PIMRC'97, Helsinki, Sept. 1997, pp 669-673

[49] A. Klein et al: FRAMES Multiple access mode 1 - Wideband TDMA with and without spreading, Proc. of PIMRC'97, Helsinki, Sept. 1997, pp 37-41

[50] P.W. Baier, P. Jung, A. Klein: Taking the challenge of multiple access for third-generation cellular mobile radio systems - A European View, IEEE Comms. Mag., Feb. 1996, Vol. 34, No. 2, pp 82-89

[51] B. Engstroem, C. Oesterberg: A system for test of multiaccess methods based on OFDM, Proc. of VTC'94, Stockholm, Sweden, 1994, pp 1843-1847

[52] F. Adachi et al: Coherent DS-CDMA - Promising Multiple Access for Wireless Multimedia Mobile Communications, Proc. IEEE ISSSTA'96, pp 351-358, Mainz, Germany, Sept. 1996

[53] J. Litva, T. K-Y Lo: Digital beamforming in wireless communications, Artech house publishers, 1996, ISBN 0-89006-712-0

[54] T. Ojanpera: Overview of research activities for third generation mobile communications; ibid pp 415-446

[55] R. Kohno: Spatial and temporal communication theory using software antennas for wireless communications; ibid, pp 293-322

[56] P.W. Baier, J. Blanz, R. Schmalenberger: Fundamentals of smart antennas for mobile radio applications; ibid pp 345-376

[57] A. Polydoros, K.M. Chugg: Per-survivor processing, ibid pp 41-72

[58] S.Verdu, "Minimum probability of error for asynchronous Gaussian multiple access channels," *IEEE Transactions on Information Theory*, vol. 32, pp. 85–96, January 1986.

[59] P.Jung and J.Blanz, "Joint detection with coherent receiver antenna diversity in CDMA mobile radio systems," *IEEE Transactions on Vehicular Technology*, vol. 44, pp. 76–88, February 1995.

[60] A.Klein, G.K.Kaleh and P.W.Baier, "Zero forcing and minimum mean square error equalization for multiuser detection in code division multiple access channels," *IEEE Transactions on Vehicular Technology*, vol. 45, pp. 276–287, May 1996.

[61] Z.Zvonar and D.Brady, "Suboptimal multiuser detector for frequency selective Rayleigh fading synchronous CDMA channels," *IEEE Transactions on Communications*, vol. 43, pp. 154–157, February-April 1995.

[62] Y.Sanada and M.Nakagawa, "A multiuser interference cancellation technique utilizing convolutional codes and multicarrier modulation for wireless indoor communications," *IEEE Journal on Selected Areas in Communications*, vol. 14, pp. 1500–1509, October 1996.

[63] P.-A.Sung and K.-C.Chen, "A linear minimum mean square error multiuser receiver in Rayleigh fading channels," *IEEE Journal on Selected Areas in Communications*, vol. 14, pp. 1583–1594, October 1996.

[64] L.Wei, L.K.Rasmussen and R.Wyrwas, "Near optimum tree-search detection schemes for bit-synchronous multiuser CDMA systems over Gaussian and two-path Rayleigh-fading channels," *IEEE Transactions on Communications*, vol. 45, pp. 691–700, June 1997.

[65] D.Dahlhaus, A.Jarosch, B.H.Fleury and R.Heddergott, "Joint demodulation in DS/CDMA systems exploiting the space and time diversuty of the mobile radio channel," in *Proceedings of the 1997 International Symposium on Personal, Indoor and Mobile Radio Communications, PIMRC'97*, pp. 47–52.

[66] S.K. Barton (Guest Editor): Special Issue on HIPERLAN of the Wireless Personal Communications Journal, Vol. 3, No.4 and Vol. 4, No. 1., 1997, pp 341-453

[67] Radio Equipment and systems (RES); High PErformance Radio Local Area Network (HIPERLAN), Type 1, Functional Specifications, ETSI Final Draft prETS 300 652

[68] W.T. Webb, L. Hanzo: Modern quadrature amplitude modulation: Principles and applications for fixed and wireless channels, IEEE Press-Pentech Press, 1994, ISBN 0-7273-1701-6, p 557

[69] J.B. Anderson, T. Aulin, C-E. Sundberg: Digital Phase modulation, Plenum Press, 1985

[70] P. Sheldon, J. Cosmas, A. Permain: Dynamically adaptive control system for MPEG-4, Proceedings of the 2nd International Workshop on Mobile Multimedia Communications, Apr. 11-13, Bristol, UK

[71] Y.-Q. Zhang, F. Pereira, T. Sikora, C. Reader (Guest Editors) Special Issue on MPEG4 of the IEEE Tr. on CSVT, Feb. 1997, Vol. 7, No.1 pp 1-256

[72] C.E. Shannon "A mathematical theory of communication", *Bell System Technical Journal*, Vol. 27, pp 379-423 and 623-656, June and Oct. 1948

[73] J. Hagenauer "Quellengesteuerte Kanalcodierung fuer Sprach- und Tonuebertragung im Mobilfunk", *Aachener Kolloquium Signaltheorie*, 23-25.03.1994, Mobile Kommunikationssysteme, pp 67-76

[74] J. Hagenauer: Source-controlled channel decoding, IEEE Tr. on Comms., Sept. 1995, Vol. 43, No. 9, pp 2449-2457

[75] A.J. Viterbi "Wireless digital communications: A view based on three lessons learned", *IEEE Comms. Mag,* Sept. 1991, pp 33-36

[76] N. Jayant: Signal compression: Technology targets and research directions, pp 796-818, in Speech and Image Coding, Special Issue of the IEEE JSAC, June 1992, Vol. 10, No. 5, pp 793-976

[77] Speech and Image Coding, Special Issue of the IEEE JSAC, Guest Editor: N. Hubing, June 1992, Vol. 10, No. 5, pp 793-976

[78] A. Gersho: Advances in speech and audio compression, Proc. of the IEEE, Vol. 82, No. 6, June 1994, pp 900-918

[79] Wong, K.H.H. and Hanzo, L., *Channel Coding*, Chapter 4, pp 347-488, in *R. Steele (Ed.) Mobile Radio Communications*, IEEE Press-Pentech Press, London, ISBN 07273-1406-8, 1992

[80] ISO/IEC 11172 MPEG 1 International Standard, 'Coding of moving pictures and associated audio for digital storage media up to about 1.5 Mbit/s, Parts 1-3

[81] ISO/IEC CD 13818 MPEG 2 International Standard, Information Technology, Generic coding of moving video and associated audio information, Parts 1-3

[82] L. Hanzo, J.P. Woodard: An intelligent multimode voice communications system for indoors communications, IEEE Tr. on Veh. Technology, Nov. 1995, Vol. 44, No. 4, pp 735-749, ISSN 0018-9545

[83] L. Hanzo, R. Salami, R. Steele, P.M. Fortune: "Transmission of Digitally Encoded Speech at 1.2 KBd for PCN", *IEE Proceedings-I*, Vol 139, No.4, pp 437-447, Aug 1992

[84] J.D. Markel and A.H. Gray, Jr.: Linear prediction of speech *Springer-Verlag, New York,* 1976.

[85] N.S. Jayant and P. Noll: Digital coding of waveforms *Prentice-Hall,* 1984.

[86] L.R. Rabiner and R.W. Shafer: Digital processing of speech signals *Prentice-Hall Int.,* 1978.

[87] D. O'Shaugnessy: Speech Communications - Human and Machine, *Addison-Wesley*, 1987

[88] Sadaoki Furui: Digital Speech Processing, Synthesis and Recognition, *Marcel Dekker*, 1989

[89] A.M. Kondoz: Digital Speech: Coding for low bit rate communications systems, John Wiley, 1994

[90] B. Kleijn, KK. Paliwal: Speech coding and synthesis, Elsevier, 1995

[91] Salami, R. A., Hanzo, L. et al *Speech Coding* , Chapter 3, pp 186-346, in *R. Steele (Ed.) Mobile Radio Communications*, IEEE Press-Pentech Press, London, ISBN 07273-1406-8, 1992

[92] J.B. Anderson, S. Mohan: Source and Channel Coding - An Algorithmic Approach, *Kluwer Academic Publishers*, 1991

[93] H.B. Law, R.A. Seymour: A reference distortion system using modulated noise, IEE paper No 399-2E, Nov 1962.

[94] K.D. Kryter: Methods for the calculation of the articulation index, ANSI 53.5 - 1965, American National Standards Institute, NY 1969

[95] A.S. House, C.E. Williams, M.H.L. Hecker, K.D. Kryter: articulation testing methods: Consonated differentiation with closed-response set, J Acoust. Soc. Am., Jan 1965, pp 158-166

[96] A.H. Gray, J.D. Markel: Distance measures for speech processing, IEEE Tr on ASSP, Vol ASSP-24, No 5, Oct 1979, pp 380-391

[97] N. Kitawaki, M. Honda, K. Itoh: Speech Quality Assessment Methods for Speech Coding Systems, IEEE Comm Magazine, Vol 22, No 10, Oct 1984, pp 26-33.

[98] N. Kitawaki, H. Nagabucki, K. Itoh: Objective Quality Evaluation for Low-bit-rate Speech Coding Systems, IEEE J-SAC, Vol 6, No 2, Feb 1988, pp 242-249.

[99] S. Wang, A. Sekey, A. Gersho: An objective measure for predicting subjective quality of speech coders, IEEE JSAC, Vol. 10, No. 5, June 1992, pp 819-829

[100] U. Halka, U. Heute: A new approach to objective quality-measures based on attribute matching, Speech Communications, Bd. 11, 1992, pp 15-30

[101] K.Y. Lee, A.M. Kondoz, B.G. Evans: Speaker adaptive vector quantisation of LPC parameters of speech, *Electr. Letters*, Vol. 24, No 22, pp 1392-93, 27th of Oct. 1988

[102] W.T.K. Wong and I. Boyd "Optimal quantization performance of LPC parameters for speech coding". *Proc. EUROSPEECH'89,* pp. 344-347, 1989.

[103] R. Laroia, N. Phamdo, N. Farvardin: Robust and efficient quantisation of speech LSP parameters using structured vector quantisers, *Proc. of ICASSP 91*, Toronto, Canada, May 1992, pp 641-644

[104] K.K. Paliwal, B.S. Atal: Efficient vector quantisation of LPC parameters at 24 bits/frame, *IEEE Tr. on Speech and Audio Processing*, vol. 1, No. 1 Jan. 1993, pp 3-14

[105] P. Kroon and E.F. Deprettere: Regular Pulse Excitation - A Novel Approach to Effective Multipulse Coding of Speech, *IEEE Trans. on Acoustics, Speech and Signal Processing*, pp. 1054–1063, 1986

[106] P. Vary, K. Hellwig, R. Hofmann, R. Sluyter, C. Galland and M. Rosso: Speech Codec for the European Mobile Radio System, *Proc. ICASSP*, pp. 227–230, April 1988

[107] I.A. Gerson and M.A. Jasuik: Vector Sum Excited Linear Prediction (VSELP) Speech Coding at 8 kbps, *IEEE Journal on Selected Areas in Communications*, pp. 461–464, 1990

[108] I.A. Gerson and M.A. Jasiuk: Vector Sum Excited Linear Prediction (VSELP)," in *Advances in Speech Coding* (Bishnu S. Atal, Vladimir Cuperman and Allen Gersho, ed., pp. 69–80, Kluwer Academic Publishers, 1991

[109] I.A. Gerson, M.A. Jasiuk, J-M. Muller, J.M. Nowack, E.H. Winter, Speech and channel coding for the half-rate GSM channel, Proceedings ITG-Fachbericht 130, VDE-Verlag, Berlin pp 225–233, Nov. 1994.

[110] I.A. Gerson, M.A. Jasiuk, Vector sum excited linear prediction (VSELP) speech coding at 8 kbps, Proceedings of IEEE ICASSP '90, pp 461–464.

[111] I.A. Gerson, M.A. Jasiuk, Techniques for improving the performance of CELP-type speech codecs, IEEE JSAC, Vol 10, No 5, June 1992, pp 858–865.

[112] Joseph Campbell, Vanoy Welch and Thomas Tremain: An Expandable Error-protected 4800 bps CELP Coder (U.S. Federal Standard 4800 bps Voice Coder), *Proc. ICASSP*, pp. 735–738, 1989

[113] Tomoyuki Ohya, Hirohito Suda and Toshio Miki, 5.6 kbits/s PSI-CELP of the half-rate PDC speech coding standard, *Proceeding of the IEEE Conference on Vehicular Technology*, pp. 1680–1684, June 1994

[114] W. Bastiaan Kleijn, Encoding Speech Using Prototype Waveforms, *IEEE Trans. on Acoustics, Speech and Signal Processing*, pp. 386–399, Oct 1993

[115] Daniel W. Griffin and Jae S. Lim, Multiband Excitation Vocoder, *IEEE Trans. on Acoustics, Speech and Signal Processing*, pp. 1223–1235, Aug 1988

[116] Dimitris J. Hiotakakos and Costas S. Xydeas, Low Bit Rate Coding Using an Interpolated Zinc Excitation Model, *Proceeding of the IEEE Singapore International Conference on Communications Systems* , pp. 865–869, Nov 1994

[117] J.H. Chen, R.C. Cox, Y.C. Lin, N. Jayant, M.J. Melchner: "A low-delay CELP codec for the CCITT 16 kb/s Speech Coding Standard", IEEE Journal on Selected Areas in Communications, Vol 10, No 5, June 1992, pp 830-849

[118] A. Kataoka, J-P. Adoul, P. Combescure and P. Kroon: ITU-T 8-kbits/s Standard Speech Codec for Personal Communication Services, *Proceedings of International Conference on Universal Personal Communications, Tokyo, Japan*, pp. 818–822, Nov 1995

[119] A. McCree et al: A 2.4 kbit/s MELP candidate for the new US federal standard, *Proc. of ICASSP'96, Atalanta, Georgia, US*, pp 200-203, 1996.

[120] R. P. Ramachandran, M. M. Sondhi, N. Seshadri, B. S. Atal: A two codebook format for robust quantisation of line spectral frequencies, *IEEE Tr. on Speech and Audio Processing*, Vol. 3, No. 3, May 1995, pp 157-168

[121] R.A. Salami, C. Laflamme, J.P. Adoul: ACELP speech coding at 8kbit/s with a 10 ms frame: A candidate for CCITT standardisation, Proc. of IEEE Workshop on speech coding for telecommunications, Hotel Le Chantecler, Sainte-Adele, Quebec, Canada, Oct. 13-15, 1993, pp 23-24

[122] R.A. Salami, C. Laflamme, J-P Adoul, D. Massaloux: A toll quality 8 Kb/s speech codec for the personal communications system (PCS), IEEE Tr. on Veh. Tech., Vol. 43, No. 3, Aug. 1994, pp 808-816

[123] Coding of Speech at 8 kbit/s Using Conjugate-Structue Algebraic Code-Excited Linear Prediction (CS-ACELP), *ITU Draft Recommendation G.729*, February 1996

[124] J.P. Campbell, Jr., T.E. Tremain and V.C. Welch: The DoD 4.8 kbps standard (Federal standard 1016), in *Advances in Speech Coding*, Kluwer Academic Publishers, pp. 121-133, 1990.

[125] J-P. Adoul, C. Lamblin: A comparison of some algebraic structures for CELP coding od speech, Proc. of ICASSP'87, pp 1953-1956

[126] J-P. Adoul et al: Fast CELP coding based on algebraic codes, Proc. of ICASSP'87, pp 1957-1960

[127] C. Lamblin, J-P. Adoul et al: Fast Celp coding based on the Barnes-Wall lattice in 16 dimension, Proc. of ICASSP'89, pp 61-64, Glasgow, UK

[128] H. C.Laflamme, J-P.Adoul, H.Y. Su, S.Morissette, "On reducing the complexity of codebook search in CELP through the use of algebraic codes," *Proc. ICASSP 1990*, Albuquerque, pp. 177–180, 1990.

[129] R.A. Salami."Binary pulse excitation: a novel approach to low complexity CELP coding" in *Advances in speech coding*, B.S. Atal et. al. (Eds) Kluwer Academic Publishers, 1991

[130] I.A. Gerson and M.A. Jasiuk. "Vector sum excitation linear prediction (VSELP) speech coding at 8 kbps". *Proc. ICASSP'90, Albuquerque, New Mexico, USA*, pp. 461-464, 3-6 Apr., 1990.

[131] X. Maitre: 7 kHz Audio Coding within 64 kbit/s, IEEE-JSAC, Vol 6, No 2, Feb 1988, pp 283-298.

[132] Quackenbush: A 7 kHz bandwidth, 32 kbps speech coder for ISDN, Proc of ICASSP '91, pp 1-4.

[133] J.D. Johnston: Transform coding of audio signals using perceptual noise criteria, IEEE-JSAC, Vol 6, No 2, 1988, pp 314-323.

[134] E. Ordentlich, Y, Shoham: Low-delay code-excited linear-predictive coding of wideband speech at 32 kbps, Proc. of ICASSP '91, pp 9-12.

[135] A.W. Black, A.M. Kondoz, B.G. Evans: High quality low delay wideband speech coding at 16 kbit/sec, Proc of 2nd Int Workshop on Mobile Multimedia Communications, 11-14 Apr 1995, Bristol University, UK.

[136] C. Laflamme, J-P. Adoul, R. Salami et al: 16 kbps wideband speech coding technique based on Algebraic CELP, Proc ICASSP '91, pp 13-16.

[137] R. Salami, C. Laflamme, J-P. Adoul: Real-time implementation of a 9.6 kbit/s ACELP Wideband speech coder, Proc GLOBECOM '92.

[138] V.E. Sanchez-Calle, C. Laflamme, R. Salami, J-P. Adoul: Low-delay algebraic CELP coding of wideband speech, Signal Processing VI: Theories and Applications, Edited by J. Vandewalle, R. Boite, M. Moonen, A. Oosterlink, 1992 Elsevier Science Publishers, pp 495-498.

[139] Proceedings of the 2nd International Workshop on Mobile Multimedia Communications, MoMuC-2, Apr. 11-13, 1995, Bristol, UK

[140] S. Sheng, A. Chandrakashan, R.W. Brodersen, *A Portable Multimedia Terminal*, IEEE Comms. Mag., Dec 1992, Vol. 30. No. 12, pp 64-75

[141] IEEE Transactions on Circuits and Systems for Video Technology, Special Issue on Very Low Bit Rate Video Coding, June 1994, Vol 4, No. 3, pp 213-357

[142] Special Issue on Image Sequence Compression, IEEE Tr. on Image processing, Sept. 1994, Vol. 3, No. 5, Guest Editors: B. Girod et al

[143] A. Netravali and B. Haskell, *Digital pictures: representation and compression.* Plenum Press, 1988.

[144] J.W. Woods (Ed.): Subband image coding, Kluwer Academic Publishers, 1991

[145] R. Stedman, H. Gharavi, L. Hanzo, R. Steele: Transmission of Subband-Coded Images via Mobile Channels, *IEEE Tr. on Circuits and Systems for Video Technology,* Feb 1993; Vol.3, No. 1, pp 15-27

[146] J. Katto, J. Ohki, S. Nogaki, M. Ohta: A wavelet codec with overlapped motion compensation for very low bit-rate environment, IEEE Tr. on Video Technology, Vol. 4, No. 3, June 1994, pp 328-338

[147] P. Strobach: Tree-Structured Scene Adaptive Coder, IEEE Trans. Commun., vol COM-38, pp 477-486, 1990

[148] J. Vaisey, A. Gersho: Image compression with variable block size segmentation, IEEE Tr. on Signal Processing, Vol. 40, No. 8, Aug. 1992, pp 2040-2060

[149] A. Gersho, R.M. Gray: Vector Quantization and Signal Compression, Kluwer Academic Publishers, 1992

[150] **L.Torres, J.Huguet** An Improvement on Codebook Search for Vector Quantisation, IEEE Tr. on Com., Vol 42, No 2, pp 208-210, Feb. 1994

[151] B. Ramamurthi, A. Gersho: Classified Vector Quantization of Images, IEEE Tr. on Com., Vol 31, No 11, pp 1105-1115, Nov. 1986

[152] L. Hanzo et al: A Portable Multimedia Communicator Scheme, pp 31-54, in R.I. Damper, W. Hall and J.W. Richards (Ed.) Multimedia Technologies and Future Applications, Pentech Press, London, 1993

[153] Sam Liu Feng-Ming Wang: Hybrid video coding for low bit-rate applications, Proc. of IEEE ICASSP, 19-22 April, 1994, pages 481–484

[154] J.F. Arnold X. Zhang, M.C. Cavenor: Adaptive quadtree coding of motion -compensated image sequences for use on the broadband ISDN, IEEE Transactions on Circuits and Systems for Video Technology, Vol 3,No 3, June 1993, pp 222–229

[155] Eli Shustermann and Meir Feder: Image Compression via Improved Quadtree Decomposition Algorithms, IEEE Transactions on Image Processing, Vol 3, No 2, March 1994, pp 207-215

[156] Proceedings of the International Workshop on Coding Techniques for Very Low Bit-rate Video, Hi-Vision Hall, Shinagawa, Tokyo, 8-10 Nov. 1995

[157] K.H. Tzou and H.G. Mussmann and K. Aizawa, (Ed.), "Special issue on very low bit rate video coding," *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 4, pp. 213–357, June 1994.

[158] N. Hubing (Ed.), "Speech and image coding," *Special Issue of the IEEE Journal on JSAC*, vol. 10, pp. 793–976, June 1992.

[159] B. Girod et al (Ed.), "Special issue on image sequence compression," *IEEE Transactions on Image Compression*, vol. 3, pp. 465–716, September 1994.

[160] **M.F Chowdhury et al**: A Switched Model-Based Coder for Video Signals, Special Issue of the IEEE Tr. on Circuits and Systems for Video Tech, June 1994, Vol 4, No. 3, pp 228-235

[161] **G. Bozdagi et al**: 3-D Motion Estimation and Wireframe Adaptation Including Photometric Effects for Model-Based Coding of Facial Image Sequences Special Issue of the IEEE Tr. on Circuits and Systems for Video Tech, June 1994, Vol 4, No. 3, pp 246-256

[162] **C.S. Choi et al** Analysis and Synthesis of Facial Image Sequences in Model-Based Image Coding Special Issue of the IEEE Tr. on Circuits and Systems for Video Tech, June 1994, Vol 4, No. 3, pp 257-275

[163] **H. Gharavi**: Subband coding of video signal, Chapter 6, pp 229-271 in J.W. Woods (Ed.) Subband image coding, Kluwer Academic Publishers, 1991

[164] **K. N. Ngan, W.L. Chooi**: Very low bit rate video coding using 3D subband approach, IEEE Tr. on Circuits and Systems for Video Technology, Vol. 4., No 3, June 1994, pp 309 316

[165] L. Hanzo, J. Streit: A Fractal Video Communicator, Proceedings of IEEE VTC '94, June 7-11, Stockholm, Sweden, pp 1030-1034

[166] M. Khansari, A. Jalali, E. Dubois, and P. Mermelstein, "Robust low bit-rate video transmission over wireless access systems," in *Proc. of International Comms. Conf. (ICC)*, pp. 571–575, 1994.

[167] N. Färber, E. Steinbach, and B. Girod, "Robust H.263 video transmission over wireless channels," in *Proc. of International Picture Coding Symposium (PCS)*, (Melborne, Australia), pp. 575–578, March 1996.

[168] E. Steinbach, A. Hanjalic, B. Girod: 3D motion and scene structure estimation with motion dependent distortion of measurement windows, Proc. of ICIP-96, 16-19 of Sept. Lausanne, Switzerland, Vol.1, pp 61-68

[169] M. Horn, B. Girod: Performance analysis of multiscale motion compensation techniques in pyramid coders, ibid, Vol. 3, pp 255-258

[170] E. Steinbach, N. Faerber, B. Girod: Standard compatible extension of H.263 for robust video transmission in mobile environments, IEEE Transactions on Circuits and Systems for Video Technology, to appear [3]

[171] B. Girod, E. Steinbach, N. Faerber: "Performance of the H.263 Video Compression Standard" Journal of VLSI Signal Processing: Systems for Signal, Image, and Video Technology. Special Issue on Recent Development in Video: Algorithms, Implementation and Applications. Invited paper, to appear in 1997 [4]

[172] B. Girod, N. Faerber, and E. Steinbach : "Standards Based Video Communication at Very Low Bitrates", Proc. VIII European Signal Processing Conference (EUSIPCO-96), Trieste, Italy, September 10-13, 1996

[173] N. Faerber, E. Steinbach, B. Girod: "Robust H.263 Compatble Transmission for Mobile Video Server Access", Proc. First International Workshop on Wireless Image/Video Communications, Loughborough University, UK, 4-5 September 1996, pp 8-13

[174] F. Eryurtlu, A.H. Sadka, A.M. Kondoz: Error robustness improvement of video codecs with two-way decodable codes, Electr. Letters, 2nd of Jan., 1997, Vol. 33, No. 1, pp 41-43

[175] A.H. Sadka, F. Eryurtlu, A.M. Kondoz: Improved performance H.263 under erroneous transmission conditions, Electr. Letters, 16th of Jan., 1997, Vol. 33, No. 2, pp 122-124

[176] P. Cherriman, L. Hanzo: Programmable H.263-based wireless video transceivers for interference-limited environments, to appear in IEEE Tr. on CSVT, May 1998

[177] P. Cherriman, L. Hanzo: H.261 and H.263 wireless videophone performance in iterference-limited scenarios, PIMRC'96, Taipei, Taiwan, 15-18 Oct., 1996, pp 158-162

[3] http://www-nt.e-technik.uni-erlangen.de/ girod/publications.html

[4] http://www-nt.e-technik.uni-erlangen.de/ girod/publications.html

[178] P. Cherriman, L. Hanzo: Power-controlled Multimode Video Telephony, submitted to IEEE Tr. on Vehicular Technology, 1997

[179] T. Keller, P. Cherriman, L. Hanzo: Orthogonal Frequency Division Multiplex transmission of H.263 encoded video over wireless ATM networks, Proceeding of ACTS Summit'97, Aalborg, Denmark, October, 1997, pp 276-281

[180] R. M. Pelz, "An unequal error protected px8 kbit/s video transmission for DECT," in *Vehicular Technology Conference*, pp. 1020–1024, IEEE, 1994.

[181] T. Chen, "A real-time software based end-to-end wireless visual communications simulation platform," in *Proc. of SPIE Conf. on Visual Communications and Image Processing*, pp. 1068–1074, 1995.

[182] K. Illgner and D. Lappe, "Mobile multimedia communications in a universal telecommunications network," in *Proc. of SPIE Conf. on Visual Communications and Image Processing*, pp. 1034–1043, 1995.

[183] Y. Zhang, "Very low bit rate video coding standards," in *Proc. of SPIE Conf. on Visual Communications and Image Processing*, pp. 1016–1023, 1995.

[184] H. Ibaraki et al, "Mobile video communication techniques and services," in *Proc. of SPIE Conf. on Visual Communications and Image Processing*, pp. 1024–1033, 1995.

[185] K. Watanabe et al, "A study on transmission of low bit-rate coded video over radio links," in *Proc. of SPIE Conf. on Visual Communications and Image Processing*, pp. 1025–1029, 1995.

[186] J. Streit: Digital Image Compression, Phd. Thesis, Dept. of Electr., Univ. of Southampton, 1996

[187] J. Streit, L. Hanzo: Vector-quantised low-rate cordless videophone systems, IEEE Tr. on Veh. Techn., Vol. 42, No.2, May 1997, pp 340-357

[188] L. Hanzo, J. Streit: Adaptive low-rate wireless videophone systems, IEEE Tr. on Video Technology, Aug. 1995, Vol. 5, No. 4, 305-319, ISSN 1051-8215

[189] J. Streit, L. Hanzo: Quadtree-based parametric wireless videophone systems, IEEE Tr. on Circuits and Systems for Video Technology, Apr. 1996, Vol. 6, No. 2, pp 225-237

[190] J. Streit, L. Hanzo: Comparative study of programmable-rate videophone codecs for existing and future multimode wireless systems, European Transactions on Telecommunications, Vol.8, Issue No 6, 1997

[191] A.K. Jain: Fundamentals of Digital Image Processing, Prentice-Hall, 1989

[192] P. Cherriman: Packet video communications, Phd Mini-thesis, Dept. of Electr., Univ. of Southampton, 1996, UK

[193] ITU-T, *ITU Recommendation H.263: Video coding for low bitrate communication.* approved in 1996

[194] M. W. Whybray and W. Ellis, "H.263 - video coding recommendation for PSTN videophone and multimedia," in *IEE Colloquium (Digest)*, pp. 6/1–6/9, IEE, England, Jun 1995.

[195] M. Khansari, A. Jalali, E. Dubois, and P. Mermelstein, "Low bit-rate video transmission over fading channels for wireless microcellular systems," *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 6, pp. 1–11, February 1996.

[196] J.C. Arnbak, J.H. Bons and J.W. Vieveen, "Graphical correspondence in electronic-mail networks using personal computers," IEEE Journal on Selected Areas in Communications, Feb. 1989, SAC-7, pp.257-267

[197] M. Klerk, R. Prasad, J.H. Bons and N.B.J. Weyland, "Introducing high-resolution line graphics in UK teletext using differential chain coding," IEE Proceedings, Part I, vol. 137, pp. 325-334, Dec. 1990.

[198] K. Liu and R. Prasad, "Performance analysis of differential chain coding, " European Transaction on Telecommunications and Related Technology, vol. 3, pp. 323-330, Jul-Aug. 1992

[199] D.L. Neuhoff and K.G. Castor, "A rate and distortion analysis of chain codes for line drawings, " IEEE Tran. Information Theory, vol. IT-31, pp. 53-68, Jan. 1985.

[200] A. B. Johannessen, R. Prasad, N.B.J. Weyland and J.H. Bons, "Coding efficiency of multiring differential chain coding, " IEE Proceedings, Part I, vol. 139, pp. 224-232, April 1992.

[201] R. Prasad, J.W. Vivien, J.H. Bons, J.C. Arnbak, "Relative vector probabilities in differential chain coded line drawings", Proc. of IEEE Pacific Rim Conference on Comms., Computers and Signal Processing, pp 138-142, Victoria, Canada, June 1989

[202] L. Yang, H.I. Cakil, R. Prasad: "On-line handwritting processing in indoor and outdoor radio environment for multimedia", Proc. of IEEE VTC'94, Stockholm, Sweden, 1015-1019

[203] R. Prasad, P.A.D. Spaargaren and J.H. Bons: "Teletext reception in a mobile channel for a broadcast teleinformation system", IEEE Tr. on VT., Vol. 42, No. 4, Nov. 1993, pp 535-545

[204] L-P.W. Niemel, R. Prasad: "A novel description of handwritten characters for use with generalised Fourier descriptors", ETT Vol. 3, No. 5, Sept-Oct. 1992, pp 455-464

[205] L-P.W. Niemel, R. Prasad: "An improved character description method based on generalized Fourier descriptors", ETT Vol. 5, No. 3, May-June 1994, pp 371-376

[206] H. Yuen, L. Hanzo: A Novel Coding Scheme for the Transmission of On-line Handwriting Proc. of Globecom'95, Singapore, 13-17 Nov. 1995, pp 2284-2288

[207] L. Hanzo, H. Yuen: Wireless multilevel graphical correspondence, European Tr. on Communications, No. 3, Vol. 8, May-June, 1997, pp 271-283

[208] C.C. Tappert, C.Y. Suen, and T. Wakahara: The state of the art in on-line handwriting recognition. IEEE Trans. on Pattern Analysis and Machine Intelligence, 12(8):787–808, 1990.

[209] R. Plamondon and G. Lorette: Automatic signature verification and writer identification — The state of the art. Pattern Recognition, 22(2):107–131, 1989.

[210] S.R. Veltman and R. Prasad: Hidden Markov models applied to on-line handwritten isolated character recognition. IEEE Trans. on Image Processing, 3(3):314–318, 1994.

[211] L. Yang and R. Prasad: On-line recognition of handwritten characters using differential angles and structural descriptors. Pattern Recognition Letters, 14(12):1019–1024, 1993.

[212] Mou-Yen Chen: Off-line handwritten word recognition using a hidden Markov model type stochastic network. IEEE Trans. on Pattern Analysis and Machine Intelligence, 16(5):481–496, 1994.

[213] L. Yang, B.K. Widjaja, and R. Prasad: Application of hidden Markov models for signature verification. Pattern Recognition, 28(2):161–170, 1995.

[214] Hanzo, L. and Steele, R., "The Pan-European Mobile Radio System, Part 1 and 2", *European Transactions on Telecommunications*, Vol. 5, No. 2, Mar.-Apr. 1994, pp 245-276

[215] Vary, P. and Sluyter, R.J., "MATS-D Speech Codec: Regular-Pulse Excitation LPC", *Proc. of Nordic Conference on Mobile Radio Communications*, 1986, pp 257-261

[216] J.D.Parsons "The mobile radio propagation channel", Pentech Press, London, 1992.

[217] D. Greenwood, L. Hanzo: Characterisation of Mobile Radio Channels, Chapter 2, pp 92-185 in R. Steele, Mobile Radio Communications, Pentech Press, 1992

[218] Goodman, D.J. and Wei, S.X., "Efficiency of packet reservation multiple access", *IEEE Transactions on vehicular technology*, vol. 40, No. 1, Febr. 1991, pp. 170-176.

[219] X. Qiu and V. K. Li, "Dynamic Reservation Multiple Access (DRAMA): A New Multiple Access Scheme for Personal Communication Systems (PCS)," *Wireless Networks*, vol. 2, pp. 117–128, 1996.

[220] David J. Goodman and Sherry X. Wei, "Efficiency of Packet Reservation Multiple Access," *IEEE Transactions. on Vehicular Technology*, vol. 40, pp. 170–176, Feb. 1991.

[221] Sanjiv Nanda, David J. Goodman, and Uzi Timor, "Performance of PRMA: A Packet Voice Protocol for Cellular Systems," *IEEE Transactions on Vehicular Technology*, vol. 40, pp. 584–598, Aug. 1991.

[222] J. DeVile, "A Reservation Multiple Access Scheme for an Adaptive TDMA Air-Interface," in *Proceedings of the Winlab Workshop, Rutgers, The State University of New Jersey*, Sep. 1993.

[223] J. Dunlop, D. Robertson, P. Cosimi and J. De Vile, "Development and Optimisation of a Statistical Multiplexing Mechanism for ATDMA," in *Proc. IEEE 44th Vehicular Technology Conference, Stockholm*, pp. 1040 – 1044, June 1994.

[224] N. Amitay and S. Nanda, "Resource Auction Multiple Access (RAMA) for Statistical Multiplexing of Speech in Wireless PCS," *IEEE Transactions on Vehicular Technology*, vol. 43, pp. 584–595, Aug. 1994.

[225] Frans J. Panken, "Multiple-Access Protocols Over the Years: a Taxonomy and Survey," in *1996 IEEE International Conference on Communication Systems (ICCS)*, pp. 2.1.1–2.1.5, Nov. 1996.

[226] J. Brecht, L. Hanzo, M. Del Buono: Multi-frame Packet Reservation Multiple Access for Variable-rate Users, Proc. of PIMRC'97, 1-4. Sept. 1997, Helsinki, Finland, pp 430-438

[227] J. Brecht, M. Del Buono, L. Hanzo: Multi-frame Packet Reservation Multiple Access Using Oscillation-scaled Histogram-based Markov Modelling of Video Codecs, to appear in Image Communications, Elsevier, Apr. 1998

[228] J. Brecht, L. Hanzo: Statistical Packet Assignment Multiple Access for Wireless Asynchronous Transfer Mode Systems, Proceeding of ACTS Summit'97, Aalborg, Denmark, October, 1997, pp 734-738

[229] J. Williams, L.Hanzo, R.Steele, J.C.S Cheung: A comparative study of microcellular speech transmission schemes, IEEE Tr. on Veh. Technology, Nov. 1994, Vol. 43, No. 4 pp 909-924

[230] J.C.S. Cheung, L. Hanzo, W.T. Webb, R. Steele: Effects of Packet Reservation Multiple Access on Objective Speech Quality, Electr. Letters, 21. Jan. 1993, Vol. 29, No. 2, pp 152-153

[231] L. Hanzo, J.C.S. Cheung, R. Steele, W.T. Webb: Performance of PRMA Schemes via Fading Channels, Proc. of IEEE VTC'93, NJ. USA, May 1993 pp 913-916

[232] M. Eastwood, L. Hanzo: Packet reservation multiple access for wireless multimedia communications, Electr. Letters, 24th of June, 1993, Vol. 29, No 13, pp 1178-1179

[233] L. Hanzo, J.C.S. Cheung, R. Steele: PRMA efficiency in adaptive transceivers, Electronics Letters, 15 Apr 1993, Vol 29, No 8, pp 697-698

[234] C.N. Campopiano, B.G. Glazer: A Coherent Digital Amplitude and Phase Modulation Scheme, IRE Trans. Commun. Systems, Vol CS-10, 1962, pp 90-95

[235] G.D. Forney et al.: Efficient Modulation for Band-Limited Channels, IEEE Journal on Selected Areas in Communications, Vol SAC-2, No 5, Sept. 1984, pp 632-647

[236] K. Feher: Modems for Emerging Digital Cellular Mobile Systems, IEEE Tr. on VT, Vol 40, No 2, pp 355-365, May 1991

[237] M. Iida, K. Sakniwa: Frequency Selective Compensation Technology of Digital 16-QAM for Microcellular Mobile Radio Communicatin Systems, Proc. of VTC'92, Denver, Colorado, pp 662-665

[238] R.J. Castle, J.P. McGeehan: A Multilevel Differential Modem for Narrowband Fading Channels, Proc. of VTC'92, Denver, Colorado, pp 104-109

[239] D.J. Purle, A.R. Nix, M.A. Beach, J.P. McGeehan: A Preliminary Performance Evaluation of a Linear Frequency Hopped Modem, Proc. of VTC'92, Denver, Colorado, pp 120-124

[240] Y. Kamio, S. Sampei: Pervormance of Reduced Complexity DFE Using Bidirectional Equalizing in Land Mobile Communications, Proc. of VTC'92, Denver, Colorado, pp 372-376

[241] T. Nagayasu, S. Sampei, Y. Kamio: Performance of 16-QAM with Decision Feedback Equalizer Using Interpolation for Land Mobile Communications, Proc. of VTC'92, Denver, Colorado, pp 384-387

[242] E. Malkamaki: Binary and Multilevel Offset QAM, Spectrum Efficient Modulation Schemes for Personal Communications, Proc. of VTC'92, Denver, Colorado, pp 325-378

[243] Z. Wan, K. Feher: Improved Efficiency CDMA by Constant Envelope SQAM, Proc. of VTC'92, Denver, Colorado, pp 51-55

[244] H. Sasaoka: Block Coded 16-QAM/TDMA Cellular Radio System Using Cyclical Slow Frequency Hopping, Proc. of VTC'92, Denver, Colorado, pp 405-408

[245] W.T. Webb, L. Hanzo, R. Steele: Bandwidth-Efficient QAM Schemes for Rayleigh-Fading Channels, IEE Proceedings, Vol 138, No 3, June 1991, pp 169-175

[246] L. Hanzo, R. Steele, P.M. Fortune: A Subband Coding, BCH Coding and 16-QAM System for Mobile Radio Speech Communications, IEEE Vehicular Technology, Nov., 1990, Vol 39, No 4, pp 327-339

[247] P.M. Fortune, L. Hanzo, R. Steele: On the Computation of 16-QAM and 64-QAM Performance in Rayleigh-Fading Channels, IEICE Tr. on Comms., Vol E75-B, No 6, June 1992, pp 466-475

[248] A.S. Wright, W.G. Durtler: Experimental Performance of an Adaptive Digital Linearized Power Amplifier, IEEE Tr. on VT, Vol 41, No 4, pp 395-400, Nov. 1992

[249] M. Faulkner, T. Mattson: Spectral Sensitivity of Power Amplifiers to Quadrature Modulator Misalignment, IEEE Tr. on VT, Vol 41, No 4, pp 516-525, Nov. 1992

[250] P.B. Kennington et al: Broadband Linear Amplifier Design for a PCN Base-Station, Proc. of 41st IEEE VTC, pp 155-160, May 1991

[251] R.J. Wilkinson et al: Linear Transmitter Design for MSAT Terminals, 2nd Int. Mobile Satellite Conference, June 1990

[252] S.P. Stapleton, F.C. Costescu: An Adaptive Predistorter for a Power Amplifier Based on Adjacent Channel Emissions, IEEE Tr. on VT, Vol 41, No 1, pp 49-57, Febr. 1992

[253] S.P. Stapleton, F.C. Costescu: An Adaptive Predistorter for a Power Amplifier Based on Adjacent Channel Emissions, IEEE Tr. on VT, Vol 41, No 1, pp 49-57, Febr. 1992

[254] S.P. Stapleton, G.S. Kandola, J.K. Cavers: Simulation and Analysis of an Adaptive Predistorter Utilizing a Complex Spectral Convolution, IEEE Tr. on VT, Vol 41, No 4, pp 387-394, Nov. 1992

[255] J.K. Cavers: An Analysis of Pilot Symbol Assisted Modulation for Rayleigh Fading Channels, IEEE Tr. on VT, Vol 40, No 4, Nov. 1991, pp 686-693

[256] A.Bateman and J.P.McGeehan, "Feedforward transparent tone in band for rapid fading protection in multipath fading", *IEE Int. Conf. Comms. Vol.68, 1986*, pp9-13.

[257] A.Bateman, "Feedforward transparent tone in band: Its implementation and applications", *IEEE Trans. Veh. Tech. Vol.39, No.3*, Aug 1990, pp235-243.

[258] J.K.Cavers, "The performance of phase locked transparent tone in band with symmetric phase detection", *IEEE Trans. on Comms., Vol.39, No.9*, Sept 1991, pp1389-1399.

[259] R. Steele and W.T. Webb, "Variable rate QAM for data transmission over Rayleigh fading channels," in *Wireless '91, Calgary, Alberta*, pp. 1–14, IEEE, 1991.

[260] W. Webb and R. Steele, "Variable rate QAM for mobile radio", "IEEE Transactions on Communications", 1995, Vol. 43, No. 7, pp 2223-2230

[261] Y. Kamio, S. Sampei, H. Sasaoka and N. Morinaga, "Performance of modulation-level-control adaptive-modulation under limited transmission delay time for land mobile communications," in 45^{th} *Vehicular Technology Conference*, pp. 221–225, IEEE, 1995.

[262] S. Sampei, S. Komaki and N. Morinaga, Adaptive Modulation/TDMA Scheme for Large Capacity Personal Multi-Media Communication Systems, IEICE Transactions on Communications, 1994, Vol. 77. No. 9, pp 1096-1103

[263] M. Morimoto, H. Harada, M. Okada and S. Komaki, A study on power assignment of hierarchical modulation schemes for digital broadcasting, IEICE Transactions on Communications, 1994, Vol. 77, No. 12, pp 1495-1500

[264] S. Otsuki, S. Sampei and N. Morinaga, "Square-QAM adaptive modulationTDMA/TDD systems using modulation level estimation with Walsh function." Electronics Letters, November 1995. pp 169-171.

[265] Soon-Ghee Chua and Andrea Goldsmith, "Variable-rate variable-power mqam for fading channels," in 46^{th} *Vehicular Technology Conference*, pp. 815–819, IEEE, 1996.

[266] D. A. Pearce, A. G. Burr and T. C. Tozer, "Comparison of counter-measures against slow Rayleigh fading for TDMA systems," in *Colloquium on Advanced TDMA Techniques and Applications*, pp. 9/1–9/6, IEEE, 1996.

[267] J. M. Torrance and L. Hanzo, "Upper bound performance of adaptive modulation in a slow Rayleigh fading channel." Electronics Letters, April 1996. pp 169-171.

[268] J. M. Torrance and L. Hanzo, "Adaptive modulation in a slow Rayleigh fading channel," in *Proceedings of the* 7^{th} *Personal, Indoor and Mobile Radio Communications (PIMRC) Conference*, pp. 497–501, IEEE, 1996.

[269] J. M. Torrance and L. Hanzo, "Optimisation of switching levels for adaptive modulation in a slow Rayleigh fading channel." Electronics Letters, June 1996. pp 1167 - 1169.

[270] J.M. Torrance, L. Hanzo: Demodulation level selection in adaptive modulation, Electr. Letters, Vol. 32, No. 19, 12th of Sept., 1996, pp 1751-1752

[271] J. Torrance, L. Hanzo: Latency considerations for adaptive modulation in slow Rayleigh fading, Proc. of IEEE VTC'97, Phoenix, USA, 1997, pp 1204-1209

[272] J. Torrance, L. Hanzo: Statistical Multiplexing for Mitigating Latency in Adaptive Modems, PIMRC'97, 1-4. Sept. 1997, Helsinki, Finland, pp 938-942

[273] M-S Alouini, A. Goldsmith, Area spectral efficiency of cellular mobile radio systems, Proc. of IEEE VTC'97, Phoenix, USA, pp 652-656

[274] M-S Alouini, A. Goldsmith, Capacity of Nakagami Multipath Fading Channels, Proc. of IEEE VTC'97, Phoenix, USA, pp 652-656

[275] K.R.Narayanan and L.J. Cimini Jr, "Equalizer Adaptation Algorithms for High Speed Wireless Communications," Proceedings of IEEE VTC '96, pp. 681 – 685, 1997.

[276] J. Wu and A. H. Aghvami, "A New Adaptive Equalizer with Channel Estimator for Mobile Radio Communications," IEEE Transactions on Vehicular Technology, vol. 45, pp. 467–474, August 1996.

[277] Y. Gu and T. Le-Ngoc, "Adaptive Combined DFE/MLSE Techniques for ISI Channels," IEEE Transactions on Communications, vol. 44, pp. 847–857, July 1996.

[278] A.P.Clark and R.Harun, "Assessment of Kalman-filter channel estimators for an HF Radio Link," IEE Proceedings, vol. 133, pp. 513–521, October 1986.

[279] R.W. Chang: "Synthesis of Band-Limited Orthogonal Signals for Multichannel Data Transmission", BSTJ, Dec. 1966

[280] M.S. Zimmermann and A.L. Kirsch: "The AN/GSC-10 /KATHRYN/ Variable Rate Data Modem for HF Radio", IEEE Trans. Commun. Techn., vol. CCM–15, no. 2, April 1967

[281] E.N. Powers and M.S. Zimmermann: "A Digital Implementation of a Multichannel Data Modem", Proc. of the IEEE Int. Conf. on Commun., Philadelphia, 1968

[282] B.R. Saltzberg: "Performance of an Efficient Parallel Data Transmission System", IEEE Trans. Commun. Techn., December 1967

[283] R.W. Chang and R.A. Gibby: "A Theoretical Study of Performance of an Orthogonal Multiplexing Data Transmission Scheme", IEEE Trans. Commun. Techn., vol. COM–16, no. 4, August 1968

[284] S.B. Weinstein and P.M. Ebert: "Data Transmission by Frequency Division Multiplexing Using the Discrete Fourier Transform", IEEE Trans. Commun. Techn., vol. COM–19, no. 5, October 1971

[285] Peled and A. Ruiz: "Frequency Domain Data Transmission Using Reduced Computational Complexity Algorithms", Proc. of the ICASSP, pp. 964–967, 1980

[286] B. Hirosaki: "An Orthogonally Multiplexed QAM System Using the Discrete Fourier Transform", IEEE Trans. Commun., vol. COM–29, no. 7, July 1981

[287] H.J. Kolb: "Untersuchungen über ein digitales Mehrfrequenzverfahren zur Datenübertragung", Ausgewählte Arbeiten über Nachrichtensysteme, Universität Erlangen-Nürnberg, no. 50

[288] H.W. Schüssler: "Ein digitales Mehrfrequenzverfahren zur Datenübertragung", Professoren-Konferenz, Stand und Entwicklungsaussichten der Daten und Telekommunikation, Darmstadt, Germany, pp. 179–196, 1983

[289] J. Cimini, "Analysis and Simulation of a Digital Mobile Channel Using Orthogonal Frequency Division Multiplexing", IEEE Trans. Commun., vol. COM–33, no. 7, pp. 665–675, July 1985

[290] K. Preuss: "Ein Parallelverfahren zur schnellen Datenübertragung Im Ortsnetz", Ausgewählte Arbeiten über Nachrichtensysteme, Universität Erlangen-Nürnberg, no. 56

[291] R. Rückriem: "Realisierung und messtechnische Untersuchung an einem digitalen Parallelverfahren zur Datenübertragung im Fernsprechkanal", Ausgewählte Arbeiten über Nachrichtensysteme, Universität Erlangen-Nürnberg, no. 59

[292] F.Mueller-Roemer, "Directions in audio broadcasting", Jnl Audio Eng. Soc., Vol.41, No.3, March 1993, pp158-173.

[293] G.Plenge: "DAB - A new radio broadcasting system - state of development and ways for its introduction", Rundfunktech. Mitt., Vol.35, No.2, 1991, pp45ff.

[294] M.Alard and R.Lassalle: "Principles of modulation and channel coding for digital broadcasting for mobile receivers", EBU Review, Technical No. 224, Aug 1987, pp47-69.

[295] Proc. 1st Int. Symp., "DAB", Montreux, Switzerland, June 1992.

[296] I.Kalet: "The multitone channel", IEEE Tran. on Comms, Vol.37, No.2, Feb 1989, pp119-124.

[297] J.P Woodard, T. Keller and L. Hanzo Turbo-coded Orthogonal Frequency Division Multiplex transmission of 8 kbps encoded speech, Proceeding of ACTS'97, Oct. 1997, Aalborg, Denmark, pp 894-899

[298] T. Keller, J.P. Woodard, L. Hanzo: Turbo-coded parallel modem techniques for Personal Communications, Proc. of IEEE VTC'97, Phoenix, USA, 1997, pp 2158-2162

[299] T. Keller, L. Hanzo: Orthogonal Frequency Division Multiplex Synchronisation Techniques for Wireless Local Area Networks, Proc. of Personal, Indoor and Mobile Radio Communications, PIMRC'96, Taipei, Taiwan, 15-18 Oct., 1996, pp 963-967

[300] F. Daffara and O. Adami, "A new frequency detector for orthogonal multicarrier transmission techniques," in *IEEE 45th Vehicular Technology Conference*, (Chicago), pp. 804–809, IEEE, July 15–28 1995.

[301] Magnus Sandell, Jan-Jaap van de Beek, Per Ola Börjesson, "Timing and frequency synchronisation in OFDM systems using the cyclic prefix," in *International Symposium on Synchronisation*, (Essen, Germany), pp. 16–19, Dec. 14–15 1995.

[302] R.V. Cox, J. Hagenauer, N. Seshadri, C.E.W. Sundberg: Subband Speech Coding and Matched Convolutional Channel Coding for Mobnile Radio Channels, IEEE Tr. on Sig. Proc., Vol 39, No 8, Aug. 1991, pp 1717-1731

[303] J. Hagenauer: Rate-Compatible Punctured Convolutional Codes (RCPC) and Their Applications, IEEE Tr. on Comms., Vol 36, No 4, Apr. 1988, pp 389-400

[304] R.A. Salami, K.H.H. Wong, R.Steele, D. Appleby: Performance of Error-Protected Binary Pulse Excitation Coders at 11.4 Kb/s over Mobile Radio Channels, Proc. of ICASSP'90, pp 473-476

[305] G. Ungerboeck: Trellis-Coded Modulation with Redundant Signal Sets, Part I and II, IEEE Comm's Mag., Vol 25, No 2, pp 5-21, Febr. 1987

[306] E. Biglieri, M. Luise: Coded Modulation and Bandwidth-Efficient Transmission, Proc. of the Fifth Tirrenia Intern. Workshop, 8-12 Sept. 1991, Elsevier, Netherlands, 1992

[307] IEEE Communications Magazine, Special Issue on Coded Modulation, Vol 29, No 12, Dec. 1991

[308] J. Hagenauer: Iterative decoding of binary block and convolutional codes, IEEE Tr. on Inf. Theory, 1995

[309] L.F. Wei: Coded modulation with unequal error protection, IEEE Tr. on Comms., Oct. 1993, vol 41, No.10, pp 1439-1450

[310] C. Berrou, A. Glavieux, and P. Thitimajshima, "Near Shannon Limit Error-Correcting Coding and Decoding: Turbo-Codes", Proc. ICC 1993, pp. 1064-1070

[311] A.J. Viterbi, "Error Bounds for Convolutional Codes and an Asymptotically Optimum Decoding Algorithm", IEEE Trans. Inform. Theory, Vol. 13, 1967, pp. 260..269

[312] J. Hagenauer and P. Hoeher, "A Viterbi algorithm with soft-decision outputs and its applications," in *IEEE GLOBECOM'89*, (Dallas, Texas), pp. 1680–1686, 1989.

[313] J. Hagenauer, E. Offer, and L. Papke, "Iterative decoding of binary block and convolutional codes," *IEEE Transactions on Information Theory*, vol. 42, no. 2, pp. 429–445, 1996.

[314] J. Hagenauer, P. Robertson, and L. Papke, "Iterative ("TURBO") decoding of systematic convolutional codes with the MAP and SOVA algorithms," in *ITG Fachtagung, Codierung für Quelle, Kanal und Übertrager*, (Muenchen), pp. 21–29, 1994.

[315] P. Robertson, E. Villebrun, and P. Hoeher, "A comparison of optimal and sub-optimal MAP decoding algorithms operating in the log domain," in *IEEE Intern. Conference on Communications*, pp. 1009–1013, 1995.

[316] P. Jung, "Comparison of turbo-code decoders applied to short frame transmission systems," *IEEE Journal on Selected Areas in Communications*, vol. 14, no. 3, pp. 530–537, 1996.

[317] P. Jung and M. Naßhan, "Performance evaluation of turbo codes for short frame transmission systems," *IEE Electronics Letters*, vol. 30, no. 2, pp. 111–112, 1994.

[318] P. Jung and M. Naßhan, "Dependence of the error performance of turbo-codes on the interleaver structure in short frame transmission systems," *IEE Electronics Letters*, vol. 30, no. 4, pp. 287–288, 1994.

[319] A. S. Barbulescu and S. S. Pietrobon, "Interleaver design for turbo codes," *IEE Electronics Letters*, vol. 30, no. 25, pp. 2107–2108, 1994.

[320] O. Joerssen and H. Mayr, "Terminating the trellis of turbo-codes," *IEE Electronics Letters*, vol. 30, no. 16, pp. 1285–1286, 1994.

[321] P. Robertson and T. Wörz, "Coded modulation scheme employing turbo codes," *IEE Electronics Letters*, vol. 31, no. 18, pp. 1546–1547, 1995.

[322] P. Robertson and T. Wörz, "A novel bandwidth efficient coding scheme employing turbo codes," in *ICC'96*, 1996. accepted.

[323] S. Benedetto, D. Divsalar, B. Montorsi, and F. Pollara, "Bandwidth efficient parallel concatenated coding schemes," *IEE Electronics Letters*, vol. 31, no. 24, pp. 2067–2069, 1995.

[324] M. Breiling, L. Hanzo: Non-iterative Optimum Supertrellis Decoding of Turbo Codes, IEE Electr. Letters, 8th May, 1997, Vol. 33, No. 10, pp 848-849

[325] M. Breiling, L. Hanzo: Optimum Non-iterative Decoding of Turbo Codes, submitted to IEEE Tr. on Inf. Theory, 1997

[326] M. Breiling, L. Hanzo: Optimum Non-iterative Turbo-Decoding, Proc. of PIMRC'97, Helsinki, Finland, 1-4. Sept. 1997

[327] D. Didascalou: The potential of Turbo codes in trellis coding schemes, Diploma Project Report, The University of Southampton, UK, 1996

[328] M. Breiling: Turbo coding simulation studies, Diploma Project Report, The University of Southampton, UK, 1997

[329] J. Brecht: Medium access control for multirate, multimedia wireless systems, Diploma Project Report, The University of Southampton, UK, 1997

[330] GSM Recommendation 05.05, Annex 3, pp.13-16, November 1988.

[331] R.Edwards and J.Durkin, "Computer prediction of service area for VHF mobile radio networks", *Proc IRE 116 (9)* pp.1493-1500, 1969.

[332] J. Cheng, R. Steele: Modified Viterbi Equaliser for Mobile Radio Channels Having Large Multi-Path Delay, Electronics Letters, Vol 25, No 19, pp 1309-1311

[333] N.S. Hoult, C.A. Dace, A.P. Cheer: Implementation of an Equaliser for the GSM System, Proc. of the 5th Int. Conf. on Radio Receivers Associated Systems, 24-26 July, 1990, Cambridge, U.K.

[334] R.D'Avella, L. Moreno, M. Sant'Agostino: An Adaptive MLSE Receiver for TDMA Digital Mobile Radio, IEEE Journal on Selected Areas in Communications, Vol 7, No 1, Jan. 1989, pp 122-129

[335] J.C.S. Cheung: Adaptive Equalisers for Wideband Time Division Multiple Acess Mobile Radio, Ph.D. Thesis, Dept. Of ECS, Univ. of Southampton, UK

[336] M.R.L. Hodges, S.A. Jensen, P.R. Tattersall: Laboratory Testing of Digital Cellular Radio Systems, BTRL Journal, Vol 8, No 1, Jan. 1990, pp 57-66

[337] M. Hata, "Empirical Formula for Propagation Loss in Land Mobile Radio", *IEEE Trans. VT-29*, pp. 317-325, August 1980

[338] Y. Okumura, E. Ohmori, T. Kawano and K. Fukuda, "Field Strength and its Variability in VHF and UHF Land Mobile Service", *Review of the Electrical Communication Laboratory, Vol 16*, pp. 825-873, Sept.-Oct. 1968

[339] J. Williams, L. Hanzo, R. Steele: Channel-adaptive voice communications, Proc. of IEE RRAS'95 Conference, 26-28 Sept., 1995, Bath, UK, Conf. Public. No. 415, pp 144-147

[340] L.Hanzo, et al: A packet reservation multiple access assisted cordless telecommunications scheme, IEEE Tr on VT., May 1994, vol. 43, No. 2, pp 234-245

[341] W.C. Wong: Dynamic allocation of packet reservation multiple access carriers, IEEE Tr. on VT, Vol. 42, No 4, NOv. 1993, pp 385-392

[342] W.C.Y.Lee: Spectrum efficiency in cellular, IEEE Trans on Vehicular Tech., Vol.38, No.2, May 1989, pp69-75.

[343] T. Keller: Orthogonal frequency division multiplex techniques for wireless local area networks, Internal Report, Dept. of Electr. and Comp. Sc., Univ. of Southampton, UK, 1996

[344] P. Cherriman, T. Keller, L. Hanzo: Orthogonal Frequency Division Multiplex transmission of H.263 encoded
video over highly frequency-selective wireless networks `http://www-mobile.ecs.soton.ac.uk/peter/robust-h263/robust.html`, submitted to IEEE Tr. on CSVT, 1997

[345] S. Pietrobon: Implementation and performance of a Turbo/Map decoder, submitted to the International Journal of Satellite Communications, 21st Feb. 1997

*Glossary

ACI Adjacent Channel Interference

ACTS Advanced Communications Technologies and Services

ADPCM Adaptive Differential Pulse Code Modulation

AGC Automatic Gain Control

AMPS Advanced Mobile Phone System

ATM Asynchronious Transfer Mode

AWGN Additive White Gaussian Noise

B-ISDN Broadband-ISDN

BCH Bose-Chaudhuri-Hocquenghem, A class of forward error correcting codes (FEC)

BER Bit error rate, the number of the bits received incorrectly

BPS Bits per Symbol

BS Base Station

CC Chain Coding

CCI Co-Channel Interference

CDMA Code Division Multiple Access

CELP Code Excited Linear Prediction

CIF Common Intermediate Format Frames containing 352 pixels vertically and 288 pixels horizontally

CSI Channel State Information

CT Cordless Telephone

CT2 British Cordless Telephone System

DAB Digital Audio Broadcasting

DAMPS Digital Advanced Mobile Phone System

DCA Dynamic Channel Allocation

DCC Differential Chain Coding

DCT A discrete cosine transform, transforms data into the frequency domain. Commonly used for video compression by removing high frequency components in the video frames

DECT Digital European Cordless Telephone

DFT Discrete Fourier Transform

DoD US Department of Defence

DRI Decoder Reliability Information

DRMA Dynamic Reservation Multiple Access

DSP Digital Siganl Processing

FDMA Frequency Division Multiple Access

FEC Forward Error Correction

FFT Fast Fourier Transform

FL-DCC Fixed-length Differential Chain Coding

FM Frequency Modulation

FPLMTS Future Public Land Mobile Telecommunications System

FS	Fixed Station
FV	Fixed-length Vector
G.722	7 kHz bandwidth wideband speech coding standard
G.728	ITU 16 kbps speech coding standard
G.729	ITU 8 kpbs speech coding standard
GOS	Grade of Service
GSM	A Pan-European digital mobile radio standard, operating at 900MHz.
H.263	A video coding standard[193], due to be published by the ITU in 1996
HIPERLAN	HIPERformance Local Area Network
HMM	Hidden Markov Model
I	In-phase component in modulation
IS-54	The Pan-American DAMPS TDMA Mobile Radio Standard
IS-95	The Pan-American CDMA Mobile Radio Standard
ISDN	Integrated Services Digital Network, digital replacement of the analogue telephone network
ISI	Inter Symbol Interference
ITU	International Telecommunications Union, formerly the CCITT, standardisation group
IZFPE	Interpolated Zinc Function Pulse Excitation
LAN	Local Area Network
LCD	Liquid Christal Display
LOS	Line of Sight
LSB	Least Significant Bit
MAC	Multiple Access
MAP	Maximum Aposteriory
MBE	Multi Band Excitation
MC	Motion Compensation
MCER	Motion Compensated Error Residual
MELP	Mixed Excitation Linear Prediction
MF-PRMA	Multi-frame Packet Reservation Multiple Access
MLH-CR	Maximum Likelihood Correlation Receiver
MLSE	Maximum Likelihood Sequence Estimation
MPEG	Motion Picture Expert Group, also a video coding standard designed by this group that is widely used
MS	Mobile Station
MSB	Most Significant Bit
MSC	Mobile Switching Centre
MV	Motion Vector, a vector to estimate the motion in a frame
NAMTS	Nippon Mobile Telephone System
NMT	Nippon Mobile Telephone
NTT	Nippon Telegraph and Telephone Company
OFDM	Orthogonal Frequency Division Multiplexing
PCM	Pulse Code Modulation
PCN	Personal Communications Network
PCS	Personal Communications System
PD	Pen-Down
PDC	Personal Digital Cellular
PHP	Personal Handy Phone
PLMR	Public Land Mobile Radio
PRMA	Packet Reservation Multiple Access
PS	Portable Station

PSAM	Pilot symbol assisted modulation, a technique where known symbols (pilots) are transmitted regularly. The effect of channel fading on all symbols can then be estimated by interpolating between the pilots
PSI	Pitch Synchroneous Innovation
PU	Pen-Up
PWI	Prototype Waveform Interpolation
Q	Quadrature-phase component in modulation
QAM	Quadrature Amplitude Modulation
QCIF	Quarter Common Intermediate Format Frames containing 176 pixels vertically and 144 pixels horizontally
QT	Quad Tree
RACE	Research in Advanced Communications Equipment
RAMA	Resource Auction Multiple Access
RF	Radio Frequency
RPE	Regular Pulse Excitation
RSC	Recursive Systematic Convolutional Code
SBC	Sub Band Coding
SDI	Soft Decision Information
SIR	Signal to Interference ratio
SNR	Signal to Noise Ratio, noise energy compared to the signal energy
SOVA	Soft Output Viterbi Algorithm
SPAMA	Statistical Packet Assignment Multiple Access
SSI	Sourse Significance Information
SV	Starting Vector
T.150	ITU handwriting coding standard
TACS	Total Access Communications System
TCM	Trellis Coded Modulation
TDMA	Time Division Multiple Access
TS	Transmission Scheme
TTIB	Transparent Tone in Band
UMTS	Universal Mobile Telecommunications System
VAD	Voice Activity Detection
VC	Vector Count
VQ	Vector Quantisation
VSELP	Vector Sum Excited Linear Prediction
WATM	Wireless Asynchronous Transfer Mode
WLAN	Wireless Local Area Network

Lajos Hanzo (http://www-mobile.ecs.soton.ac.uk) graduated in Electronics in 1976 and in 1983 he was conferred a Phd. During his 20-year career in telecommunications he has held various research and academic posts in Hungary, Germany and the UK. Since 1986 he has been with the Department of Electronics and Computer Science, University of Southampton, UK and has been a consultant to Multiple Access Communications Ltd., UK. Currently he holds a chair in Telecommunications. He co-authored two books and over 200 research contributions, organised and chaired conference sessions, presented overview lectures and was awarded a number of distinctions. Currently he is managing a research team, working on a range of research projects in the field of wireless multimedia communications under the auspices of the Engineering and Physical Sciences Research Council (EPSRC) UK, the European Advanced Communications Technologies and Services (ACTS) Programme and the Mobile Virtual Centre of Excellence (VCE). For further information on research in progress and associated publications please refer to the above www home-page.