Chapter 3

Communication and Modulation

3.1 Introduction

Surprisingly, the earliest paper in this chapter originates from as late as 1986, testifying that the ”transmission and modulation community” within the Benelux at first did not identify itself with the WIC. Actually, the advent of coded modulation and the interest in modulation issues of people having a background in coding and information theory, led to a growing stream of WIC papers in this field. Also upcoming industrial applications like digital storage and transmission in the eighties (e.g., CD, GSM and DAB), stimulated research and publications within the WIC.

The chapter on Communication and Modulation is subdivided into the sections Transmission, Recording and Networking. The papers in each section are rubricated according to their subject. Within each subject the papers are dealt with in chronological order, so that the developments in time of the subjects can be followed (in sofar as results have been published within the WIC). Background information and extensive bibliographies can be found in standard texts like [2, 5, 3, 4].

3.2 Transmission

The section Transmission is subdivided into the subjects Coded Modulation, Single Carrier Systems and OFDM (multi-carrier or multi-tone systems). Coded modulation [1, 6] found and finds its main applications in transmission systems, where the channel is known (due to soundings) relatively well to both the transmitter and receiver and which needs to have a high spectral efficiency, e.g., by now classical modems (19.6 kbit/s) and other cable transmission systems such as ADSL and DVB-C. Within the Benelux, research in this particular field was mainly academic of nature. On the other hand, communication-theoretic aspects of single carrier systems (among which we also count digital optical communication), channel estimation, equalization and synchronization issues were and are of interest to a wide-spread community within the Benelux, which began to see the WIC as a forum for which they could present the more theoretical results. OFDM [8, 9] was studied because of its applications, first in DAB (Digital Audio Broadcast) and later in DVB-T (Terrestrial Digital Video Broadcast), where these types of modulation systems, in combination with appropriate channel coding systems, are used for efficiently transmitting digital information via a frequency-selective (broadcast) channel. Also for cable transmission, (trellis-coded) OFDM is used, but this did not lead to a WIC paper. By the end of the nineties, we see OFDM also being used in WLAN systems such as the IEEE802.11a and upcoming MIMO systems.

3.2.1 Coded Modulation

In 1988, Dekker and Smit [207] first explain that a hexagonal packing of signal points achieves asymptotically a 0.58 dB gain w.r.t. a rectangular signal set because of the denser packing of signal points in D2. Next, they consider trellis-coded modulation (TCM) using a 4-dimensional lattice D4. As in [1], they find that doubling the number of signal points, combined with a set-partitioning approach, where the last 2 bits are encoded using a convolutional encoder leads to approximately 3 dB coding gain on an AWGN channel.

In 1990, a low-complexity approach is taken by De Bot and Vinck [210], achieving basically also an asymptotic coding gain of 3dB on an AWGN channel. An example explaining their idea applied to 4-PSK works as follows. First, double the number of signal points by taking 8-PSK. Next, partition a block of m 8-PSK symbols into an even and an odd set of m 4-PSK symbols each, where the odd set differs from the even set by a rotation of $\pi/4$ for each symbol. A total of $2m$ user bits is transmitted using these m symbols, where the coding is realized by picking the even set if the parity of the $2m$ user bits is even, and picking the odd set otherwise. Note that the intra-set Euclidean distance in each set is $\sqrt{2}$ larger than for 4-PSK because the parity in each set is prescribed. It turns out that the Euclidean distance between the sets is at least as large as the intra-set distance for $m \geq 8$.

In 1995, De Bart and Willems [231] introduce enumerative techniques for obtaining shaping gain and simultaneously combat intersymbol interference in a PAM signaling scheme. As trellises are being used, this shaping technique can be combined with error correcting codes, thus providing both coding and shaping gain. The computational complexity is rather high.

Bargh and Schalkwijk [233], present in 1997 an extension of low rate noiseless feedback coding strategies for the BSC to AWGN channels, in order to achieve coding gain as in coded modulation. They consider sequences of transmitted QAM symbols, using a set partitioning along each of the transmitted dimensions. In traditional coded modulation, the ”weakest” bits may be protected by a distance providing code while the ”strong” bits remain uncoded. Similarly, the authors propose to apply a temporal binary feedback coding strategy on the weakest bit in each dimension in order to ensure a reliable decision for these weak bits, while the remaining bits are uncoded, thus aiming at a coding gain of 6 dB. The main advantage claimed is an enormous complexity reduction compared to traditional coded modulation for a comparable performance. Of course, the existence of a virtually error-free feedback channel is required.

Peek [239], introduces in 1999 multirate block codes which simultaneously may provide spectral shaping, Hamming distance and change of sampling frequency. The input x to the coding system is assumed to be a binary string $x_i \in \{-1, +1\}$, which is partitioned into blocks of equal size L. Each such block is multiplied by a $K \times L$ matrix A, which is $\{-1, +1\}$-nonsingular, to obtain a coded output block of L symbols, where the output alphabet depends on A. Depending on the column properties of A, one can enforce spectral nulls, e.g., at zero frequency or the Nyquist frequency. It turns out that such spectral nulls may lead to an increased minimum Hamming distance between the possible output sequences of a given block.

Gorokhov and Van Dijk [249], consider in 2001 the effect of choosing different bit labelings for a bit-interleaved (convolutionally) coded modulation scheme, while using iterative demodulation. In this setup the combination of convolutional code, bit interleaver and (QAM or PSK) mapper is considered as a serial concatenated coding system, where the mapper acts as an inner code. The bit labelling defines the code properties of the inner code. For non-iterative decoding, a Gray mapping is known to be good as it minimizes the number of bit errors of the demapper for the SNR region of interest. For iterative decoding however, it turns out that it is beneficial to choose the mapping such that it maximizes the minimum Euclidean distance between signal points that have labels with Hamming distance 1. In this way, the inner decoder is better capable of improving the LLR's after the first iteration, where it is mostly faced with single errors for interesting SNR's.

3.2.2 Single Carrier Systems

In 1991, De Bot [213] presents a simple phase recovery algorithm for the detection of M-PSK. In particular, he is interested in the detection of differentially encoded PSK (DPSK). It is known that coherent detection of DPSK asymptotically performs 3 dB better than incoherent detection (i.e., looking at phase differences between two successive symbols only). Let ϕ be the unknown common phase deviation of a sequence of received signal values. For each received signal $r_i = |r_i|e^{j\vartheta_i}$ with $\vartheta = \frac{2k_i\pi}{M} + \phi + \theta_i$, where θ_i is the phase deviation caused by the AWGN, he considers r_i^ϕ which is obtained from r_i by rotating it a suitable multiple of $2\pi/M$ such that $\arg r_i^\phi \in (\phi - \pi/M, \phi + \pi/m)$. By simple operations using r_i^ϕ, he obtains estimates of ϕ that are ML-like for a series of consecutive observations i, thus leading to almost coherent detection. He also introduces an adaptive variant for time-varying channels or channels having frequency offsets.

Van Linden, De Bot and Baggen [215] present in 1993 an analytical derivation of the error rate performance of 2-DPSK using non-coherent detection on a Ricean fading channel. It is shown that, both for 2-PSK (coherent detection) and for 2-DPSK (incoherent detection), the performance on a Ricean channel resembles the performance on a Gaussian channel for low SNR, while it is more like a Rayleigh fading channel for large SNR, the transition point depending on the K-factor of the channel. An intuitive physical explanation for this phenomenon is given.

Krapels and Jansen [229], expand in 1995 on previous work of Jansen, which considers a dual signal receiver using successive interference cancellation, for simultaneous reception of two BPSK modulated co-channels. They investigate various alternative detection schemes, among which a joint ML detection scheme, for improving the performance in the notoriously difficult situation, where the two co-channels have about equal strength at the joint receiver. They find that even joint ML detection gives little improvement over conventional successive interference cancellation for uncoded BPSK.

In 1999, Gerrits et al. [237] present the Philips proposal for an adaptive multi-rate (AMR) GSM system. The AMR system comprises a set of speech and channel coders where, for a fixed given channel bit rate and depending on the channel quality, the combination of speech and channel coder giving the best speech quality is selected. A solution for a fast and seamless adaptation to a time-varying channel quality is explained and demonstrated. Although the system did not end up in the standard, several of its ideas can be found in the current GSM-AMR.

In 2000, Jansen and Slimana [243], consider the BER performance of successive interference cancellation (SIC) or ”onion peeling” of a received signal being a sum of N independently modulated M-PSK signals (using the same carrier frequency) and AWGN. Assuming that the amplitude and phase of each signal is known at the receiver, the performance of a coherent SIC system is approximated analytically and simulated. Assuming that the signal amplitudes A_i are geometrically related, $A_k = \alpha^{k-1}A_1$, $k = 2, \ldots, N$, they find that such a system can work reliably for all N if α and A_1 are sufficiently large, depending on M. They also consider the extra margin in α that is required if the amplitude and phase of the received signals are not perfectly known at the receiver.

In 2001, Meijerink, Heideman and Van Etten [246], consider an optical communication system using Optical Code Division Multiple Access (OCDMA). In this set-up, the phase noise of each transmit laser (assumed to be independent between M different transmitters) is effectively used as its signature. Such a system is known to suffer from so-called beat noise of which the power is proportional to M^2. The authors replace the delay elements traditionally used in OCDMA, by a bank of filters and delay elements, both at the sender and the receiver such that the arrangement at the receiver forms a matched filter for the arrangement at the (wanted) transmitter. In this way they can make the beat noise to be proportional with M.

Levendovszky, Kovacs and Van der Meulen [256], analyze in 2002 the performance of a blind adaptive equalizer (DFMMSE) compared to an equalizer using a training set (MMSE). Both use the Robbins-Monroe stochastic approximation for adapting the equalizer coefficients, where the blind equalizer replaces the assumed known transmitted symbols (in case of presence of a training sequence) by the hard-decisions made at the output of the equalizer for the blind case. They confirm, both from computations and from simulations, that the DFMMSE equalizer converges to the same performance as the MMSE equalizer, provided that the initial error rate is less than 10%.

Desset [181] considers in 2002 error control coding for Wireless Personal Area Networks (WPAN). In a Wireless Personal Area Network power consumption plays a very important role. High performance channel coding strategies can be used to obtain coding gain and thus reduce transmit power. The average energy required per bit in a typical situation is about 15 nJ/bit. In addition, power consumption due to the complexity of encoding and decoding has to be considered. The complexity of Hamming codes, Reed-Muller codes, Reed-Solomon codes and Convolutional and Turbo codes has been analyzed. Both constraints are contradicting and an optimum solution has to be found. The paper proposes a strategy to select error correcting codes for WPANs. Considering applications with different average bit energies ranging from 100 pJ/bit to 10 nJ/bit the authors successively recommend Hamming codes, short constraint length convolutional codes and turbo coding.

Meijerink, Heideman and Van Etten [255], consider in 2002 again optical communication using OCDMA. They note that, e.g., because of temperature drift, it is difficult to accurately match the delays of the transmitter and receiver, which is required for coherent detection using BPSK. They analyze as a function of the number of users M, the performance of OOK and DPSK, which are less sensitive to drifts in phase. They find that DPSK using phase diversity detection performs almost as well as BPSK using balanced detection , while OOK has several disadvantages leading to a performance degradation with respect to DPSK.

Janssen [265] presents in 2003 a method to increase spectral efficiency in the downlink of a cellular system by addressing multiple users simultaneously using a single compound QAM signal. The technique is based on stacking a number of M-PSK modulated signals, each intended for a different user. The signal amplitudes and phases are optimized for given link gains and interference levels, in order to obtain a required symbol error probability performance at each of the user locations with minimum transmit power. The QAM compound signal and a successive cancellation detection structure are described. Comparisons with alternative signaling methods show the power gain of the presented scheme, especially in the situation where system capacity is basically interference limited. The scheme is very similar to the hierarchical modulation scheme suggested for DVB and the degraded broadcast channel [13].

Levendovszky et al. [266] consider in 2003 a bit detector for an ISI channel, where the bit detector consists of a FIR equalizer followed by a threshold detector. Classical equalizers use ZF or MMSE algorithms for optimizing the tapweights of the equalizer. The authors propose an algorithm that chooses the tapweights such that the resulting BER is minimized. The algorithm considers all binary sequences of length L, where L has to be sufficiently large considering the memory length of the channel and equalizer. Therefore, the algorithm is exponentially complex in L. They also propose a simplified (sub-optimal) algorithm which only considers those binary sequences of length L that most influential in determining the BER. Although much more complex than ZF or MMSE algorithms, the new algorithms are shown to have a better performance on two examples of two-tap channels for equalizer lengths from 2 to 10.

3.2.3 OFDM

In 1993, De Bot [221] considers (spatial) antenna diversity for OFDM systems. He first discusses various antenna combining techniques for a flat Rayleigh fading channel. Next, he observes that in the context of DVB-T, the channel is severely frequency-selective which is the reason for using OFDM. He also observes that all of the considered wide-band combining techniques give little improvement for the frequency-selective channel using OFDM. The reason is that different OFDM subchannels that for each antenna have their own (independent) fading parameters need to be combined in a different manner. The solution for OFDM is to apply a baseband combining technique for each of the subchannels separately, giving large performance improvements for the frequency selective channel.

Koppelaar [220], considers in 1993 an OFDM system where the channel impulse response is larger than the guard interval or where the guard interval is even absent. In this case, successive OFDM symbols suffer from intersymbol interference. He develops a formalism based on a vector channel (a vector corresponding to an OFDM symbol), using it to describe a (vector) DFE equalizer, the (LMS-type) algorithms that are required to compute the equalizer coefficients and to compute their performances. It turns out that, for reducing the complexity, one can use band-matrices. In an example, excellent results are obtained by using only 2 tri-diagonal matrices for the OFDM DFE.

Van Linden [219] presents in 1993 an attempt to analytically derive the performance of a coded OFDM system on a frequency-selective Raleigh fading channel. Because of the limited delay spread, the signal quality of different subcarriers of the OFDM system are correlated, leading to burst errors in the frequency domain. Comparing computations with simulations, it turns out that a generalization of the Gilbert-Elliott burst-noise model can be used to fairly predict the performance of an interleaved algebraic code for SNR's up to 30 dB. It also turns out that an interleave depth of about twice the coherence bandwidth is required for approximating the performance on an infinitely interleaved Rayleigh fading channel. For high SNR's, the behavior of the error rates is not correctly described by the theoretical model, for which an explanation is given.

In 1994, Van de Wiel and Vandendorpe [224] consider a combination of OFDM and DS/SS, where the spreading is applied to the composite OFDM signal. Furthermore, because of spectral efficiency, the guard interval is removed which leads to inter symbol interference (ISI between successive OFDM symbols) and inter channel interference (ICI between different subcarriers). At the receiver these interferences can be mitigated using 2-dimensional (time-frequency) equalizers. Modelling this problem as a MIMO equalization problem, they consider 2-D MMSE equalization leading to the LMS algorithm, and the also consider an RLS-type of equalization leading to a Kalman filter. They find that the RLS-type of equalizer performs much better than the LMS-type, in particular if the search space becomes large.

In 2002, Taubock [254] considers an equivalent baseband transmission system, where the complex additive (Gaussian) noise is not circular complex (i.e., having uniform phase distribution), which they call rotationally variant complex noise. First the author shows that, for a given noise power, the entropy is maximal if it is circular. Next, he shows that the capacity of an additive noise channel having an average input power constraint (and an average noise power) is increased, if the noise turns out to be rotationally variant. This capacity increase however, is only to be found and used if one considers the "pseudo-covariance" matrix of the noise. Essentially, one has to exploit the rotationally invariance of the noise by using a proper loading of the real and imaginary components of the channel ("waterfilling"). An application would be in OFDM transmission, where the presence of non-white noise at the input of the FFT leads to rotationally variant additive noise at the subcarriers.

Cendrillon, Rousseaux and Moonen [264] consider in 2003 a MIMO channel with channel state information available at the transmitter. They explain that an optimal TX/RX structure can be found by considering the eigen decomposition of the channel. By using the corresponding eigenvectors, the MIMO channel is decomposed into a set of parallel channels for which ”water filling” can be applied and for which the capacity is easily found. Furthermore they show that, in case the spread of the eigen values of the channel is large, a power constraint per transmitter is more detrimental to the capacity than a power constraint on the total transmitted power as the latter leaves more freedom to the power allocation.

In 2003, Van Houtum [260] first explains the physical layer of the IEEE802.11a system. Next, he compares the performance obtained from simulations of this system on an AWGN channel with information theoretic bounds and union bounds. Finally, he gives plausible reasons for the differences (13 dB) between theoretical obtainable curves and simulated performances.

3.3 Recording

Research in the area of recording within the Benelux is mainly related to Philips activities in the area of optical and magnetic recording [10, 11, 12]. One typically sees the application of runlength-limited modulation (RLL) codes, at first both in optical and magnetic recording. In high density magnetic recording, one shifts to the application of PRML detection, while in optical recording (d,k)-constrained codes are used much longer (e.g., for DVD and Blu-ray), because of the robustness required for removable media in combination with simple detectors.

In 1986, Bergmans [203] studies the optimum performance of the decision feedback equalizer (DFE) for partial response (PR) channels with D-transform in the form $g(D) = (1 - D)^n(1 + D)^m$. He derives a closed-form expression for the minimum mean-square error (MMSE) at the bit detector input. From the expression we see that the MMSE depends on g_0. Since $g_0 = 1$ for all PR channels of the above mentioned form, as well as for the non-partial response channel ($g(D) = 1$) he concludes that unlike for the linear equalizer, the optimum performance of the DFE is independent of the PR used.

In 1987, Bergmans and Jansen [204] derive the DFE with an optimum mean-square performance in the presence of a mixture of intersymbol interference (ISI), noise and channel parameter variations. They use a transform introduced in 1967 by J. Zak in the field of quantum mechanics. The Zak transform of a continuous-time signal is the discrete Fourier transform of a sampled version of the signal with a specified sampling phase. The Zak transform therefore results to be a natural tool to introduce the timing errors into the optimization of the DFE and permits the authors to find a closed-form solution. The superior performance of the DFE with an optimum resistance to uniformly distributed timing errors w.r.t. the conventional MMSE DFE is demonstrated by means of computer simulations.

Schouhamer-Immink [206] proposes in 1988 to code digitized audio samples $\mathbf{s}$ with a rate $n - 1/n$ binary code, where n is a power of 2. The coding has several interesting properties. Simple decoding: **s** can be recovered from the binary codeword **x** by performing a Hadamard transform $\mathbf{y} = H_n\mathbf{x}$ followed by a slicer. The Hadamard transforms has low complexity since H_n is a binary matrix. Error resilience: the code is constructed in such a way that the MSB of **s** is placed in the most reliable frequency band of **y**, and so on until the LSB which is placed in the most unreliable frequency band. In this way an increase in additive noise or a reduction of bandwidth result in a graceful degradation of the audio SNR.

Vleuten and Schouhamer-Immink [208] describe in 1989 the implementation and performance of a class IV $(1 - D^2)$ PR magnetic recording system. The authors built two detectors: the classical threshold detector and the maximum likelihood (ML) Viterbi detector (VD). Experiments were performed in order to assess the better performance of the VD as predicted by theoretical analysis (3 dB improvement w.r.t. the threshold detector for AWGN). The $(1-D^2)$ VD consisted of two independent $(1-D)$ VD used in *ping-pong*. Two experiments were performed. In the first experiment, the system was optimally adjusted to achieve the smallest possible bit error rate (BER). The VD achieved a reduction of the BER by a factor of 2.9 w.r.t. the threshold detector. In the second experiment a tracking error was introduced which increased the BER. The VD showed to be more robust than the threshold detector and achieved a reduction of the BER by a factor of 9.3.

In 1990, Bergmans [211] shows that run-length-limited (RLL) codes lead to poorer pre-detection SNR's than uncoded recording, for a high-density recording system with optimum mean-square DFE. More specifically he shows that the merit factor due to the spectral shaping introduced by the use of RLL codes is not enough to compensate for the loss in minimum mean-square error that results from the fact that the RLL codes have a rate $R < 1$. Losses are lower bounded for a number of practical codes as well as for maxentropic (d, k) sequences.

In 1991, Bergmans [212] revisits the implications of binary modulation codes on PR channels. He considers a continuous-time transmission system with ISI and noise in which signaling occurs by means of non-overlapping rectangular pulses and binary modulation codes with rate $R = 1/N$ (N is a positive integer). He shows that the common assumption that the effect of coding on the channel is a SNR loss by a factor of R does not necessarily apply to PR channels. He computes the actual loss for most PR channels and shows it to differ from R. Furthermore he shows that coding implies more ISI for some PR channels.

Ribeiro [218] considers in 1993 the robustness of frame synchronization for a digital magnetic tape recorder (S-DAT). Each frame starts with a sync pattern, which does not appear elsewhere in the frame. Experimental error analysis shows that the main source of synchronization errors are due to deletions and insertions. Burst and random errors are scarcely found. His synchronization strategy uses a flying wheel, a search window and a number of sync levels. The flying wheel memorizes the position where the next sync pattern is expected. The search window defines how many bits around the expected position are checked for the sync pattern. At sync level 0 the search window is always open. When the pattern is found the sync level jumps to 1. If the sync level $L \neq 1$ and the sync pattern is found at the expected position the synchronizer jumps to level $L + 1$, otherwise it jumps back to $L - 1$. Simulations results showed that this strategy improves robustness against false alarms (due to the search window) and that the optimum is to consider $L = 1$ levels.

In 1994, Siala and Kaleh [223] derive bounds on the total SNR loss due to equalization and coding. Furthermore they derive the cut-off rate for the normalized information density $\delta = \tau/T$, where $1/T$ is the user bit rate and τ represents the impulse width of the Lorentzian channel model. Both bounds are in function of m, which defines the PR channel $g_m(D) = (1 - D)(1 + D)^m$. They conclude that for magnetic recording little equalization is required for the channel to match the class-4 PR channel ($m = 1$). At higher recording densities $m = 2$ represents a better choice. From the plot of the cut-off rate they conclude that for high SNRs it is more interesting to work with large values of m (neglecting the nonlinearities). They also conclude that for a great interval of SNRs the system equalized to $m = 1$ outperforms the one equalized to $m = 0$. They therefore recommend to equalize to $m = 1$ since it offers a good compromise between efficiency and complexity and presents low nonlinearity effects compared to $m > 1$.

Riani *et al.* [261] derive in 2003 the MMSE linear equalizer for a Two Dimensional Optical Storage (Two-DOS) system. Data is stored in a hexagonal two-dimensional lattice. Furthermore they consider the design of an optimum 2D target response. They derive an expression for the BER of the 2D PRML system. By means of numerical simulations they are able to find the optimal 2D target response in the sense of minimizing the resulting BER.

3.4 Networking

3.4.1 Packet transmission

Prasad et. al. [217] propose in 1993 to enhance the throughput of the slotted ALOHA by using more than one transmitting frequency (channel). Transmitted packets are distributed at random over a number of frequencies. It is assumed that a packet is received correctly if its power exceeds the total interfering power by the capture ratio. An expression for the total network throughput is derived and evaluated for different channel conditions, like uncorrelated log-normal shadowing, Rician and Rayleigh fading.

In 1994, Ruszinko and Vanroose [225] study the ALOHA protocol with multiplicity feedback. The ALOHA collision resolution scheme is based on using feedback at the end of each time slot to signal that a collision occurred. Several types of feedback can be considered. This paper develops the situation of multiplicity feedback, where all users are informed of the multiplicity of the collision. The capacity of the multiplicity feedback scheme is 1 (proved by Pippenger in 1981), and can be obtained by random coding. In their contribution they describe a constructive protocol that has throughput arbitrarily close to 1. The protocol is based on earlier work by Gyrfi and Vajda using protocol sequences.

Vvedenskaya and Linnartz [230] consider in 1995 a wireless network with two base stations and many mobile users transmitting packets. The users in a particular cell compete for random access, using the stack algorithm with feedback from the respective base station. Two different cases are considered: both base stations share and thus interference may occur; both base stations use different channels and thus no interference is assumed. To avoid interference in the second situation requires two different channels, each with half the bandwidth. The first situation is modeled with a 2-state Markov channel model with a "good" (no interference) and a "bad" (interference) state. Performance of this two cell system is analyzed. Simulations show that splitting bandwidth into two separate channels performs worse than using only a single channel system for both base stations handling all traffic. The results suggest that it might be advantageous, to allow nearby cells to use the same channel in lightly loaded wireless networks with bursty traffic.

Levendovszky et.al. [257] remark in 2002 that a major bottleneck in multicast communications stems from the number of NACKs generated by the receivers if the sender's packet is received erroneously. Flooding the network with these signaling packets can considerably decrease the throughput. To circumvent the effect, a suppression mechanism of NACKS is introduced by sampling a stochastic timer. They design optimal stochastic timers for feedback mechanisms in multicast communication. The sender is assumed to include a timer p.d.f. in the message to a receiver. When sending feedbacks, the receiver samples the timer p.d.f. and waits accordingly. If no feedback from other nodes arrive during the waiting period, then a feedback is generated. Otherwise the feedback is suppressed. The challenge is to prevent the network from flooding with NACKs but, at the same time ensure a secure feedback to the sender. The goal of the paper is to develop optimal timer distributions that lead to specified properties of the distribution of the aggregated NACKs. Results are given in the case of uniform distances between the sender and receiver and among the receivers themselves. For nonuniform distances the central limit theorem is used to derive the results. An optimal feedback mechanism is presented using a Markovian control scheme.

3.4.2 Routing and Queuing

In 1998, Boxma [235] gives a performance analysis of communication networks in a tutorial presentation. Especially, he focuses on the congestion problems that are assumed not to disappear with the introduction of fast networking. The distributed structure of modern computer-communication networks as well as the nature of traffic arrival processes and service request offered to those networks pose new challenges to queuing theory. Queuing models also lead to accurate predictions of the behaviour of complex computer systems. As an example, the performance analysis of ATM networks gives rise to stochastic networks that still comprise traditional single or multiple server queues, but with often complicating features like intricate priority structures. In order to take full advantage of the available network bandwidth, it is desirable to make use of statistical multiplexing effects. LAN, Internet, WAN, VBR video are examples of networks with traffic that is self-similar or with traffic that has a long-range dependence. The occurrence of heavy-tailed active (and/or silent) periods of sources seems to provide the most natural explanation of long-range dependence and self-similarity in aggregated packet traffic. The changing traffic distributions forces one to consider novel non-exponential stochastic networks. An example is the investigation of the effect of non-exponential service time distributions in ordinary single server queues.

Vvedenskaya [226] investigates in 1995 the distribution of message delay in a network with many multiple routes. As a network model a single input node is connected to N server nodes, where N goes to infinity. An arriving packet is transferred to the least busy server out of a randomly selected set of m servers. This means that the node is informed about the server queues. The probability distribution for the message delay is computed for the case when N goes to infinity, making queues independent. Simulation results are presented that suggest the existence of a stationary probability distribution of the queue length at a server.

In 1996, Vvedenskay [232] gives another example of optimal message routing in a complete graph network model with N nodes. The model forwards a message of length m from node I to node J with probability p or it divides the message into unit-length packets and forwards the packets individually on one of the two-link connections for the path from node I to node J. Each two-link path is selected with probability 1/(N-2). The end-to-end delay of a message is the delivery time of its last packet. The asymptotic performance is defined as the mean end-to-end delay as N goes to infinity. For a given message length distribution and flow intensity, the optimal value of p that minimizes the mean end-to-end delay is investigated. The optimum value for p is shown to be p = 0 or p = 1, depending on system parameters. Simulations support numerical results.

Giannakouros and Laloux [209] describe in 1989 a system of multiple queues served by a single server under the exhaustive service discipline. They first analyze priority polling systems and give explicit approximated formulas for the mean waiting times at individual stations for a given group of polling sequences. Then, they propose an elegant definition of a special group of polling sequences, which enables both performance and system optimization.

Levendovszky and van der Meulen [236], argue in 1998 that efficient traffic control is imperative in ATM networks when statistical multiplexing results in bursty aggregate traffic. ATM cell loss occurs when there is a buffer overflow. To maintain a previously negotiated level of Quality of Service (QoS), a Call Admission Control (CAC) function must be performed. They model an ATM switch as a buffer connected to a single server with deterministic service time. Their interest is to develop a fast algorithm that evaluates the tail of the stationary distribution of the underlying queuing system. The algorithm is expected to support real-time operation. Based on the outcome of the algorithm, user calls can be admitted or rejected.

Vitale et.al. [250] present in 2002 a new diffuse data routing concept based on multi-path signal propagation aided with adaptive beam-forming methods. The multi-path data flow incorporates redundancy and therefore increases resilience. The beam-forming provides efficient utilization of energy within the multi-path channel. To increase the energy efficiency further for low-power operation, multi-path channels are bounded within a diffusive data flow region determined by the strength of the signals. The operation of the multi-path diffuse routing algorithm is demonstrated with a simple example network topology. The multi-path diffuse routing has the potential to provide low power and resilient communications in dense networks of low cost devices in changing and noisy environments.

Levendovszky et.al. [245] investigate in 2001 the problem of ensuring Quality of Service (QoS) in packet communication networking. That is, the selected route has to satisfy given end-to-end delay or bandwidth requirements. In this contribution a path is selected which guarantees the end-to-end QoS criteria with maximum probability. This type of selecting is called Maximum Likely Path Selection (MLPS) procedure.

If link parameters are random variables, the problem becomes an NP-hard problem. The MPLS is reduced to a quadratic optimization that can be carried out by a Cellular Neural Network (CNN). As a result, the QoS requirements are met, even in the case of incomplete information.

In 2002, Bargh et.al. [253] study the role and define the functionality of a service broker in a Personal Service Environment (PSE). The PSE has to take care of the integration of complex and distributed heterogeneous entities such as wireless and fixed networks, terminals, services users and organizations. In the PSE two planes deliver personalized mobile services, a data or service plane and a brokerage or control plane. The data plane contains service components, governed by the brokerage plane, that store, forward and adapt the data units and logic in mobile services. A broker is in charge of the control plane and handles all issues of mobility. The PSE system reaches satisfaction or an acceptable QoS level for all involved agents, and thus the actors (end-users, end-devices, network operators, service providers and policy makers) they represent, when they are pleased with the proposed settings of mobile services in the service plane. The paper studies the role and functionality of a service broker in the PSE by investigating the basic mechanisms from a privacy and from a distribute QoS management perspective.

3.4.3 Multiple Access

In 1993, Prasad [216] reviews CDMA systems for future universal personal communication systems. One of the important topics considered is the choice of a multiple access technique. Performance results are presented for a DS CDMA network in macro, micro and pico cellular systems using DPSK and BPSK modulation and perfect power control, in terms of throughput and delay for fast and slow Rician fading channels. The paper further summarizes the research performed at the TU Delft in the Traffic Control Systems Group.

Rodrigues et al. [222](1994) and Jacquemin et al. [228](1995) combine multi-h continuous-phase modulation (CPM) with DS-CDMA in order to exploit the benefits of both principles. These benefits include low cost non-linear receivers, interference rejection and multiple access capabilities. As a result, a finite state description for the signal structure permits to define a periodic trellis and thus enables maximum likelihood sequence detection by means of the Viterbi Algorithm. In [222] simulation results are presented for the AWGN channel and several types of indoor channels. In [228] the authors develop an analytical model for the performance evaluation in a multipath Rayleigh fading indoor channel corrupted by multiple user interference. Previously, results were obtained for the AWGN channel. The evaluation is based on the constructed trellis and its transfer function, see also [222]. Simulations validate the model.

Çamkarten [214] studies in 1992 the design of an optimum CDMA receiver for a fixed number of fixed or mobile terminals. An accurate statistical model of a multiple-access Rayleigh fading channel and of the receive signal is developed to optimize the use of the allocated channel bandwidth and to maximize the throughput of a packet radio network. Single-user coherent and partially coherent multi-user base station receiver structures have been designed for uncoded BPSK packet transmissions over uncorrelated Rayleigh fading linear channels using CDMA. The corresponding exact bit error rates (BER) are evaluated and the feasibility and robustness of the new systems developed are discussed.

Vanhaverbeke et. al. [252] investigate in 2002 CDMA for the situation where the users are divided into two groups. This is called OCDMA/OCDMA (O/O). Set-1 contains as many users as the spreading factor of the CDMA system. The rest of the users are supposed to be in set-2. The perfect synchronized users of the two orthogonal signature sets are allowed to have a different average-input-energy constraint. The sum capacity of the O/O system can be made arbitrarily close to the upper bound imposed by the Gaussian Multiple-Access Channel if the set-1 users are assigned a higher power than the set-2 users. Making the power of the set-2 users higher than that of the set-1 users drastically reduces the sum capacity of the O/O system.

In 2000, Vinck [241] considers Frequency Hopping (FH) as alternative to DS CDMA. He generalizes a binary FH scheme to M-ary symbols and calculates the maximum throughput that can be obtained. He shows that uncoordinated M-ary Frequency Hopping gives rise to an efficiency of about 70%. The same paper discusses transmission of signatures in a multi user environment where the set of active users is small compared to the total amount of users. Two classes of signatures are described: uniquely decipherable where the individual signatures are detected uniquely from the composite signature; uniquely distinguishable signatures where the presence of a particular signature can be detected uniquely. Upper and lower bounds on the length of these signatures are given.

In 2003, Lathauer, Vandewalle and De Moor [262, 263] discuss an algebraic technique for blind signal separation of constant modulus (CM) signals, received on multiple antennas. They apply this technique for estimating (blindly) a MIMO equalizer that separates a convolutive mixture of multiple CM signals. Another application is for the separation of a mixture of DS-CDMA signals (also being of CM-type), received on multiple antennas. Their approach consists of a matrix formulation of the MIMO channel model, where the CM property is used to infer that a solution for the separation problem can be found by looking for dominant singular values and a simultaneous diagonalization of a set of matrices.

Tang et al. [247] consider in 2001 Link Adaptation (LA) to maximize the spectral efficiency in high-speed wireless networks. To approach the instantaneous channel capacities, the adaptation of the system parameters needs a general optimal LA switching scheme. Using a block by block adaptation mode instead of a symbol by symbol approach, channel quality thresholds obtaining a target Bit Error Rate and spectrum efficiency are determined. These parameters lead to the optimization problem that maximizes throughput for a given average power budget, or minimizes power under an average throughput constraint. The paper also presents numerical calculations verified by simulations. For a study case, 18 dB gain can be provided with the presented scheme using adaptive modulation as an example.

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