Voice-Band Data Communication Modems—A Historical Review: 1919–1988

Kaveh Pahlavan
Jerry L. Holsinger

In spite of repeated predictions over the past decade or more that modems would soon be eliminated by all digital networks, the modem business continues to prosper (although at a reduced growth rate). Today, most data communication is carried out by voice-band modems over analog telephone networks, and the modem business has revenues on the order of a billion dollars.

For the past few decades, voice-band data communication has been an active area for both scientific investigation and commercial product development. The data modem field has been the scene not only of theoretical advances but also of early commercial application of adaptive equalizers, digital filtering, multilevel combined amplitude and phase modulation, channel coding with multilevel and multiphase signals, and various synchronization techniques. Many of these advances later found their way into radio communication modems and other applications. Today, multidimensional coding, multicarrier systems, and the effects of voice-band quantizers [pulse-code modulation (PCM), adaptive differential PCM (ADPCM)] on the performance of modems are areas of active investigation.

The nearly 70-year history and an almost explosive growth in both research and product development over the past 15 years make it difficult to cover all key topics without treating them superficially. It is hoped that a compromise has been suitably chosen. It is also important to note that, for competitive reasons, many companies are reluctant to publish the results of their work, thus making it difficult to properly determine the sequence of events. The information presented here relies on both the open literature and our own personal knowledge. Finally, our view is primarily oriented toward full-duplex modems for operation on four-wire leased lines. However, aside from the use of echo cancelers, which we discuss, two-wire public switched telephone-line modems operate on basically the same underlying principles as private-line modems, so this focus should not be unduly limiting.

The history of the voice-band data modems falls quite naturally into four chronological eras:

1. From roughly 1919 to the mid-1950s, work grew out of a need for transmission of low-speed telegraphic information. Research efforts were focused primarily on basic characterization of the lines and basic theories of data communications. The maximum data rates in this era were on the order of 100 bits/s.
2. Starting in the mid-1950s, a growing military need, followed by commercial interest, for transmitting much larger amounts of data focused efforts on obtaining higher data rates. These efforts led to a much better understanding of telephone-line characteristics and a quantum increase in understanding of efficient modulation and receiver design techniques for band-limited channels. This, in turn, led to an increase in data rates from 100 bits/s at the start of the era to 9,600 bits/s by the late 1960s.
3. During the 1970s, the maximum practical speed remained at 9,600 bits/s, but extensive research efforts by those working in this area improved

0163-6804/88/0001-0016$01.00

methods of implementing modems and affected investigations in other areas, such as adaptive filtering, digital signal processing, and design of integrated circuits. This era led to significant reductions in size and power requirements for modems and laid the groundwork for the most recent advances.

4. The 1980s has seen the exploitation of improvements in telephone lines, introduction of the first practical coding techniques for these lines, and continuing research. Taken together, this has led to modems operating at speeds up to 19.2 kbits/s.

The remainder of this paper expands on activities in these eras. Those fortunate enough to have been involved in this business over the past 20 years or so have had a stimulating experience and an unusual opportunity to see the practical application of sophisticated theory and research. It is hoped that some of this will show through in what follows.

First Era: The Early Beginnings

As early as 1919, the transmission of teletype and telegraph data had been attempted over both long-distance land lines [1] and transoceanic cables [2]. During these experiments, it was recognized that data rates would be severely limited by signal distortion arising from the nonlinear phase characteristics of the telephone line. This effect, although present in voice communication, had not been noticed previously due to the insensitivity of the human ear to phase distortion. Recognition of this problem led to fundamental studies by Carson [3], [4], Nyquist [5], [6], and Mead [7]. From these studies came techniques, presented around 1930, for quantitatively measuring phase distortion [8] and for equalizing lines with such distortion [9]. This work apparently resolved the existing problems and little or no additional work appears to have been done until the early 1950s.

Second Era: Growth in Demand, Knowledge, and Speed

The advent of the digital computer in the early 1950s and the resulting military and commercial interest in large-scale data processing systems led to a new interest in using telephone lines for transmitting digital information. The government need for transmitting aircraft radar data to central locations resulted in development of Bell System data communication products for voice-band networks [10], and these products soon found their way into commercial applications. In the early 1960s, the commercial importance of voice-band modems was being established. Magazines such as *Fortune, Business Week,* and *U.S. News and World Report* reflected this growing commercial interest [11]-[14], and numerous vendors responded with products. With the FCC's Carterfone Decision in 1968, which opened the switched network to interconnection of non-Bell modems, the growth of data communication products increased even more rapidly. As a result, the demand for higher-speed data communications increased, and it became desirable to attempt a more efficient utilization of the telephone channel.

Research Efforts

Starting in about 1954, researchers at both the MIT Lincoln Laboratory [15], [16] and Bell Telephone Laboratories [17]-[19] began to investigate more efficient utilization of the telephone network. These and other studies were continued by a moderate but ever-increasing number of people during the late 1950s [20]-[26]. By 1960, numerous modulation techniques for obtaining higher rates (over 500 bits/s) had been proposed, built, and tested [27]-[35]. However, these systems were still quite poor relative to what many people felt to be possible. Because of this, and also due to a growing interest in the application of coding techniques to this problem, work continued at a rapidly increasing pace [36]-[49].

In this period, carrier and timing recovery techniques were quite primitive and included the use of pilot tones or a significant carrier component. In addition, since the transmitted data was not scrambled, automatic gain control (AGC) circuits were often unreliable because of possible extended idle periods between bursts of data. For these reasons, noncoherent modulation techniques such as frequency-shift keying (FSK), differential phase-shift keying (DPSK), and amplitude modulation with vestigial side band (AM-VSB) (with a low-modulation index) were the central focus for most of the modems at that time.

In the early 1960s, some theoretical work on the design of optimum signal constellations for multiamplitude and multiphase systems appeared in the literature. Lucky [49] and Lucky and Hancock [50] introduced a new class of "optimum" two-dimensional signal constellations; among them, the most interesting was an eight-point constellation, which was later recognized to have excellent immunity to marginal phase distortion in an additive noise channel [49]. Campopiano and Glazer [51] suggested a set of rectangular-grid-type quadrature amplitude modulated (QAM) signal constellations for transmission of various numbers of bits per symbol. At that time, due to extensive intersymbol interference (ISI) for two-dimensional modulation, these multilevel structures did not find immediate application. Later on, constellations similar to that of Lucky were reinvented for the design of 4,800-9,600-bits/s modems, and, recently, structures similar to that of Campopiano and Glazer were adopted by the International Telegraph and Telephone Consultative (CCITT) for standard coded and uncoded constellations for higher-speed modems [52].

In this time frame, one of the more unusual schemes was one tested at Lincoln Laboratory [48]. In this system, the transmitted signal was "matched" to the telephone line so that the effect of phase distortion was essentially eliminated. Use of this technique with the SECO [53] machine (a sequential coder-decoder) and a feedback channel allowed virtually error-free transmission at an average rate of 7,500 bits/s. The experiment was done over a closed loop and synchronization signals were shared between transmitter and receiver. While this was, by no means, a product, it proved certain principles in communication theory and coding and was probably the first demonstration that such speeds were feasible on telephone lines.

By the mid-1960s, it was clear that channel amplitude and delay distortion were the major impairments to be focused on for further improvements. Two methods were considered to fight these impairments. One was the application of a multicarrier system; the other was the use of an automatic equalizer to cancel ISI, which resulted from amplitude and phase distortions of the channel.

The idea of multicarrier systems was investigated by Holsinger in his Ph.D. thesis at MIT [54]. He concluded that a practical optimum signal was a set of time-limited sinusoids whose duration and frequency made them orthogonal over an appropriate time interval. However, he observed that multitone systems had several problems relative to timing recovery and frequency offset, and, as a result, the approach did not appear practical. Multicarrier systems were further investigated by Chang [55], Smith [56] and Chang and Gibby [57], and, as we will see in the paper, this idea is still current.

In the area of automatic equalization, Lucky used a steepest descent algorithm for automatic adjustment of tap gains [58]. In a subsequent paper [59], he proposed a method for continuous adaptation of the tap gains; this method is known as the zero-forcing algorithm. By the late 1960s, single-carrier VSB or SSB systems using digitally implemented adaptive transversal equalization appeared to be the best approach for realizing high speeds (4,800–9,600 bits/s).

Commercial Products

The earliest commercially important modems were released in the late 1950s by AT&T and included the Bell 103 and Bell 202 for 300 bits/s and 1,200 bits/s, respectively. Both of these modems used FSK modulation. In the early 1960s, the application of four-phase modulation resulted in the Bell 201, which operated at 2,400 bits/s. This modem essentially used 2,400 Hz of transmission bandwidth with 1,200 Hz for data and the rest for additional timing recovery signals. The Bell 201 did not include a manual equalizer, as did the other 2,400-bits/s modems of that time; however, AT&T used this modem with specially conditioned lines (C2). The receiver in the modem used differential demodulation, which avoided carrier recovery circuits. The bit timing was extracted from the demodulated signal, and there was no scrambling of the transmitted signal.

Also in this time frame, a system operating at rates of 2,500–3,500 bits/s was marketed by Rixon [29], [34]. This modem used binary AM-VSB transmission and, for proper operation of its built-in manually adjustable equalizer, required telephone lines having maximum differential delays of 200–400 μs. This requirement necessitated special equalization of all but the highest quality telephone lines and prevented the use of the system on dial-up lines.

The Collins Kineplex [32], [60] modem used a multitone design to obtain data rates of 4,000–5,000 bits/s. This modem was one of the first to use signal design techniques in an attempt to overcome some of the special problems encountered on the telephone channel. Basically, this system used four-phase differential modulation of several sinusoidal carriers spaced in frequency throughout the passband of the telephone line. The differential delay requirements for this system was essentially the same as for the Rixon system at high rates, i.e., about 200 μs.

In 1966, Holsinger developed what was to become the first commercially available 9,600-bits/s modem. This modem used adaptive equalization, VSB modulation, and pilot tones for timing and carrier recovery. However, extensive testing over various channels showed a wide variation in error-rate performance. It was subsequently discovered that this was due to phase jitter, an unknown impairment at that time. A second-generation modem was then developed, which used partial response signaling to provide spectral nulls in the band edge. This allowed a much faster phase-locked loop to be used for carrier recovery, thus dealing better with phase jitter. While this led to significantly better performance, it did cost nearly 6 dB (roughly 3 dB for partial response and 3 dB for tones) and, given the quality of lines at that point in time, the performance was still marginal.

The Milgo 4400/48 operating on conditioned lines was announced in 1969. This modem operated at 4,800 bps using eight-phase PSK in 1,600-Hz nominal bandwidth and used a manually adjustable equalizer. This modem was the first commercially successful 4,800-bits/s modem.

In 1971, Codex introduced the 9600C based on work done by Forney and Gallager. This modem used QAM modulation with a 16-point signal constellation to send 4 bits/Hz in a 2,400-Hz nominal bandwidth and was the first 9,600-bits/s modem with broad success. The Codex 9600C used the band-edge components of the received signal to extract timing recovery. Carrier phase was recovered from the detected signal, and a scrambler was used to provide reliable timing and carrier recovery as well as AGC operation. The modem operated on C2 conditioned lines.

Third Era: Extensive Research and Technology Advances

During the 1970s, 9,600 bits/s remained the limit for the highest speed voice-band modems over four-wire private lines. However, research activities into various aspects of voice-band data communication modems continued extensively. A number of new topics were investigated in timing recovery, adaptive filtering, pulse shaping, optimization of signal constellations, and hardware implementation using semiconductor large-scale integration (LSI) devices.

Timing Recovery

The use of a pilot tone to transfer timing information is a waste of transmission power, and, as a result, significant work was done to enable derivation of timing information directly from the information-bearing signal. In this area, early unpublished work by Codex preceded many of the published techniques. In the open literature, several techniques were suggested and examined for this purpose. In 1971, Kobayashi [61] adopted *minimum-mean-square error at the equalizer output* as the criteria for a decision-aided algorithm for finding the optimum sampling phase. This type of scheme generally yielded a relatively complicated implementation and a slow convergence, and, consequently, did not meet widespread acceptance.

A second approach was to use narrowband filters turned to half the baud rate followed by a square-law device and a passband filter [62]. Lyon [63] analyzed the relationship of this method to the band-edge component maximization, which has been shown by Mazo [64] to yield the optimum sampler phase for infinite tap equalizers. As compared with the method suggested by Koboyashi, this technique yielded more economical implementation while giving comparable performance. Today, extracting timing from band-edge components of the received signal is widely used in modems employing standard baud-rate equalization.

A third approach was to use a fractionally spaced equalizer (FSE) rather than standard equalizers. In standard equalizers, tap spacing is at the baud rate, i.e., every T sec, while in FSEs *the tap spacing is a fraction of* the baud rate. Ungerboeck [65] showed that a $3T/4$ FSE had minimal sensitivity with respect to timing, and that a simple timing recovery procedure could be used with this type of equalizer. Qureshi and Forney [66] made extensive simulations for $T/2$ FSEs over various telephone lines and concluded that, with the same number of taps, the FSE performs at least as well as the ordinary equalizer, and, for channels with severe band-edge delay distortion, the FSE performs noticeably better. Later on, analysis and simulations by Gitlin and Weinstein [67] showed that the $T/2$ FSE performs better than conventional equalizers if they both span the same time interval of the received signal. The reason for the superior performance of the FSE is that it samples the received signal faster than an ordinary equalizer, and, thus, aliasing of the received signal can be eliminated. Otherwise stated, a FSE can incorporate a matched filter, whereas a T-spaced equalizer cannot.

Fractionally spaced equalizers perform in computer simulations with floating-point arithmetic. However, in real-time implementation with fixed-point arithmetic and limited word length, the tap gains tend to diverge. Gitlin, Meadors, and Weinstein [68], through analysis and implementation, related tap-gain blowup to characteristics of the channel. They showed that a FSE generally has many sets of tap weights, which result in nearly equal values of mean-squared error. Some of these solutions are large, and, in a digital implementation, the tap gains can drift toward these due to bias and round-off errors in the tap updating algorithm [68], which can lead to overflow of the tap-gain registers. In general, the simple solution to tap gain blowup is the so-called tap-leakage algorithm [65]-[68]. This is an ad hoc solution, which introduces small corrections in tap gains to prevent them from excessive growth. Codes marketed its first FSE modem in 1975, and today FSEs are widely used in various modems. Other notable activities in timing recovery techniques are reported in [69]-[74].

Adaptive Filtering

The major work in adaptive filtering applied to voice-band modems was concerned with: fast-converging algorithms for equalization, application of decision feedback equalizers (DFE), adaptive channel measurement for maximum likelihood sequence estimation (MLSE), passband equalization, blind equalization, and echo cancellation. Motivations for using these algorithms in voice-band data communications and an overview of the work in these areas will be given in this section.

- *Fast Poll Modems*

The effective response time in a multipoint polling system is dependent on the start-up time of the central site modems in the network. The time required by the equalizer to set up its tap coefficients represents the main part of the modem start-up time. Therefore, it is desirable to find methods for rapidly setting up the equalizers.

The least-mean-square (LMS) algorithm, today's dominant algorithm for tap-gain adjustments, was originally presented by Widrow and Hoff [75] in 1960. During the 1960s, research was mostly concerned with steady-state behavior of LMS algorithms, as applied to adaptive equalizers. Around 1970, the transient behavior of LMS equalizers was studied [76]-[78], and it was concluded that the convergence behavior of LMS adaptive equalizers is dependent on the number of taps and the eigenvalue spread of the covariance matrix of the received signal. In the early 1970s, the first techniques for fast start-up equalization were introduced [79], [80]. These algorithms were intended to minimize the dependence on eigenvalue spread of the channel covariance matrix and, for this reason, were referred to as orthogonalized LMS algorithms.

In 1974, Godard [81] introduced an application of Kalman filtering to obtain an adaptive orthogonalizing algorithm for fast-converging equalization. He showed that the convergence rate of this adaptive algorithm is only proportional to the length of the equalizer and is independent of eigenvalue spread. However, the computational complexity of the algorithm was on the order of the square of the length of the equalizer. Gitlin and McGee [82] provided another algorithm with reduced computational complexity while retaining the same speed of convergence. Later on, Falconer and Ljung [83] introduced the idea of fast Kalman filtering, which took advantage of the structure of the data vector and reduced complexity of computation to multiples of length of the equalizer. Then, Satorius et al. [84], [85] used orthogonality of backward prediction error for lattice filters as a basis to design fast-converging lattice equalizers. Other notable theoretical works in this area include [86]-[88]. To the authors' knowledge, none of the preceding techniques are currently used in a successful voice-band modem product. However, a variation of the Kalman filtering algorithm has been tried for fast tracking of a radio modem working over fading multipath HF channels [89]. The work on fast poll modems motivated extensive research into the general structure of adaptive filters [90], [91].

Today, to the best of our knowledge, the two most widely used techniques for fast start-up are tap storage and derivation of the tap gains in the frequency domain. In tap storage, the tap gains related to each connection in a multipoint connection are stored in the receiver along with a unique identifying code. Upon reception of the code for a particular connection, the receiver recovers and updates the tap gains used for that connection. This approach tends to offer a very rapid start-up (less than 5 m/s) but has "system-level" issues concerning retraining of the individual stored equalizer values, which have to be carefully thought out. With the frequency-domain technique, a cyclic training sequence with a flat spectrum is transmitted. The receiver takes the Fourier transform of the received signal, inverts the result, and takes the inverse Fourier transform to determine the tap gains of the equalizer. This method is straightforward and the details of implementation are given in [92]-[94]. It offers the advantage of eliminating the system-level problems of the tap storage technique but generally requires a greater start-up time (around 25 ms). A comprehensive survey of various techniques evolved for fast start-up equalization is given in [95].

- *Beyond 9,600 bits/s*

In the 1970s, the quality of the telephone network prevented transmission of more than 16 points in the signal constellation. Therefore, to increase the data rate, it was necessary to think about increasing of the symbol rate by expanding the nominal bandwidth of the channel beyond the commonly used value of 2,400 Hz. To attain wider bandwidth, it looked promising to consider more advanced equalization techniques such as DFE or MLSE, rather than standard linear equalizers.

A DFE consists of two tapped delay lines, one fed by the received signal and the other by the detected signal. A DFE [96], [97] is more successful than a transversal equalizer in handling channels with a deep null in the passband, and, for this reason, it has been applied to various fading multipath radio channels [98], [99]. Therefore, one may think of using a DFE to equalize the edges of the transmission channel to attain a wider equalized bandwidth. MLSE, as suggested by Forney [100], applies the Viterbi algorithm [101] to obtain the optimum receiver in the presence of ISI. In conjunction with an adaptive channel estimator, the MLSE can be used over slowly time-varying channels [102]. A computationally efficient version of this method was suggested by Ungerboeck [103] for communication over telephone channels. An increase in the bandwidth of the transmitted pulses increases the ISI, but one may hope to cancel this excess ISI with MLSE.

Falconer and Magee [104], [105] investigated the DFE and MLSE to extend the nominal bandwidth beyond 2,400 Hz. They tried 16-point constellations with nominal bandwidths of 3,000 Hz and 3,600 Hz for 12 kbits/s and 14.4 kbits/s, and a 32-point constellation with a nominal bandwidth of 2,880 Hz for 14.4 kbits/s. Their conclusion was that, for a real channel with amplitude and delay distortion as well as phase jitter, a reliable 14.4-kbits/s modem was not attainable at that time. The MLSE is extremely sensitive to phase jitter and frequency offset, and the DFE with a large number of feedback taps suffers from error propagation; therefore, neither of these techniques had much to offer over a transversal equalizer with a large number of tap gains.

- *Blind Equalization*

In a multipoint network, if one of the remote modems needs retraining without interruption of the normal data transmission, a retraining algorithm is needed that does not require any knowledge of the transmitted symbols for the equalizer. Such equalization is usually referred to as blind equalization. In 1973, Godard [106] introduced a simple algorithm that is based on the square of the error signal used for normal adjustments of the tap weights, and today it is widely used to retrain the equalizers after interrupts. In these algorithms, a reference to transmitted symbols is not available at the receiver so the convergence is about ten times slower than standard LMS algorithms. More analysis of the transient behavior of blind equalizers, as applied to radio communications, is available in [107], [108].

- *Bandpass Equalization*

Bandpass equalization [109], [110] is performed before demodulation, as opposed to standard baseband equalization, which is done after demodulation. When the phase reference of the oscillator in the demodulator is obtained from the detected data, the delay between demodulation and phase recovery is smaller with passband equalizers. This results in a faster tracking of phase variations and, consequently, a more robust modem. However, in modern digital designs using programmable, very large scale integrated (VLSI) chips, it is possible to demodulate without a phase reference and adjust the phase by multiplying the demodulated symbol with a numerical phasor, which shifts the point in the constellation into its proper place. For these implementations, there is no difference between baseband or bandpass equalization. In radio modems working on high-frequency carriers, sometimes passband equalization either at IF or RF is more cost-efficient; for this reason, it is preferred to baseband equalization.

- *Echo Cancelers*

Echoes are the result of impedance mismatch in the communication circuit. A substantial amount of echo energy comes from the mismatching of impedance at the hybrid couplers, which are located at two-wire to four wire interfaces in the telephone circuit. There is one coupler at each end of a two-wire line; consequently, each user suffers from near and distant echoes created by the coupler at its side and the coupler at the other side of the telephone line. Shorter echoes are actually desirable and their existence keeps the telephone voice from sounding dead. Most of the echo cancellation in the telephone network is devoted to improving the quality for voice transmission. Therefore, cancelers are normally located near the two-wire to four wire interface, which does not eliminate the near echoes.

For full-duplex two-wire communication modems, the near echoes are just as damaging as the distant echoes. For this reason, and also because echo cancelers are not deployed throughout the telephone network, echo cancelers have been proposed [111], [112] to improve the performance of full-duplex two-wire networks. Full-duplex operation of two-wire modems up to 2,400 bits/s is feasible without echo cancellation; for higher data rates, 4,800 bits/s and 9,600 bits/s, satisfactory operation requires echo cancellation.

Echo cancelers used in full-duplex modems are adaptive transversal filters similar to those used for channel measurement, and their principles of operation and mathematical structure are very similar to those of adaptive equalizers. Details of existing problems and an overview of echo-cancellation techniques are given in [113]-[116].

Semiconductor Technology and Modem Design

Due to rapid improvements in the semiconductor technology, the past two decades witnessed simultaneous reduction in the size and price of modems. Digital technology and LSI devices resulted in more cost-effective implementation with lower internal processing noise. Microprocessors facilitated the design and opened the door for inclusion of ever-growing network control and management features. And, finally, the superior number-crunching capabilities of digital signal processors opened the way for low-cost implementation of sophisticated trellis coding techniques, which require a Viterbi decoder at the receiver. In the mid-1970s, custom MOS/LSI implementation of 9,600-bits/s modems was announced [117], [118]. In the late 1970s, microprocessor-based designs were of growing interest [119]-[121]; however, off-the-shelf microprocessors were not fast enough for implementation of a 9,600-bits/s modem and special-purpose microprocessors turned out to be expensive [122]. Even design of 4,800-bits/s modems required a series of processing-cycle reduction tactics [123]. Hardware implementation of a typical 9,600-bits/s modem in this period included a bit-slice processor, a fast multiplier (number cruncher), and an ordinary microprocessor for network control.

Today, for low-speed modems, simple general-purpose microprocessors, such as the Intel 8085 or Motorola 6800, are used. For higher speeds, inclusion of other digital signal processor chips, such as the TI TMS320 or NEC 7720 [124] or custom VLSI chips [125], [126], are customary. Surveys of modem integrated circuits (ICs) and digital signal processing chips are available in [127], [128].

Design of Pulse-Shaping Filters

In band-limited data communications, the time duration of each symbol extends much beyond the symbol duration. The distortion caused by the overlap of these pulses results in ISI. Pulse-shaping filters are used to limit the bandwidth and, at the same time, minimize the ISI. Superior design of the pulse-shaping filter reduces the burden on the equalizer to cancel the ISI. In 1975, a careful design of the pulse-shaping filter allowed Intertel (now Infinet) to design the first 9,600-bit/s modem operating on unconditioned lines.

Raised cosine pulses, as described by Nyquist [6], [8], are the traditional zero-ISI filters, and approximation to raised cosine pulses had been the conventional technique in designing these filters. In 1969, Spaulding [129] proposed a method to design a pulse-shaping filter that minimized the stopband gain and ISI simultaneously. Using this technique, one could design a better filter than by direct approximation of raised cosine pulses.

Later on, there were extensive investigations of digital implementation of pulse-shaping filters. Finite-impulse response (FIR) filters are used for this purpose to provide more flexibility in shaping the spectrum. Standard techniques such as the Remez exchange algorithm or the windowing technique [130] may be used for design. However, these techniques do not minimize the ISI and stopband attenuation simultaneously; for example, a designer needs to try several designs to find the optimum solution. Muller [131] designed a digital linear phase FIR filter with special attention to zero-ISI and minimum stopband attenuation. Chevillat and Ungerboeck [132] developed an iterative technique using the steepest descent algorithm to design a pair of zero-ISI matched filters with maximum spectral power in the passband. Others used linear programming [133], [134] and a modified Remez exchange algorithm [135] for the same purpose.

Design of Signal Constellation

Other interesting work during this period was in the design of the optimum signal constellation, by Foschini, *et al.* [136], [137]. They used an iterative steepest descent algorithm to design optimum signal constellations in the presence of additive noise and phase distortion. Their work confirmed the optimality of several signal constellations designed previously and introduced some interesting new constellations.

Fourth Era: Ultrahigh-Speed Modems

In the late 1970s and early 1980s, the quality of average telephone lines had been significantly improved by the telephone company, especially in regard to phase jitter. Paradyne was the first to recognize this situation and to introduce a 14.4-kbit/s modem. The technique was quite simple: a 64-point QAM modulation (6 bits/symbol) provided a 10^{-5} probability of error at a signal-to-noise ratio (SNR) of 26.5 dB (Special D-conditioned lines from the telephone company delivered a SNR of better than 28 dB). State-of-the-art phase recovery techniques were used to reduce the small remaining phase jitter to a negligible level, and digitally implemented equalizers were able to equalize the channel to near perfection. Several other companies introduced 14.4-kbit/s modems in rapid succession after the the Paradyne introduction.

The performance of these modems over the poor lines was marginal. In an attempt to improve performance, Paradyne moved the corner points of the constellation to the middle of the sides. This improved the SNR by about 0.1 dB. Codex used a hexagonal constellation, with 0.6-dB improvement in SNR [52]. The cause of the problem was investigated at Infinet, and it was realized that it was due to multiplicative noise created by PCM systems.

[138]. This led to the design of an expanded signal constellation [139], which decreased the SNR for about 0.4 dB on additive noise channels, but improved it about 3 dB on multiplicative noise channels. Since most telephone lines today use PCM facilities, multiplicative noise is the dominant source of the noise in these channels. Consequently, the new constellation outperforms the other structures on lines with significant PCM noise.

Coding for Performance Improvement

In classical communication systems, error control is done by coding the input data bits and then modulating the coded signal. To keep the data rate unchanged, one should compensate for the bits used for error correction by increasing the transmission rate. In band-limited channels such as voice-band telephone channels, an increase in transmission rate requires an increase in the number of points in the constellation, which results in a higher transmission error rate. Until recently, it was believed that practical codes could not compensate for the loss caused by an increase of the number of points.

In the early 1970s, several companies made unsuccessful attempts to incorporate classical coding techniques in the design of voice-band modems. in 1982, a different type of coding attracted more serious attention. The motivation was the work of Ungerboeck [140] on combining modulation and coding, which was originally started in 1976 [141]. In the current literature, his coding technique is referred to as an example of Trellis-Coded Modulation (TCM). Various versions of TCM can improve performance of a modem by 3 to 6 dB. A variant of Ungerboeck's eight-state trellis code, with a nominal gain of 4 dB, has been the most widely used to date.

In 1982, a working group was formed to establish new CCITT standards for a family of full-duplex two-wire modems operating at data rates up to 9,600 bits/s in each direction on the public switched telephone network. Several new features were considered by this group, including the use of channel coding, which was felt necessary for an adequate performance margin. Much of the group's effort therefore went into developing a standard coding for modems. This was the first time in the history of voice-band modem standards that a CCITT standard was developed prior to appearance of products in the market. In the past, all CCITT recommendations for voice-band data modems had been developed based on proven records of successful modems.

In January 1983, the group decided to consider combined coding and modulation. During 1983, one and two-dimensional TCM with four and eight states was examined for various QAM signal constellations. Also, block codes based on lattice sphere packings [142] were considered, including an eight-dimensional code suggested by Lang [143]. In the end, an eight-state TCM code with immunity to 90-degrees phase ambiguities, invented by Wei and introduced by AT&T [144], was accepted as in recommendation V.32 for two-wire 9,600 bits/s, as well as in recommendation V.33 for four-wire 14.4 kbits/s. AT&T provided extensive laboratory results to support the standard, which helped to convince most of the delegates to support the use of trellis coding for voice-band data transmission. With the 4-dB coding gain obtained with eight-state TCM, a reliable 14.4-kbits/s or even 16.8-kbits/s modem can be designed.

During the period that the group was working on the standards, a new generation of TCM modems was under development in various companies for reliable 14.4 kbits/s (or possibly 16.8 kbits/s) modems. Taking account of the 4-dB gain from eight-state TCM, a 16.8-kbits/s modem seems to perform slightly better than uncoded 14.4-kbits/s modems. However, it should be mentioned that for 16.8 kbits/s, 7 bits/symbol for the data and one additional bit for coding are required. This requires 256 points in the signal constellation, as compared with 64 points for 64-QAM uncoded 14.4 kbits/s. Since the harmful effects of PCM compandors [138] increase with the increase in density of points in the constellation, one might be concerned about a marginal performance for such a 16.8-kbits/s modem. Codex announced the first 16.8-kbits/s modem in 1984.

19.2-Kbit/s Modems

Disregarding the effects of PCM systems, another 3 dB is needed to operate at 19.2 kbits/s. There have been two approaches to this goal. One uses the idea of orthogonal multiplexed QAM [145]-[147] and has been adopted by NEC [145]; the second uses multidimensional TCM [148]-[150] and has been adopted by Codex.

In orthogonal QAM, the idea is to transmit several orthogonal subchannels over the 3-kHz bandwidth of the telephone channel. For a large number of subchannels, the amplitude and delay distortion of each subchannel remain relatively linear, resulting in small dispersion. As mentioned earlier, the first multicarrier system was developed by Collins Kineplex [52], [53] around 1960. At that time, the idea was examined by Holsinger [55], and it was realized to be impractical for implementation due to time and phase recovery problems. In 1967, the idea was reexamined by Salzberg [151] and, in 1971, Weinstein and Ebert [152] gave an implementation based on the use of discrete Fourier transforms. In 1981, the digital implementation was further investigated by Hirosaki [146], and, finally, in 1985, NEC introduced a 19.2-kbit/s modem based on orthogonal QAM modulation [145]. This modem employs sophisticated timing and phase recovery techniques and uses pilot tones for deriving synchronization signals.

Recently, the idea of multidimensional coding has been discussed in several publications [52], [148]-[150], [153]-[155]. The design of signal constellations as well as techniques for adaptive equalization and phase recovery for multidimensional signals are discussed by Gersho and Lawrence [153]. In multidimensional communications, various pairs of coordinates of a point in the multidimensional constellation are transmitted in sequential time slots as in ordinary QAM.

In two-dimensional TCM, one redundant bit is added for each QAM symbol, resulting in a constellation of twice the original size. The cost of doubling the constellation, which must be offset by the coding gain, is a 3-dB loss in signal-to-noise ratio. In multidimensional coding, one redundant bit is added every $2N$ dimensions, which corresponds to N QAM symbols. Therefore, the number of redundant bits per QAM symbol is reduced, resulting in a smaller QAM constellation size; consequently, a smaller penalty due to constellation expansion in the coded system. For example, the 3-dB penalty for a two-dimensional constellation is reduced to 1.5 or 0.75 dB [149] if a four or eight-dimensional constellation is used. In comparison with two-dimensional TCM systems, the number of points in the constellation are reduced, resulting in better performance over channels with PCM noise. An eight-dimensional 64-state trellis code with a gain of 5.4 dB has been adopted by Codex for their 19.2-kbit/s modem. This modem uses 7 bits per symbol in 2,743-Hz nominal bandwidth.

Summary and Conclusions

Interest in problems associated with voice-band data communications has existed for about 70 years. A significant amount of theoretical work and scientific investigations and a large amount of capital have been invested by businesses in this area in the last 25 years. The maximum data rate supportable by the telephone channel was once predicted by Holsinger [156] to be bounded at 23.5 kbits/s, and state-of-the-art modems are now running at 19.2 kbits/s. Further improvements in data rate seem unlikely, unless the condition of the telephone lines is improved (except possibly by using up to 3-kHz nominal bandwidth). However, the market demand for higher rates never seems to end, and demand is likely to expand in view of the worldwide demand for data processing and the extensive investments in analog switched telephone networks. In terms of research, the effects of nonlinearities in the telephone network on the performance of modems, multidimensional codes for band-limited channels, and multicarrier communications are currently under investigation. On the development side, further applications of VLSI circuits are to be expected. In the standards arena, it is expected that CCITT work will continue on ultra high-speed modems.

Acknowledgments

The authors thank Dr. G. D. Forney, Jr., for his careful reading and constructive comments, and Dr. V. B. Lawrence, the guest editor of this series, for his encouragements and patience.

References

[1] A. B. Clark, "Telephone transmission over long cable circuits," *Bell Syst. Tech. J.*, vol. 2, pp. 67-94, 1923.

[2] O. E. Buckley, "High-speed ocean cable telegraphy," *Bell Syst. Tech. J.*, vol. 7, pp. 225-267, 1928.

[3] J. R. Carson, "Theory of transient oscillations of electrical networks and transmission systems," *Trans. AIEE*, Feb. 1919.

[4] J. R. Carson, "The building up of sinusoidal currents in long periodically loaded lines," *Bell Syst. Tech. J.*, vol. 3, pp. 558-566, 1924.

[5] H. Nyquist, "Certain factors affecting telegraph speed," *Bell Syst. Tech. J.*, vol. 3, pp. 324-346, 1924.

[6] H. Nyquist, "Certain topics in telegraph transmission theory," *Trans. AIEE*, vol. 47, p. 617, Apr. 1928.

[7] S. P. Mead, "Phase distortion and phase distortion correction," *Bell Syst. Tech. J.*, vol. 7, pp. 195-224, 1928.

[8] H. Nyquist and S. Brand, "Measurement of phase distortion," *Bell Syst. Tech. J.*, vol. 9, pp. 522-566, 1930.

[9] O. J. Zobel, "Distortion correction in electrical circuits with constant resistance recurrent networks," *Bell Syst. Tech. J.*, vol. 7, p. 438, 1928.

[10] E. R. Kretzmer, *The New Look in Data Communication, Data Communication*, edited by R. W. Lucky *et al.*, 1975.

[11] "New UNIVAC device provides an inexpensive way to link some of its computers by telephone," *Business Week*, p. 150, Nov. 9, 1963.

[12] "New challenge to IBM," *Business Week*, p. 30, Dec. 7, 1963.

[13] G. Burck, "The boundless age of the computer," Part I of a Series, *Fortune*, p. 230, March 1964.

[14] "What's coming tomorrow from today's telephones," *U.S. News and World Report*, pp. 60-64, March 9, 1964.

[15] J. V. Harrington, P. Rosen, and D. A. Spaeth, "Some results on the transmission of pulses over telephone lines," *Proc. Symp. Inform. Networks*, Polytechnic Institute of Brooklyn, NY, pp. 115-130, 1954.

[16] K. E. Perry, "Phone line data transmission system," MIT, Lincoln Laboratory Report M24-54, Sept. 1955.

[17] P. Mertz, "Transmission line characteristics and effects on pulse transmission," *Proc. Symposium on Information Networks*, Polytechnic Institute of Brooklyn, NY, pp. 85-114, 1954.

[18] E. D. Sunde, "Theoretical fundamentals of pulse transmission," *Bell Syst. Tech. J.*, vol. 33, p. 721, 1954.

[19] A. W. Horton and H. E. Vaughn, "Transmission of digital information over telephone circuits," *Bell Syst. Tech. J.*, vol. 34, pp. 511-528, 1955.

[20] W. D. Cannon, "Delay distortion correction," *Communication and Electronics*, p. 55, March 1956.

[21] B. P. Bogert, "Demonstration of delay distortion correction by time-reversal techniques," *Trans. IRE*, vol. CS-5, pp. 2-7, Dec. 1957.

[22] P. Mertz and D. Mitchell, "Transmission aspects of data transmission service using private line voice telephone channels," *Bell Syst. Tech. J.*, vol. 36, p. 1451, 1957.

[23] A. D. Fowler and R. A. Gibby, "Assessment of effects of delay distortion in data systems," *Commun. Electron.*, pp. 918, Jan. 1959.

[24] R. G. Enticknap and E. F. Schuster, "SAGE data system considerations," *Commun. Electron.*, pp. 824-832, Jan. 1959.

[25] E. D. Sunde, "Ideal binary pulse transmission by AM and FM," *Bell Syst. Tech. J.*, vol. 38, p. 1357, 1959.

[26] R. G. Enticknap, "The testing of digital data transmission channels and circuits," *Proc. Natl. Electron. Conf.*, vol. 16, pp. 66-71, Oct. 1960.

[27] E. J. Hoffman and G. E. Masters, "Error statistics on dual A-1 digital data transmission circuits from Kingston, New York to Cape Canaveral, Florida," MIT, Lincoln Laboratory Report 2G-25-4, June 6, 1958.

[28] L. A. Weber, "A F-M digital subset for data transmission over telephone lines," *Commun. Electron.*, p. 867, Jan. 1959.

[29] G. Holland and J. C. Myrick, "A 2500-band time-sequential transmission system for voice-frequency wire line transmission," *Trans. IRE*, vol. CS-7, p. 180, Sept. 1959.

[30] K. E. Perry, "An error-correcting encoder and decoder for phone line data," *IRE Wescon Convention Record*, Part 4, pp. 21-26, Aug. 1959.

[31] E. J. Hoffman, "Error statistics on dual A-1 digital data transmission circuits from Lexington, MA to S. Truro, MA," MIT, Lincoln Laboratory Report 2G-25-12, Feb. 1959.

[32] E. J. Hoffman, "Error statistics on the collins kineplex

over various media," MIT, Lincoln Laboratory Report 25G-0024, Oct. 31, 1960.
[33] E. J. Hoffman and P. L. Grant, "Error statistics on a long haul looped telephone circuit utilizing the milgo data system," MIT, Lincoln Laboratory Report 25G-0010, June 1960.
[34] J. L. Hollis, "Digital data fundamentals and the two level vestigial sideband system for voice bandwidth circuits," *IRE Wescon Convention Record*, Part 5, pp. 132-145, 1960.
[35] R. A. Gibby, "An evaluation of AM data system performance by computer simulation," *Bell Syst. Tech. J.*, vol. 39, Part 1, pp. 55, March 1960.
[36] H. L. Yudkin, "Some results in the measurements of impulse noise on several telephone circuits," *Proc. Natl. Electron. Conf.*, vol. 16, pp. 222-231, 1960.
[37] R. P. Pfeiffer and H. L. Yudkin, "Impulse noice on an H-44 telephone circuit looped from Lexington, MA via West Haven, CT," MIT, Lincoln Laboratory Report 25G-0012, Apr. 1960.
[38] A. A. Alexander, R. M. Gryb, and D. W. Nast, "Capabilities of the telephone network for data transmission," *Bell Syst. Tech. J.*, vol. 39, Part 1, p. 431, 1960.
[39] H. L. Yudkin, "Experiments in the improvement of the impulse response of telephone circuits," MIT, Lincoln Laboratory Report 25G-4, Nov. 27, 1961.
[40] B. Reiffen, W. G. Schmidt, and H. L. Yudkin, "The design of an error-free transmission system for telephone circuits," *Commun. Electron.*, pp. 224-230, July 1961.
[41] E. J. Hoffman, "Error statistics utilizing the code translation data system over various media," MIT, Lincoln Laboratory Report 25G-0026, March 28, 1961.
[42] E. Hopner, "Phase reversal data transmission for switch and private telephone line applications," *IBM J. Res. Develop.*, vol. 5, Apr. 1961.
[43] J. E. Toffler and J. N. Buterbough, "The HC-270, a four-phase digital data transceiver," *IRE Wescon Convention Record*, Paper 36 1, 1961.
[44] A. B. Fontaine and R. G. Gallager, "Error statistics for coding for binary transmission over telephone circuits," *Proc. IRE*, p. 1059, June 1961.
[45] P. Mertz, "Model of impulse noise for data transmission," *Trans. IRE*, vol. CS-9, p. 130, June 1961.
[46] E. D. Sunde, "Pulse transmission by AM, FM and PM in the presence of phase distortion," *Bell Syst. Tech. J.*, vol. 40, p. 353, March 1961.
[47] G. Comstock, "Low cost terminals triple phone TV link data rates," *Electron. Des. II*, p. 8, Jan. 18, 1963.
[48] I. I. Lebow *et al.*, "Application of sequential decoding to high-rate data communication on a telephone line," *IEEE Trans. Inform. Theory*, vol. IT-9, pp. 124-126, Apr. 1963.
[49] R. W. Lucky, "Digital phase and amplitude modulated communication systems," Ph.D. Dissertation, Purdue Univ., Lafayette, IN, 1961.
[50] R. W. Lucky and J. C. Hancock, "On the optimum performance of N-Ary systems having two degree of freedom," *IRE Trans. Commun. Syst.*, vol. CS-10, pp. 185-192, June 1962.
[51] C. N. Campopiano and B. G. Glazer, "A coherent digital amplitude and phase modulation scheme," *IRE Trans. Commun. Syst.*, vol. CS-10, pp. 90-95, June 1962.
[52] G. D. Forney, Jr. *et al.* "Efficient modulations for band-limited channels," *IEEE J. Selec. Areas Commun.*, pp. 632-647, Sept. 1984.
[53] K. E. Perry and J. M. Wozencraft, "SECO: A self-regulating error correcting coder-decoder," *Trans. IRE*, vol. IT-8, pp. 128-135, Sept. 1962.
[54] J. L. Holsinger, "Digital communication over fixed time continuous channels with memory," Ph.D. Thesis, MIT, Oct. 1964.
[55] R. W. Chang, "Synthesis of band-limited orthogonal signal for multichannel data transmission," *Bell Syst. Tech. J.*, pp. 1775, Dec. 1966.
[56] J. W. Smith, "A unified view of synchronous data transmission system design," *Bell Syst. Tech. J.*, pp. 273-300, March 1968.
[57] R. W. Chang and R. A. Gibby, "A theoretical study of performance of an orthogonal multiplexing data transmission scheme," *IEEE Trans. Commun. Technol.*, pp. 529-540, Aug. 1968.
[58] R. W. Lucky, "Automatic equalization for digital communication," *Bell Syst. Tech. J.*, pp. 547-588, Apr. 1965.
[59] R. W. Lucky, "Techniques for adaptive equalization of digital communication," *Bell Syst. Tech. J.*, pp. 255-286, Feb. 1966.
[60] M. L. Doelz, E. T. Heald, and D. L. Martin, "Binary data transmission techniques for linear systems," *Proc. IRE*, p. 656, May 1957.
[61] H. Kobayashi, "Simultaneous adaptive estimation and detection algorithm for carrier modulated data communication systems," *IEEE Trans. Commun. Technol.*, pp. 268-280, June 1971.
[62] L. E. Franks and J. P. Bubrouski, "Statistical properties of timing jitter in a PAM timing recovery scheme," *IEEE Trans. Commun.*, pp. 913-920, July 1974.
[63] D. L. Lyon, "Envelope-driven timing recovery in QAM and SQAM systems," *IEEE Trans. Commun.*, pp. 1327-1331, Nov. 1975.
[64] J. E. Mazo, "Optimum timing phase for an infinite equalizer," *Bell Syst. Tech. J.*, pp. 189-201, Jan. 1975.
[65] G. Ungerboeck, "Fractional tap-spacing equalizer and consequences for clock recovery in data modems," *IEEE Trans. Commun.*, pp. 856-864, Aug. 1976.
[66] S. H. Qureshi and G. D. Forney, "Performance and properties of T 2 equalizers," *NTC*, pp. 11:1-1, 11:1-9, Dec. 1977.
[67] R. D. Gitlin and S. B. Weinstein, "Fractionally-spaced equalization: An improved digital transversal equalizer," *Bell Syst. Tech. J.*, pp. 275-296, Feb. 1981.
[68] R. D. Gitlin, H. C. Meadors, and S. B. Weinstein, "The tap-leakage algorithm: An algorithm for the stable operation of a digitally implemented, fractionally-spaced adaptive equalizer," *Bell Syst. Tech. J.*, pp. 1817-1839, Oct. 1982.
[69] R. D. Gitlin and J. Salz, "Timing recovery in PAM systems," *Bell Syst. Tech. J.*, p. 1645, May 1971.
[70] R. W. Chang, "Joint equalization, carrier acquisition and timing recovery for data communications," *ICC*, pp. 34-43, June 1970.
[71] K. H. Mueller and M. Muller, "Timing recovery in digital synchronous data receivers," *IEEE Trans. Commun.*, pp. 561-521, May 1976.
[72] D. N. Godard, "Passband timing recovery in all-digital modem receiver," *IEEE Trans. Commun.*, pp. 517-523, May 1978.
[73] S. H. Qureshi, "Timing recovery for equalized partial-response systems," *IEEE Trans. Commun.*, pp. 1326-1330, Dec. 1976.
[74] J. Steel and B. M. Smith, "Carrier and clock recovery from transversal equalizer tap settings for a partial response system," *IEEE Trans. Commun.*, pp. 976-979, Sept. 1975.
[75] B. Widrow and M. E. Hoff, "Adaptive switching circuits," *IRE Wescon Convention Record*, pp. 96-104, Aug. 1960.
[76] J. G. Proakis and J. H. Miller, "An adaptive receiver for

digital signaling through channels with intersymbol interference," *IEEE Trans. Inform Theory*, pp. 484-497, July 1969.

[77] A. Gresho, "Adaptive equalization of highly dispersive channels for data transmission," *Bell Syst. Tech. J.*, pp. 48-55, 1969.

[78] G. Ungerboeck, "Theory on the speed of convergence in adaptive equalizers for digital communications," *IBM J. Res. Devel.*, pp. 546-555, Nov. 1972.

[79] R. W. Chang, "A new equalizer structure for fast start-up digital communication," *Bell Syst. Tech. J.*, p. 50, 1971.

[80] H. Koboyashi, "Application of Hestens-Stiefel algorithm to channel equalization," *ICC*, pp. 21:25-30, 1971.

[81] D. Godard, "Channel equalization using a Kalman filter for fast data transmission," *IBM J. Res. Devel.*, pp. 267-273, May 1974.

[82] R. D. Gitlin and F. R. Magee, Jr., "Self-orthogonalizing algorithms for accelerated convergence of adaptive equalizers," *IEEE Trans. Commun.*, pp. 666-672, July 1977.

[83] D. D. Falconer and L. Ljung, "Application of fast Kalman equalization to adaptive equalization," *IEEE Trans. Commun.*, pp. 1439-1446, Oct. 1978.

[84] E. H. Satorius and S. T. Alexander, "Channel equalization using adaptive lattice algorithms," *IEEE Trans. Commun.*, pp. 899-905, June 1979.

[85] E. H. Satorius and J. D. Pack, "Application of least squares lattice algorithms to adaptive equalization," *IEEE Trans. Commun.*, pp. 136-142, Feb. 1981.

[86] S. U. H. Qureshi, "Fast start-up equalization with periodic training sequences," *IEEE Trans. Inform. Theory*, pp. 553-563, Sept. 1977.

[87] K. H. Mueller and D. A. Spaulding, "Cyclic equalization—A new rapidly converging equalization technique for synchronous data communication," *Bell Syst. Tech. J.*, pp. 369-406, Feb. 1975.

[88] J. J. Liu and D. L. Lyon, "A fast start-up equalizer algorithm using an auto-correlation method," *ICC*, pp. 14.8.1-14.8.5, 1981.

[89] F. M. Hsu, "Square root Kalman filtering for high-speed data received over fading dispersive HG channels," *IEEE Trans. Inform. Theory*, pp. 753-763, 1982.

[90] J. G. Proakis, *Digital Communications*. New York: McGraw-Hill, 1983.

[91] S. Haykin, *Adaptive Filter Theory*. Englewood Cliffs, NJ: Prentice-Hall, 1986.

[92] M. Choquet, "Channel equalization apparatus and method using Fourier transform technique," U.S. Patent 4,152,649, May 1979.

[93] D. N. Godard, "A 9600 bps modem for multipoint communication systems," *NTC*, pp. B3.3.1-B3.3.5, 1981.

[94] A. Milewski, "Periodic sequences with optimal properties for channel estimation and fast start-up equalization," *IBM J. Res. Devel.*, pp. 426-431, Sept. 1983.

[95] S. H. Qureshi, "Adaptive equalization," *IEEE Proc.*, pp. 1349-1387, Sept. 1985.

[96] C. A. Belfiore and J. H. Park, Jr., "Decision feedback equalization," *Proc. IEEE*, pp. 1143-1156, Aug. 1979.

[97] J. Salz, "Optimum mean-square decision feedback equalization," *Bell Syst. Tech. J.*, pp. 1341-1373, Oct. 1973.

[98] P. Monsen, "Theoretical and measured performance of a DFE modem on a fading multipath channel," *IEEE Trans. Commun.*, pp. 1144-1153, Oct. 1977.

[99] P. A. Bello and K. Pahlavan, "Adaptive equalization for SQPSK and SQPR over frequency selective microwave LOS channels," *IEEE Trans. Commun.*, pp. 609-615, May 1984.

[100] G. D. Forney, Jr., "Maximum-likelihood sequence estimation of digital sequence in the presence of intersymbol interference," *IEEE Trans. Inform. Theory*, pp. 363-378, May 1972.

[101] A. J. Viterbi, "Error bounds for convolutional codes and an asymptotically optimum decoding algorithm," *IEEE Trans. Inform. Theory*, pp. 260-269, Apr. 1967.

[102] R. F. Magee and J. G. Proakis, "Adaptive maximum likelihood sequence estimation for digital signaling in the presence of ISI," *IEEE Trans. Inform. Theory*, pp. 120-124, Jan. 1973.

[103] G. Ungerboeck, "Adaptive Maximum Likelihood Receiver for Carrier Modulated Data Transmission Systems," *IEEE Trans. Commun.*, pp. 624-636, May 1974.

[104] D. D. Falconer and F. R. Magee, "Evaluation of decision feedback equalization and Viterbi algorithm detection for voiceband data transmission—Part I," *IEEE Trans. Commun.*, pp. 1130-1139, Oct. 1976.

[105] D. D. Falconer and F. R. Magee, "Evaluation of decision feedback equalization and Viterbi algorithm detection for voiceband data transmission—Part II," *IEEE Trans. Commun.*, pp. 1238-1245, Nov. 1976.

[106] D. Godard, "A 9600 bps modem for multipoint communications systems," *Proc. IEEE NTC*, New Orleans, LA, pp. B3.3.1-5, Dec. 1981.

[107] D. Godard, "Self-recovering equalization and carrier tracking in two-dimensional data communication systems," *IEEE Trans. Commun.*, vol. COM-28, pp. 1867-1875, Nov. 1980.

[108] G. J. Foscihini, "Equalizing without alternating or detecting data," *Bell Syst. Tech. J.*, pp. 1885-1911, Oct. 1985.

[109] R. D. Gitlin, E. Y. Ho, and J. E. Mazo, "Passband equalization for differentially phased-modulated data signals," *Bell Syst. Tech. J.*, vol. 52, no. 2, pp. 219-238, Feb. 1973.

[110] D. D. Falconer, "Analysis of a gradient algorithm for simultaneous passband equalization and carrier phase recovery," *Bell Syst. Tech. J.*, pp. 409-428, Apr. 1976.

[111] V. G. Koll and S. B. Weinstein, "Simultaneous two-way data transmission over a two wire line," *IEEE Trans. Commun.*, pp. 143-147, Feb. 1973.

[112] K. M. Mueller, "A new digital echo canceller for two-wire full duplex data transmission," *IEEE Trans. Commun.*, pp. 956-967, Sept. 1976.

[113] S. B. Weinstein, "A passband data-driven echo canceller for full duplex transmission on two-wire circuits," *IEEE Trans. on Comm.*, pp. 654-666, July 1977.

[114] S. B. Weinstein, "Echo cancellation in the telephone network," *IEEE Commun. Soc. Mag.*, pp. 9-15, Jan. 1977.

[115] D. G. Messerschmitt, "Echo cancellation in speech and data transmission," *IEEE J. Select. Areas Commun.*, pp. 283-298, March 1984.

[116] D. D. Falconer, "Adaptive reference echo canceller," *IEEE Trans. Commun.*, pp. 2083-2094, Sept. 1982.

[117] H. Harris, T. Saliga, and D. Walsh, "An all digital 9600 bps LSI modem," *NTC*, pp. 279-284, Dec. 1974.

[118] H. L. Logan and G. D. Forney, "A MOS LSI multiple configuration 9600 bps data modem," *ICC*, pp. 48.7-48.12, June 1976.

[119] P. J. VanGerwen, N. A. M. Verhoeckx, H. A. VanEssen, and F. A. M. Snijders, "Microprocessor implementation of high-speed data modem," *IEEE Trans. Commun.*, pp. 238-250, Feb. 1977.

[120] D. Godard and D. Pilost, "A 2400 bps microprocessor-based modem," *IBM J. Res. Devel.*, pp. 17-24, Jan. 1981.

[121] M. Kaya, K. Ishizuka, and N. Maeda, "High speed data modem using digital signal processor," *ICC*, pp. 14.7.1-14.7.5, June 1981.

[122] K. Watanabe, K. Inoue, and Y. Sato, "A 4800 bps microprocessor data modem," *IEEE Trans. Commun.*, pp. 493-498, May 1978.
[123] J. L. Holsinger, "Where are modems going?," *IEEE Commun. Soc. Mag.*, pp. 3-5, Sept. 1977.
[124] H. Haas, E. A. Fuchs, H. Sailer, and H. Schenk, "Digital high speed modem using only a few standard components," *ICC*, p. A5.8.1, June 1983.
[125] S. U. H. Qureshi and H. M. Ahmed, *VLSI Signal Processing*. New York: IEEE Press, 1984.
[126] Y. Mochida *et al.*, "VLSI high speed data modem," *Globecom*, pp. 45.8.1-45.8.6, 1983.
[127] W. Twaddell, "Modem IC's," *EDN*, pp. 160-172, March 1985.
[128] R. H. Cushman, "Third-generation DSP's put advanced functions on chip," *EDN*, pp. 59-67, July 1985.
[129] D. A. Spaulding, "Synthesis of pulse-shaping networks in the time domain," *Bell Syst. Tech. J.*, vol. 48, pp. 2425-2444, Sept. 1969.
[130] L. R. Rabiners and B. Gold, *Theory and Application of Digital Signal Processing*. Englewood Cliffs, NJ: Prentice-Hall, 1975.
[131] K. Muller, "A new approach to optimum pulse shaping in sampled systems using time-domain filtering," *Bell Syst. Tech. J.*, vol. 52, pp. 723-729, May-June 1973.
[132] P. Chevillat and G. Ungerboeck, "Optimum FIR transmitter and receiver filters for data transmission over band-limited channels," *IEEE Trans. Commun.*, vol. COM-30, pp. 1909-1915, Aug. 1982.
[133] A. C. Salazar and V. B. Lawrence, "Design and implementation of transmitter and receiver filters with periodic coefficient nulls for digital systems," *Proc. IEEE ICASSP*, Paris, France, pp. 306-310, May 1982.
[134] J. K. Liang, R. J. P. DeFigueiredo, and F. C. Lu, "Designing of optimum Nyquist, partial response, Nth band, and nonuniform tap spacing FIR digital filters using linear programming technique," *IEEE Trans. Circuits Syst.*, vol. CAS-32, pp. 386-392, Apr. 1985.
[135] T. Saramaki and Y. Neuvo, "A class of FIR filters with zero intersymbol interference," *IEEE Trans. Circuits Syst.*, 1986.
[136] G. J. Foschini, R. D. Gitlin, and S. B. Weinstein, "On the selection of a two-dimensional signal constellation in presence of phase jitter and Gaussian noise," *Bell Syst. Tech. J.*, Jul-Aug. 1973.
[137] G. J. Foschini, R. D. Gitlin, and S. B. Weinstein, "Optimization of two-dimensional signal constellation in the presence of Gaussain noise," *IEEE Trans. Commun.*, pp. 28-37, Jan. 1974.
[138] K. Pahlavan and J. L. Holsinger, "A model for the effects of PCM compandors on the performance of high speed modems," *Globecom*, pp. 28.8.1-28.8.5, Dec. 1985.
[139] K. Pahlavan and J. L. Holsinger, "A method to counteract the effects of PCM systems on the performance of ultra high speed modems," *ICC*, pp. 50.1-50.5, June 1986.
[140] G. Ungerboeck, "Channel coding with multilevel phase signals," *IEEE Trans. Inform. Theory*, pp. 55-67, Jan. 1982.
[141] G. Ungerboeck and I. Csajka, "On improving data-link performance by increasing the channel alphabet and introducing sequence coding," *Int. Symp. Inform. Theory*, Ronneby, Sweden, June 1976.
[142] J. H. Conway and N. J. A. Sloane, "Fast quantization and decoding algorithms for lattice quantizers and codes," *IEEE Trans. Inform. Theory*, pp. 227-232, March 1982.
[143] J. D. Brownlie and E. L. Cusack, "Duplex transmission at 4800 and 9600 bps on the general switched telephone network and the use of channel coding with a partitioned signal constellation," *Proc. Int. Zurich Seminar on Digital Commun.*, pp. 113-120, 1984.
[144] L. F. Wei, "Rotationally invariant convolutional channel coding with expanded signal space—Part II: Nonlinear codes," *IEEE J. Select. Areas Commun.*, pp. 672-686, Sept. 1984.
[145] B. Hirosaki *et al.*, "A 19.2 Kbps voice-band data modem based on orthogonally multiplexed QAM techniques," *ICC*, pp. 21.1.1-21.1.4, June 1985.
[146] B. Hirosaki, "An orthogonally multiplexed QAM system using the discrete Fourier transform," *IEEE Trans. Commun.*, pp. 982-989, 1981.
[147] B. Hirosaki, "An analysis of automatic equalizers for orthogonally multiplexed QAM systems," *IEEE Trans. Commun.*, pp. 73-83, Jan. 1980.
[148] A. R. Calderbank and N. J. A. Sloane, "Four-dimensional modulation with an eight-state trellis code," *Bell Syst. Tech. J.*, pp. 1015-1017, May-June 1985.
[149] L. F. Wei, "Trellis-coded modulation with multidimensional constellations," *IEEE Trans. Inform. Theory*, pp. 483-501, July 1987.
[150] G. Ungerboeck, "Trellis-coded modulation with redundant signal sets—An overview," *IEEE Commun. Soc. Mag.*, Feb. 1987.
[151] B. R. Saltzberg, "Performance of an efficient parallel data transmission system," *IEEE Trans. Commun. Technol.*, pp. 805-811, 1967.
[152] S. B. Weinstein and P. M. Ebert, "Data transmission by frequency multiplexing using discrete Fourier transform," *IEEE Trans. Commun. Technol.*, pp. 628-634, Oct. 1971.
[153] A. Gersho, and V. B. Lawrence, "Multidimensional signal constellations for voice-band data transmission," *IEEE J. Select. Areas Commun.*, pp. 687-702, Sept. 1984.
[154] R. Fang and W. Lee," Four-dimensional coded PSK systems for combating effects of severe ISI and CCI," *Globecom*, pp. 30.4.1-30.4.7, Dec. 1983.
[155] S. G. Wilson et al., "Four-dimensional modulation and coding: An alternative to frequency reuse," *ICC*, pp. 919-923, June 1984.
[156] R. W. Lucky, J. Salz, and E. J. Weldon, Jr., *Principles of Data Communication*. New York: McGraw-Hill, 1968.

Kaveh Pahlavan was born in Teheran, Iran, on March 16, 1951. He received the M.S. degree in Electrical Engineering from the University of Teheran, in 1975, and the Ph.D. degree from Worcester Polytechnic Institute, Worcester, MA, in 1979.

He was a Graduate Assistant at Worcester Polytechnic Institute until 1979. From 1979, he was an Assistant Professor in the Electrical and Computer Engineering Department at Northeastern University, Boston, MA. He was consulting with CNR, Inc., Needham, MA, and GTE Laboratories, Waltham, MA, during the same period. He joined Infinet, Inc., in 1983 and served as the Senior Staff Engineer and later as the Director of Advanced Developments until 1985. During this period, he was teaching digital signal processing courses at the Graduate School of Northeastern University. In September 1985, he joined the faculty of Worcester Polytechnic Institute (WPI), where he is currently an Associate Professor. During services at WPI, he has consulted with GTE Laboratories, and other local industry. His present areas of interest include voice-band data communications, wireless intraoffice communication networks, and fading channel communications. Recently, he has contributed in numerous papers and two patents in these areas.

Jerry L. Holsinger was born in Lansing, MI, on October 14, 1935. He received the B.S.E.E. degree from Indiana Institute of

Technology, Fort Wayne, IN, in 1957, the M.S.E.E. from Purdue University, Lafayette, IN, in 1962, and the Ph.D. in Electrical Engineering from the Massachusetts Institute of Technology, Cambridge, Massachusetts, in 1965.

From 1957 to 1960, he worked at ITT Laboratories developing slow-scan TV systems, which transmitted digital signals over telephone lines using some of the earliest data modems. From 1960 to 1962, he taught at Purdue University and performed research on coding and telephone-line data transmission techniques. He then worked for the MIT Lincoln Laboratories while attending MIT, where his thesis involved a theoretical and experimental study of high-speed digital transmission techniques for telephone lines. He then joined the Defense Research Corporation (DRC), where, in the period from 1965 to 1967, he developed the first commercially available 9,600 bps data modem and formed the Teledata Division of DRC to exploit this project. In 1965, the Teledata Division was bought by Codex Corporation, where he became Vice President of R&D and was responsible for research programs in both high-speed adaptively equalized modems and error-correction techniques. In 1969, he founded and became President of Intertel, Inc. (now Infinet, Inc.), where the initial focus was on application of active filter techniques to slow-speed modems to reduce the size and cost. Intertel pioneered in introducing the first 9,600-bps modem to operate on unconditioned lines and the first Network Control System for problem diagnosis and restoral in on-line computer networks. In addition, he supervised significant research on signal space design and timing recovery techniques for QAM modems, which led to numerous published papers on this work. More recently, his interest has centered on modeling the noise encountered on "T-Carrier" telephone lines (which involve nonlinear PCM techniques) and on the application of forward error correcting and signal design to such channels. He has received numerous patents, and has several pending, for his work in these areas. He is currently a private investor pursuing the application of statistical modeling and prediction theory to the management of investments. ■