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MODEMS FOR DATA TRANSMISSION OVER TELEPHONE NETWORK

This paper is dedicated to prof. Ilija Stojanović on the occasion of his 75^{th} birthday and the 50^{th} anniversary of his scientific work

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Abstract. In this paper a short historical account concerning the modem development is presented. An extensive table summarizing the ITU Recommendations V.21 to V.34 is given and commented. The basic modems features are analysed. Some results from original experiments are also commented. The Recommendation V.90 concerning 56K modem is shortly duscussed.

1. Introduction

An important characteristic of telephone circuits is the wide range of different types of noise and distortion that can be encountered. Some of them are inherent to the medium used, but some are introduced by specific signal processing. Moreover, a telephone circuit is often a cascade of several different links, each of them having pecularities of its own. It is interesting to note that a telephone network, built up at the beginning to transmit speech signals in an analogue form, is used now to transmit data signals, digital by their nature. To accomplish this, corresponding modulation is performed to obtain a signal suitable for transmission over the voiceband channel. Modems are now standardized to transmit up to 33.6 kb/s over the telephone network. The new and special member of the modem family is the "56K" modem, allowing data rates up to 56 kb/s, being standardized (V.90) last year.

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Now, almost all main analogue links are replaced by digital (PCM) ones. Moreover, on some local links PCM4 systems are used to increase capacity. These systems are based on 32 kb/s ADPCM. PCM systems as well as ADPCM systems are designed for speech signal transmission (A or μ compression curves for PCM, corresponding time constants for ADPCM), so they are not suitable to process modem signals, being by their nature highly different from speech signals. The quantization noise may degrade modem performances significantly. It should be noted also that the quantization noise introduced by PCM is uncorrelated with the signal, while that one introduced by ADPCM is highly correlated. Finally, the switching centers are now mainly digital with corresponding A/D and D/A conversions.

In such a way, an interesting paradox occurred. Data signals are digital. By the aid of a modem an analogue signal is obtained, suitable for analogue transmission. Now, such signals are digitized on some links or in some switchng centers. Thus, there may be even more successive A/D and D/A conversions (PCM or even ADPCM). At the end of the circuit, the modem signal has to be reconstructed in an analogue form before delivered to the modem. The modem itself has to produce digital data for the user. So, data transmission over the existing telephone network is a very specific kind of data communications.

A suitable basis for studying the historical development of modems for data transmission over the telephone network (PSTN - Public Switched Telepnone Network) are the corresponding ITU-T (ex CCITT) Recommendations (Series V). They reflect the contemporary state of technology as well as the corresponding stage of the telephone network development. It should be noted that the recomendations give the "rules" concerning the modem behavior to the "outer world", while the manufacturers have the liberty to develop modem using any technology they can.

In this paper an extensive table summarizing the Recommendations V.21 to V.34 will be presented as well as the appropriate analysis of basic features. Some interesting results from original experiments will be commented. A short explanation concerning the "philosophy" of the "56K" modem will be given and the corresponding V.90 Recommendation will be discussed.

2. ITU-T Recommendations

The basic features of ITU-T (ex CCITT) Recommendations V.21 to V.34 are presented in Table 1. These Recommendations apply to "classical" modems with bit rates up to 33.6 kb/s. It is practically impossible to include all relevant features into the table. The most relevant features are included only. But, it suffices for following a historical development of modems as well as to get insight concerning the impact of technology on their performances. Besides these Recommendations, V.41 and V.42 should also be mentioned. They concern the error control as well as block (packet) retransmission. Recommendation V.42 *bis* includes data compression procedures for Data circuit terminating equipment (DCE) using error correction procedures.

Table 1. Basic features of ITU-T (ex CCITT) Recommendations V.21 to V.34

ITU-T (CCITT) Recom.	Bit rate $[b/s]$	Mod. type	Circuit type	Principal characteristics
V.21	< 300	FSK	switched	duplex, synchronous and asynchronous operation
V.22	1200	DPSK	switched, leased	duplex, scrambling, synchronous and start-stop operation
V.22 *bis*	2400	16QAM	switched, leased	duplex, scrambling, synchronous and start-stop operation, compromise equalizer in transmitter
V.23	600/ 1200	FSK	switched	synchronous and asynchronous operation, optional inclusion of backward channel (up to 75 Bd) for error control
V.26	2400	4DPSK	4-wire, leased	duplex, optional inclusion of backward channel (up to 75 Bd) for error control
V.26 *bis*	2400/ 1200	4DPSK/ 2DPSK	switched	optional inclusion of backward channel (up to 75 Bd) for error control
V.26 *ter*	2400	4DPSK	2-wire switched, leased	duplex, channel separation by echo cancellation, scrambling, compromise or adaptive equalizer
V.27	4800	8DPSK	4-wire, leased	duplex, optional inclusion of backward channel (up to 75 Bd) for error control, manually adjustable equalizer
V.27 *bis*	4800/ 2400	8DPSK/ 4DPSK	2-wire, 4-wire, leased	full-duplex or half-duplex (4-wire), half-duplex (2-wire), optional inclusion of backward channel (up to 75 Bd) for error control, automatic adaptive equalizer

Table 1. Continue

ITU-T (CCITT) Recom.	Bit rate $[b/s]$	Mod. type	Circuit type	Principal characteristics
V.27 *ter*	4800/ 2400	8DPSK/ 4DPSK	2-wire, 4-wire, switched	full-duplex or half-duplex (4-wire), half-duplex (2-wire), optional inclusion of backward channel (up to 75 Bd) for error control, automatic adaptive equalizer
V.29	< 9600	combined AM and DPSK	4-wire, leased	full-duplex or half-duplex, fallback rates 7200 and 4800 b/s, optional inclusion of multiplexer to combine data subchannels, automatic adaptive equalizer
V.32	< 9600	QAM (with optional TCM)	2-wire, switched, leased	duplex, channel separation by echo cancellation, fallback to 4800 b/s, optional trellis coding (TCM), automatic adaptive equalizer
V.33	< 14400	QAM (with TCM)	4-wire, leased	Fallback to 12000 b/s, optional inclusion of multiplexer to combine data subchannels, automatic adaptive equalizer
V.34	< 33600	QAM (with TCM)	2-wire, switched, leased	full-duplex or half-duplex, channel separation by echo cancellation, adaptive techniques to establish data rate (14 data rates from 2400 to 33600 b/s), optional auxilary channel (200 b/s)

Before commenting Table 1, a short recapitulation of recommendations concerning error control and data compression is necessary. According to Rec. V.41 (Mar del Plata 1968) error control is achieved using ARQ (Automatic repetition upon request) procedure. The modems must provide simultaneous forward (synchronous) and backward (asynchronous) channels. Storage for at least two data blocks must be provided at the transmitter. Transmitted blocks contain 260, 500, 980 or 3860 bits. Sixteen check bits are generated by dividing service and information bits with the generating polynomial $x^{16} + x^{12} + x^5 + 1$. If scrambling with self-synchronizing scramblers is used, the scrambler polynomial and the generating polynomial must have no common factors. Rec. V.42 (Melbourne 1988) considers error control procedures for DCE-s using asynchronous-to-synchronous conversion. Frame check sequence (FCS) consists either of 16 bits generated using the above mentioned generating polynomial or of 32 bits generated using the generating polynomial $x^{32}+x^{26}+x^{23}+x^{22}+x^{16}+x^{12}+x^{11}+x^{10}+x^{8}+x^{7}+x^{5}+x^{4}+x^{2}+x+1$. This Recommendation, twenty years "younger", is very extensive, splitting control function from error control function, including negotiation etc. As a final touch Rec. V.42 *bis* (Geneva 1990) describes data compression procedures to be followed by DCE-s using the error control procedures defined in Rec. V.42. The data compression function is added besides error control function.

The traditional estimate of the capacity of an analogue voiceband telephone channel is in the range od 30 kb/s [1]. This result is obtained when considering data transmission over voiceband channel with aditive white Gaussian noise (assuming a nominal bandwidth of about 3.5 kHz and a signal-to-noise ratio of about 30 dB). In the past decades much effort has been made to approach this capacity. The restricted bandwith limits the symbol rate by Intersymbol interference (ISI). Even, if ISI is completely eliminated, the signal-to-noise ratio dictates the number of levels having in view the permitted Bit error rate (BER). Almost all modems described in Table I have a symbol rate of 2400 Bd, except V.34 modem having a range of symbol rates from 2400 to 3429 Bd (some rates are optional!). So, after choosing the symbol rate so as to satisfy the Nyquist creterion, thus allowing the possibility of transmission without ISI, one should try to eliminate ISI in the established connection. This procedure is known as "equalization". The need for equalization is obvious. Modems are not designed for one and just one specific link. Better to say, they are designed for an "ensamble" of links (having similar characteristics). So, they will operate over links with various transfer functions (even the connection obtained by switching between the same two modems that do not change location may be established for every call over a different physical path). For low bit rates (say under 1200 b/s) there is no need for equalization. For higher rates some form of equalization is necessary. At the beginning it was the so called "compromise equalization". The equalizer is used with fixed coefficients obtained for a hypothetical link averaging over all possible transfer functions. It should also be mentioned that Lucky [2] from the very beginning recognized the important role of a transversal filter as an equalizer for data transmission. A variety of transfer functions that can be encountered on various links implies an equalizer with easily adjustable coefficients just being the case for a transversal filter. So, "manual" equalization was used in some modems, where coefficients were adjusted manually at the beginning of transmission. Of course, the further development of technology made it possible, the use an "automatic equalizer" leading at the end to an "automatic adaptive equalizer" being active not only at the beginning of transmission, but also during the transmission.

The problem of duplex mode was resolved at the beginning either by using 4-wire circuits or by channel separation by frequency division on 2-wire circuits. Later, channel separation was obtained by echo cancellation. Also there was the optional inclusion of multiplexer to combine data subchannels. The optional inclusion of a backward channel (up to 75 Bd) for error control was provided for almost all modems.

Rec. V.29 (Geneva 1976) made possible data transmission at a very "high" rate of 9600 b/s over 4-wire leased circuits. It was still far from the nominal capacity (cca. 30 kb/s). A new breakthrough was needed, and it become possible! At the beginning, modulation and error control coding were completely separated. It means that the modulation type was chosen according to some knowledge about channel characteristics (transfer function, interference, noise). Later, if needed, a suitable error control code was implemented, having in view the expected (or measured) error statistics. Of course, the type of errors depends on the channel bahaviour as well as on the signal processing implemented. For example, differential coding introduces often double (successive) errors instead of single ones. Ungerboek [3] was first to propose a practical implementation of the combined choice of modulation and coding suitable for channels with restricted bandwith. The result was Trellis coded modulation (TCM). The beginning was modest. Rec. V.32 (Malaga-Torremolinos 1984) made possible data transmission at the same rate (up to 9600 b/s), but over 2-wire switched curcuits. At the rate of 9600 b/s two alternatives were provided, one using 16 carrier states QAM scheme and one using trellis coding with 32 carrier states. The next modem was defined by Rec. V.33 (Melbourne 1988) for use over 4-wire leased circuits with data rate of 14400 b/s including an eight state convolutional coder for trellis coded modulation. The last (?) modem of the family was defined by Rec. V.34 (Geneva 1996). It operates at data signalling rates up to 33600 b/s on 2-wire switched circuits using TCM for all data rates with 16 state convolutional coder. Possible data rates are multiples of 2400b/s from 2400 to 33600 b/s (the last two rates 31200 and 33600 being optional). The symbol rates are 2400, 3000 and 3200 Bd with the optional rates of 2743, 2800 and 3429 Bd (note that the highest rate is just very close to the theoretical maximum of 3500!). The carrier frequency changes accordingly from 1600 to 2000 Hz (two values being possible for every symbol rate). There is an adaptive technique to establish the data rate the channel can support on each connection. Line probing signal is used to analyse channel characteristics. It consists of a set of tones spaced 150 Hz apart at a frequencies from 150 to 3750 Hz. Some tones are omitted. So, new methods as well as the technology developement enabled practically a hundredfold increase in data rates! It is also a big step from compromise or manualy adjusted equalizer to fully automatic modem according to Rec. V.34.

So, it looked like the end of the story about modems for voiceband channels. But, a new special member of a modem family (may be not from the same family, but also not a distant cousin) appeared at the market. Last year Rec. V.90 introduced "56K" modem allowing data rates up to 56 kb/s. The corresponding story will be postponed for the section at the end of this paper.

3. Some experimental results

The problems concerning modem data signal transmission over the telephone network may be a subject of theoretical analysis. They can be also investigated by computer simulation as well as experimentaly, under either laboratory or real conditions. Theoretical approach is rather complicated having in view a series of possible non-linear signal transformations. Using fast processors, computer simulation is promising, but often some idealised conditions are supposed (synchronization etc.). Laboratory experiments are just one step to the reality, but the "best" results should be expected from experiments under real conditions. Of course, the last ones can not be conducted twice under identical conditions. In this section some interesting results from two original specific experiments will be commented. The first one was conducted about 20 years ago, while the other took place just two years ago. The difference in experiment type as well as in the obtained results is obvious.

3.1. Voiceband data over PCM and DM channels

The laboratory configuration for experiments is shown in Fig. 1[4]. Data source as well as error counter was the standard instrument in that period - TREND 1-4. It can measure quantities relevant to assess the quality of data transmission over telephone lines (BER, BLER - Block error rate etc.). Only results concerning BER will be shown. The measurements were carried out using a PN generator (with period of 511 bits) for a time interval of 15 minutes (according to contemporary valid CCITT recommendations). There was a possibility to attenuate the modem (SRT 2084, working according to Recs. V.26 and V.26 bis at rates 1200 and 2400 b/s) signal. In such a way the various distances from user to coder were simulated. The nominal signal level was 0 dBm and the attenuation steps were 6 dB. The pairs coder-decoder (codec) allowed to investigate the use of "standard PCM" (64 kb/s, compression according to A-law) and the use of CVSDM (Continuously Variable Slope Delta Modulation) with rates of 16, 32 and 64 kb/s. The corresponding abbreviations are PCM, DM16, DM32 and DM64. There was also a possibility to introduce the channel errors. They were generated using a white noise generator. When its output (sampled once for every bit) was greater than a given (variable) threshold, the corresponding bit was complemented. The number of errors was registered by the error counter. Division of this number by the number of transmitted (PCM or DM) bits gave the line error rate (P_{lin}). After that, PCM (DM) signal was decoded and fed into the modem. In such a way BER for data signal (P_{dat}) was obtained. The configuration used allowed to introduce more (up to five) sections with the A/D and corresponding D/A conversions, but without the possibility to insert further line errors.

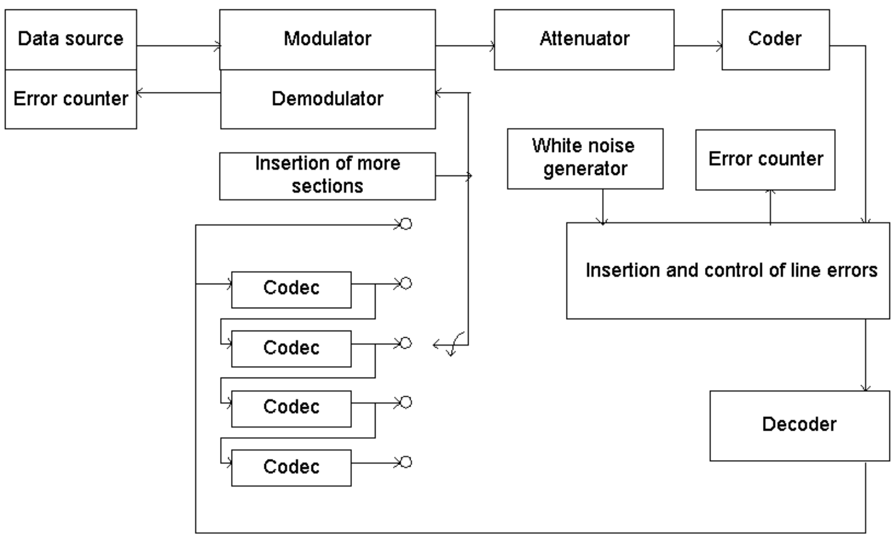


Figure 1. Laboratory configuration for experiments with PCM and DM

It was concluded that the modem signal level at the PCM coder input (L_{in}) practically does not affect the quality of received modem signal at the rate of 2400 b/s. Also, the modem signal of acceptable quality was obtained after more (up to five) A/D and D/A conversions. On the other hand, the resistance to line errors was small. The acceptable quality (P_{dat}) was obtained only if line error rate (P_{lin}) was smaller than 10^{-5}, according to

Table 2. Concerning the delta modulation, only the basic conclusions will be cited. Using DM64 allows reliable transmission of modem signals (2400 b/s) over one or more sections. The coder input level is not important. There is a very good resistance to line errors (even to $P_{lin} = 10^{-3}$ and greater). DM32 is acceptable for 2 or 3 sections with careful choice of coder input level. The resistance to line errors is smaller than for the DM64 case, but still greater than that for PCM. The use of DM16 to transmit modem signals with rate of 2400 b/s is practically impossible. It should be noted that the results were obtained using a modem from that era (the modem under investigation had a compromise equalizer), so with modern modems better results should be expected.

Table 2. BER (P_{dat}) for some values of signal level at the coder input (L_{in}) and line error rate (P_{lin})

$L_{in}[dBm]$	$P_{lin} = 10^{-3}$	$P_{lin} = 10^{-4}$	$P_{lin} = 5 \times 10^{-5}$	$P_{lin} = 10^{-5}$
0	10^{-3}	10^{-4}	3×10^{-5}	2×10^{-5}
-6	2×10^{-3}	10^{-4}	3×10^{-5}	6×10^{-4}
-12	2×10^{-3}	10^{-4}	5×10^{-5}	2×10^{-5}
-18	3×10^{-3}	3×10^{-4}	5×10^{-5}	3×10^{-5}
-24	4×10^{-3}	4×10^{-4}	8×10^{-5}	2×10^{-5}

3.2. Voiceband data over PCM and ADPCM systems

During the experiments [5] the transmission of modem signals over ADPCM links was investigated. Now, 32 kb/s ADPCM is standardized (ITU-T G.726) for doubling the number of voice channels over the existing digital links built up for standard 64 kb/s PCM. The main difference between the quantization noise introduced by PCM and that one introduced by ADPCM is that the first one can be considered as a white Gaussian noise (uncorrelated with the signal), while the latter one is highly correlated with the signal and should be considered as a type of nonlinear distortion. So, an analytic approach is not so easy and experimental investigation is often done instead.

Several configurations were used (Figs. 2-5). TT 9400 stands for ADPCM transcoder Twin Trunk 9400, PCM4 for four channel ADPCM unit and 2H2 for Siemens equipment for BER and BLER measurements. Some configurations were realized in the laboratory, while some comprised of actual telephone lines. Some configurations chosen were in fact a part of the existing national network, while ADPCM units were configured according to topology which can be expected in such a network.

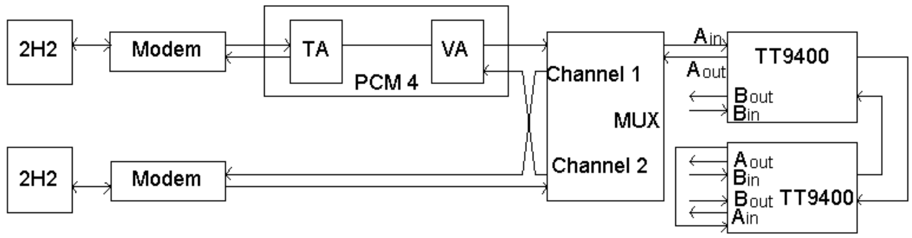


Figure 2. Laboratory configuration for simultaneous work of PCM and two ADPCM transcoders

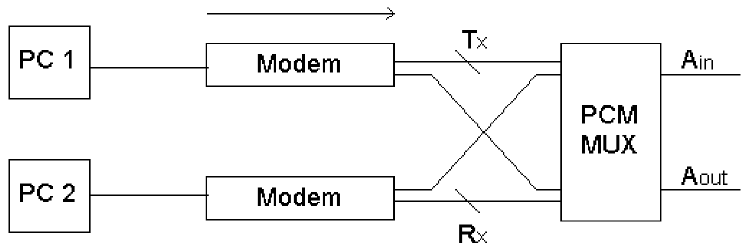


Figure 3. The basic configuration for testing the influence of more ADPCM transcoders

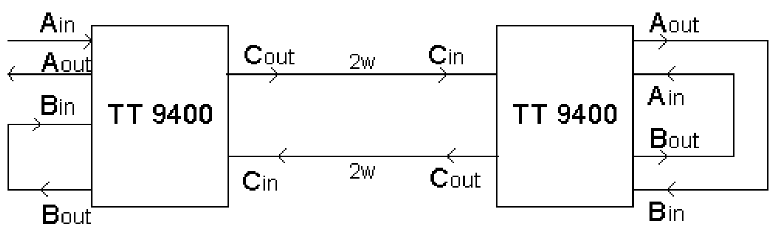


Figure 4. Aditional laboratory configuration for testing the influence of more ADPCM transcoders

All results were obtained by averaging over more sessions. Some results are shown in Table 3. Data signals were generated according to Recs. V.26 bis, V.27 ter, V.29, V.32, V.32 bis and V.33. The duration of one session was 5 minutes, so during the session more million bits were transmitted. Note that during the preceeding experiments (subsection 3.1), the BER was

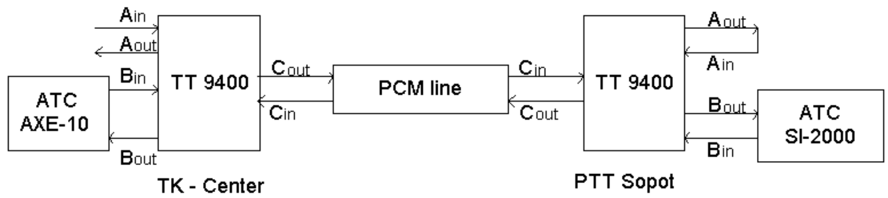


Figure 5. Aditional configuration for testing the influence of more ADPCM transcoders including real line

calculated on the basis of 15 minutes sessions.

Table 3. Results for various tested configurations

Configuration	ITU-T Rec.	Bit rate [b/s]	BER
Figure 2	V.29	7200	7.4×10^{-6}
Figure 2	V.29	4800	3.9×10^{-6}
Figures 3 and 4 concatenated	V.33	14400	4.05×10^{-6}
Figures 3 and 4 concatenated	V.33	12000	1.17×10^{-6}
Figures 3 and 5 concatenated	V.32	9600 (TCM)	9×10^{-6}

For V.32 modem TCM stands for using TCM option. If the value of BER less than 10^{-5} is considered satisfactory, all tested configurations for given data rates are acceptable. Of course, the results for laboratory configurations should be taken with some caution. The general conclusion is that for an international telephone connection, where one or more ADPCM links can be expected, modems would at 4800 b/s operate successfully, as well as, in most cases, at 9600 b/s. These results are in good agreement with conclusions from [6] where similar (but not identical!) configurations were tested. It is also to be expected that new generation modems (more sophisticated) would show slightly better performance under described circumstances.

4. The "56K" modem

4.1. General

The appearance of the "56K" modem is a result of a logical continuation of a line of thoughts outlined in the Introduction. To repeat it shortly:

Speech signal is an analogue signal. So, PSTN was developed for its transmission. Now, a great part of the telephone network is digitized, having in view its basic goal - to transmit speech signals. PCM was introduced with suitable A or μ characteristic as well as sampling at a rate of 8000 samples per second and each sample represented by 8-bits code word yielding channel rate of 64 kb/s. Now, in many countries, PSTN is practically everywhere digital except the line from the subscriber to the switching centre ("the last mile"), which is usually a twisted pair. In the meantime the great neeed for data transmission found an economic solution in using PSTN as a medium. Modem signals, mentioned in previous sections, were designed for transmission over an analogue telephone network. In the digital networks these signals were digitized using procedures specially developed for speech signals. So, quantization noise may be a limitting factor for modem performance.

If we are not allowed to change network characteristics, we have to change modem properties. So, modem signals should be designed in such a way to minimize quantization distortion [7]. The result of such an approach is the "56K" modem, sometimes denoted as "2X" or "PCM" modem.

The use of these new modems is expected in cases where a subscriber is connected via an analogue line to a switched digital network. Of course, if the subscriber buys an ISDN connection, he can use basic configuration (2B+D channels with total bit rate of 144 kb/s) and there is no need for a 56K modem.

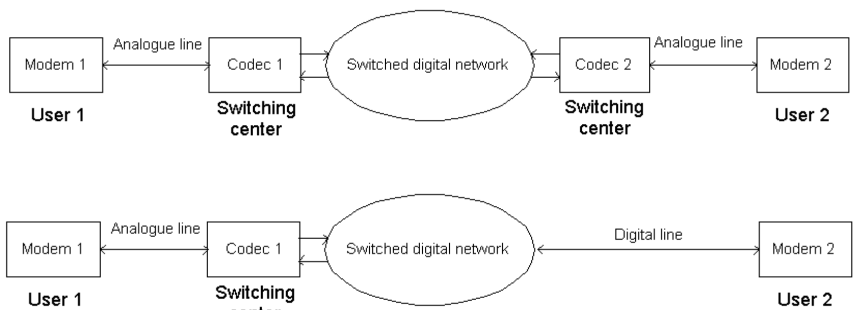
The basic features of Rec. V.90 are as follows. The modem consists of a digital modem and a an analogue modem pair. Tha data signaling rates are up to 56000 b/s downstream (from PSTN to subscriber) and up to 33600 kb/s upstream (from subscriber to PSTN). Channel separation by echo cancellation techniques allows duplex mode of operation. In fact, communication in the upstream direction is according to Rec. V.34. Symbol rate in the downstream direction is 8000 Bd (corresponding to 8000 samples per second in PCM systems). Downstream data rates are from 28000 to 56000 b/s in increments of 8000/6 b/s. Adaptive techniques enable modems to achieve close to the maximum data rates the channel can support in each connection. If a connection does not support V.90 operation, full duplex V.34 operating mode can be negotiated.

The first manufacturer of a 56K modem (under the name "X2") was 3Com (US Robotics) and a little latter Rockwell and Lucent Technologies started with the "K56flex" modem. At first, these modems could not interoperate.

It should be noted that the data rate on actual lines of good quality is 40000 to 45000 b/s for more than 90 % connections. So, the 56K modem is probably the last one in the series of modems for data transmission over telephone network.

4.2. Some details

In. Fig. 6. [7] some possible connections through PSTN are shown. In. Fig. 6a the first and the last sections are conventional twisted pairs which transport the analogue signals to the switching centers. In the switching center these signals are digitized and converted to 64 kb/s PCM data streams, transported through a switched digital network and after corresponding D/A delivered to destination.



(a)

(b)

Figure 6. Two connections through a PSTN [7].
(a) analogue end–to–end; (b) hybrid

This configuration has two codecs. A hybrid connection, shown in Fig. 6b, is very common as residental users access the Internet via service providers. So, only a single codec is present. That introduce only one A/D or D/A conversion.

Now, the problem may be stated as follows. Data, at a rate of 64 kb/s, are avaliable from the provider in switching center (before D/A conversion). Why not transmit data at the same rate via the analogue line? It is well known that more than 1 Mb/s may be transmitted over twisted pairs at short distances. The answer is that the codec must not change. For Fig. 6b from the standpoint of Modem 1, Modem 2 could be as well as in the switching center. The codec, designed for speech transmission, will convert every 8-bits word into an analogue value and send a contionuous signal to Modem 1 ("thinking" that it reconstructs a speech signal). If bits entering the D/A convertor (codec) are data bits, then the generated signal will remind to Pulse amplitude modulation (PAM) having 256 levels. So, the idea is clear. If Modem 1 can distinguish received amplitudes without errors, it will easily reconstruct 8-bits data words. Of course, there are some "technical difficulties". Filters for voiceband channels are passbands filters having a bandwidth of about 3500 Hz. The codecs use A or compression. Because of the restricted bandwidth (3500 Hz) the transmission without ISI is possible using signal rates up to 7000 Bd. In fact, actual symbol rate of such a modem can be 8000 Bd, but during the 125 s interval, from eight equidistant symbol positions, seven positions can be used (each symbol representing a 8-bits code word), while the position eight can not used. So, the maximum possible data rate is 56000 b/s. The equalization is used to achieve transmission without ISI for mentioned seven successive symbol positions. If ISI is eliminated, the number of levels is determined teoretically with acceptable probability of error depending only on signal-to-noise ratio. It should be mentioned that signal-to noise ratio in this case (twisted pairs between centre and user) is much higher. The main part of noise is here due to nearby crosstalk from other voiceband channels. According to [1] it is above 62 dB, while for the capacity of 64 kb/s 55 dB will sufice. The levels are not equidistant, but they are generated according to A or law. The number of levels is at most 256, so probability of error can be reduced only by omitting some levels. Also, the number of information carrying positions can be reduced. The combination of levels and information carrying positions omitting results in different data rates. The receiver can be realized as 7 paralel adaptive receivers, each of them for one of the information carrying position. In such a way, the corresponding algorithm for equalizer (usually LMS) is easy to implement, because of periodicity of the samples. For equalization and voltage levels adjusting a good "cooperation" between modems must be realized using training sequences and negotiation (and renegotiation!). So, the connection shown in Fig. 6a is not suitable, because the presence of two tandemed codecs and the absence of any side information which could be used for training equalizers and echo cancelers.

5. Conclusion

Data transmission over a telephone network is a very specific kind of data communication. The main reason is that a network for transmission of analogue (speech) signals is used for transmission of digital signals, under the condition that network characteristics could not be changed. Only a modem can be put instead of a phone. Respecting this constraint modems were designed to give best performances. With the development of technology, some new ideas can be put into practice. But, the new technology led also to digital telephone networks. So, modems had to take into account new network behavior. The answer was the 56K modem. Now, we are maybe at the end of long journey where speech and data transmission will meet and start a new "married life".

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