ADAPTING TELECOMMUNICATIONS MODEMS TECHNIQUES FOR USE IN UNDERWATER COMMUNICATION SYSTEMS

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Underwater channel data transmission is one of telecommunications most difficult problems. One way to optimise the transmission may be by adapting transmission techniques used in telecommunications modems. The articles presents two ITU-T standards: V.34 and G.992.1, known as ADSL. The possibilities and limitations of these techniques for underwater communication systems are discussed. The ADSl standard using the DMT modulation seems to be the best.

INTRODUCTION

The existing PSTN (Public Switched Telephone Network) infrastructure was designed for human voiceband transmission. Adapted to carrying analogue signals with limited bandwidths, the infrastructure does not allow fast data transmission. Rather than replacing the entire analogue telecommunications infrastructure, a new device was devised, called the modem. The job of the modem is to adjust digital data to the type of data normally carried in a telecommunications transmission channel. It compresses and modulates the data leading to substantial transmission speed gains in analogue telephony, as is the case with analogue modems or, in the wide useful band of copper cable, in the case of digital modems.

The underwater communication channel is not only band limited, but stationary as well and particularly difficult because of strong distortions. Therefore, underwater communication is in the focus of intensive research aiming to reach higher rates and better quality of transmission. The mainstream research is on data processing and transmission techniques. Thus the question – which of the techniques used in telecommunications modems can be used for underwater communication systems?

The article gives an overview of the methods used in the modems V.34 (section 1) and ADSL (section 2). Section 3 discusses ways to adapt the selected methods of coding and modulation.

1. TECHNIQUE V.34

The first telecommunications modem was built in 1958 in Bell laboratories. Its speed was 0.3 kbit/s in an analogue telephony channel with a frequency range of 300Hz-3400Hz. With the progress made in acoustic band data transmission, the ITU-T V.34 modem could be designed, offering a transmission speed at 33.6 kbit/s. It could be possible by applying complex techniques of compression and error-correction encoding.

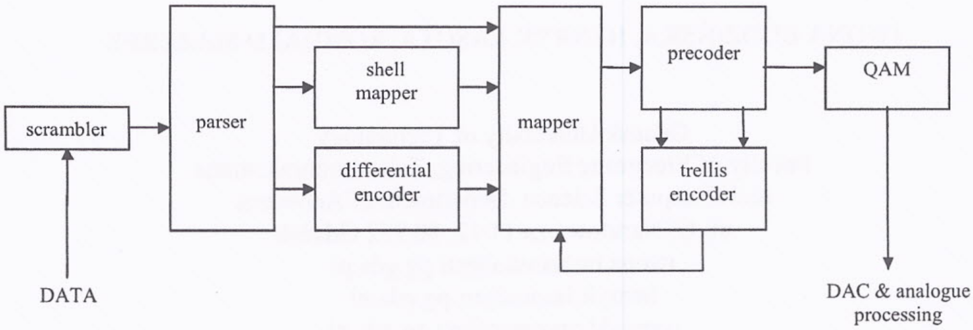

Figure 1. Functional block diagram of the V.34transmitter.

In the transmitter of modem V.34 (Fig. 1) the input stream of bits is processed by the scrambler, a shift register, which changes the sequences 0 and 1 by modulo 2 sum operation. It changes any possible serial errors into single random errors and eliminates long periodic sequences 0 and 1, which can generate a periodic signal on the transmitter's output. The periodic signal has pics in its spectrum and can negatively impact the performance of the receiver's adaptation filters. The scrambler eliminates the effects of pulse distortions, which may produce serial errors. The signal spectrum at transmitter output fits in better with the channel characteristics because the pics are cut out.

The parser divides the block of bits from the scrambler into three groups, which from there on are processed parallel by the point select unit consisting of the shell mapper, differential encoder and mapper.

The point select unit forms signal constellation points corresponding to integers. The process ensures maximal compression of the information and protection from AWGN noise and other distortions in the telecommunications channel.

Information is compressed using a shell mapping algorithm, which maps bits of information into integer co-ordinates, to form a signal constellation. Signal constellation is a subset of a multidimensional array of integer points, called a superconstellation. Each integer point has its complex representation in the constellation. It is set by co-ordinates on the complex plane and the rotation factor computed by the differential encoder.

The size of the constellation depends on transmission parameters. For the fastest transmission, the biggest constellation, i.e. the entire superconstellation, is chosen. For the lowest transmission speed, only central points of the superconstellation are chosen.

Using the shell mapping algorithm, the shell mapper computes integers, so called ring indexes, from the first group of bits. The mapper uses these ring indexes to determine integer points in signal constellation and rotation factors which are computed by the differential encoder, to determine complex signal samples.

In determining minimal distances between the powers of constellation points, the shellmapping algorithm makes sure that they are greater than the power of channel noise. The selected transmission speed determines the number of signal constellation points, i.e. the resolution of signal point power.

The precoder pre-emphasises signal points, i.e. it carries out complex filtration of a sequence of complex signal samples. Filter coefficients are computed in the receiver by the LMS adaptation filter during transmission of probing sequences. The filtered signal component is added to the signal in the receiver.

The trellis coder introduces the so called trellis sequence into the sequence of complex signal points. The sequence secures the distances in between the points protecting the resolution of signal point power. In the trellis coder each signal point is dependent on a previous point. When signal points are the result of signals from the past, errors caused by noise and inter-symbol interference can be detected in the receiver and corrected by the Viterbi decoder.

The modulator forms the analogue signal to match the conditions of the telephone channel. Raised cosine filters smooth out the analogue signal and protect it from inter-symbol interference. The QAM modulator modulates the quadrature signal on 1800 Hz.

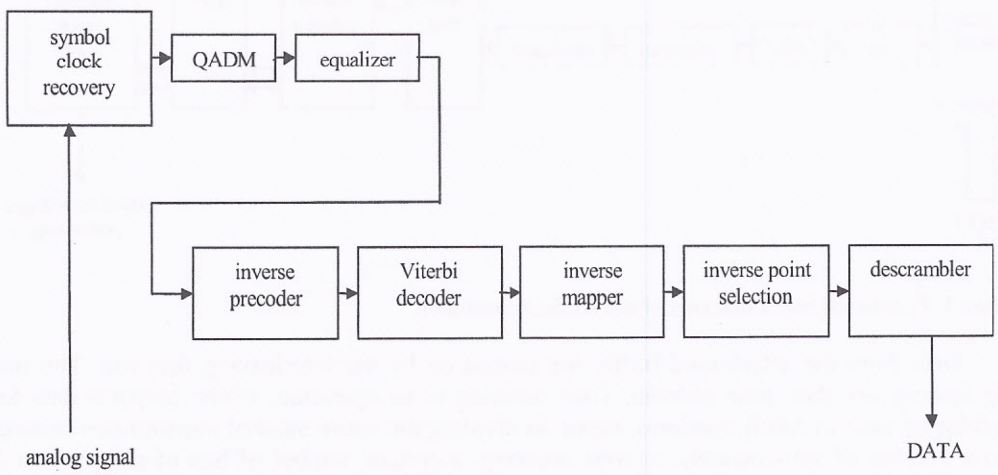

Figure 2. Functional block diagram of the V.34 receiver.

Figure 2 shows the V.34 receiver. The clock recovery system is responsible for transmitter-receiver synchronisation. It corrects the receiver's sampling rate to ensure that it matches the transmission speed. The echo canceler reduces inter-symbol interference caused by line impedance inhomogeneities. It identifies and removes echo from the signal received. . The equaliser cleans the signal of interference caused by non-linearity in the propagation path. It filters the signal received via LMS adaptation filters whose coefficients are computed during training sequence transmission. The coefficients are used by complex filters in the precoder and inverse precoder.

The inverse precoder introduces to the signal a component which was previously filtered from the signal by the transmitter's precoder. The Viterbi decoder checks the trellis sequence which was previously introduced into the signal by the trellis coder in the transmitter and corrects errors detected in a signal point sequence. The inverse mapper and inverse point selection unit are responsible for performing inverse functions in the transmitter. Also descrambler is a shift register, which realize the inverse function of scrabler.

2. THE ADSL TECHNIQUE

The ADSL technology, described in ANSI T1.413 and ITU-T G.992.1 recommendations, is designed to transmit signals in a telephone network above the band of traditional human voiceband.

Fig. 3 shows a functional block diagram of an ADSL transmitter. The data stream is multiplexed into two separate buffers of data (fast and interleaved). Next, the two streams undergo a cyclic redundancy check (CRC) and Forward error correction (FEC), based on Reed-Solomon codes. Each of the streams is scrambled using the shift register, as for the V.34 modem.

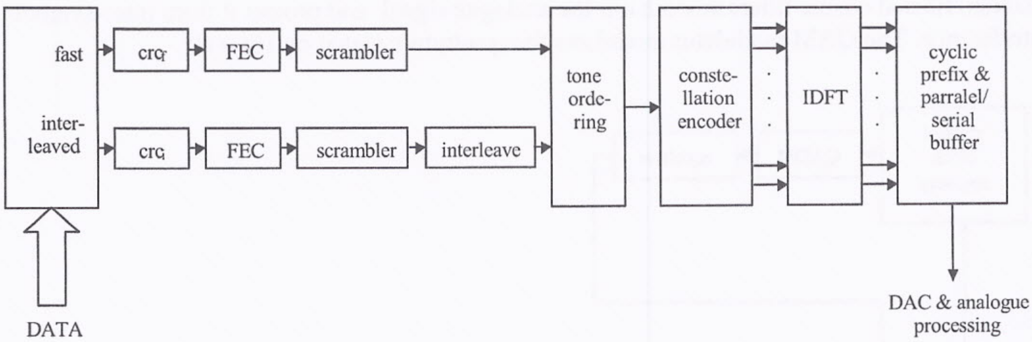

Figure 3. Functional block diagram of the ADSL transmitter.

Data from the interleaved buffer are passed on by the interleaving function. The two data streams are then tone ordered. Tone ordering is an operation, which prepares data for modulation used in ADSL modems, based on dividing the entire band of transmission channel into a number of sub-channels. In tone ordering, a certain number of bits of information is assigned to each of sub-channels. All sub-channels are tested during training and actual transmission. Those with the biggest degree of interference receive the smallest number of bits or no use is made of them at all. Those with the best transmission performance receive the biggest amount of information to carry. On the other hand, sub-channels that offer the fastest transmission are susceptible to clipping errors occurring during ADC and DAC conversion. The errors can be corrected by FEC coding, as long as sub-channels with the biggest number of bits received data from the interleaved buffer.

Tone ordered bits are recorded in the bit allocation table to form data symbols which constitute the constellation encoder's input. The constellation encoder can be used with or without the trellis coder. The output data from the constellation encoder, i.e. signal points, are modulated using one of the two modulations DMT or CAP. DMT is used more commonly.

Discrete-MultiTone is a variation of OFDM (Orthogonal Frequency-Division Multiplexing). It offers a very good level of radio frequency interference and pulse noise resistance. In DMT modulation the main channel is divided into a number of sub-channels. The frequency range 0..26kHz is not used. The entire range of the ADSL modem frequencies, in this case 26 .. 1130 kHz, is divided into 256 sub-channels, 4,3125 kHz each. The distance between the centres of adjacent channels is 4,3125 kHz as well.

In each of the sub-channels the signal is formed using QAM signal point modulation. The points were produced using the bit allocation table recorded during tone ordering. Depending on the transmission rate, which can vary from 2 to 15 b/s/Hz, one of the following modulations will be applied: 128-QAM, 64-QAM, 32-QAM, 16-QAM, 8-QAM, QPSK.

QAM signals are interpreted as frequency domain samples and processed using the IDFT (Inverse Discrete Fourier Transform) processor into time domain samples. Next, the parallel block of samples is transformed into a serial sequence, followed by the addition of a cyclic prefix, which delineates the DMT symbol. The signal undergoes the digital to analogue conversion and filtration to be then sent to the telephone network.

The ADSL received signal is converted in the analogue equaliser and ADC converter. Next, the cyclic prefix is removed from the stream of signal samples and the stream itself is formed into a parallel block. Following the DMT demodulation, where the signal samples block undergoes DFT transformation, the data are processed by the decision-making system responsible for constellation and trellis encoding, if applicable. Following FEC decoding and descrambling, the information is reproduced.

3. MODEM TECHNIQUES AND UNDERWATER COMMUNICATION

A unique feature of underwater communication is that the acoustic wave signal is transmitted by a time-varying medium. Because of absorption the channel is band-limited. The majority of underwater communication systems operate below 30 kHz. The maximal range and transmission rate are functions of channel physics.

The interference inherent to underwater data transmission depends on the type of communication channel. For a shallow channel, the interference mainly involves reverberation and time domain (travel time) and frequency (Doppler) spreading. In deep water, however, white Gaussian noise is the dominating interference.

There are two factors causing time domain spreading. The first is the motion of water surface. It leads to multi-paths and reverberations. The second is the relative motion of the transmitter and receiver. In addition, the motion causes Doppler shift in the signal being transmitted. The shift is described using mean transmitter and receiver speed and random fluctuation around the mean. The Doppler shift is the result of phase-delay in several of the paths.

When frequencies undergo continuous spreading, the result is Doppler-spread in the signal, which compared to Doppler shift is much more difficult to compensate for in the receiver.

Research on underwater communication systems mainly aims to develop error-correction and reverberation compensation techniques in the time (multipath) and frequency (Doppler spreading) domain, to improve system reliability .

Telecommunications modems V.34 and ADSL follow a similar functional design. The first element is the scrambler. There is every indication that the scrambler would work very well in underwater communication modems. The probability of serial errors here is greater than in the case of telephone cables. The signal needs to be protected from such interference and the scrambler seems to be best suited for this task.

The second element of the functional design of telecommunications modem transmitters is that the data stream is divided into several sub-streams, which are processed independently from each other on this part of processing. The processing of each sub-stream follows a different algorithm. In V.34 modems the data stream is divided into three sub-streams. The first is processed by the algorithm of maximum complexity, producing decimal integer numbers, the second binary sequence is converted into quarternary numbers while the third stream remains binary. In ADSL modems the data stream is divided into two sub-streams which undergo similar processing. The only difference in how the sub-streams are processed is the interleaving function.

While proper signal compression is ensured by multi-dimensional parallel processing, its significance for error-correction encoding is secondary. Compression of information is an important aspect of underwater communication and by developing relevant techniques data transmission rates can be improved.

The individual data streams are joined into multi-dimensional constellation points by way of constellation encoding. In addition to data compression, another role of constellation encoding is to manage the signal point's energy. In selecting signal points the idea is to pick signals with minimum energy distance strong enough for two adjacent signal points to be discriminated by the receiver. It seems there is every reason to use constellation encoding in underwater communication as it helps to improve transmission rates by compressing the information and maintaining discernible distances between the levels of signal power.

Proper distances between signal constellation points can be assisted by trellis encoding which introduces a known by receiver structure into the sequence of the points. Consequently, the decoder in the receiver (e.g. Viterbi decoder in a V.34) can detect or even fix errors in the sequences of samples it received. Methods like these can be successfully used for underwater communication systems. The only downside is the redundancy of this coding technique. On the other hand, however, it is quite unlikely for a data transmission system in an underwater environment to have a redundancy lower than the one introduced by trellis encoding.

While V.34 modems use echo cancellation all the time, ADSL treats it as an option. Reflections and reverberations are the primary aspects of acoustic wave propagation in underwater environments, and consequently echo cancellation devices have to be much more complex than in transmission via cable.

Digital encoding techniques used in telecommunications modems can be applied in underwater communication systems without any significant limits. This is because there is no direct connection of this techniques with channel physics. The complexity of digital processing is largely dependent on the modulation technique used.

Because of the linear nature of propagation in an underwater channel, signal frequency characteristics remains within its original band. Signal amplitude and phase, however, can vary largely both in time and space because of reverberation and fluctuation.

The QAM modulation involving the encoding of amplitude and phase, can be used to transmit acoustic signals on a single carrier frequency in underwater communication, but it requires complex decision feedback equalisers with adaptation filters. Time domain and frequency domain spreading leads to effects similar to those of inter-symbol interference in telecommunications. To help with that, adaptive equalisers must be used because of their good performance in fighting signal symbol spreading in telecommunications.

DMT systems use a number of tonal pulses to transmit different amounts of information. The decision to choose a particular sub-channel is based on signal energy detection at the output of narrow-band filters in the receiver. The band edges of the entire channel can be modified, if only a Doppler shifts occur. This type of modulation seems best suited for underwater communication, rather than QAM. The band it uses is wide allowing a wide choice of sub-channels and ensuring the best transmission characteristics. The choice can be verified during the actual transmission. The DMT modulator and demodulator consist of a number of QAM modulators and a DFT processor. QAM modulation in DMT technology is used in narrow sub-channels where transmission rates are low and no complex filters or equalisers are necessary.

In telecommunications systems the ADSL transmitter terminates with a DAC converter (and analogue filters), which converts all signal samples into an analogue current signal consisting of all carrier frequencies used by the DMT modulator. Underwater acoustic transducers have a narrow band of transmission and the signal they generate has at maximum several component frequencies. Consequently, no digital DFT processing would be required for an underwater ADSL system. The frequency-to-time-domain transformation could be an analogue process using a number of acoustic converters. DMT modulation's primary problem would then be the design of converter lay-outs.

The LinkQuest Inc. - one of the world's leading producer of acoustic communication systems - offers underwater acoustic modems with a bit error rate less than 10^-7, a result comparable to the quality offered by cable telecommunications. The transmission rate of these modems is up to 19200 bits/sec at up to 1,5 km on the depth of 1 km. This was made possible by using an ADSL based modulation and mobile telecommunications techniques.

Range [m]	Depth [m]	Data rate [bits/sec]
350	200	9600-19200
1500	1000	9600-19200
3000	6000	2500-5000
4000	6000	4800-9600
10000	7000	2750-5500

Table 1. Data rates offered by LinkQuest modems.

4. CONCLUSION

The article gave an overview of techniques used in telecommunications modems. There is every reason to believe that the signal encoding methods they use can be used successful in underwater communication, if only the simulation results are good. More caution and in-depth analysis is recommended when it comes to adapting modulation techniques. It would be worthwhile to test DMT modulation in a natural environment. It should be expected that, if the modulation results are positive, the encoding methods described in the article can be applied in underwater communication without restraint.

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