

Chapter 2

Topics Bordering the Physical Layer

2.1 Time Synchronization

The actual need for time synchronization within an underwater acoustic network is not always present. It can be argued that given a network with an operation time of hours or a few days, any standard equipment will have a clock drift that is negligible given most applications and network protocol stacks. Given this argument, synchronization of clocks can be done on board, before deployment. This might be true for some cases, even though from a practical and logistical point of view, especially when the number of nodes gets large, it gets time consuming to access all nodes individually through their electrical interface to set their clock manually. Another option is to synchronize clocks through switching the power on simultaneously for all nodes, but this can also be impractical. Common for both approaches is that the accuracy will vary and errors might occur (human in the loop). From this point of view it would be beneficial to be able to have the nodes doing time synchronization through the actual acoustic network.

The accuracy of a clock is inflicted by factors as temperature, supply voltage, shock [1] and ageing, all which an underwater network node is experiencing. The accuracy of the clock crystal is given in parts per million, ppm. Typical accuracies found in simulations for time synchronization methods are 40 ppm [1, 2], 50 ppm [3] and 80 ppm [4]. For example, given a clock with an accuracy of 40 ppm this effectively means a clock skew of 40 microseconds per second. Given an operation time of a week, the resulting clock offset will be 24 s. Such a figure might also result in a need for re-synchronization after deployment.

The application of the network is important when deciding the need for synchronization. Sensor networks might be divided into basically three categories in this respect [2]. The first group of applications merely requires the order of events, while the second requires the time interval of each of the events, whereas third require the absolute time of the event. Same type of division might also be true if any actuators are connected to the nodes. Delivery of packets in an underwater

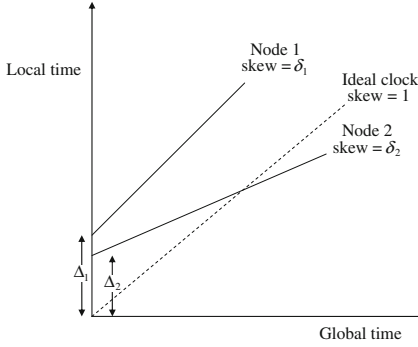


Fig. 2.1 Clock inaccuracies in two nodes

network generally has a high and non-deterministic latency, so time-stamping of sensor and actuator data with a global clock might be beneficial for many applications. Applications as target tracking and positioning requires time stamping. Also sleep scheduling for saving power will need time synchronization between nodes within a network. When it comes to the implementation of the network protocol stack, TDMA-based MAC protocol schemes benefit strongly from time synchronization.

2.1.1 Clock Inaccuracy Model

To avoid frequent re-synchronization between nodes it is beneficial to both estimate the clock skew and offset. The local time of any node i is related to the true global time, t by

$$t_i(t) = \delta_i \cdot t + \Delta_i$$

where, $t_i(t)$ denotes the local time of node i at time t , δ_i the clock skew and Δ_i the clock offset. Figure 2.1 illustrates the clock inaccuracies for two nodes. Generally it is assumed that clocks are short term stable, which is that they do not vary while doing estimation of clock skew [1]. This means that the clock drift can be represented with straight lines in the figure.

2.1.2 Time Synchronization Protocols

As for most of the other aspects in underwater networks, solutions and methods in wired and terrestrial networks can not be directly applied [1]. The Network Time Protocol (NTP) used for time synchronization on the internet copes with latencies

but does not consider energy consumption issues. In RF-based sensor networks it is usually considered that the propagation delay is negligible, assuming nearly instantaneous and simultaneous reception and ignoring movement of nodes during synchronization. In underwater networks we know that propagation delays are large and variable.

Minimizing overhead of signalling for time synchronization is important due to the generally low data rate in underwater acoustic networks. The re-synchronization frequency should be minimized, thus the synchronization algorithm should be able to maintain a certain accuracy without the need for frequent re-synchronization. When re-synchronization is required the system performance should not degrade substantially. Any mobile nodes in the network introduce the need for the synchronization algorithm to compensate for the movements during synchronization.

Cross layer design with time stamping at the MAC layer is suggested by the work performed on synchronization within underwater acoustic networks [1, 2]. Utilizing data from the PHY layer in the protocol also shows to be beneficial [4].

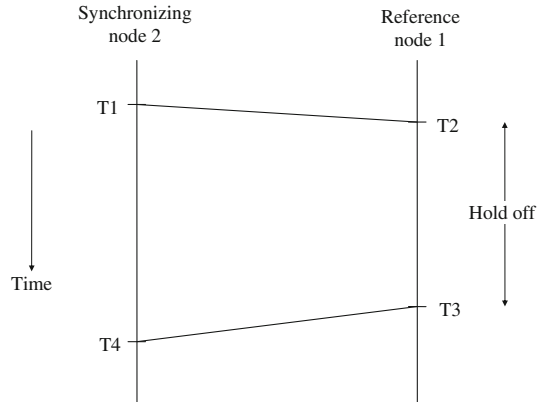
Generally, what seems not to have been studied in detail in the literature found is how the synchronization is achieved network wide. Great amount of detail can be found on synchronization between two nodes or within a cluster, but things can get complicated when nodes start to move, extra nodes are deployed, or nodes are taken out of the network. Reference [2] is mentioning that a cluster needs to select its cluster head, but does not discuss it in detail. This might introduce some additional overhead for time synchronization, and is a point of further study. The degradation of time accuracy as a function of number of hops in a multi-hop network is suggested in [1] to degrade as the square-root of the number of hops. This is based on the assumption that the error per hop follows a Gaussian distribution of equal standard deviation. This might be a viable first order assumption.

The Mobi-Sync [3] method is one of the latest time synchronization protocols and is getting some attention for synchronization within networks with mobile nodes. What is special with this method is that it assumes that nodes are spatially correlated. That means that when one node moves, the other nodes also move in a related pattern. Even though this is the case for e.g. free floating drifters in a sea current, this generally does not hold when having gliders and AUVs in the network. The method also requires a dense network with every node having contact with at least three or more super nodes, a super node having correct time, in order to perform well. For static networks there are more energy efficient methods.

2.1.2.1 Time Synchronization for High Latency (TSHL)

The protocol TSHL [1] was proposed to compensate for high latency in acoustic networks. The method estimates and compensates both for clock skew and offset. This work assumes static nodes, and performance is strongly degraded when nodes are moving. It is shown in [2] that the method performs even worse than no

Fig. 2.2 Two way message exchange (T1-4 are time stamps)



synchronization when nodes move. This is due to the fact that the estimation of clock skew is inflicted by the movement.

This method is among the most energy efficient in the literature found. For estimation of clock skew, a beacon node is first broadcasting a number of messages to the neighbouring nodes. Every neighbouring node is then using linear regression on the times of arrival to estimate its clock skew. After that a single two way message exchange, see the scheme in Fig. 2.2, is used for estimating the clock offset. This is done as the reference node is informing the synchronizing node about time stamp T2 and T3 in the message sent back to the synchronizing node.

2.1.2.2 MU-Sync

The MU-Sync method [2] is designed for mobile networks. It assumes a cluster based network, and in contrast to the TSHL method it is the cluster head that takes responsibility for initiating and calculating the clock skew and offset for the nodes in the cluster. The cluster assumption does not exclude the method from working within a sparse network with maybe only one neighbour node to the cluster head.

The method is relying on two way message exchange for acquisition of clock skew and offset. The number of messages suggested is 25, same as for TSHL. The cluster head is then using linear regression to calculate clock skew and offset. Finally these parameters are distributed to each node.

2.1.2.3 D-Sync

Integrating the Doppler estimate of the PHY-layer for relative velocity estimates with the time stamps, preferably also at the PHY-layer, the D-Sync method [4] represents a novel approach for time synchronization in mobile underwater

acoustic networks. Similar to Mu-sync it is the beacon or cluster head that initiates and calculates the clock skew and offset relying on two way message exchange. At the end the clock skew and offset is distributed to the synchronized node.

Reference is made to [4] for details of the method. There are two main sources of error in the method: the error due to Doppler measurements and the error due to the fact that Doppler measurements are not available continuously.

In a coherent transmission scheme accurate estimation of Doppler of the received signal is important to be able to equalize and decode the transmission. So the actual Doppler measurements tend to be very accurate. The work assumes a nominal error of 0.1 m/s, but simulations with up to 0.5 m/s are performed.

The Doppler is measured at T2 and T4 in Fig. 2.2. The time between these two measurements will in a dense network with a slotted contention based MAC protocol be governed by the Hold off time (T3–T2). This time might be several tens of seconds. This leads to a potential under-sampling of the Doppler and thereby the actual movement of the node. For slower moving nodes under water such as AUVs and gliders, this might not have such a severe effect. But it can be imagined that for gateway buoys on the surface submerged nodes in the splash zone exposed to wave motion, this under-sampling will degrade the performance of the algorithm.

Anyhow, simulations show that for a given set of parameters, including a network of 10 nodes distributed within a square of 1000 m sides, the error of the time sync two hours after the synchronization is 20 ms. This maps into an error of 2 s after a week of operation after the synchronization.

There is also described a light weight protocol B-D-Sync that has the same power consumption as TSHL. The performance of this protocol introduces a degradation of 5 times compared to the full D-sync.

2.1.3 Summary

Time synchronization is not always needed in an underwater acoustic network, but might be required given a long deployment, applications as target tracking or TDMA based protocols. Handling and logistics of nodes might also be simplified if they can be synchronized after deployment.

There exist a few time synchronization protocols in the literature. They all estimate both clock skew and offset in order to be able to minimize the need for re-synchronization. TSHL is suitable only for static networks, while Mu-Sync and D-sync are suitable for mobile networks. Even though designed for mobile networks it might be stated that even these methods would benefit from avoiding movement of nodes during synchronization.

All work on the methods considers local time synchronization between two nodes or within a cluster of nodes. Further work must be done to find optimal ways of getting network-wide synchronization in a multi-hop network.

2.2 Full-Duplex Links

A full-duplex link allows communication in both directions simultaneously. Full-duplex over the same physical medium is often emulated using the methods of Time-Division Duplex (TDD) or Frequency Division Duplex (FDD). TDD is bordering Time Division Multiple Access (TDMA) in functionality where separate time slots are used for sending and receiving signals. FDD is bordering Frequency Division Multiple Access (FDMA) where separate frequency bands are used for sending and receiving signals.

Full-duplex links are common in the cabled and radio frequency domain, included in systems as ADSL (cabled), UMTS (mobile) and satellite communication systems, while half-duplex links are predominant in underwater communication systems. No commercial full duplex modems seem to be available and a limited number of experiments has been conducted [5–7].

Obtaining full-duplex in a network is affecting the complexity of the link layer as well as the physical layer: The link layer may become simpler while the physical layer will be more complex.

2.2.1 Link Layer

Many half duplex underwater acoustic network protocols use collision avoidance by reserving the channel through a request to send and clear to send (RTS/CTS) session before accessing the channel. Further, flow control is often implemented using some kind of stop-and-wait flow control mechanisms. Given the large propagation delay of the acoustic channel this will lead to a lot of time waiting with potential low resource use efficiency. Study of channel reservation and flow control is done in Sects. 3.3.2 and 4.2, respectively.

If the available bandwidth is channelized and nodes are assigned unique channels within their respective two-hop neighborhoods, the need for collision avoiding coordination prior to message transmission is eliminated as each node is effectively operating over point-to-point links with its neighbors. The resulting full-duplex communications also allow for more efficient flow control mechanisms, such as sliding-window based methods [8]. The large propagation delay of the channel will result in the channel to act as a virtual buffer of data waiting to be read by the receiver.

The down-side of allocating two unidirectional channels for each connection is that it may result in very low bandwidth efficiency, unless the traffic from each node is regular and constant. However, in most data communications exchanges, the data is irregular and bursty, resulting in periods where allocated bandwidth is unused. This will again lead to potential low resource use efficiency.

In [8], this is mitigated by techniques from satellite communications capacity management: Demand Assigned Multiple Access (DAMA) controls and

Bandwidth-on-Demand (BoD) techniques. DAMA allocates channels to users when the users request an allocation. These channels are typically fixed in size. Alternately, BoD provides users a variable sized allocation depending on the request of the individual user. By allocating multiple channels to a user on demand, these channels may be used to inverse multiplex two or more message frames, thus providing a coarse version of BoD. It is this coarse BoD implemented via DAMA that is suggested.

2.2.2 Physical Layer

Time division duplexing is no option in an underwater network with the large propagation delays of the acoustic channel. Frequency division duplexing was demonstrated in [5] where data was transmitted between the shore and a ship. Using two transducers on the ship, one for transmission and one for reception, with a distance of 33 m between them, they managed to transmit and receive simultaneously in adjacent frequency bands. The distance between the ship and shore was up to 4500 m. Reception was good on both the ship and the shore. The frequency bands are not known, but Chebyshev filters with 80 dB out-of-band rejection were used for side-lobe suppression.

Obtaining full-duplex can also be achieved by using Code Division Multiple Access (CDMA) based techniques, see Sect. 3.2. In [7] a test using CDMA based channelization schemes was performed in a bucket and a small lake with a distance up to 5 meter. Separate Tx and Rx transducers were used with a spacing of 30 cm. Several channelization schemes were tested including frequency hop CDMA, time hop CDMA, direct sequence CDMA, and also hybrids thereof. Pulse position modulation was used for keying the data onto data symbols. In these tests, frequency hop CDMA performed best.

2.2.2.1 Transducers

All demonstrations utilize a separate transducer for transmitting and receiving signals, and apparently the most successful [5] having a long distance of 33 m between them. On a node like an AUV this kind of distance is not available, and preferably it should be a single transducer both transmitting and receiving. No literature could be found on full-duplex transducers. Radio systems like maritime VHF and maritime ship radio stations have FDD channels using only a single antenna.

2.2.3 Concluding Notes

Full-duplex has this far not been demonstrated to work well for underwater acoustic communications, and the required hardware is not commercially

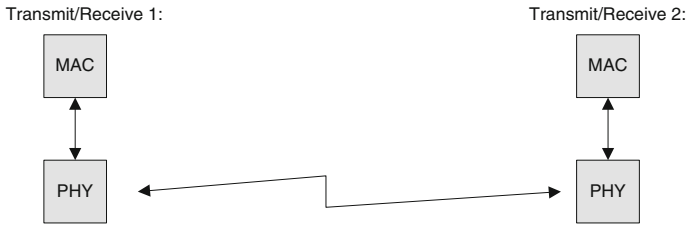


Fig. 2.3 The traditional MAC/PHY division (the OSI-model)

available. But if a good full-duplex solution is found in the future, it could significantly improve the performance of underwater acoustic network protocols.

2.3 Adaptive Data Rate

By Adaptive Data Rate in this context it is meant that the communication system is able to utilize some knowledge about the present state of the communication channel so that both the coding and modulation methods can be adapted to this state. The goal is to maximize the system throughput under varying channel conditions.

One way to achieve this is to employ a technique which in the telecom industry is known as “Adaptive Coding and Modulation”, or ACM. This is used today in both wired and wireless communication systems.

In order to use adaptive coding and modulation effectively, it is necessary to establish a close interaction between the operations of the physical layer (PHY) and the medium access control layer (MAC).

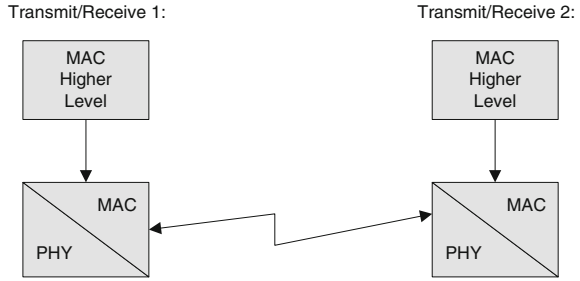
In a traditional telecom environment, where circuit-switched networks (ISDN and ATM) were the norm, there were clear distinctive lines between the responsibilities of the PHY and the MAC, as it is laid out in the OSI model and as shown in Fig. 2.3.

With the introduction of packet switched networks (Ethernet and others), these lines have been considerably blurred, and this has lead to a simplified model where some of the layers have reduced functionality and others are removed.

In a communication scenario that involves ACM, this model will have to be changed further in that some of the traditional MAC functionality, such as the selection of the modulation format and coding scheme, will have to be moved down to the PHY layer in order to be able to respond to the (quickly) changing channel conditions. This part of the MAC functionality is sometimes referred to as the “lower-level MAC functionality” (Fig. 2.4).

In this scenario the “higher level MAC functionality” is responsible for establishing the overall system parameters like quality-of-service (QoS) requirements for the individual links, and to organize the network for maximum system capacity, the latter being important in an ad-hoc/mesh network scenario.

Fig. 2.4 A re-organized MAC/PHY division



In this discussion of adaptive data rate, only the PHY and lower-level MAC functionality will be considered, and it is easier to take the bottom-up approach and identify the requirements of the PHY first.

2.3.1 The Physical Layer

The discussion of the PHY-layer will be based on a conceptual transmitter/receiver pair, as shown in Fig. 2.5, below. Note that more detailed discussions on physical layer technology for underwater acoustic communications are not part of the present study.

In a communication system, each terminal will at least have one such transmitter/receiver pair (or “transceiver”). In addition, a transmit/receive switch circuitry is needed if time-division duplex (TDD) operation (not shown in the figure) is required.

In order to achieve maximum system capacity, this hypothetical system would have to be able to utilize all access techniques, such as frequency division multiple access (FDMA), time division multiple access (TDMA), code division multiple access (CDMA) and space division multiple access (SDMA). These access techniques are described elsewhere in this document, and will not be repeated here.

The various access techniques all have different requirements with respect to clock stability, frequency stability, linearity of up-/down-conversion chains and number and size of transducer elements, and will eventually be dictated by a cost/benefit trade-off.

A short description of the various functions carried out in the building blocks of Fig. 2.5 is given below.

From the information source comes the user data to be transmitted over the link. A forward error correction (FEC) encoder protects the data before transmission by adding special bits (parity) or bit-patterns that can later be utilized in the receiver to extract the original information bits. There are a number of different coding schemes that can be applied for this purpose, and they range from simple to advanced block-coding structures (from Hamming to Reed-Solomon) to convolutional and Turbo codes, and various concatenations of the above.

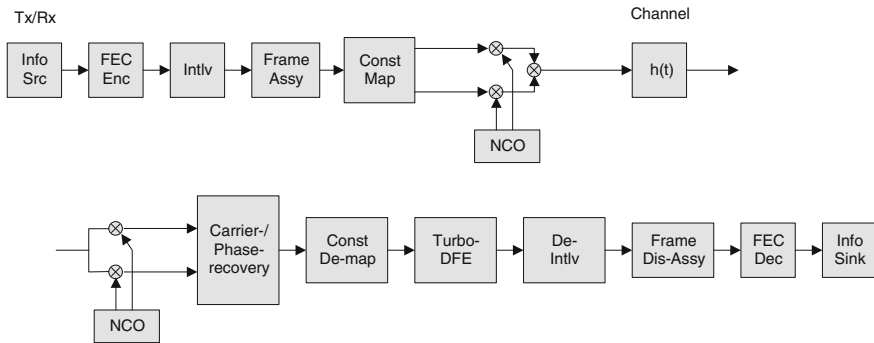


Fig. 2.5 Conceptual transmit/receive functionality. See text for explanation

In the interleaver, the encoded data are repositioned in the data-stream according to a predefined structure. This is to avoid loss of data caused by impulsive noise which would be detrimental to a convolutional code decoding process. (This is not so much of a problem with block-codes.)

In the frame-assembly block, the information is inserted into a frame where preambles like unique words (for frame alignment) or various pilot-assisted modulation (PSAM) bits and post-ambls like CRC's and/or end-of-frame (EOF) delimiters are placed.

In the constellation mapper, the individual bits of the frame are mapped onto a suitable alphabet of symbols, later to be modulated onto two orthogonal waveforms (for a quadrature modulated signal).

For a coherently modulated signal, the alphabets can be a set of symbols belonging to a modulation format like e.g. binary phase shift keying (BPSK), quaternary phase shift keying (QPSK) and higher order like quadrature amplitude modulation (QAM).

For a code division multiple access system based on a direct sequence spread spectrum (DSSS) technique the alphabet is a set of orthogonal spreading codes.

For a non-coherently modulated signal, the alphabet is e.g. a set of frequencies in a multiple frequency shift keying (MFSK) modulation format.

A code division multiple access system can also be constructed by using a set of frequencies in a frequency hopping pattern orthogonal for each code.

The symbol set is then modulated onto two orthogonal carrier waveforms (cosine and sine), where the carrier frequency is generated through a numerical controlled oscillator (NCO). In Fig. 2.5, a direct-to-carrier type of system is shown. The transmitted signal, $s(t)$, is a real band-pass signal.

At the receive side, the signals are converted directly from carrier (real band-pass signal) to complex base-band, the demodulation frequency (and phase, in case of a coherent demodulation scheme) are again controlled by a NCO. The exact frequency and phase are controlled by the carrier-frequency- and phase-recovery sub-system.

Not shown in the figure are the necessary signal/burst acquisition sub-systems.

The information bearing signals are then extracted from the signal constellation and fed to an equalizer to remove inter-symbol-interference (ISI) induced by the channel.

In the frame disassembly operation, the information bearing signal is extracted and fed to the de-interleaver before FEC decoding and the result is then fed to the end-user.

In [9], it is claimed that a coherent modulation scheme based on Phase Shift Keying (BPSK/QPSK), in conjunction with an adaptive decision feedback equalizer (DFE) and a spatial diversity receiver is an effective way of combating the effect of multipath fading in a shallow water environment. It is, however, admitted in the article that the excessive delay spread, often several hundred symbols, make it too computational complex for real-time operation for such a system.

2.3.2 Medium Access Control, Lower Level

In order to be able to utilize the communication channel effectively, an optimally configured system will have to be able to change both the modulation format and the coding scheme in order to adapt to the current channel conditions.

To make this work, it would be necessary to collect information about the current channel conditions, convey this information back to the transmitter side and use this to select the modulation and coding schemes for the next outgoing burst.

The information to be used in this process could be e.g. an estimate of the signal-to-noise ratio and/or the delay and Doppler spread of the channel, and this information could be extracted from known symbols in the frame structure like the pre-amble (e.g. a unique-word) or the PSAM symbols (if they are used).

However, any addition of extra symbols to aid in these processes would incur an overhead and thus reduce the information data rate.

Other, more indirect means of extracting the same information could be to monitor the states of an equalizer or the decoding depth of the trellis in an (convolutional code decoder) FEC decoding process.

A different way of obtaining information about the channel state could be to use a special burst sent in advance as a channel probe, as proposed in [10]. In their system, such a probe is sent just before the actual information burst is transmitted, as shown in Fig. 2.6.

Here station 1 is transmitting a probe before the actual data is transmitted, and this is used at station 2 to extract some key properties of the channel state at the moment of reception. This information is then affixed to the data in the next outgoing burst from the station 2. If the turn-around time is short this is a more or less correct description of the channel, and it is used at station 1 for the next outgoing burst from this station.



Fig. 2.6 Using a channel probe to extract information about the channel

In addition, if the channel can be treated as reciprocal, station 2 can itself use this information to set the parameters for the next outgoing burst, and/or to construct a better probe for more detailed channel measurements.

The approach described in the paper is a very simple adaptation of this, but it lacks a conclusive statement about the system improvements, if any, that can be achieved using this technique. It is not clear why a special probe is required to do this, as the same result can be achieved by simply embedding the same information into the pre-amble of the burst itself.

2.3.3 Adaptive Data Rate in ARQ Systems

Adaptive Coding and Modulation is something that is normally closely connected to the physical layer processes of a communication system, as described in the previous sub-sections. This is due to the fact that very low latency is required in order for the system to respond to the changing channel conditions.

If ACM methods based on direct measurements of the current channel conditions are not possible, e.g. where the time-delays in the system are so severe that the channel measurements are obsolete by the time the system can respond to these, an ACM approach based on the use of information extracted from the ARQ system (see [Chap. 4](#)) can be foreseen.

One way to achieve this could be that instead of using an approach where the previous packet is blindly repeated when an ACK time-out occurs or a NAK is received, one would use a scheme where the first re-transmission is just a re-transmission of the packet. If this transmission is also unsuccessful, the packet is reformatted into a longer burst where the modulation and coding schemes are strengthened (more energy per bit and/or more protection bits added).

If the transmission still fails, this back-off procedure is repeated until all possibilities are exhausted and the system finally breaks down.

When the ACK packets start to arrive, the modulation and coding overhead is gradually reduced until the maximum possible information bandwidth is re-established on the channel. The transmit side will always be aware of the current maximum channel capacity based on the reception of the ACK-packets, since this will confirm that the receiver is able to decode the packets.

2.3.4 Summary and Conclusions

In terrestrial wireless communication systems, ACM has been in widespread use for some time. But as is pointed out in [11], the key difference between a terrestrial radio-based communications system and an underwater acoustic communication system is the large propagation delay, low bandwidth and high bit-error-rate. Their conclusion is that a direct adaptation of a terrestrial radio protocol may not provide acceptable results, and that protocols for this purpose need to be developed from the ground up.

The research into adaptive data rates in underwater communication systems seems to be in its infancy, and in order to move this research forward and in the end create a successful system employing ACM the following factors need to be addressed:

The special requirements for an underwater acoustic communication system need to be taken into account already at the design stage. There are currently no existing solutions that can directly be applied.

Measures for estimating the (strongly time varying) channel impulse response need to be built into the burst structure in order to reduce measurement latency and overhead in the communication link.

In order to have maximum flexibility in the selection of possible signalling waveforms to use at any given time, a software defined radio (SDR) approach is necessary. In any case, lessons learnt in the SDR arena should be taken into consideration in a new design.

In order to be able to realize a fully software defined radio, the hardware platform need to be as flexible as possible, and that means that all the necessary functions must be realized in digital domain. Especially important in this respect is to ensure that the digital-to-analog (DAC) and analog-to-digital (ADC) conversion subsystems have the speed and precision needed.

Fortunately, the latter is steadily improving with the advances in ADCs, DACs, field-programmable-gate-arrays (FPGA) and microcontrollers (uC), and in the end the transducers and the required signal conditioning circuitry will be the limiting factors.

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