UNDERWATER TELEPHONY: PAST, PRESENT AND FUTURE

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Abstract - Underwater telephones allow divers to talk to each other or to a supervisor on the surface. This paper outlines the principles of various systems used in the past, how they have improved for use at present, and what developments are likely in the future.

1 - INTRODUCTION

Conventional telecommunication techniques are now very sophisticated, particularly with the advent of mobile telephones, but techniques for communicating underwater are still comparatively crude. As with any other communications system, all that is required in principle is a method of conveying a meaningful message from one person to another. The essential parts are a source (a voice signal sent by a microphone), a channel (a wire link or the water itself), and a destination (the voice signal received by a headphone).

But there are particular problems associated with an underwater system that are not shared with a conventional system. One of the main problems is the poor quality of the source signal caused by the diver speaking, often unclearly, into a limited bandwidth microphone placed in a small resonant cavity. This changes the frequency spectrum of the speech, and therefore distorts the signal from the microphone. The signal is further degraded by the introduction of noise at the source, in the transmission channel, and at the receiver /1/.

At the source there is breathing noise, noise from free-flowing air (in helmets) and, for divers breathing oxy-helium, the well-known 'Donald Duck' effect due to the increased speed of sound in helium. Non-linear microphone distortion, together with indistinct articulation, adds to these problems.

In directly-wired systems there is noise from crackle due to the ingress of water into connectors and umbilicals, and electrical noise from extraneous sources such as motors and switches. In through-water acoustic systems, there is noise due to wave action, biological sources, shipping and many other mechanisms. In addition, there is the problem of multipath interference which gives rise to delayed versions of the signal reaching the receiver. At the receiver, there is often audible noise from engines, generators, telephones and conversation.

The quality and reliability of an underwater telephone system are important to diver safety. In most underwater work, a communication link is a statutory requirement because it allows coordination with surface activities, monitoring and control of decompression stops, warnings of dangers, calls to assist other divers and diagnosis of medical problems. The problems outlined above indicate that it is difficult to design a system having acceptable quality of received speech. Present systems, both hard-wired and acoustic, are virtually unchanged from those developed many years ago. Any future development will depend on cost and perhaps the imposition of new standards, rather that any technical breakthrough.

2 - PAST AND PRESENT

Divers have used hard-wired telephones since 1905. Early versions used direct transmission of the voice signal without modulation, which made them prone to noise pick-up from audible sources. One advantage was that the diver's microphone and speaker worked in the comparatively large, dry volume of a traditional "hard hat". The voice signal would therefore have been better than that from a modern band mask which has a very small air volume in front of the diver's mouth.

The majority of versions developed since use amplitude modulation. These systems work in the same way as conventional telephones and are still widely used because they have proven reliability. For many commercial, police, navy and scientific divers, the wire link can be conveniently included in the umbilical line supplying the breathing gas.

The two basic types are two-wire and four-wire configurations. With the two-wire system, each diver is connected to the surface unit by two wires. This allows transmission in one direction at a time, i.e. half-duplex operation. The supervisor is normally on receive mode, allowing him to monitor the divers. To talk to them he uses a Press-to-Talk (PTT) switch to reverse the direction of communication. This means that while he is talking he cannot hear the divers, an obvious potential hazard.

With a four-wire system, each diver is connected to the surface by four wires in a 'round-robin' configuration. This allows communication in both directions at the same time, i.e. full-duplex operation.

Of the many acoustic systems that have been designed and manufactured, the commonest type uses the principle of amplitude modulation with single sideband suppressed carrier (SSBSC). This has the advantage of having half the bandwidth of earlier conventional Amplitude Modulation (AM) systems and Double Sideband (DSB) systems (i.e. AM without carrier). Typically, the speech is transmitted as a band-limited signal that modulates a carrier frequency centred on the resonant frequency of an omnidirectional electrostrictive transducer. There is no legislation on the use of frequency bands underwater, although certain frequencies have been adopted unofficially for communications /2,3/. The lower the frequency, the greater the range of transmission achievable, because range is inversely proportional to the square of frequency.

Another factor that governs the choice of frequency is the effect of ambient noise, which is most noticeable at low frequencies, particularly in the audible range. While some systems use frequencies as low as 8 kHz and as high as 70 kHz, the most common frequency used in commercially available systems is 42 kHz. This is high enough to avoid excessive extraneous noise, and low enough to give a range of about 1 km, depending on the power output of the transmitter.

Any diver breathing an oxy-helium mixture requires a helium speech unscrambler, first used in 1970. This is necessary because the speed of sound in helium, being nearly three times higher than in air, causes the speech to become garbled. The principle of the device is to write samples of speech into a memory and subsequently to read them back at a slower speed to render it intelligible.

According to a recent estimate by the U.K. Department of Energy, there are probably some 250-500 communication systems used for oxy-helium diving, and some 2000-5000 for air diving /4/.

3 - FUTURE

The main scope for improving both hard-wired systems and acoustic systems is by improving the quality and reliability of components, especially microphones and headphones /5/. Speech can be digitally encoded but this will not improve the quality and intelligibility of the received signal if the diver does not speak clearly and if all the noise sources mentioned earlier are still present. There is an obvious need to reduce noise input from breathing and air flow, possibly by noise cancellation techniques, although this does not seem to be economically viable at present.

Since acoustic systems are usually self-contained there is scope for miniaturising circuits. With microchip technology this is easily achievable. There is also scope for optimising the system design. But there are other problems that are not so easily surmountable, for example multipath interference. The other factor that need to be considered is the harsh environment (ingress of salt water and physical abuse).

All these considerations are important, but often the main cause of poor quality reception is the poor quality of the diver's speech. Garbled or indistinct speech, together with interfering noise, reverberation, resonance, band limitation and non-linear distortion are bound to lead to a degraded received signal. If a diver with a clear, accentless voice can only just be heard when he is fresh and relaxed, he may become unintelligible when tired, stressed or in panic. The mental processing of distorted speech is a strain for any listener, whether diver or supervisor, which may lead to a breach of safety. One suggested solution is to improve communication discipline by training divers to speak more clearly, adopt standard phrases, and employ the phonetic alphabet. But these suggestions have not been adopted by commercial divers /6/.

Most presently available commercial systems comprise electronic circuits enclosed in a waterproof housing attached to the diver's aqualung, with connections to a microphone and bone conduction headphone. A better arrangement would be for the diver to carry the entire system on his head. The electronics, batteries, microphone, earphone and acoustic transducer can be mounted in a helmet or attached to a mask. This would obviate the need for long wires that tend to get in the way or get damaged. Advances in technology now allow the use of large scale integrated circuits, designed by computer aided engineering, and miniature long-life batteries.

Most acoustic systems use half-duplex transmission controlled by a PTT switch. An alternative to this is a voice-operated switch (VOX), which allows transmission automatically when the diver speaks. These may be used more in the future if they can be used with a transmitter circuit that can distinguish the voice signal from breathing noise. For safety reasons, full-duplex transmission would be preferable, although this leads to greater circuit complexity.

In principle, digital encoding of speech can be achieved by a variety of well-known modulation techniques that are widely used in telecommunications. Techniques include pulse width modulation (PWM), pulse position modulation (PPM), pulse code modulation (PCM) and delta modulation (DM), each of which has particular advantages and disadvantages. These are perfectly feasible for hard-wired systems, although they do not seem to have been incorporated in presently available systems.

For through-water acoustic systems, there are severe problems preventing future development, apart from the ever-present phenomenon of multipath interference. One of these is the low sampling rate achievable. With a speech bandwidth of about 3 kHz, the Nyquist sampling theorem dictates that the minimum sampling rate must be at least twice this, i.e. 6 kHz. The corresponding sampling period is therefore 167 µs, so whatever modulation method is chosen this is the maximum time allowed to transmit the encoded information. This is a typical duration for a pulse transmitted from a resonant transducer, which means that any kind of encoding requires much shorter pulses than this. Since the transducer is a resonant device that rings down from its steady state amplitude after excitation by a "rectangular" pulse from the modulator, the transmitter will generate a much longer acoustic pulse into the water. The precise length of the acoustic pulse is difficult to control, which would appear to limit the application of PWM, in which the pulse width represents the voice signal amplitude. For the same reason, PPM may be difficult to apply, since PPM can be derived from PWM.

Generating a multibit code in such a short period also appears to rule out PCM and DM. This is chiefly because each of these techniques requires the generation of a multi-bit code to represent an analogue signal amplitude. The 1's and 0's would have to be transmitted as two different frequencies, i.e. by frequency shift keying (FSK), which would be technically extremely difficult at a practical sampling rate /7/.

One possible technique that is feasible is pulse time modulation (PTM). This is very similar to PPM, and sometimes referred to as such. The analogue amplitude is represented by the time separation between pulses, so if this can be achieved all the transmitted pulses can have the same duration and the same carrier frequency. The receiver then has to measure the time between consecutive pulses to reconstitute the original signal. This technique should allow different sampling rates to be tried to optimise the quality of the received signal.

4 - CONCLUSIONS

Presently available underwater telephone technology has not moved ahead with the microchip revolution and other advances. There is some scope for improving the design of through-water acoustic systems, by miniaturising circuits, by re-thinking the physical configuration, by designing better microphones and headphones, and by digitising the speech signal, notably by pulse time modulation. The biggest breakthrough would be to incorporate these improvements and also to suppress the various extraneous noise sources and offset multipath interference.

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