

SPREAD-SPECTRUM COHERENT ACOUSTIC COMMUNICATION BETWEEN A SUBMARINE AND A SURFACE SHIP

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This paper describes an underwater acoustic communication experiment executed in the North Sea near Norway. A submarine broadcast direct-sequence spread-spectrum signals with data rates of 167 and 33 bit/s. The receiver station was a surface ship with a towed hydrophone array. As the platforms sailed away from each other the SNR at the receiver progressively dropped, with a continuously changing channel impulse response. It is shown that these complicating factors are largely overcome by beamforming of the hydrophone signals in combination with chip-level channel equalization. After the spread-spectrum gain, applied at a late stage in the reception chain, the transmissions are virtually free of errors up to a distance of 12 km.

1. INTRODUCTION

To investigate the possibilities of acoustic communication during multistatic submarine searches, a communication run was planned for the French/Dutch/Norwegian multistatic sea trials in September 2002 [1]. The signals were designed for transmission with a low-frequency active sonar (LFAS) towed by a surface ship. However, owing to hardware problems the LFAS was not available for the run and the trials team improvised by broadcasting the communication signals with a smaller source onboard a submarine. This source had a 30 dB lower source level than the intended LFAS. Fortunately, part of the SNR thus lost could be regained by employing a towed hydrophone array as the receiver station. Unlike vertical arrays, which have proven to be useful for underwater communication by exploiting spatial diversity [2, 3], i.e. an impulse response that varies with the depth, horizontal arrays hardly benefit from such a diversity. For the channel impulse response varies much faster in the vertical plane than it does in the horizontal plane. Consequently, spatial filtering in the horizontal plane (beamforming) is mostly useful to filter out noise, thereby increasing the signal-to-noise ratio (SNR) while leaving the channel impulse response more or less intact.

2. EXPERIMENTAL DETAILS

The acoustic source was mounted in the sail of the submarine and was operated at a source level of 180 dB re 1 μ Pa @ 1 m. The receiver array comprised 64 hydrophone triplets with a longitudinal spacing of 36 cm. During the run, which lasted 2 hours, the submarine sailed at a speed of 8.5 knots, while the surface ship sailed at 6 knots. At an angle of 45 degrees between their headings, the mutual distance increased from 3 to 25 km throughout the experiment (see Fig. 1). The depths of both the submarine and the towed receiver array were 80 m, which roughly corresponded to the sound channel axis revealed by in-situ measurements of the sound speed profile. As to the water depth, the bathymetry varied between 250 and 350 m during the run.

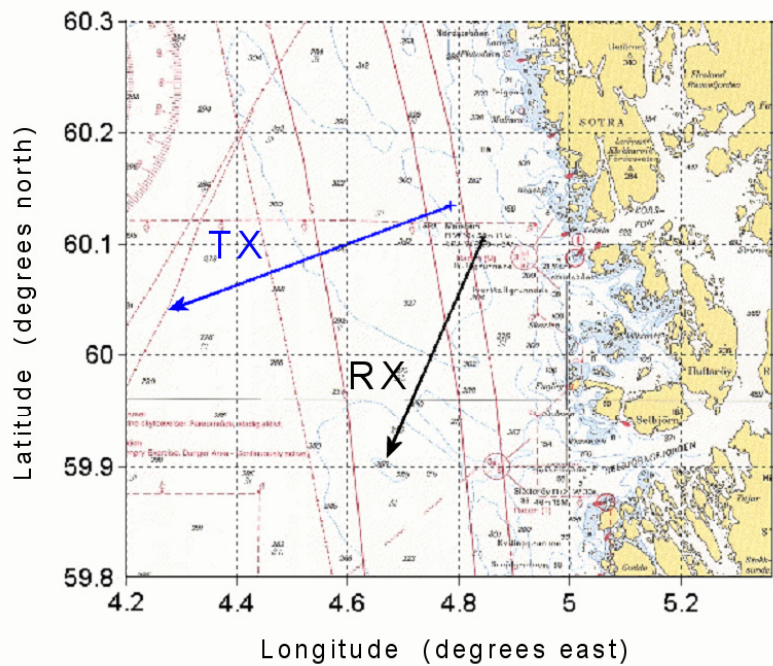


Fig.1: Trajectories of the platforms. TX = submarine; RX = surface ship.

3. SIGNAL PROCESSING

The communication signals are described by the following parameters: passband 1-2 kHz, BPSK modulation, carrier frequency 1500 Hz, chip duration 2 ms, maximum-length spreading codes of 3 & 15 chips [4], corresponding data rates of 167 & 33 bit/s, signal duration ~ 8 seconds. In addition, the signals were preceded by a 0.5-s LFM pilot for detection and synchronization. Once every 40 seconds a signal was emitted, the data rate alternating between 167 and 33 bit/s between transmissions. Every now and then a few transmissions were skipped when the submarine required silence for its navigation.

On the receiver platform the hydrophone signals were digitized at a sampling rate of 5120 Hz and stored on tape. After the sea trials, the data were recovered and prepared for offline processing. Beamforming was performed with a single line array of 64 hydrophones,

that were given equal weights. Owing to the larger gain in SNR, this rectangular window yielded better results (i.e. lower bit error rates) than, for example, a Hanning window.

The receiver structure is described as follows. To find the start of a signal the recorded waveforms were matched-filtered with a replica of the LFM pilot. Coherent demodulation of the data segment was achieved with a squaring loop. That is to say, the time signals were squared and a narrow bandpass filter was applied to recover the (doubled) carrier. To allow for Doppler shifts, the centre frequency of the filter was placed at the peak of the power spectral density in a 20-Hz search window surrounding the nominal squared carrier at 3000 Hz. After recovery, the carrier wave was used to bring the signal to complex baseband. Subsequently, the baseband signal was downsampled to one sample per chip. For the chip timing use was made of the known ratio between the carrier frequency and the chip rate. (Clearly, a reception chain set up in this manner is critically dependent on proper carrier recovery.) To deal with the -occasionally severe- multipath propagation, the interchip interference was combatted with a linear adaptive equalizer based on the mean square error criterion [5] with up to 160 feedback taps. The first second (500 chips) of the incoming signal was used to train the equalizer. Finally, the symbols were obtained by correlating the equalized chips with the known maximum-length spreading sequence.

4. RESULTS AND DISCUSSION

To illustrate the favourable effects of beamforming and equalization on the chips, an example is shown in Fig. 2. Here, the chip and symbol constellations are plotted for a single-hydrophone reception and for the beamformed signal. At the hydrophone level the equalizer improves the chip constellation, but it is only after the spread-spectrum gain that the clouds are nicely separated with zero bit errors. For the same received signal, but beamformed, the output SNR is considerably improved. As a matter of fact, the constellation of equalized chips demonstrates that pure BPSK at 500 bit/s would have been an option in this case.

The bit error rate (BER) of all received signals is shown in Fig. 3 for a transmission range up to 20 km. After a difficult start at the very beginning of the run, the transmissions are virtually error free up to 12 km. At 167 bit/s this requires beamforming, but at 33 bit/s a single hydrophone suffices. At 12 km the communication link collapses with the BER rising to 50% for nearly all transmissions. Beamforming results in a considerable improvement, although bit errors are still a regular occurrence. At 33 bit/s there is a cluster of transmissions with a single bit error between 11 and 12 km, but real deterioration sets in at 16 km. Beyond 18 km error free communication is no longer possible with the current receiver architecture.

To interpret the quality of the communication link in relation to the acoustic channel, it is necessary to consider the two chief complicating factors, viz., ambient noise and multipath propagation. The SNR was determined by comparing a portion of noise preceding the signals with a section of the signals itself (which actually consists of signal plus noise). This procedure was performed before and after beamforming: Fig. 4. It is observed that the SNR is quite poor on the hydrophone level, dropping from +10 dB at the start of the experiment to below -10 dB at a 20-km distance. Beyond 16 km the signal level has fallen to a point where the single-hydrophone SNR can no longer be determined accurately; nonetheless the trend remains clear. By contrast, the SNR after beamforming averages 26 dB higher over the run. This is considerably more than the theoretical gain of 16 dB at 1500 Hz for omnidirectional noise. An explanation is found in the tow ship noise, which makes a significant contribution to the total noise. The tow ship noise is directional and can be cancelled more efficiently than omnidirectional noise, depending on the configuration.

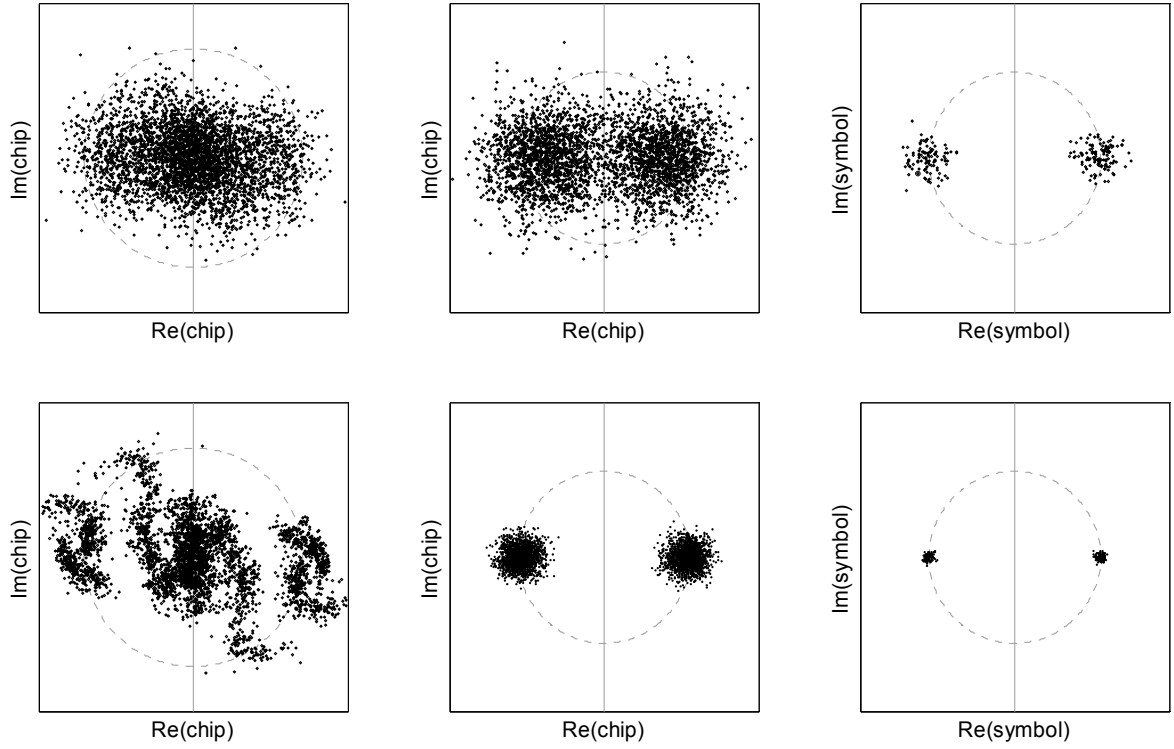


Fig.2: Chip constellation degraded by multipath and noise (left column); chip constellation after channel equalization (middle column); symbol constellation after the spread-spectrum gain (right column). Top row: single hydrophone. Bottom row: beamformed data.

Transmission range = 5.4 km; data rate = 33 bit/s.

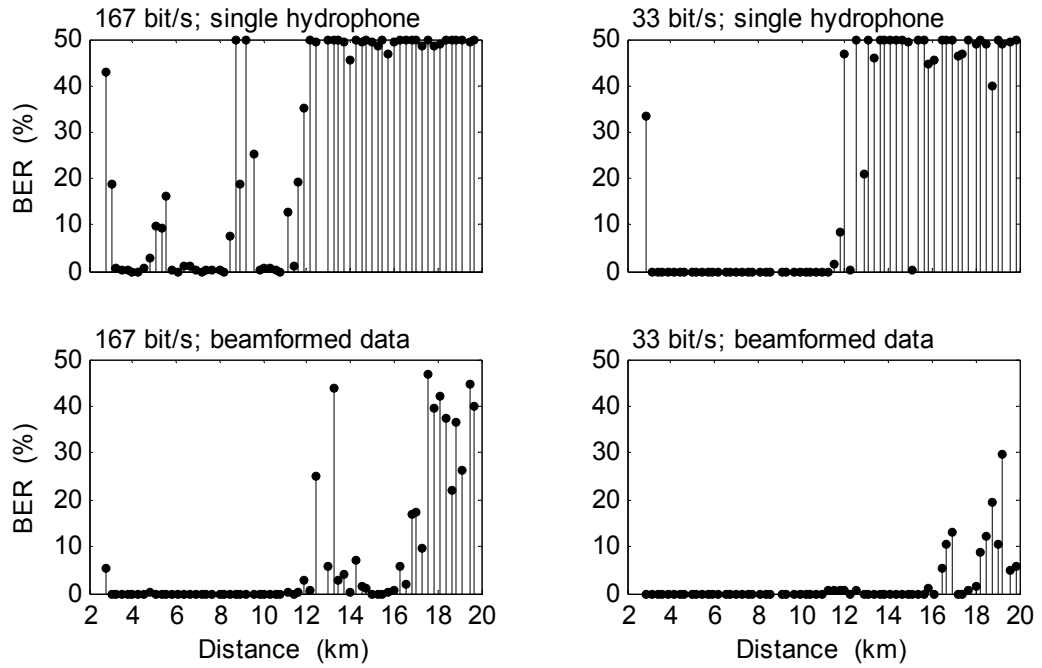


Fig.3: Bit error rate versus distance, for a single hydrophone and after beamforming.

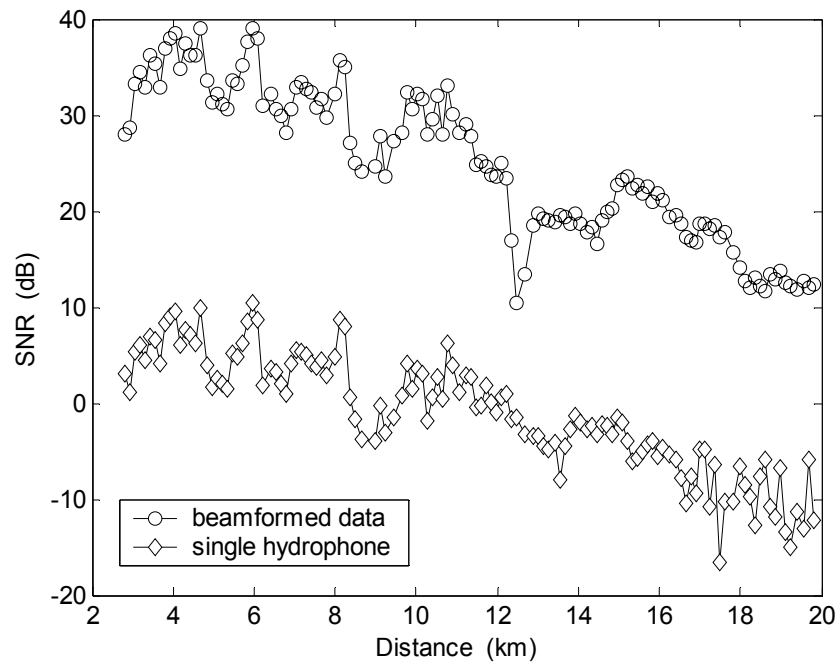


Fig.4: The signal-to-noise ratio versus distance.

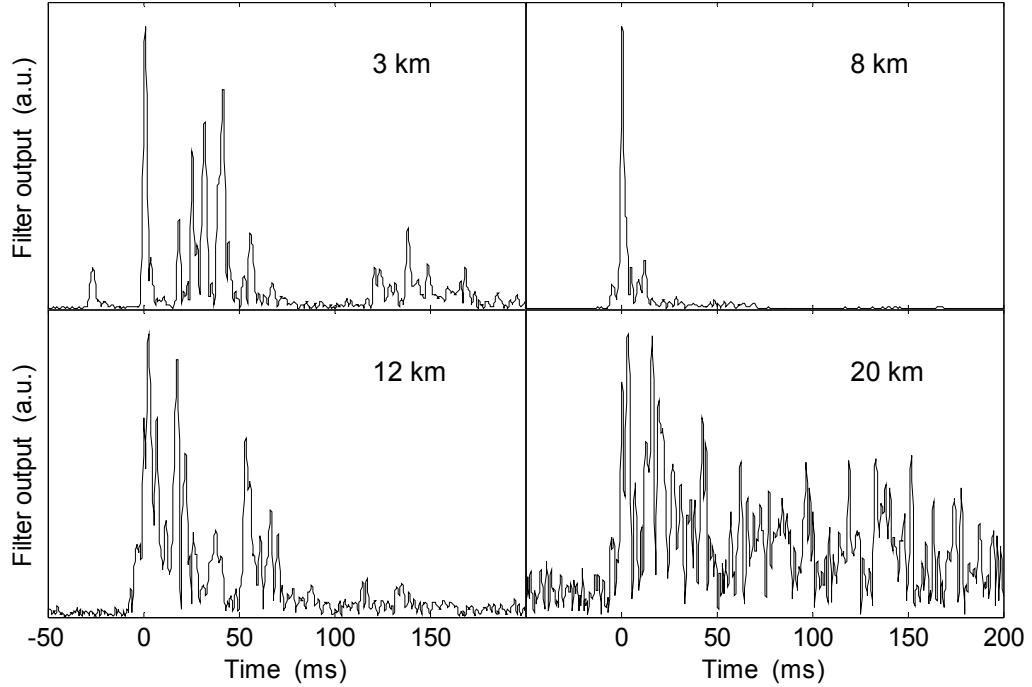


Fig.5: Four characteristic impulse responses throughout the run (after beamforming).

As to the channel impulse response, the multipath arrival pattern changes drastically throughout the run (Fig. 5). At the start the impulse response extends over 200 ms, gradually improving towards a more or less isolated dominant path at 8 km, after which the timespreading increases again until, at 20 km, the receiver faces reverberation stretching over hundreds of milliseconds.

The SNR in Fig. 4 and the channel impulse responses in Fig. 5 bring some understanding to the bit error rates of Fig. 3. At 3 km the impulse response is the main difficulty, although the combined effects of beamforming, chip equalization and the spread-spectrum gain already do a proper job at 33 bit/s. At a distance of 12 km there is an (unexplained) dip in the SNR of the beamformed signals, while at the same time the impulse response deteriorates. These two effects cause the bit errors around 12 km. As the distance increases further the single-hydrophone SNR becomes so small that synchronization with the LFM precursor frequently fails, which is the cause of a great deal of the ~50% error rates. Beyond 18 km the extremely poor impulse response and the low SNR terminate the communication link.

5. CONCLUDING REMARKS

This paper described a direct-sequence spread-spectrum transmission scheme for underwater acoustic communication with LFAS. It is shown that the communication link greatly benefits from beamforming with a horizontal hydrophone array and a linear adaptive, chip-level equalizer. Nonzero error rates could be related to time spreading and/or a low SNR. Future work on acoustic communication with LFAS systems will continue in the same direction, with a PSK instead of an LFM pilot and with a QPSK data modulation.

6. ACKNOWLEDGEMENTS

The author wishes to thank his TNO colleagues who improvised to carry out the communication experiment, as well as the submarine crew who endured the ‘horrendous’ noise. Arnaud Miller is acknowledged for preprocessing of the data.

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