FFI-RAPPORT

16/01378

The Watermark manual and user's guide

version 1.0

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Paul van Walree
Roald Otnes
Trond Jenserud

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Roald Otnes
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Norwegian Defence Research Establishment (FFI) 29 November 2016

Keywords

Akustisk telekommunikasjon
Referansemåling
Modem
Kanalmodeller
Simulering

FFI-rapport
FFI-RAPPORT 16/01378

Project number
1402

ISBN
P: ISBN 978-82-464-2886-4
E: ISBN 978-82-464-2887-1

Approved by
Connie Elise Solberg, *Research Manager*
Elling Tveit, *Director*

Summary

WATERMARK is a benchmark for physical-layer schemes for underwater acoustic communications. It allows researchers and modem manufacturers to develop, test and compare algorithms for the physical layer under highly realistic and reproducible conditions. WATERMARK is a shell around the validated FFI channel simulator MIME, which is driven by at-sea measurements of the time-varying impulse response. WATERMARK is programmed in MATLAB and can be used under the Windows and Linux operating systems.

This report serves as a user manual for WATERMARK version 1.0. It describes the available test channels, the format of the channel data, and the format of user-defined input signals. It shows how to pass these signals through the test channels, and how to retrieve the received signals. Examples of WATERMARK usage are provided.

The simulation possibilities are governed by the available test channels. The initial release of WATERMARK is issued with channels measured in Norway (two sites), France, and Hawaii, offering three frequency bands (4–8, 10–18, and 32.5–37.5 kHz), single-hydrophone and array receivers, and run times varying from 33 seconds to 33 minutes. The benchmark can be extended with channels from different environments and frequency bands, either for own use or for general distribution, depending on the willingness of third parties to perform suitable measurements and share the data.

Sammendrag

WATERMARK er en benchmark for fysiske lag innenfor undervannskommunikasjon. Denne benchmarken tillater forskere og modemprodusenter å utvikle, teste og sammenligne algoritmer for det fysiske lag under realistiske og reproduserbare forhold. WATERMARK er et skall rundt den validerte FFI kanalsimulatoren MIME, som drives av målinger av undervannskanalens tidsvarierende impulsrespons. WATERMARK er programmert i MATLAB og kan brukes under operativsystemene Windows og Linux.

Denne rapporten tjener som brukermanual for WATERMARK versjon 1.0. Den beskriver de tilgjengelige testkanalene, deres format, samt formatet til brukerdefinerte inngangssignaler. Det eksemplifiseres hvordan disse signalene blir sendt gjennom testkanalene og hvordan man får tak i de mottatte signalene. Eksempler på bruk av WATERMARK er inkludert.

Simuleringsmulighetene begrenses av de tilgjengelige testkanalene. Den første utgaven av WATERMARK er utstyrt med kanaler målt i Norge (to steder), Frankrike og Hawaii, og tilbyr tre frekvensbånd (4–8, 10–18 og 32,5–37,5 kHz), enkelthydrofon- og antennemottakere, og run-tider som varierer fra 33 sekunder til 33 minutter. WATERMARK kan enkelt utvides med kanaler fra andre omgivelser og frekvensbånd, enten for eget bruk eller for generell distribusjon, avhengig av tredje parts vilje til å gjøre høvelige målinger og dele data.

Contents

1 Introduction

Numerous modulation schemes have been proposed for underwater acoustic communications during the past decades. In the majority of cases it is impossible to compare the performances of these schemes in a meaningful way, because modem manufacturers and researchers normally perform their field tests in different environments. The variation in channel characteristics is large even within a given environment [1, 2]. Proposed channel models and channel simulators have widely varying degrees of realism, completeness and availability. Unlike terrestrial radio-frequency communications, underwater acoustic communications is a technological field deprived of standard test channels. Nobody "knows" what is the best scheme, and in all likelihood the picking order depends on the type of channel and the chosen performance metrics.

The underWater AcousTic channEl Replay benchMARK (Watermark) is a realistic simulation tool which is now made available to the underwater communications community. It is a shell around the validated FFI channel simulator "Mime" [3, 4], and comes with five test channels to get started. The primary objective of the present initiative is to provide an *easy-to-use* benchmark that enables development, testing, and comparison of physical-layer algorithms under *realistic* and *reproducible* conditions.

Watermark was presented at UCOMMS 2016 [5] with two test channels for a single receiver. Following the presentation at UCOMMS, Telecom Bretagne generously offered array data [6] for release with the benchmark, and, soon after, Scripps Institution of Oceanography approved a request to include array data from the Kauai Acomms MURI 2011 (KAM11) experiment [7, 8]. The Watermark channels thus come in two varieties, single-input single-output (SISO) and single-input multiple-output (SIMO). Both types of channels are represented by a collection of so-called channel files. A channel file is a MATLAB (.mat) file with a continuous single-hydrophone time-varying impulse response (TVIR) estimate. The collection represents consecutive soundings (TVIR measurements) in the SISO case, and a single sounding on consecutive hydrophones in the SIMO case. The SISO channels offer a long but discontinuous play time. SIMO channels are continuous but with a relatively short play time. SISO channels are of practical interest since most acoustic modems use a single receiver. SIMO channels are more of academic interest, but can also be used for SISO testing by considering a single hydrophone.

1.1 The Mime channel simulator

Mime distorts input waveforms by convolving them with measured channels. Its operation principle is known as channel replay

$$y(t) = \int_{-\infty}^{\infty} \hat{h}(t,\tau)\, x(t-\tau)\, \mathrm{d}\tau + n(t)\,, \tag{1.1}$$

where $x(t)$ is the input signal, $\hat{h}(t,\tau)$ a TVIR estimate, $n(t)$ a noise term, and $y(t)$ the distorted output signal. Channel replay can be performed in several ways [3, 4, 9, 10, 11], and recently the term has also been used for network simulations with measured packet error statistics [12].

Mime has a direct-replay mode, where the quantity $\hat{h}(t,\tau)$ is an estimate of the true channel $h(t,\tau)$, with a maximum simulation time that equals the measurement time. Mime also has a stochastic replay mode, which generates TVIR realizations that are statistically consistent with the measured channel. This mode can run forever on a single input measurement. The realism of direct-replay simulations depends primarily on the quality of the channel measurement. The measurement errors have been well documented in the case of correlative channel sounders [13], and the combination of Mime and correlative sounders has been well validated [3, 4]. Simulated bit and packet error ratios, and receiver output SNR, are close to the corresponding values measured at sea. The stochastic replay mode has also yielded realistic simulation, but it has some limitations regarding the reproduction of non-stationary channels, time-varying delays, and correlated scattering. To guarantee realistic simulations under all conditions, Watermark is released only with the direct-replay mode of Mime. This mode faithfully reproduces all effects mentioned in [2], including channel non-stationarity, correlated scattering, time-varying delays, and ultra-wideband channels. The one type of channel that cannot be reproduced is the overspread channel, because it cannot be measured properly.

Watermark channels include acoustic propagation effects and hardware effects. Users can transmit arbitrary waveforms through the available test channels, which will be distorted in precisely the same way as they would have been distorted had they been transmitted at sea under the conditions of the sounding campaign. This distortion includes the transfer functions of source and receiver, and possible clock frequency offsets. This hardware distortion is included in the channel estimates obtained with the correlative sounder, and is naturally reproduced by direct replay.

2 Test channels

The initial release of Watermark comes with sounding data for five channels. Table 2.1 summarizes measurement conditions and test channel parameters. The transmitter is bottom-mounted in NOF1 and NCS1, suspended in the water column in BCH1, and towed by a surface ship in KAU1 and KAU2. Receivers are either bottom-mounted hydrophones or vertically suspended hydrophone arrays.

Table 2.1 *Test channel parameters and sounding conditions.*

Name	NOF1	NCS1	BCH1	KAU1	KAU2
Environment	Fjord	Shelf	Harbour	Shelf	Shelf
Time of year	June	June	May	July	July
Range	750 m	540 m	800 m	1080 m	3160 m
Water depth	10 m	80 m	20 m	100 m	100 m
Transmitter depl.	Bottom	Bottom	Suspended	Towed	Towed
Receiver depl.	Bottom	Bottom	Suspended	Suspended	Suspended
Probe signal type	LFM train	Pseudonoise	Pseudonoise	LFM train	LFM train
Frequency band	10–18 kHz	10–18 kHz	32.5–37.5 kHz	4–8 kHz	4–8 kHz
Roll-off fact.	1/8	1/8	1/10	1/8	1/8
Sounding duration	32.9 s	32.6 s	59.4 s	32.9 s	32.9 s
Delay coverage	128 ms	32 ms	102 ms	128 ms	128 ms
Doppler coverage	7.8 Hz	31.4 Hz	9.8 Hz	7.8 Hz	7.8 Hz
Type	SISO	SISO	SIMO	SIMO	SIMO
# hydrophones	1	1	4	16	16
Element spacing	—	—	1 m	3.75 m	3.75 m
# cycles	60	60	1	1	1
Cycle time	400 s	600 s	—	—	—
Total play time	33 min	33 min	1 min	33 s	33 s

The frequency band and sounding duration are properties of the employed channel probe signal, but they also give the operating band and the maximum signal duration for Watermark simulations, as explained in Sec. 3.1. The given frequency band is the −3 dB bandwidth of the probe signal. However, the probe signals have a (root) raised-cosine spectrum with a small roll-off factor [14, pp. 607–608], which is a near-rectangular spectrum that is flat over a large part of the given band. The importance of a flat probe signal spectrum is discussed in Sec. 3.1.

The delay coverage is the tracking period of the channel sounder, and the Doppler coverage is its reciprocal. These numbers give the maximum possible delay spread and Doppler spread in the replay channel, respectively. Energy in the delay-Doppler spread function of the true channel that is outside these boundaries, is aliased in the channel estimate [13, 15]. Some degree of aliasing is often unavoidable in acoustic channels, but the aliased energy may be a small fraction of the total energy. All channels selected for Watermark are well behaved and offer realistic replay.

Watermark channels are not necessarily stationary. There are many reasons for channel non-stationarity, such as platform motion, changing weather conditions, movement of the tides, etc. For a fair and reproducible comparison between modulation schemes, performance results reported for WATERMARK channels should be averaged over the full play time of the replay channel.

The test channels are described in more detail in the following sections. See [2, 15] for definitions of the quantities plotted in the multipanel figures, and note that the mean Doppler shift has been removed in this analysis. Figures are shown for a single, representative sounding / hydrophone, but for completeness WATERMARK is distributed with the corresponding figures for all soundings / hydrophones.

2.1 Norway – Oslofjord (NOF1)

NOF1 is a channel measured in a shallow stretch of Oslofjorden between a stationary source and a stationary single-hydrophone receiver. The data consist of a 32.9-s channel estimate, repeated 60 times at 400-s intervals, yielding a total simulation time of ≈ 33 minutes. Note that the play time is discontinuous. For instance, a 5-s communication packet fits 6 times in a single channel estimate (see Sec. 5.2) and yields $60 \times 6 = 360$ independent packets in simulation. The first 6 packets are transmitted between t_0 and t_0 + 32.9 s, packets 7–12 between t_0 + 400 s and t_0 + 432.9 s, etc.

Figure 2.1 characterizes the channel. Following a rapid onset, the delay profile decays quickly. A cluster of discrete arrivals is observed between 50 and 80 ms, but at −30 dB these paths carry a small fraction of the total energy and can be tolerated as interference with most communication systems. Figure 2.2 zooms in on the start of the impulse response and reveals a few stable paths, as well as more clutter-like arrivals due to surface interactions. The stable paths are responsible for the sharp peak in the Doppler spectrum in Fig. 2.1, which also reveals weak sidelobes reflecting the spectrum of the surface gravity waves. The autocorrelation function has a rapid initial decay, remaining at a value of 0.8 afterwards. Stable paths carry 80% of the received signal energy, and on the whole NOF1 is a benign channel.

NOF1 remarks:

- Use serial fetch (`sfetch.m`, see Sec. 5.3) to retrieve packets.
- The channel is not stationary over its entire replay duration.
- The data are not calibrated with respect to propagation loss, but variations in propagation loss over time are reproduced (i.e., the signal level may vary between packets).

2.2 Norway — Continental Shelf (NCS1)

NCS1 was measured on Norway’s continental shelf between a stationary source and a stationary single-hydrophone receiver. The data consist of a 32.6-s channel estimate, repeated 60 times at 600-s intervals, yielding a total simulation time of ≈ 33 minutes. The play time is discontinuous. For instance, a 5-s communication packet fits 6 times in a single channel estimate and yields $60 \times 6 = 360$ independent packets in simulation. The first 6 packets are transmitted between t_0 and t_0 + 32.6 s, packets 7–12 between t_0 + 600 s and t_0 + 632.6 s, etc.

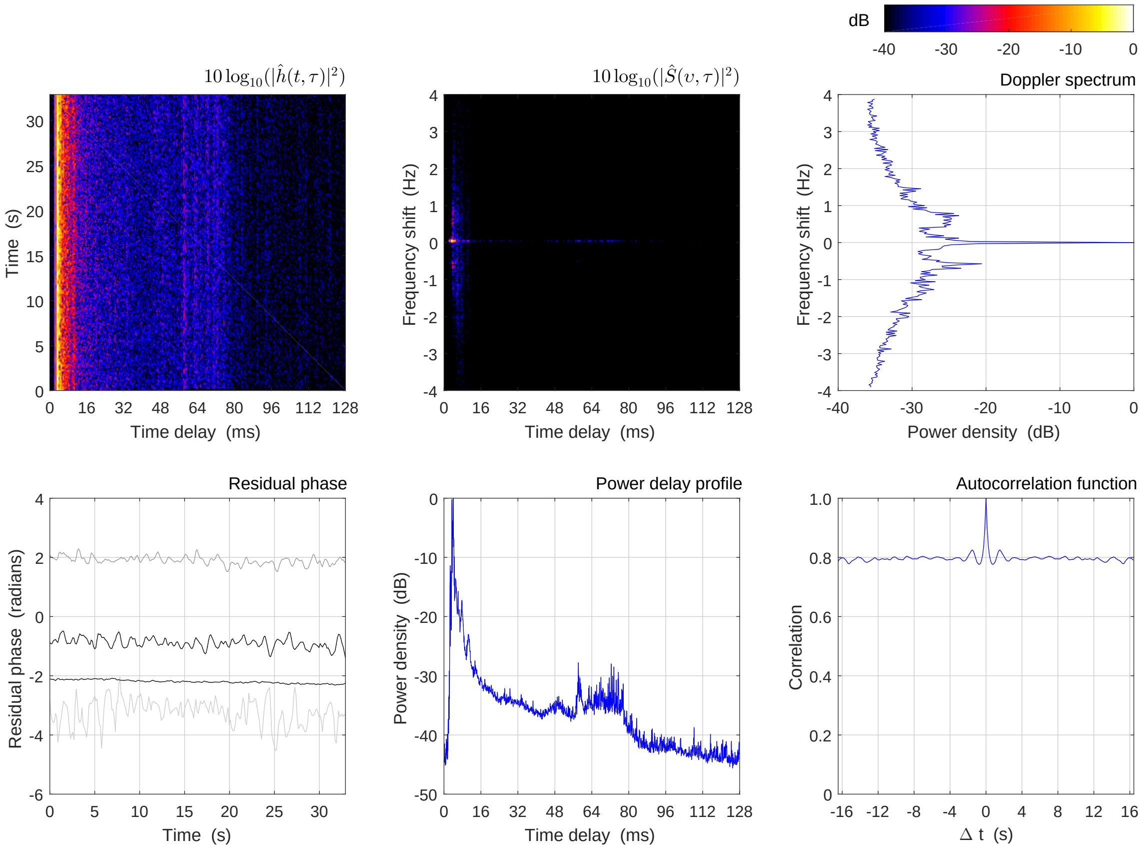

Figure 2.1 *Channel analysis of a characteristic NOF1 sounding. The quantities $h(t,\tau)$, $S(\upsilon,\tau)$, the Doppler spectrum, and the power delay profile are given in dB relative to their peak value. Zero delay is arbitrarily placed near the start of the impulse response.*

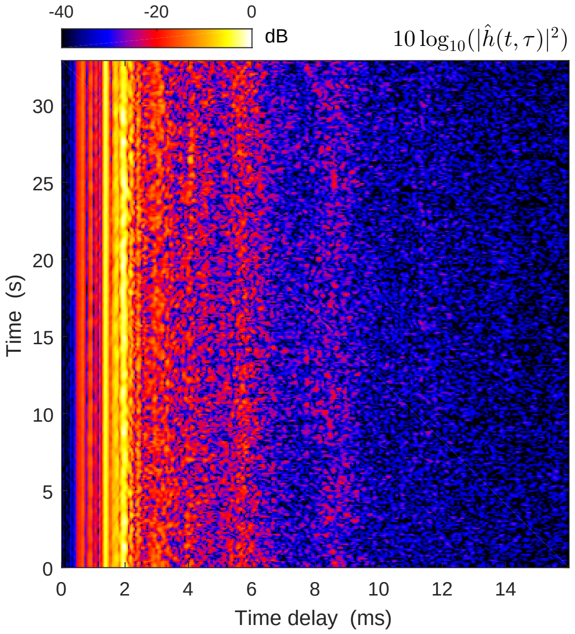

Figure 2.2 *The top left panel of Fig. 2.1, zoomed in on the first 16 ms.*

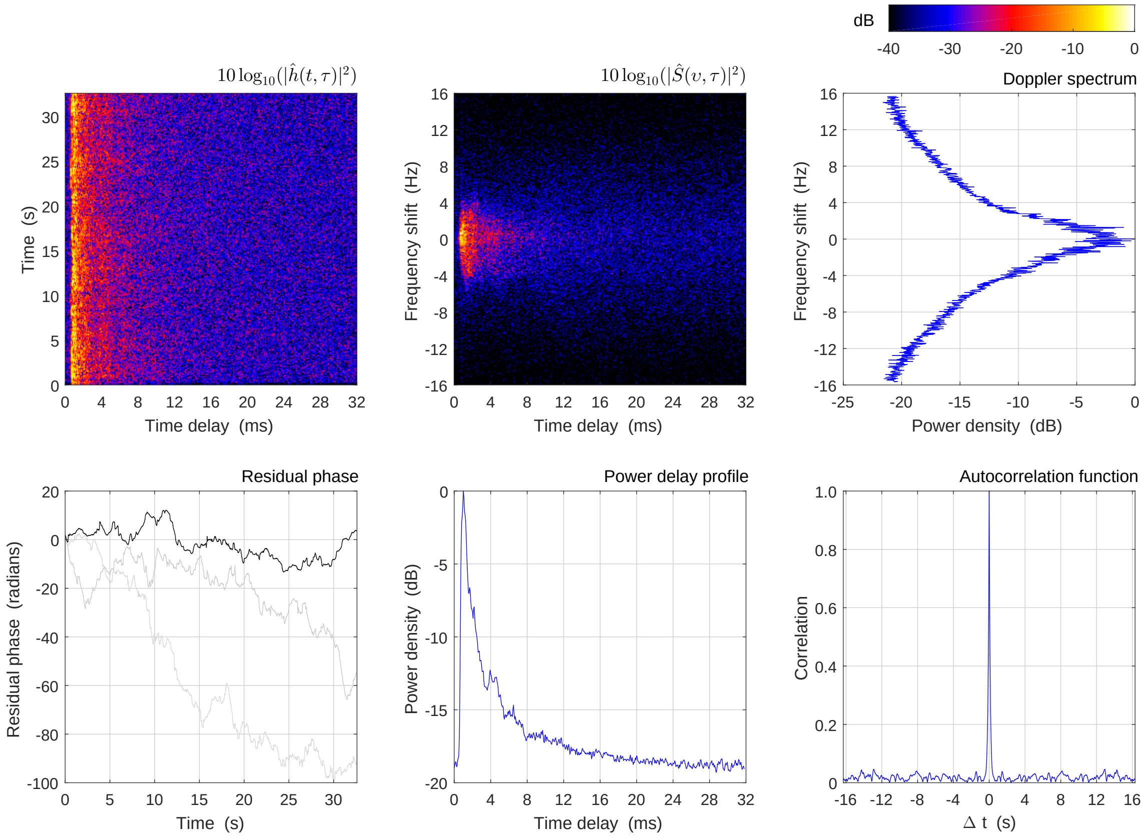

Figure 2.3 *Channel analysis of a characteristic NCS1 sounding.*

Figure 2.3 characterizes the channel, which has in common with NOF1 that most energy is concentrated at the start of the impulse response. In other regards it is a completely different channel. There are no specular paths worth mentioning, yielding a wide Doppler spectrum and a correlation function that rapidly drops to zero. The coherence (half) time is of order 100 ms. NCS1 is a challenging channel for coherent communication schemes, despite the short impulse response.

Challenging channels are difficult to measure, and from all test channels in Table 2.1, NCS1 is the one with the largest measurement errors. The delay profile is “only” at -18 dB at its boundaries, and the Doppler spectrum at -20 dB. The true channel has long, diffuse tails in both profile and spectrum, which are aliased in the measurement. However, the fraction of the total energy that is aliased is not necessarily large. A validation has been performed to assess the impact of aliasing and other measurement errors for NCS1.

Figure 2.4 shows a validation result for a few tested communication schemes, using the validation methodology of [4]. These schemes are JANUS[1], direct-sequence spread spectrum (DSSS), and 8-ary phase-shift keying (8PSK). JANUS uses frequency-shift keying and energy detection, DSSS uses a chip equalizer and phase detection, and 8PSK is a high-rate scheme with turbo equalization. The JANUS and DSSS implementations are described in [17] and 8PSK is a scheme from

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[1] This is not the JANUS standard [16], but an early development version.

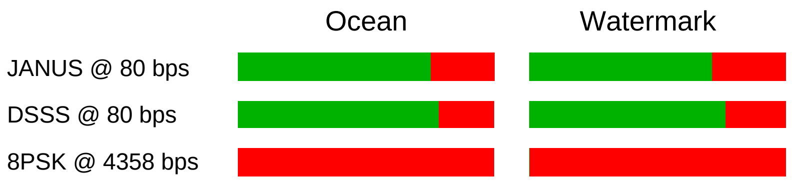

Figure 2.4 *WATERMARK validation for NCS1. Green is the fraction of packets received without bit errors, red the fraction with one or more errors.*

[18] adapted to the 10–18 kHz band.

The validation involves 300 probe signals and 300 communication packets, which were transmitted alternatingly in the NCS1 sounding set-up. Recorded communication packets were demodulated, and 300 additional packets were "WATERMARKED" with channel estimates extracted from the recorded probe signals. Measured and simulated packet error ratios are close for JANUS and DSSS at 80 bits/s, which coincidentally also have a similar performance in this channel. Both the oceanic channel and the simulated channel are too difficult for 8PSK.

NCS1 remarks:

- Use serial fetch (`sfetch.m`, see Sec. 5.3) to retrieve packets.
- The channel is not stationary over its entire replay duration.
- The data are not calibrated with respect to propagation loss, but variations in propagation loss over time are reproduced (i.e., the signal level may vary between packets).

2.3 Brest Commercial Harbour (BCH1)

BCH is a SIMO channel from the commercial harbour of Brest, France. A source and a 4-element array were lowered into the water column from two docks [6]. A single probe transmission over a range of 800 m resulted in a 59.4 s channel estimate, simultaneously recorded on the four hydrophones. Figure 2.5 characterizes the channel for one of the receivers. Similarly to NOF1, the channel is a mixture of stable and fluctuating arrivals, but with a larger number of distinct trailing paths.

Time synchronization between hydrophones is preserved in the channel files, and reproduced in direct replay. This means, for example, that beamforming algorithms yield the same results as beamforming applied to data recorded at sea.

BCH1 remarks:

- Use parallel fetch (`pfetch.m`, see Sec. 5.3) to retrieve packets for array processing.
- Data channel 1 is the *top* hydrophone, located 3 m below the surface. The transmitter depth is 2 m.
- The data are not calibrated with respect to propagation loss, but variations in propagation loss over time are reproduced (i.e., the signal level may vary between packets).

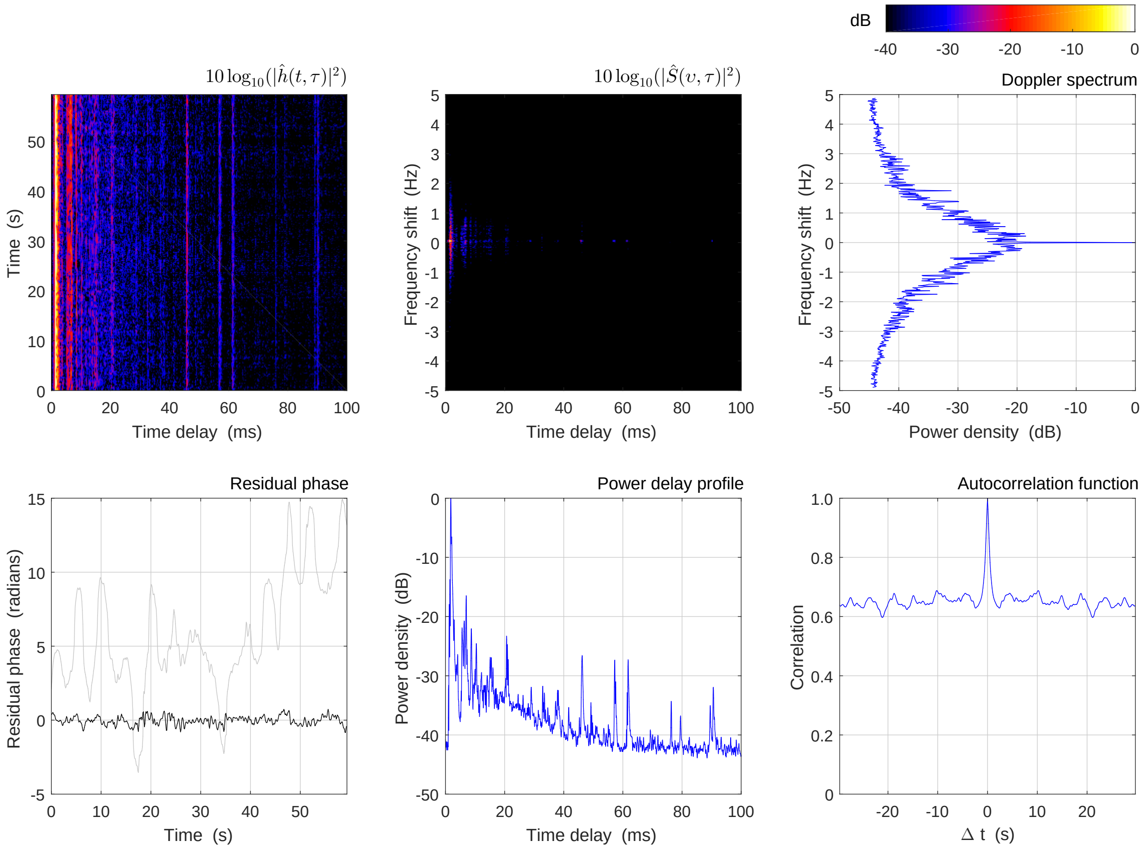

Figure 2.5 *Channel analysis of a characteristic BCH1 sounding.*

- Propagation loss differences between hydrophone channels are reproduced.
- Time synchronization between hydrophone channels is preserved.

2.4 Kauai 1 (KAU1)

KAU1 is a SIMO channel from the Kauai Acomms MURI 2011 (KAM11) experiment. This experiment was conducted in shallow water off the western side of Kauai, HI, USA, over the period June 23–July 12, 2011 [7, 8]. KAU1 was recorded in the band of 4–8 kHz, between a towed source and a vertically suspended array with 16 receivers. The run time is 33 s.

Figure 2.6 characterizes the channel for a hydrophone near the center of the array. The first arrival (direct path) is a sharp peak in the delay profile. Later arrivals become weaker, broader, and are characterized by an increasing Doppler spread. This is caused by an increasing number of surface bounces at increasing grazing angles. In addition to Doppler spreading due to variability of the medium, there is a time-varying Doppler (TVD) shift due to tow ship motion. This also affects the direct path and explains why there is no specular peak in the Doppler spectrum.

TVD can be removed to a large extent by resampling the recorded probe signal with a time-varying resampling factor [3, 15]. Figure 2.7 shows the Doppler spectrum after TVD removal,

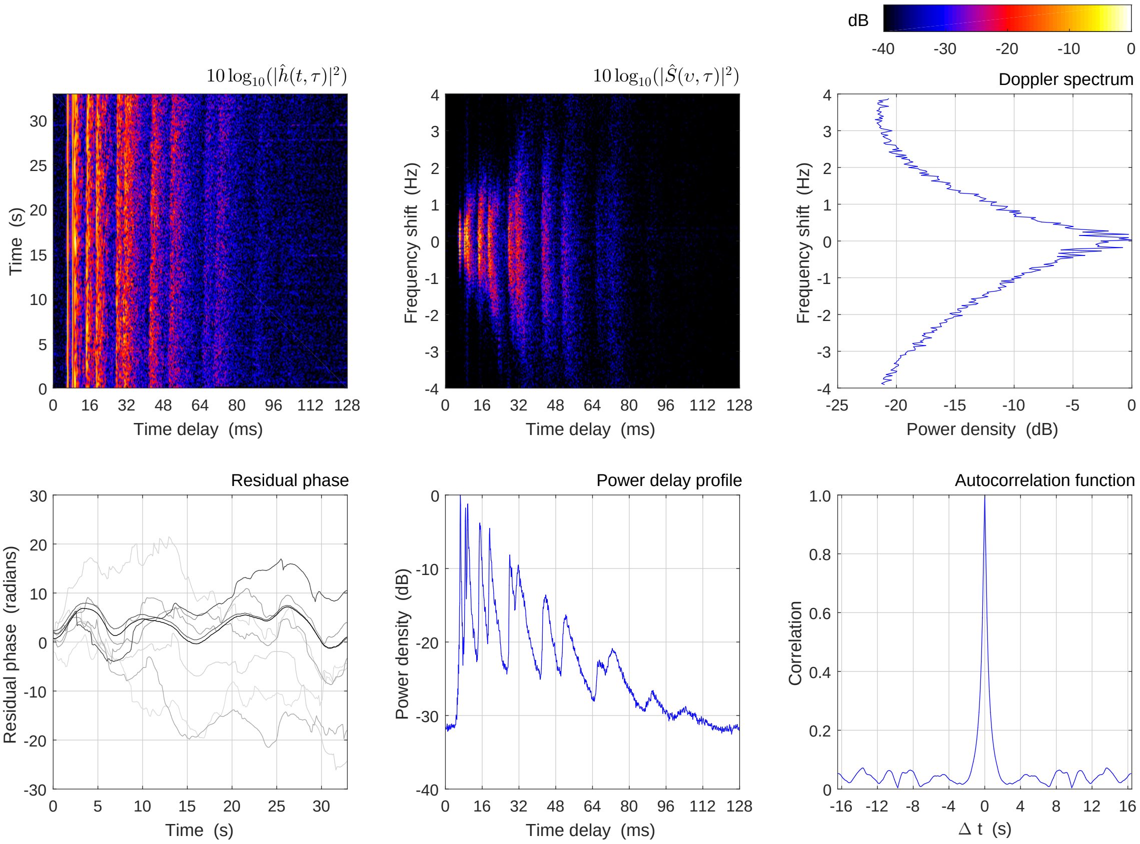

Figure 2.6 *KAU1 channel analysis (hydrophone channel 8).*

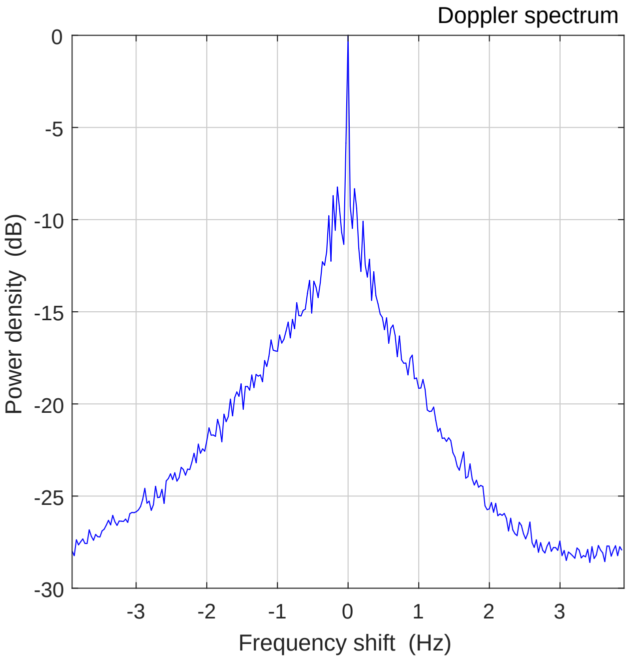

Figure 2.7 *KAU1 Doppler spectrum after TVD removal by resampling (hydrophone channel 8).*

which has a specular peak. Coherent communication systems operating in this channel may benefit from an adaptive resampling routine, or a phase-locked loop [19], to deal with the TVD. Note that TVD removal is not included in WATERMARK. It is applied separately here to illustrate a channel property.

KAU1 remarks:

- Use parallel fetch (`pfetch.m`, see Sec. 5.3) to retrieve packets for array processing.
- Data channel 1 is the *bottom* hydrophone, located 9 m above the seafloor at a water depth of about 106 m. The source depth is 45–50 m.
- Hydrophone channel 2 was broken. Nonetheless the channel estimation has been performed, resulting in a weak (nonsense) signal in this data channel.
- The data are not calibrated with respect to propagation loss, but variations in propagation loss over time are reproduced (i.e., the signal level may vary between packets).
- Propagation loss differences between hydrophone channels are reproduced.
- Time synchronization between hydrophone channels is preserved.

2.5 Kauai 2 (KAU2)

KAU2 is another SIMO channel from the KAM11 experiment. It concerns the same signal transmission as KAU1, but recorded on a different array. The main difference with KAU1 is a larger range, which attenuates delayed arrival clusters more rapidly: Figure 2.8. TVD contributes significantly to the total Doppler spread in this channel, similarly to KAU1.

KAU2 remarks:

- Use parallel fetch (`pfetch.m`) to retrieve packets for array processing.
- Data channel 1 is the *bottom* hydrophone, located 7 m above the seafloor at a water depth of about 100 m. The source depth is about 45–50 m.
- The data are not calibrated with respect to propagation loss, but variations in propagation loss over time are reproduced (i.e., the signal level may vary between packets).
- Propagation loss differences between hydrophone channels are reproduced.
- Time synchronization between hydrophone channels is preserved.

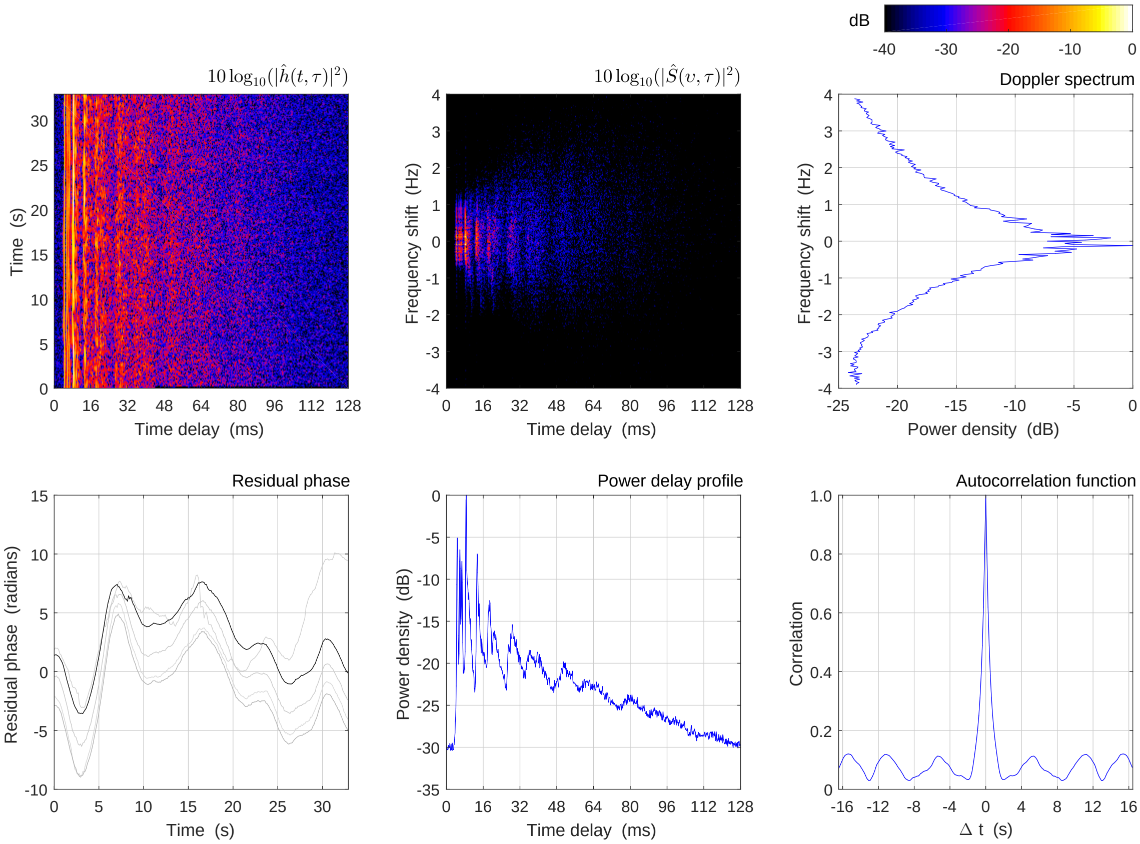

__Figure 2.8__ KAU2 channel analysis (hydrophone channel 8).

3 Technical remarks

3.1 Transfer function

Input signals have to comply with the frequency bands listed in Table 2.1. WATERMARK does not use explicit filters, but the channel files contain an implicit bandpass filter. Correlative sounders yield ultra-wideband channel estimates $\hat{h}(t, \tau)$ [20], where the transfer functions of the channel, the sounder hardware, and the probe signal are superimposed on one another. The composite transfer function is reproduced in direct replay. It is for this reason that the WATERMARK probe signals use a root raised-cosine spectrum with a small roll-off factor (Table 2.1). The probe signal spectrum of NOF1 and NCS1, for instance, is flat between 10.5 and 17.5 kHz. This implies that, in this band, the spectrum of signals passed through the replay channels is not affected by the probe signal. The resulting distortion is entirely due to the acoustic channel (frequency-dependent propagation loss) and the employed hardware.

The probe signal spectrum, and thus the composite transfer function, falls off rapidly outside the nominal band (10–18 kHz in the case of NOF1 and NCS1). It makes no sense to apply WATERMARK to signals with a nominal bandwidth exceeding that of the probe signal, because there is no response beyond the spectrum of the probe signal. Or more precisely, there may be a response due to sidelobes of the channel estimate, but this is not a meaningful response. Communication schemes using a different band than the nominal band of a test channel should be adapted, because WATERMARK will not adapt to these schemes.

The frequency response is revealed by passing white noise through a test channel. Figure 3.1 illustrates this for a NCS1 sounding. The top panel shows a spectrogram of the received probe signal, and the middle panel that of the input signal $x(t)$ (see Eq. 1.1).[2] A spectrogram of the output signal $y(t)$ is shown in the bottom panel. It strongly resembles that of $x(t)$, as it should, and shows that simulation is only possible where there is signal energy in the probe signal spectrogram. This makes complete sense, as it is impossible to reproduce something which has not been measured. The main difference between the spectrograms is the weak reverberation tail, seen at the end of the probe signal, which is not reproduced in simulation. This energy is aliased in the channel estimate.[3]

3.2 Doppler effects

The mean Doppler shift V_0 is removed from the sounding data prior to the channel estimation. This is done by resampling the raw acoustic data by a resampling factor $1 - V_0/c$, where $c = 1500$ m/s is the nominal sound speed. The used sign convention is such that a positive velocity corresponds to a positive range rate, i.e. time dilation of the signal. This Doppler shift is thus missing from the

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[2] Note that the white noise is the input signal $x(t)$ in this case, not the additive term $n(t)$.

[3] Actually WATERMARK does not allow input signals whose length exceed that of the probe signal; the parts of $y(t)$ before $t = 4$ s and after $t = 37$ s have not been simulated.

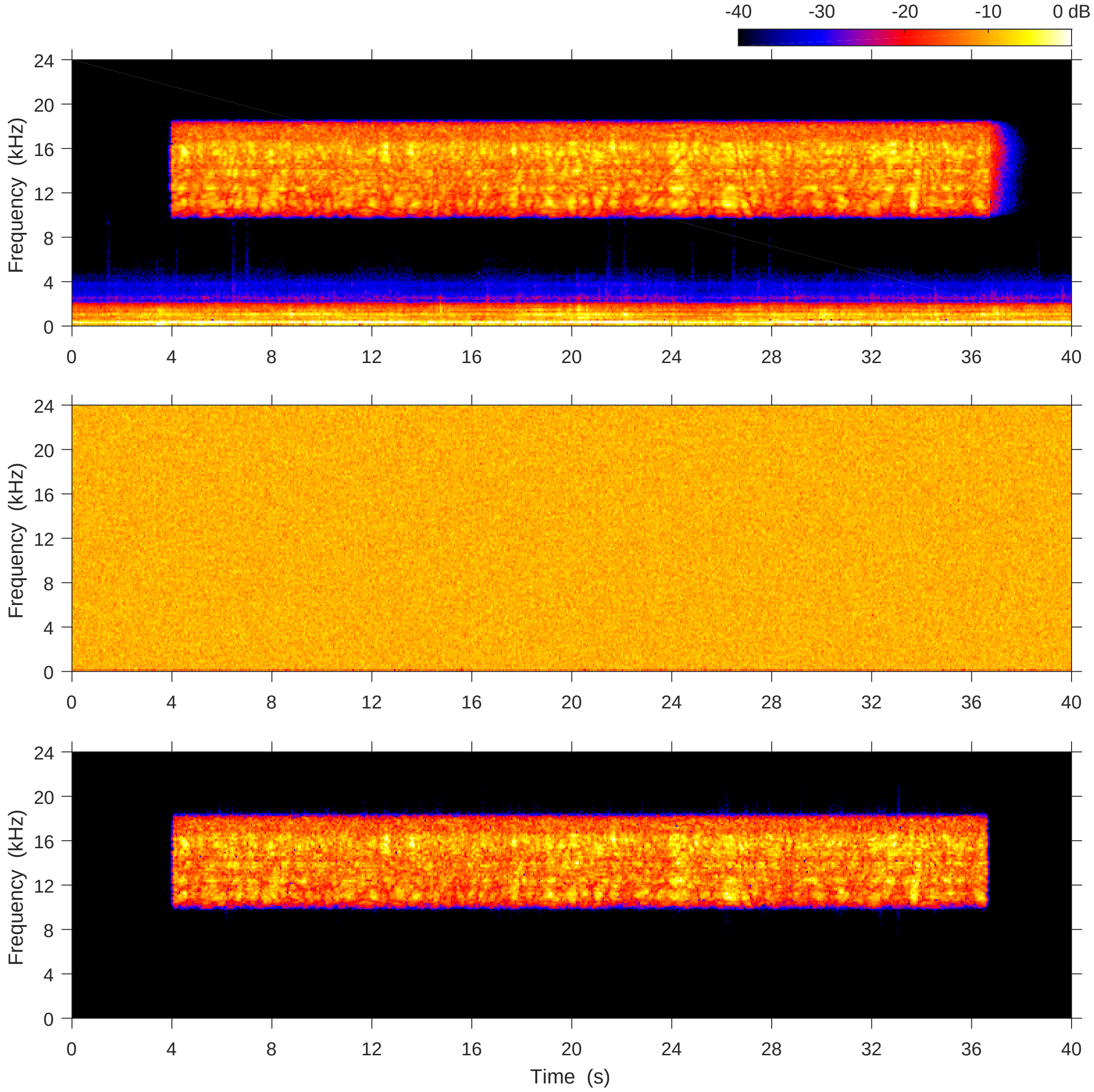

Figure 3.1 *Spectrograms of a recorded probe signal (top), white Gaussian noise as the input $x(t)$ to* Watermark *(middle), and the output signal $y(t)$ (bottom).*

estimates $\hat{h}(t,\tau)$ in the channel archives,[4] but instantaneous Doppler spreading and time-varying Doppler shifts around the mean value are preserved.

The reason for removing the mean Doppler shift is that this minimizes the measurement errors of the channel sounder. Instantaneous Doppler spreading and time-varying Doppler shifts around the mean value are reproduced in direct replay, and the mean shift is reinstalled when the functions `sfetch.m` and `pfetch.m` are used to retrieve packets. Retrieved packets are resampled by a factor $1/(1 - V_0/c)$. This ensures that all Doppler effects experienced by the channel probe signal in the real channel are transferred onto the input signal $x(t)$ in the simulated channel.

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[4] Note that the mean Doppler shift is also omitted from the channel analysis figures in Section 2.

WATERMARK does not discriminate between true Doppler shifts due to platform motion, and apparent Doppler shifts due to clock frequency offsets (CFOs). Both effects affect the duration of received acoustic signals. A CFO is a mismatch between the nominal (assumed) and the true sample rate of a device. There can be an offset in both sender and receiver, resulting in an apparent Doppler shift V_0 whose magnitude depends on the quality of the clocks. Values of several cm/s are not uncommon.

WATERMARK reproduces the at-sea conditions, including hardware effects, and hence simulated packets can be subject to a small Doppler shift even between bottom-mounted senders and receivers. Communication systems need to deal with such shifts in simulation, just like they need to deal with such shifts in at-sea signalling.

4 File structures

4.1 Channel files

The test channels are stored in `\Watermark\input\channels\`. For instance, NOF1 is stored in `\Watermark\input\channels\NOF1\`. `\Watermark\input\channels\NOF1\mat\` contains the channel estimates and `\Watermark\input\channels\NOF1\png\` the corresponding analysis figures. The channel estimate files are named `NOF1_001.mat`, `NOF1_002.mat`, ... `NOF1_060.mat`, corresponding to the 60 probe signal transmissions (cf. Table 2.1). In the case of array data, the file numbers give the hydrophone number. For instance, `KAU1_001.mat`, `KAU1_002.mat`, ..., `KAU1_016.mat` contain the channel estimates for 16 hydrophones.

The contents of the mat files are illustrated with `NOF1_001.mat`:

```
>> load('\Watermark\input\channels\NOF1\mat\NOF1_001.mat')
>> whos
  Name          Size                Bytes  Class     Attributes

  V0            1x1                     8  double
  fc            1x1                     8  double
  fs_t          1x1                     8  double
  fs_tau        1x1                     8  double
  h           258x2048            8454144  double    complex
  meta          1x1                  2018  struct
```

where

- V_0 is the mean Doppler shift (in m/s) discussed in Sec. 3.2;
- f_c is the probe signal center frequency (Hz);
- $f_{s,t}$ is the sampling rate (in Hz) of $\hat{h}(t,\tau)$ in time;
- $f_{s,\tau}$ is the sampling rate (in Hz) of $\hat{h}(t,\tau)$ in delay;
- $h = \hat{h}(t,\tau)$;
- `meta` is a struct with additional information. This struct may contain whatever the creator of the mat file finds useful. It is not used by Watermark.

WATERMARK starts its processing with the first file (`_001`) of a given test channel and assumes that the other files use identical parameters. One should not mix channel estimates obtained with different probe signals or different processing methods in the archive of a given channel.

4.2 Input signals

Input signals to be used in simulation are stored in `\Watermark\input\signals\` as `.mat` files. The name of each `.mat` file should be unique and descriptive, as it will be used as a signal identifier string throughout the benchmarking. For instance, if simulations are carried out with DSSS signals at four different data rates, they could be named `dsss1.mat`, `dsss2.mat`, `dsss3.mat`, `dsss4.mat`. If these signals are tested in multiple frequency bands, the names could be `dsss1_band1.mat`, `dsss1_band2.mat`, ..., etc.

The input signal file contains three obligatory variables, illustrated by the example workspace

```
Name           Size            Bytes  Class     Attributes

fs_x           1x1                 8  double
nBits          1x1                 8  double
x          68366x1            546928  double
```

where

- $f_{s,x}$ is the sampling rate (in Hz) of the signal;
- `nBits` is the number of information bits;
- $x = x(t)$ is the signal in passband.

Avoid needless samples before the start or after the end of the signal, such as trailing zeroes, because such samples lower the effective bit rate and the number of packets in simulation. It also pays to not use excessively high sampling rates, because the output packets are stored and returned at the same sampling rate as the input signal. Oversampling increases computation time and the use of disk space.

`nBits` is the number of physical-layer information bits, available to higher layers in the protocol stack. Hence, `nBits` does not include overhead due to synchronization, training, physical-layer header, and error-correcting codes. But if a communication protocol stack is considered, `nBits` does include overhead due to layers above the physical layer. WATERMARK uses the value of `nBits` when packets are retrieved with AWGN at a specified E_b/N_0 value (`sfetch.m`, see Sec. 5.3), and when it reports the effective data rate.

5 Using Watermark

5.1 System requirements

Watermark runs in MATLAB and works under the Windows and Linux operating systems. It uses the Signal Processing toolbox and requires release R2012b or more recent. The operating scripts for Watermark are located in `\Watermark\matlab\`.

5.2 Channel simulation

The first step is sending your signal(s) through the replay channel(s). This is done with the function `watermark.m`.

```
% Synopsis: watermark(signal, channel, howmany);
%
% Input :  signal      : string variable denoting the signal (e.g. 'my_signal')
%          channel     : string variable denoting the channel (e.g. 'NOF1')
%          howmany     : use 'all' to process all channel files
%                      : use 'single' to process a single channel file
%
%
% This function takes an input signal, stacks multiple copies of it, and passes the
% stack through replay channels. Results are stored in Watermark's output directory.
```

The hypothetical signal `dsss4.mat` uses $f_{s,x} = 48000$ Hz and `nBits` = 256, and its duration is 1.42 s. When it is passed through NOF1, the following text appears on the Matlab prompt:

```
>> watermark('dsss4', 'NOF1', 'all');

Watermark V1.0
--------------
Signal parameters for dsss4:
message size = 256 user bits
total duration = 1.424 s
effective bit rate = 179.74 bit/s

Channel parameters for NOF1:
center frequency = 14000 Hz
sounding duration = 32.9 s
number of soundings = 60

Number of packets per sounding = 21
Total number of simulated packets = 1260

Deleting previous results, if any ... done.

Filtering dsss4 x NOF1_001 ... done.
Filtering dsss4 x NOF1_002 ... done.
Filtering dsss4 x NOF1_003 ... done.
Etc.
```

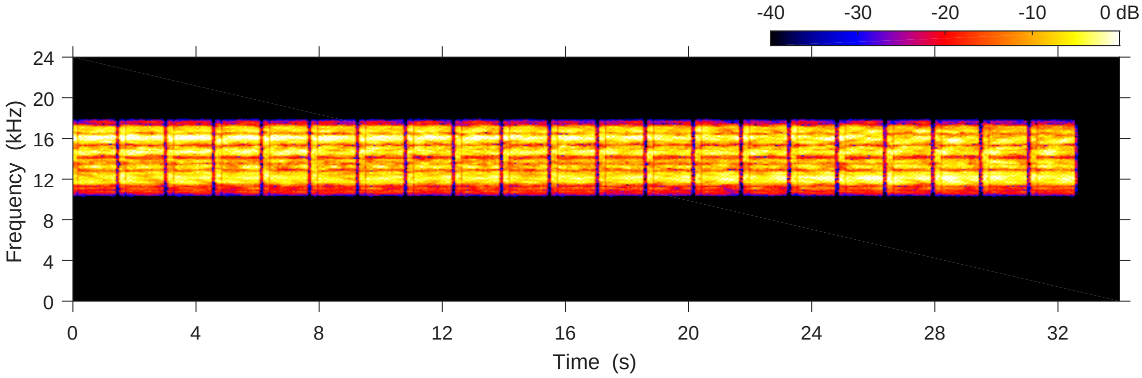

Figure 5.1 *Spectrogram of a Watermark output file featuring a packet train in NOF1.*

WATERMARK stacks multiple copies of the input signal packet, as many as fit into a channel estimate. It uses a spacing between successive packets to avoid reverberation from one packet spilling into the next one. Since it is a replay channel with a measurement-imposed upper limit on the possible delay spread, the required spacing is precisely known. This packet train is subsequently convolved with the channel estimates using the direct replay mode of MIME. The output is stored in `Watermark\output\NOF1\dsss4\`, and consists of 60 wave files with received signal trains. These files are named `NOF1_001.wav`, `NOF1_002.wav`, ... `NOF1_060.wav`. Figure 5.1 depicts such a file. Watermark also stores a file `bookkeeping.mat` containing a struct `bk` with bookkeeping information.

The number of independent packets returned by WATERMARK depends on i) the signal duration; ii) the play time of a channel estimate; iii) the probe signal tracking period; iv) the number of channel estimates. The DSSS4 packet with length 1.424 s fits 23 times in a NOF1 sounding with length 32.9 s, but with the required spacing of 128 ms (the tracking period) it fits $\lfloor 32.9/(1.424 + 0.128 + \Delta) \rfloor = 21$ times in a single sounding. Here, $\Delta = 4$ ms is a safety margin to accomodate possible sidelobes occurring in the filtering process. Since there are 60 NOF soundings, the total number of packets becomes $60 \times 21 = 1260$. Figure 5.2 illustrates the number of packets as a function of signal duration.

An example for array data may look as follows:

```
>>watermark('ofdm', 'BCH1', 'all');

Watermark V1.0
--------------
Signal parameters for ofdm:
message size = 2800 user bits
total duration = 3.466 s
effective bit rate = 807.83 bit/s

Channel parameters for BCH1:
center frequency = 35000 Hz
sounding duration = 59.4 s
number of soundings = 4

Number of packets per sounding = 16
Total number of simulated packets = 64
```

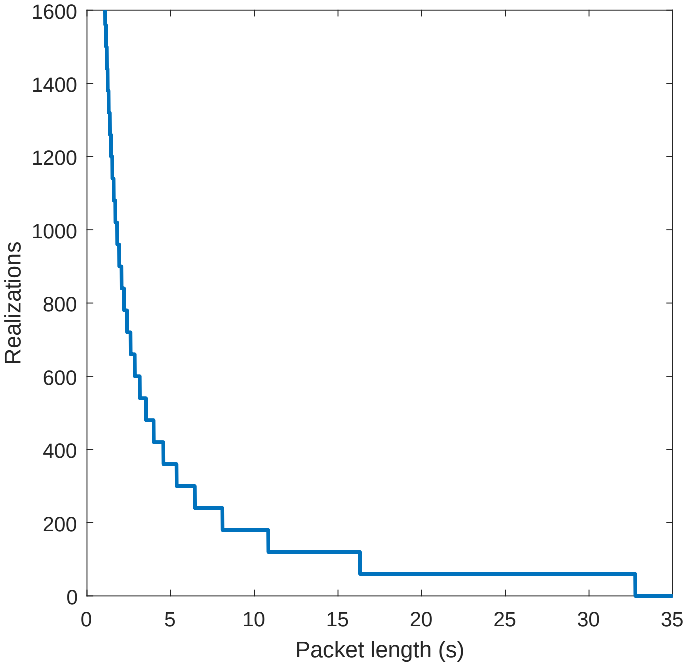

Figure 5.2 *Number of independent packets in NOF1 as a function of the packet length.*

```
Deleting previous results, if any ... done.

Filtering ofdm x BCH1_001 ... done.
Filtering ofdm x BCH1_002 ... done.
Filtering ofdm x BCH1_003 ... done.
Filtering ofdm x BCH1_004 ... done.
>>
```

The total number of packets returned by WATERMARK is 64, but since `BCH1_001 ... BCH1_004` represent hydrophone channels the number of packets for array processing is 64/4 = 16.

For rapid test results, the `watermark.m` function can also called with the `howmany = 'single'` argument. This limits the output to the yield of the first channel file. Note that this makes no sense for array processing, as all channel files have to be processed in order to obtain array data.

5.3 Retrieving packets

After a signal has been passed through a channel, the resulting packets can be retrieved with the functions `sfetch.m` and `pfetch.m`.

5.3.1 Serial fetch

```
% Synopsis: [y, fs] = sfetch(signal, channel, packetNumber, SNR);
```

```
%
% Serial fetch: retrieve packets for temporal data.
%
% Input :  signal      : string variable denoting the signal (e.g. 'my_signal')
%          channel     : string variable denoting the acoustic channel (e.g. 'NOF1')
%          packetNumber: an integer value in the range [1, Nmax], where Nmax is the total number
%                        of filtered packets (bk.nPackets)
%          SNR         : numerical value -> Eb/N0 value in dB
%                        Omit this argument to retrieve the packet without additive noise
%
% Output:  y  : time series of the distorted signal
%          fs : sampling frequency
%
%          NB. When an SNR is specified, the returned time series is ~10 seconds longer than
%              the actual packet, which arrives at a random offset between ~ 4 and 6 s.
```

This function retrieves packets between a source and a single-hydrophone receiver (NOF1 and NCS1). Examples:

```
» [y, fs] = sfetch('dsss4', 'NOF1', 1);
```

returns the first packet received in `NOF1_001`. The sampling rate equals the sampling rate of the transmitted waveform.

```
» [y, fs] = sfetch('dsss4', 'NOF1', 22);
```

returns packet 22, which is the first packet in `NOF1_002`.

```
» [y, fs] = sfetch('dsss4', 'NOF1', 1260);
```

returns packet 1260, which is the last packet received in `NOF1_060`.

```
» [y, fs] = sfetch('dsss4', 'NOF1', 1, 30);
```

returns the first packet, while adding white Gaussian noise (AWGN) at an SNR of 30 dB (E_b/N_0), where E_b is the signal energy per information bit and N_0 the noise power spectral density.[5] When an SNR is specified in the function call, the returned waveform is ≈ 10 seconds longer than the actual packet, which arrives with a random delay after 4–6 seconds. This is the preferred use, because detection and synchronization are essential tasks of modems and modulation schemes. WATERMARK aims to enable a realistic comparison of autonomous receivers.

The characteristics of ambient noise in the oceans vary greatly with environment and conditions [21, 22, 23, 24]. Gaussian statistics may occur in regions where the noise is dominated by sea surface agitation, but it will generally be coloured. The reason for using AWGN in WATERMARK is that this permits the use of the unambiguous E_b/N_0 metric for SNR. Alternatively, users can retrieve packets without noise and apply different noise models, or measured noise, if available.

Figure 5.3 shows a clean input packet, and the same packet after transmission through NOF1. The distorted signal without noise reveals reverberation. Delayed signal energy is observed between the detection preamble and the message, and at the end of the message.

Serial fetch can be applied to array data, but returns a single signal. Packet 1 is the first packet received on the first hydrophone, and the last available packet is the last transmission received on the last hydrophone.

---

[5] Retrieving multiple packets at a constant SNR is not equivalent to at-sea signalling with a constant source level, because both signal level and noise level may vary in the field. Signal level variations are reproduced by WATERMARK, which simply scales the noise accordingly to realize the desired SNR for each packet.

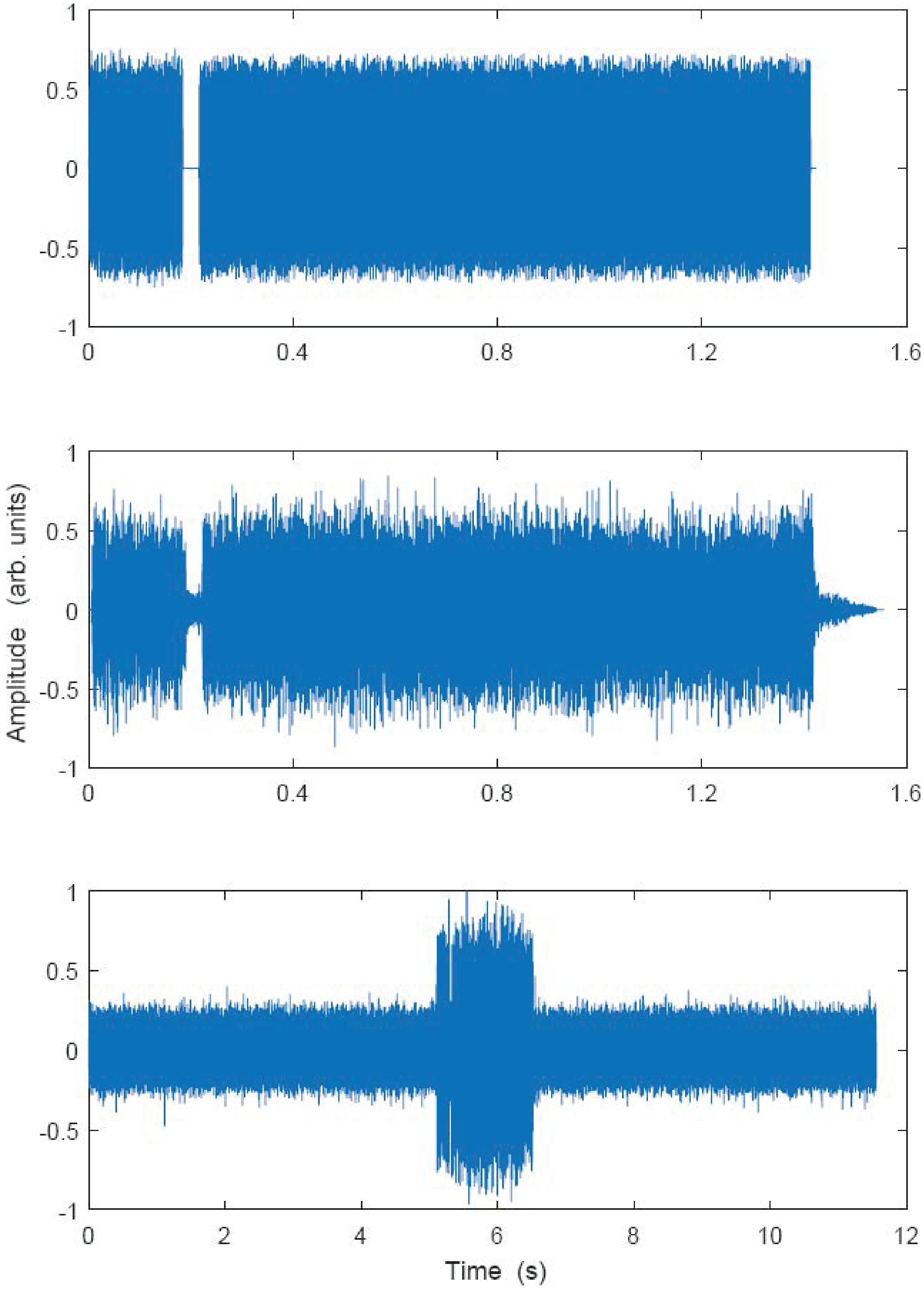

Figure 5.3 *Example of a Watermark input signal (top), and an output signal without noise (middle) and with noise and a delay (bottom). Note the different time scales.*

5.3.2 Parallel fetch

```
% Synopsis: [y, fs] = pfetch(signal, channel, packetNumber);
%
% Parallel fetch: retrieve packets for array data.
%
% Input :  signal      : string variable denoting the signal (e.g. 'my_signal')
%          channel     : string variable denoting the acoustic channel (e.g. 'KAU1')
%          packetNumber: an integer value in the range [1, Nmax], where Nmax is the number
%                        of packets per channel file (bk.nPacketsPerSounding)
%
% Output:  y  : multichannel time series of the distorted signal
%          fs : sampling frequency
```

This function retrieves packets received on a hydrophone array (BCH1, KAU1, and KAU2). The function has no argument specifying an SNR, because Watermark V1.0 lacks a noise model with realistic spatial properties. You need to add noise yourself.

Examples:

```
» [y, fs] = pfetch('ofdm', 'BCH1', 1);
```

returns the first OFDM packet (of the BCH1 example in Sec. 5.2) passed through this channel. The output y is a matrix with four columns representing the signals received on the four hydrophones.

```
» [y, fs] = pfetch('ofdm', 'BCH1', 16);
```

returns the 16-th and last OFDM packet.

If parallel fetch is applied to non-array channels, e.g.

```
» [y, fs] = pfetch('dsss4', 'NOF1', 1);
```

a 60-column matrix is returned with the reception of the first packet in each of the 60 transmission cycles.

6 Reporting results

When Watermark is used to demonstrate the performance of a modulation scheme or a modem, it is good practice to produce an unambiguous report that allows other people to compare their systems in a meaningful way. There is a lack of standard terminology in underwater acoustic communications, causing people to use different definitions for apparently trivial quantities such as SNR, bandwidth and bit rate. It is recommended to report, in addition to whatever you would like to share about the proposed communication algorithms:

- The Watermark version number
- The number of information bits in the packet
- The effective bit rate reported by Watermark
- The signal bandwidth
- The total number of packets in the test channel
- The type of noise in simulation (e.g. the default AWGN of `sfetch.m` or something else)
- Whether communication results are averaged over all packets or a specific subset
- Used definitions of bandwidth, input SNR, output SNR, etc., as appropriate.
- Whether pre-existing knowledge is used about signal start or Doppler shift, or whether the receiver has to figure out everything itself (e.g. with a Doppler-bank preamble detector)
- To what extent signal and receiver parameters are tuned to the test channel under consideration
- Anything else which helps to reproduce or appreciate the results

7 Example of use

Table 7.1 specifies signals of two modulation schemes which are tested and compared in NOF1 and NCS1. The first signal is a single-carrier waveform with a quadrature phase-shift keyed (QPSK) symbol constellation. The second signal is also single-carrier QPSK, but uses a 15-chip spreading code for increased robustness. It is DSSS, but a different implementation than the scheme used for the validation in Figure 2.4. A description of the receiver algorithms is not given, because the objective is to illustrate the use of WATERMARK and not to present novel modulation schemes. They are routine FFI schemes with standard parameter settings, not optimized for the channels under consideration.

Both signals carry a message of 256 bits. They are equipped with a detection preamble and training symbols, which are a considerable overhead at this short message length. The effective bit rates are thus relatively low. The QPSK packet lasts only 0.21 s, which results in thousands of independent packets in NOF1 and NCS1.

The `sfetch` function is used to retrieve packets with AWGN and a variable start time. The receivers use a filter matched to a bank of Doppler-shifted preamble replicas (with velocities spanning the range from −5 to +5 m/s) for detection, synchronization and Doppler estimation. The knowledge that NOF1 and NCS1 are channels between stationary stations is thus not used. The employed detection threshold results in about 1 false alarm per day in AWGN.

The `sfetch` function is called from a receiver batch job cycling through a range of (E_b/N_0) SNR values. At each SNR, all WATERMARK packets are retrieved and fed to the communication receiver. The packet error ratio (PER) is considered as the performance metric for the present investigation. A packet error occurs when one or more bits are in error at the decoder output, or when the packet is not detected.

The outcome of the simulations is shown in Fig. 7.1. There are several observations:

- NCS1 is a more challenging channel than NOF.
- DSSS is more robust than QPSK.
- DSSS works well in both channels, but requires a 3–6 dB higher SNR in NCS1 to achieve the same performance as in NOF1.
- The performance of QPSK in NCS1 is limited by delay-Doppler spread. It is not possible to improve the robustness of this link by increasing the modem source level.

Table 7.1 Signal parameters.

	QPSK	DSSS
Carrier frequency	14 kHz	14 kHz
−3 dB bandwidth B	5.6 kHz	5.6 kHz
# information bits	256	256
Effective bit rate R_{eff}	1222 b/s	180 b/s
# packets in NOF1	5760	1260
# packets in NCS1	7920	1320

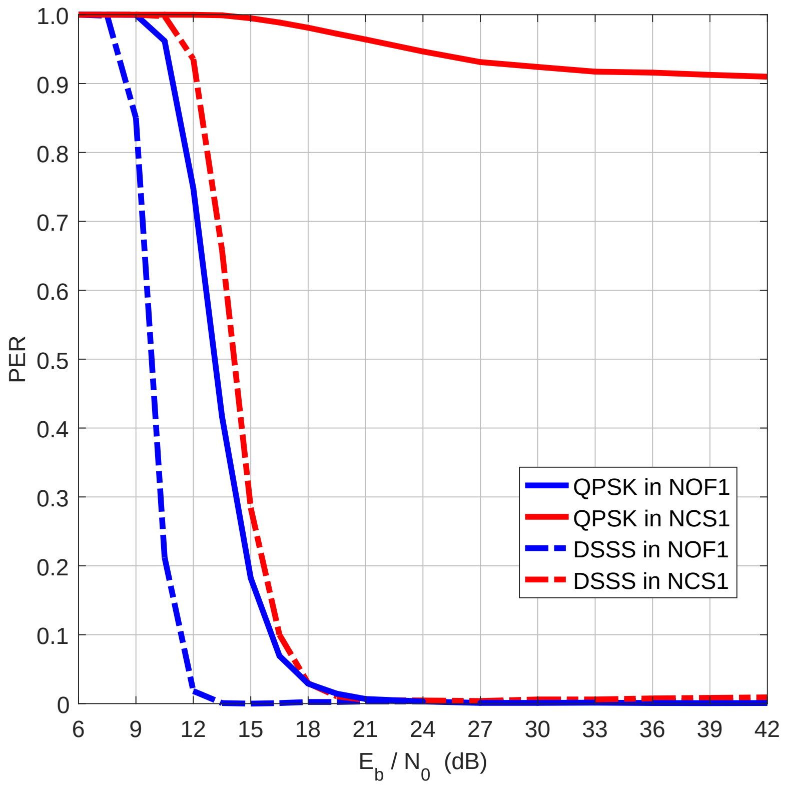

Figure 7.1 *PER versus E_b/N_0 in NOF1 and NCS1.*

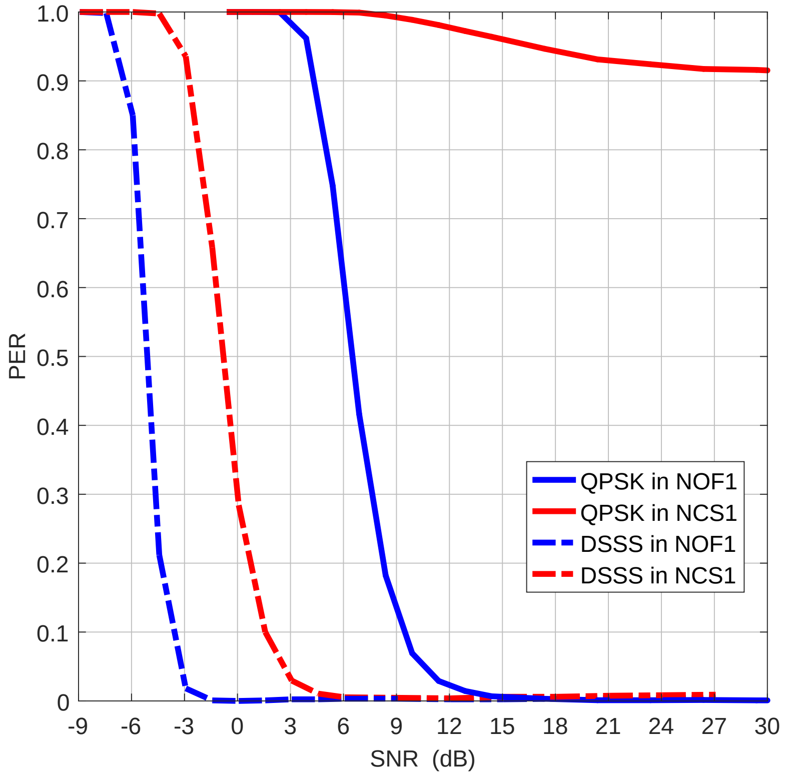

Figure 7.2 *PER versus the acoustic SNR in NOF1 and NCS1.*

Results can also be plotted versus the “acoustic SNR” (ratio of signal power to noise power, measured in the frequency band of the signal) with the conversion SNR $= (R_{\text{eff}}/B) \times (E_b/N_0)$. Fig. 7.2 shows the outcome and confirms the notion that data communication at low SNR requires low-rate signalling.

The observation that a spreading code increases the robustness to channel distortions and noise is not surprising, but without realistic simulations it would be difficult to tell how big the differences are in a given channel.

8 Conclusions

A realistic benchmark is now available for underwater acoustic communications. It is based on a replay channel simulator driven by measurements of the time-varying impulse response. Its initial library of five test channels comprises four geographical areas and three frequency bands. Two of these test channels offer reception on a vertical line array. The primary use of WATERMARK is benchmarking of the physical layer of acoustic communication systems, but in principle it can also be used for different sonar applications involving the one-way transmission of sound.

WATERMARK brings the realism of at-sea signalling into the office while adding reproducibility. Applications include:

- Develop, test and improve algorithms
- Document the performance of modulation schemes and parameter settings, both for in-house evaluations and scientific publications
- Compare different modulation schemes in the same channel
- Examine the performance of a given scheme in different channels
- The quest for a robust high-rate system
- Extract error statistics for network simulations
- Study channel characteristics

The benchmark can be extended with channels from different environments and frequency bands, either for own use or for general distribution, depending on the willingness of third parties to perform suitable measurements and share the data.

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9 Abbreviations

AWGN	Additive White Gaussian Noise
BCH	Brest Commercial Harbour
DSSS	Direct-Sequence Spread Spectrum
FFI	Norwegian Defence Research Establishment
KAM	Kauai Acomms MURI
KAU	Kauai
LFM	Linear Frequency Modulation
Mime	Name of the FFI channel simulator
m-sequence	Maximum-Length Sequence
NCS	Norway – Continental Shelf
NOF	Norway – Oslofjord
PER	Packet Error Ratio
QPSK	Quadrature Phase-Shift Keying
SIMO	Single Input Multiple Output
SISO	Single Input Single Output
SNR	Signal-to-Noise Ratio
TVD	Time-Varying Doppler (Shift)
TVIR	Time-Varying Impulse Response
OFDM	Orthogonal Frequency Division Multiplexing

About FFI

The Norwegian Defence Research Establishment (FFI) was founded 11th of April 1946. It is organised as an administrative agency subordinate to the Ministry of Defence.

FFI's MISSION

FFI is the prime institution responsible for defence related research in Norway. Its principal mission is to carry out research and development to meet the requirements of the Armed Forces. FFI has the role of chief adviser to the political and military leadership. In particular, the institute shall focus on aspects of the development in science and technology that can influence our security policy or defence planning.

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FFI turns knowledge and ideas into an efficient defence.

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Creative, daring, broad-minded and responsible.

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FFIs FORMÅL

Forsvarets forskningsinstitutt er Forsvarets sentrale forskningsinstitusjon og har som formål å drive forskning og utvikling for Forsvarets behov. Videre er FFI rådgiver overfor Forsvarets strategiske ledelse. Spesielt skal instituttet følge opp trekk ved vitenskapelig og militærteknisk utvikling som kan påvirke forutsetningene for sikkerhetspolitikken eller forsvarsplanleggingen.

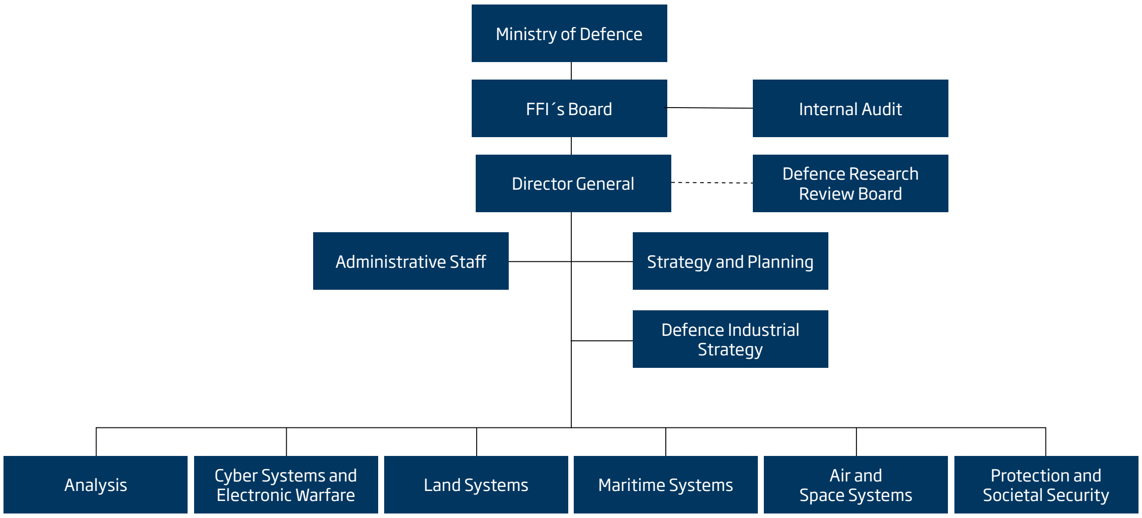
FFIs VISJON

FFI gjør kunnskap og ideer til et effektivt forsvar.

FFIs VERDIER

Skapende, drivende, vidsynt og ansvarlig.

FFI's organisation



Forsvarets forskningsinstitutt
Postboks 25
2027 Kjeller

Besøksadresse:
Instituttveien 20
2007 Kjeller

Telefon: 63 80 70 00
Telefaks: 63 80 71 15
Epost: ffi@ffi.no

Norwegian Defence Research Establishment (FFI)
P.O. Box 25
NO-2027 Kjeller

Office address:
Instituttveien 20
N-2007 Kjeller

Telephone: +47 63 80 70 00
Telefax: +47 63 80 71 15
Email: ffi@ffi.no

FFI Forsvarets forskningsinstitutt
Norwegian Defence Research Establishment