



BSI Standards Publication

Underwater acoustics — Measurement of underwater ambient sound

National foreword

This British Standard is the UK implementation of ISO 7605:2025.

The UK participation in its preparation was entrusted to Technical Committee EH/1/7, Underwater acoustics.

A list of organizations represented on this committee can be obtained on request to its committee manager.

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© The British Standards Institution 2025
Published by BSI Standards Limited 2025

ISBN 978 0 539 18608 6

ICS 17.140.01

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This British Standard was published under the authority of the Standards Policy and Strategy Committee on 31 July 2025.

Amendments/corrigenda issued since publication

Date	Text affected
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**International
Standard**

ISO 7605

**Underwater acoustics —
Measurement of underwater
ambient sound**

Acoustique sous-marine — Mesurage du son ambiant sous-marin

**First edition
2025-07**

Reference number
ISO 7605:2025(en)

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Published in Switzerland

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Foreword

ISO (the International Organization for Standardization) is a worldwide federation of national standards bodies (ISO member bodies). The work of preparing International Standards is normally carried out through ISO technical committees. Each member body interested in a subject for which a technical committee has been established has the right to be represented on that committee. International organizations, governmental and non-governmental, in liaison with ISO, also take part in the work. ISO collaborates closely with the International Electrotechnical Commission (IEC) on all matters of electrotechnical standardization.

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This document was prepared by Technical Committee ISO/TC 43, *Acoustics*, Subcommittee SC 3, *Underwater acoustics*.

Any feedback or questions on this document should be directed to the user's national standards body. A complete listing of these bodies can be found at www.iso.org/members.html.

Introduction

This document describes a procedure for the measurement and analysis of underwater ambient sound. The development of this document made use of resources previously developed by the UK National Physical Laboratory^[20], the US-funded project ADEON¹⁾ and by the EU-funded Interreg programme JOMOPANS²⁾.

1) (University of New Hampshire online) Atlantic Deepwater Ecosystem Observatory Network (ADEON): An Integrated System for Long-Term Monitoring of Ecological and Human Factors on the Outer Continental Shelf (<https://adeon.unh.edu/standards>)

2) Joint Monitoring Programme for Ambient Noise North Sea (JOMOPANS) (<https://northsearegion.eu/jomopans/>)

Underwater acoustics — Measurement of underwater ambient sound

1 Scope

This document provides requirements and recommendations for measuring and reporting ambient sound in water, as characterized by sound pressure and selected quantities that can be derived from sound pressure. “Ambient sound” implies sound from any source except sources of self-noise.

The scope includes equipment performance, calibration and deployment, digital data acquisition and data processing. Data processing is the process of converting raw data into a form and context necessary to be interpreted by people and computers. The scope includes data analysis and reporting of recordings of duration one day or longer.

Five data processing stages are considered: raw digital acquisition data³⁾, sound pressure time series, sound pressure level time series, sound pressure spectra and their statistics.

The scope excludes measurement of particle motion.

2 Normative references

The following documents are referred to in the text in such a way that some or all of their content constitutes requirements of this document. For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments) applies.

ISO 80000-1, *Quantities and units — Part 1: General*

ISO 80000-2, *Quantities and units — Part 2: Mathematics*

ISO 80000-3, *Quantities and units — Part 3: Space and time*

ISO 80000-8, *Quantities and units — Part 8: Acoustics*

IEC 80000-13, *Quantities and units — Part 13: Information science and technology*

ISO 18405, *Underwater acoustics — Terminology*

IEC 60565-1, *Underwater acoustics — Hydrophones — Calibration of hydrophones — Part 1: Procedures for free-field calibration of hydrophones*

IEC 60565-2, *Underwater acoustics — Hydrophones — Calibration of hydrophones — Part 2: Procedures for low frequency pressure calibration*

BIPM 2019, *9th (2019) edition*

3 Terms and definitions

For the purposes of this document, the terms and definitions given in BIPM 2019, ISO 80000-1, ISO 80000-2, ISO 80000-3, ISO 80000-8, IEC 80000-13, ISO 18405 and the following apply.

3) The word data is generally used as a collective noun in this document; the plural form is reserved for cases where the constructive relationship to individual observations or measurements is to be emphasized.

ISO and IEC maintain terminology databases for use in standardization at the following addresses:

- ISO Online browsing platform: available at <https://www.iso.org/obp>
- IEC Electropedia: available at <https://www.electropedia.org/>

3.1 Concepts and quantities: Hardware

3.1.1 Acoustic measuring systems: Concepts and components

3.1.1.1

hydrophone

electroacoustic transducer that produces electrical voltages in response to water borne pressure signals

Note 1 to entry: A hydrophone is designed to respond principally to underwater sound pressure.

Note 2 to entry: In general, a hydrophone can also produce a signal in response to non-acoustic pressure fluctuations (e.g. those existing in a turbulent boundary layer during conditions of high water flow).

Note 3 to entry: Hydrophone types include reference hydrophones and measuring hydrophones. Measuring hydrophones are used in general measurements of sound fields. Reference hydrophones are principally used for *calibration* ([3.1.1.11](#)) purposes (e.g. in comparison calibrations with measuring hydrophones).

Note 4 to entry: Hydrophones are principally used as listening devices, but in reciprocity calibration, a hydrophone is used as reciprocal transducer, not only acting as a hydrophone, but also as a projector (sound source [IEC 60565-1]).

Note 5 to entry: A compact, integrated *underwater acoustic measuring system* ([3.1.1.6](#)) is sometimes called a “digital hydrophone” from its superficial resemblance to a standalone analogue hydrophone.

Note 6 to entry: If a hydrophone is connected to a charge amplifier, the sensitivity of the hydrophone is sometimes described in terms of charge sensitivity, which is related to the voltage sensitivity of the hydrophone by its electrical capacitance.

[SOURCE: IEC 60565-1:2020, 3.15, modified.]

3.1.1.2

pre-amplifier

amplifier located immediately after the sensing element to increase the amplitude of the voltage (or current)

Note 1 to entry: A preamplifier often takes the form of a low-noise amplifier located as near as possible to the sensing element or another signal source. Other amplifiers can be placed later in the measurement chain.

Note 2 to entry: Adapted from IEC 60050-713, 10-45 and ISO 18406:2017.

[SOURCE: adapted from Robinson and Wang (2021), Table 9^[18]]

3.1.1.3

aliasing

false representation of a signal caused by mixing of spectral components above the *Nyquist frequency* ([3.1.4.2](#)) with those spectral components below the Nyquist frequency

[SOURCE: ISO 18431-1:2005, 3.1]

3.1.1.4

anti-alias filter

AAF

low-pass filter designed to avoid *aliasing* ([3.1.1.3](#))

Note 1 to entry: An anti-alias filter removes frequencies above the *Nyquist frequency* ([3.1.4.2](#)) of the *digital acquisition system* ([3.1.1.6](#)).

[SOURCE: adapted from Robinson and Wang (2021), Table 9^[18]]

3.1.1.5

analogue-to-digital converter

ADC

electronic component that converts an analogue input signal to a digital output signal

Note 1 to entry: The input signal is typically an analogue electric voltage. The output is a digitized representation of the input sampled at finite time intervals. Adapted from ISO 18406:2017 and IEC 60050-723, 10-04.

[SOURCE: based on Robinson and Wang (2021), Table 9^[18]]

3.1.1.6

digital acquisition system

measuring system

sequence of electronic components designed for data acquisition that may be considered together as whole, with a single overall performance

Note 1 to entry: An underwater acoustic acquisition system typically comprises a *hydrophone* (3.1.1.1), a *pre-amplifier* (3.1.1.2), an *AAF* (3.1.1.4), an *ADC* (3.1.1.5) and some form of digital interface.

[SOURCE: Robinson and Wang (2021), Table 9^[18]]

3.1.1.7

ADC input

analogue input signal to an *ADC* (3.1.1.5) such as electric voltage

Note 1 to entry: Adapted from Ainslie et al. (2020)^[8].

[SOURCE: Robinson and Wang (2021), Table 9^[18]]

3.1.1.8

ADC output

digital representation of the *ADC input* (3.1.1.7) sampled at finite time intervals

Note 1 to entry: An ADC output is suitable for storage in a digital storage medium or processing on a digital computer. Adapted from Ainslie et al. (2020)^[8].

[SOURCE: Robinson and Wang (2021), Table 9^[18]]

3.1.1.9

full-scale signal

signal spanning the entire range of input values representable by an *ADC* (3.1.1.5), from *minimum unsaturated voltage* (3.1.4.11) to *maximum unsaturated voltage* (3.1.4.10), without clipping

Note 1 to entry: Compare IEEE (STD-1241-2010) (p 13)^[4]: “A full-scale signal is one whose peak-to-peak amplitude spans the entire range of input values recordable by the *analogue-to-digital converter* (3.1.1.5) under test.”

[SOURCE: Ainslie et al. (2020)^[8]]

3.1.1.10

digital autonomous acoustic recorder

underwater acoustic *measuring system* (3.1.1.6) designed to operate without human intervention for sustained periods, with storage to record acoustic signals in a digital format

Note 1 to entry: Typically, digital autonomous acoustic recorders can be programmed to operate autonomously and sample over a range of duty cycles. Adapted from ISO 18406:2017.

[SOURCE: adapted from Robinson and Wang (2021), Table 9^[18]]

3.1.1.11 calibration

method of using known inputs, possibly using physical stimuli (such as a known, calibrated and traceable acoustic source) or electrical input (charge or voltage signal injection) at the input (or other stage) of an underwater acoustic *measuring system* (3.1.1.6)

[SOURCE: ISO 17208-1:2016, 3.1, modified — "field" was deleted from the term and the definition slightly reworded.]

3.1.2 Hydrophone characteristics

3.1.2.1 free-field receive sensitivity free-field voltage sensitivity

FFRS

FFVS

$M_f(f)$

<of a hydrophone> quotient of the Fourier transform of the hydrophone open-circuit voltage signal and the Fourier transform of the acoustic pressure signal, for specified frequency and specified direction of plane wave sound incident on the position of the reference centre of the *hydrophone* (3.1.1.1) in the undisturbed free-field if the hydrophone was removed

Note 1 to entry: In formula form, $M_f(f) = \frac{\int_{-\infty}^{+\infty} \exp(-2\pi i f t) v(t) dt}{\int_{-\infty}^{+\infty} \exp(-2\pi i f t) p(t) dt}$, where $v(t)$ is the hydrophone open-circuit voltage at time t and $p(t)$ is the sound pressure of the plane wave at time t .

Note 2 to entry: Free-field receive sensitivity is expressed in volt per pascal (V Pa⁻¹).

Note 3 to entry: The hydrophone free-field receive sensitivity is a complex-valued parameter. The modulus of the free-field receive sensitivity of a *hydrophone* (3.1.1.1) is expressed in units of volt per pascal, V·Pa⁻¹. The phase angle is the argument of the sensitivity and represents the phase difference between the hydrophone electrical voltage and the sound pressure. The unit of phase angle is the radian.

Note 4 to entry: The term "response" is sometimes used instead of "sensitivity".

[SOURCE: IEC 60565-1:2020, 3.4]

3.1.2.2 equivalent bandwidth self-noise pressure

$p_{N,eq}$

square root of the integral over a specified frequency range of the quotient of the noise voltage power spectral density present at the electrical terminals of the *hydrophone* (3.1.1.1), in the absence of pressure fluctuations at the hydrophone input, and the squared modulus of its *free-field receive sensitivity* (3.1.2.1)

Note 1 to entry: In formula form, $p_{N,eq} = \sqrt{\int_{f_1}^{f_2} \frac{V_{N,f}^2}{|M_f(f)|^2} df}$, where $V_{N,f}^2$ is the noise voltage power spectral density and $M_f(f)$ is the *free-field receive sensitivity* (3.1.2.1) at frequency f . The frequency range is f_1 to f_2 .

Note 2 to entry: Equivalent bandwidth self-noise pressure is expressed in pascals (Pa).

Note 3 to entry: If the *hydrophone* (3.1.1.1) contains an integral preamplifier, the output terminals considered are those at the output of the *pre-amplifier* (3.1.1.2) and the sensitivity in question includes the pre-amplifier gain.

3.1.2.3**self-noise voltage****RMS self-noise voltage** $v_{N,self}$

RMS voltage at a specified location in an acoustic receiver in the absence of sound pressure input at the *hydrophone* (3.1.1.1), for a specified frequency range

Note 1 to entry: Self-noise voltage is expressed in volts (V).

[SOURCE: Ainslie et al. (2020)^[8]]

3.1.3 Analogue acquisition system characteristics**3.1.3.1****noise current**

current at a specified location in an acoustic receiver in the absence of an acoustic signal at the *hydrophone* (3.1.1.1)

3.1.3.2**noise voltage**

voltage at a specified location in an acoustic receiver in the absence of an acoustic signal at the *hydrophone* (3.1.1.1)

3.1.3.3**signal current**

current that would exist at a specified location in an acoustic receiver in the absence of noise

3.1.3.4**signal voltage**

voltage that would exist at a specified location in an acoustic receiver in the absence of noise

3.1.3.5**signal-plus-noise current**

current at a specified location in an acoustic receiver

3.1.3.6**signal-plus-noise voltage**

voltage at a specified location in an acoustic receiver

3.1.3.7**electrical noise power**

time-averaged product of *noise current* (3.1.3.1) and *noise voltage* (3.1.3.2), for a specified position in the processing chain and a specified averaging time

Note 1 to entry: Electrical noise power is expressed in watts (W).

Note 2 to entry: In an electrical circuit of resistance, R , electrical noise power is given by mean-square noise voltage divided by R or mean-square noise current multiplied by R .

Note 3 to entry: The electrical noise power depends on the position in the processing chain at which it is determined.

Note 4 to entry: Compare IEEE (STD-1241-2010) ('noise (total)', p15)^[4]: "Any deviation between the output signal (converted to input units) and the input signal except deviations caused by linear time-invariant system response (gain and phase shift), or a DC level shift. For example, noise includes the effects of random errors (random noise), fixed pattern errors, nonlinearities (e.g. harmonic or intermodulation distortion), and aperture uncertainty. See also: random noise."

[SOURCE: adapted from Ainslie et al. (2020)^[8]]

3.1.3.8**electrical signal power** W_s

time-averaged product of *signal current* (3.1.3.3) and *signal voltage* (3.1.3.4), for a specified position in the processing chain and a specified averaging time

Note 1 to entry: Electrical signal power is expressed in watts (W).

Note 2 to entry: In an electrical circuit of resistance, R , electrical signal power is given by mean-square signal voltage divided by R or mean-square signal current multiplied by R .

Note 3 to entry: The electrical signal power depends on the position in the processing chain at which it is determined.

[SOURCE: Ainslie et al. (2020)[8]]

3.1.3.9**electrical signal-plus-noise power**

time-averaged product of *signal-plus-noise current* (3.1.3.5) and *signal-plus-noise voltage* (3.1.3.6), for a specified position in the processing chain and a specified averaging time

Note 1 to entry: Electrical signal-plus-noise power is expressed in watts (W).

Note 2 to entry: In an electrical circuit of resistance, R , signal-plus-noise power is given by mean-square signal-plus-noise voltage divided by R or mean-square signal-plus-noise current multiplied by R .

Note 3 to entry: The electrical signal-plus-noise power depends on the position in the processing chain at which it is determined.

3.1.3.10**signal-to-noise power ratio****signal-to-noise ratio** R_{SN}

ratio of *electrical signal power* (3.1.3.8) to *electrical noise power* (3.1.3.7), for a specified position in the processing chain

Note 1 to entry: The signal-to-noise power ratio depends on the position in the processing chain at which it is determined.

[SOURCE: Ainslie et al. (2020)[8]]

3.1.3.11**signal-plus-noise-to-noise power ratio****signal-plus-noise-to-noise ratio** R_{S+NN}

ratio of *electrical signal-plus-noise power* (3.1.3.9) to *electrical noise power* (3.1.3.7), for a specified position in the processing chain

Note 1 to entry: The signal-plus-noise-to-noise power ratio depends on the position in the processing chain at which it is determined.

3.1.4 Sampling system characteristics**3.1.4.1****sampling rate****sampling frequency** f_s

number of samples of a signal taken per unit time

Note 1 to entry: Sampling frequency is expressed in hertz (Hz).

[SOURCE: IEC 60050-704:1993, 23-03]

3.1.4.2**Nyquist frequency** f_{Nyq} maximum usable frequency available in data taken at a given *sampling rate* (3.1.4.1)Note 1 to entry: In formula form, $f_{\text{Nyq}} = f_s / 2$, where f_s is the sampling rate.

Note 2 to entry: Nyquist frequency is expressed in hertz (Hz).

Note 3 to entry: Except for the in-phase component of continuous signals, frequencies for analysis are lower than the Nyquist frequency.

Note 4 to entry: This document imposes additional restrictions on the maximum useful frequency due to practical limitations in the realization of anti-aliasing filters.

Note 5 to entry: An alternative definition of Nyquist frequency is '1/2 times the inverse of the sampling period' (see ISO 19262:2015, 3.173).

[SOURCE: ISO 18431-1: 2005, 3.7, modified — Notes to entry added.]

3.1.4.3**ADC input voltage** V_{ADC} voltage at input to *ADC* (3.1.1.5)

[SOURCE: Ainslie et al. (2020)[8]]

3.1.4.4**integer ADC output** N_{ADC} integer representation of *ADC output* (3.1.1.8), defined such that a unit change in integer ADC output corresponds to a change in the least significant bit from 0 to 1 or from 1 to 0Note 1 to entry: The integer ADC output is equal to the product of *ADC input* (3.1.1.7) voltage and *ADC sensitivity to voltage* (3.1.4.9).Note 2 to entry: The integer ADC output is one of $2^{N_{\text{bit}}}$ consecutive integers, where N_{bit} is the *bit depth* (3.1.4.5).Note 3 to entry: The integer ADC output is the *ADC output* (3.1.1.8) when the *ideal code bin width* (3.1.4.13) is equal to 1.

[SOURCE: Ainslie et al. (2020)[8]]

3.1.4.5**bit depth** N_{bit} number of bits at *ADC output* (3.1.1.8) used to represent one value of *ADC input* (3.1.1.7)

[SOURCE: Ainslie et al. (2020)[8]]

3.1.4.6**maximum integer ADC output** $N_{\text{ADC,max}}$ maximum possible value of the *integer ADC output* (3.1.4.4)

[SOURCE: Ainslie et al. (2020)[8]]

3.1.4.7**minimum integer ADC output** $N_{\text{ADC,min}}$ minimum possible value of the *integer ADC output* (3.1.4.4)Note 1 to entry: The *integer ADC output* (3.1.4.4) can be positive or negative. If at least one value of the *integer ADC output* (3.1.4.4) is negative, the *minimum integer ADC output* (3.1.4.7) is also negative.

Note 2 to entry: The *integer ADC output* (3.1.4.4) is one of $2^{N_{\text{bit}}}$ consecutive integers, where N_{bit} is the *bit depth* (3.1.4.5).

[SOURCE: Ainslie et al. (2020)[8]]

3.1.4.8 full-scale ADC output

$N_{\text{ADC,FS}}$

one more than the difference between the *maximum integer ADC output* (3.1.4.6) and the *minimum integer ADC output* (3.1.4.7)

Note 1 to entry: The *integer ADC output* (3.1.4.4) can be positive or negative. If the *maximum integer ADC output* (3.1.4.6) is positive and the *minimum integer ADC output* (3.1.4.7) is negative, then the *full-scale ADC output* (3.1.4.8) is the sum of the *maximum integer ADC output* (3.1.4.6) and the magnitude of the *minimum integer ADC output* (3.1.4.7).

[SOURCE: adapted from Ainslie et al. (2020)[8]]

3.1.4.9 ADC sensitivity to voltage

$M_{\text{ADC},v}$

quotient of root-mean-square *integer ADC output* (3.1.4.4) ($N_{\text{ADC,rms}}$) and root-mean-square *ADC input voltage* (3.1.4.3) ($v_{\text{ADC,rms}}$)

Note 1 to entry: In formula form, $M_{\text{ADC},v} = \frac{N_{\text{ADC,rms}}}{v_{\text{ADC,rms}}}$. The ADC sensitivity to voltage is equal to 2^N bit divided by the *full-scale signal* (3.1.1.9) peak to peak voltage.

Note 2 to entry: ADC sensitivity to voltage is expressed in reciprocal volt (V^{-1}).

[SOURCE: Ainslie et al. (2020)[8]]

3.1.4.10 maximum unsaturated voltage

v_{max}

maximum *ADC input voltage* (3.1.4.3) for which the *ADC sensitivity to voltage* (3.1.4.9) is independent of *ADC input* (3.1.1.7)

Note 1 to entry: Maximum unsaturated voltage is expressed in volts (V).

Note 2 to entry: The maximum unsaturated voltage is the maximum *ADC input voltage* (3.1.4.3) for which the *ADC sensitivity to voltage* (3.1.4.9) is linear with input voltage.

[SOURCE: Ainslie et al. (2020)[8]]

3.1.4.11 minimum unsaturated voltage

v_{min}

minimum *ADC input voltage* (3.1.4.3) for which the *ADC sensitivity to voltage* (3.1.4.9) is independent of *ADC input* (3.1.1.7)

Note 1 to entry: Minimum unsaturated voltage is expressed in volts (V).

Note 2 to entry: The *ADC input voltage* (3.1.4.3) can be positive or negative. If at least one value of the integer ADC input voltage is negative, the minimum unsaturated voltage is also negative.

[SOURCE: Ainslie et al. (2020)[8]]

3.1.4.12**full-scale input range****full-scale range****full-scale voltage****FSR**

difference between *maximum unsaturated voltage* (3.1.4.10) and *minimum unsaturated voltage* (3.1.4.11)

Note 1 to entry: Full-scale input range is expressed in volts (V).

Note 2 to entry: The *ADC input voltage* (3.1.4.3) can be positive or negative. If the *maximum unsaturated voltage* (3.1.4.10) is positive and the *minimum unsaturated voltage* (3.1.4.11) is negative, then the full-scale input range is the sum of the maximum unsaturated voltage and the magnitude of the minimum unsaturated voltage.

Note 3 to entry: Compare IEEE (STD-1241-2010) ('full scale range', p13)^[4]: "The difference between the most positive and most negative analog inputs of a converter's operating range. For an N -bit converter, FSR is given by: $FSR = (2^N - 1) \times (\text{ideal code width})$ in analog input units."

[SOURCE: Ainslie et al. (2020)^[8]]

3.1.4.13**ideal code bin width****ideal code width****ADC output step size** **Q**

full-scale input range (3.1.4.12) divided by $N_{\text{ADC,FS}}$, where $N_{\text{ADC,FS}}$ is the *full-scale ADC output* (3.1.4.8)

Note 1 to entry: The *full-scale ADC output* (3.1.4.8) is equal to the total number of code bins plus one.

Note 2 to entry: Compare IEEE (STD-1241-2010) ('ideal code bin width', p14): "The ideal full-scale input range divided by the total number of code bins."

Note 3 to entry: Compare IEEE (STD-1241-2010) ('least significant bit', p14)^[4]: "With reference to analog-to-digital converter input signal amplitude, an LSB [least significant bit] is synonymous with one ideal code bin width."

[SOURCE: adapted from Ainslie et al. (2020)^[8]]

3.1.4.14**pre-amplifier voltage gain** **$G_{\text{PA,V}}$**

ratio of root-mean-square pre-amplifier output voltage to root-mean-square pre-amplifier input voltage

Note 1 to entry: The pre-amplifier voltage gain can vary with frequency.

Note 2 to entry: For a tonal signal, the pre-amplifier voltage is equal to the ratio of peak pre-amplifier output voltage to peak pre-amplifier input voltage.

[SOURCE: Ainslie et al. (2020)^[8]]

3.1.4.15**AAF voltage gain** **$G_{\text{AAF,V}}$**

ratio of root-mean-square AAF output voltage to root-mean-square AAF input voltage

Note 1 to entry: The AAF voltage gain can vary with frequency.

[SOURCE: Ainslie et al. (2020)^[8]]

3.1.4.16 equivalent root-mean-square ADC noise voltage

$v_{N,eq}$

quotient of the root-mean-square integer ADC noise output and the *ADC sensitivity to voltage* ([3.1.4.9](#))

Note 1 to entry: In formula form, $v_{N,eq} = \frac{N_{ADC,rms}}{M_{ADC,V}}$.

Note 2 to entry: Equivalent root-mean-square ADC noise voltage is expressed in volts (V).

Note 3 to entry: Equivalent root-mean-square ADC noise voltage includes contributions from acoustic and non-acoustic noise.

[SOURCE: Ainslie et al. (2020)^[8]]

3.1.4.17 equivalent ADC self-noise voltage

$v_{N,eq,self}$

equivalent root-mean-square ADC noise voltage ([3.1.4.16](#)) in the absence of *ADC input* ([3.1.1.7](#))

Note 1 to entry: Equivalent root-mean-square ADC self-noise voltage is expressed in volts (V).

[SOURCE: Ainslie et al. (2020)^[8]]

3.1.5 Acquisition system characteristics

3.1.5.1 system sensitivity frequency response

M_{sys}

quotient of the root-mean-square output of the *measuring system* ([3.1.1.6](#)) and the incident root-mean-square sound pressure that would be present at the position of the reference centre of the *hydrophone* ([3.1.1.1](#)) in the undisturbed free field if the hydrophone was removed, for specified frequency and specified direction of harmonic plane wave sound

Note 1 to entry: System sensitivity is expressed in reciprocal pascal (Pa⁻¹).

Note 2 to entry: The system sensitivity is defined here as a free-field sensitivity for an underwater acoustic *measuring system* ([3.1.1.6](#)) designed to measure sound pressure signals in water. The response of the system will in general vary with acoustic frequency.

Note 3 to entry: For digital systems, the *calibration* ([3.1.1.11](#)) of the *ADC* ([3.1.1.5](#)) is typically incorporated into the overall sensitivity of the whole system, which is the number of digital counts per unit change in sound pressure (unit Pa⁻¹).

Note 4 to entry: If the *hydrophone* ([3.1.1.1](#)) is physically attached to the body of an acoustic recorder (rather than deployed on an extension cable), diffraction and scattering of sound by the recorder body can affect the free-field sensitivity at kilohertz frequencies, causing enhanced directivity compared to the response of the free hydrophone.

Note 5 to entry: In general, the *measuring system* ([3.1.1.6](#)) can introduce a phase delay into the measured signal. This can be accounted for by representing the system sensitivity as a complex valued quantity, the modulus of which represents the magnitude-only response (and is described by the definition above) and the phase of which describes the phase response of the system.

Note 6 to entry: In general, the system sensitivity varies with acoustic frequency, but deconvolution can be difficult or impractical working in the time domain. M_{sys0} ([3.2.4.2](#)) is the measured sensitivity at a frequency above the hydrophone's low-frequency cutoff, but where the wavelength of sound is much longer than the physical dimensions of the system and/or calibration apparatus.

[SOURCE: ISO 18406:2017, 3.18, modified — Definition slightly rephrased.]

3.1.5.2**nominal system sensitivity** $M_{\text{sys}0}$ *system sensitivity* (3.1.5.1) at a nominal frequency

Note 1 to entry: Nominal system sensitivity is expressed in reciprocal pascal (Pa⁻¹).

3.1.5.3**system filter function** $W_{\text{sys}}(f)$ ratio of *system sensitivity* (3.1.5.1) to *nominal system sensitivity* (3.1.5.2)

Note 1 to entry: The *system sensitivity* (3.1.5.1) is equal to the product of the *system filter function* (3.1.5.3) and the *nominal system sensitivity* (3.1.5.2).

3.1.5.4**calibration factor**reciprocal of *nominal system sensitivity* (3.1.5.2)

Note 1 to entry: Calibration factor is expressed in pascals (Pa).

3.1.5.5**system frequency range**usable frequency range of the underwater acoustic *measuring system* (3.1.1.6)

Note 1 to entry: The lowest usable frequency of the *measuring system* (3.1.1.6) often depends on a high-pass filter built into the hydrophone preamplifier to reduce non-acoustic flow and movement noise. The highest usable frequency is generally dependent on anti-aliasing filters and the *Nyquist frequency* (3.1.4.2). The *Nyquist frequency* (3.1.4.2) depends on the *sampling rate* (3.1.4.1).

3.1.5.6**system non-acoustic self-noise output**system digital output in the absence of sound pressure at the *hydrophone* (3.1.1.1) input

Note 1 to entry: Adapted from: Ainslie et al. (2020)[8].

[SOURCE: Robinson and Wang (2021), Table 12[18]]

3.1.5.7**system mean-square non-acoustic self-noise output spectral density****system self-noise output spectral density**

distribution as a function of non-negative frequency of the mean-square system non-acoustic self-noise output per unit bandwidth of a sound having a continuous spectrum

Note 1 to entry: System mean-square non-acoustic self-noise output spectral density is expressed in reciprocal hertz (Hz⁻¹).

[SOURCE: Adapted from Ainslie et al. (2020)[8]]

3.1.5.8**input-referred system non-acoustic self-noise sound pressure spectral density****system self-noise sound pressure spectral density****system spectral self-noise**quotient of *system mean-square non-acoustic self-noise output spectral density* (3.1.5.7) and the squared system sensitivity

Note 1 to entry: Input-referred system non-acoustic self-noise sound pressure spectral density is expressed in pascal squared per hertz (Pa² Hz⁻¹).

[SOURCE: Ainslie et al. (2020)[8]]

3.1.5.9 system-filtered sound pressure

p_{sysw}

weighted sound pressure when the linear filter is a specified underwater acoustic *measuring system* (3.1.1.6)

Note 1 to entry: System-filtered sound pressure is expressed in pascals (Pa).

Note 2 to entry: See ISO 18405:2017, 3.7.1.1.

Note 3 to entry: The *measuring system* (3.1.1.6) might include a *hydrophone* (3.1.1.1), a *pre-amplifier* (3.1.1.2) and an *ADC* (3.1.1.5) (including *anti-alias filter* (3.1.1.4)).

Note 4 to entry: A filter removes from a signal some component or feature.

Note 5 to entry: The linear filter might be chosen to characterize a specified frequency-dependent transfer function of a mechanical or electrical system.

Note 6 to entry: The system-filtered sound pressure and derivatives are sometimes reported without further filtering.

[SOURCE: based on Ainslie et al. (2020)^[8]

3.1.5.10 zero-to-peak system-filtered sound pressure peak system-filtered sound pressure

$p_{\text{sysw},0\text{--pk}}$

$p_{\text{sysw},\text{pk}}$

greatest magnitude of the *system-filtered sound pressure* (3.1.5.9) during a specified time interval, for a specified frequency range

Note 1 to entry: Zero-to-peak system-filtered sound pressure is expressed in pascals (Pa).

Note 2 to entry: A zero-to-peak system-filtered sound pressure can arise from a positive or negative *system-filtered sound pressure* (3.1.5.9). For example, if the maximum and minimum sound pressures in a sound signal are +5 Pa and -10 Pa, respectively, the peak sound pressure is 10 Pa, determined by the minimum value, which is negative.

3.1.6 Levels and other logarithmic quantities

3.1.6.1 system maximum recording level

L_{max}

maximum sound pressure level that the *measuring system* (3.1.1.6) can measure with a specified amount of distortion.

Note 1 to entry: System maximum recording level is expressed in decibels (dB).

Note 2 to entry: The system maximum recording level depends on the *full-scale signal* (3.1.1.9), the *full-scale ADC output* (3.1.4.8) and the *full-scale input range* (3.1.4.12).

Note 3 to entry: The reference value of sound pressure is 1 µPa.

Note 4 to entry: Generally 1 dB of distortion is the maximum acceptable, so that L_{max} occurs at the maximum unsaturated voltage input (see 3.1.4.10).

3.1.6.2 free-field receive sensitivity level

L_M

twenty times the logarithm to the base 10 of the ratio of the modulus of the *free-field receive sensitivity* (3.1.2.1) M_f to a reference value of sensitivity, M_0 , in decibels

$$L_M = 20 \log_{10} \frac{|M_f|}{M_0} \text{ dB}$$

Note 1 to entry: Free-field receive sensitivity level is expressed in decibels (dB).

Note 2 to entry: The reference value of *free-field receive sensitivity* (3.1.2.1), M_0 , is 1 V/ μ Pa.

Note 3 to entry: Free-field receive sensitivity level is a property of a *hydrophone* (3.1.1.1).

[SOURCE: IEC 60565-1:2020, 3.5]

3.1.6.3

hydrophone spectral noise floor level

$L_{N,eq,f, hp}$

level of the hydrophone self-noise spectral density

Note 1 to entry: In formula form, $L_{N,eq,f, hp} = 10 \log_{10} \frac{(p_{N,eq}^2)_f}{p_0^2 / f_0}$ dB, where $(p_{N,eq}^2)_f$ is the hydrophone self-noise spectral density.

Note 2 to entry: Hydrophone spectral noise floor level is expressed in decibels (dB).

Note 3 to entry: The reference value of hydrophone self-noise spectral density is $p_0^2 / f_0 = 1 \mu\text{Pa}^2 / \text{Hz}$.

Note 4 to entry: The hydrophone spectral noise floor level contributes to the *system spectral noise floor level* (3.1.6.4).

Note 5 to entry: The dynamic range depends on the hydrophone spectral noise floor level.

[SOURCE: Ainslie et al. (2020)[8]]

3.1.6.4

system spectral noise floor level recorder spectral noise floor level

$L_{N,eq,f, sys}$

level of the system self-noise spectral density

Note 1 to entry: In formula form, $L_{N,eq,f, sys} = 10 \log_{10} \frac{(p_{N,eq}^2)_f}{p_0^2 / f_0}$ dB, where $(p_{N,eq}^2)_f$ is the system self-noise spectral density.

Note 2 to entry: System spectral noise floor level is expressed in decibels (dB).

Note 3 to entry: The reference value of system self-noise spectral density is $p_0^2 / f_0 = 1 \mu\text{Pa}^2 / \text{Hz}$.

Note 4 to entry: The system spectral noise floor level depends on the *hydrophone spectral noise floor level* (3.1.6.3).

[SOURCE: adapted from Ainslie et al. (2020)[8]]

3.1.6.5

ADC dynamic range

$\Delta L_{DR, ADC}$

quantity given by $\Delta L_{DR, ADC} = 10 \log_{10} \frac{\overline{v_{FS}^2}}{v_{N,eq,self}^2}$ dB, where $\overline{v_{FS}^2}$ is the mean-square voltage of a sinusoidal *full-scale signal* (3.1.1.9) and $v_{N,eq,self}$ is the equivalent root-mean-square ADC self-noise voltage

Note 1 to entry: ADC dynamic range is expressed in decibels (dB).

[SOURCE: Ainslie et al. (2020)[8]]

3.1.6.6 system dynamic range

$\Delta L_{DR,sys}$

quantity given by $\Delta L_{DR,sys} = 10 \log_{10} \frac{\overline{v_{FS}^2}}{v_{N,eq,self}^2 + v_{N,self}^2}$ dB, where $\overline{v_{FS}^2}$ is the mean-square voltage of a sinusoidal full-scale signal (3.1.1.9), $v_{N,eq,self}$, is the equivalent root-mean-square ADC self-noise voltage and $v_{N,self}$ is the non-acoustic self-noise voltage at the ADC input (3.1.1.7), for a specified measurement frequency range

Note 1 to entry: System dynamic range is expressed in decibels (dB).

Note 2 to entry: The system dynamic range depends on *system maximum recording level* (3.1.6.1) and *system spectral noise floor level* (3.1.6.4).

[SOURCE: adapted from Ainslie et al. (2020)[8]]

3.1.6.7 total harmonic distortion THD

quantity given by $\Delta L_{THD} = 10 \log_{10} \frac{v_2^2 + v_3^2 \dots + v_{n+1}^2}{v_1^2}$ dB, where v_1 is the root-mean-square voltage of a sinusoidal full-scale signal (3.1.1.9) and $v_2 \dots v_{n+1}$ are the root-mean-square ADC voltage of the first n harmonics

Note 1 to entry: Total harmonic distortion is expressed in decibels (dB).

Note 2 to entry: Compare IEEE (STD-1241-2010) (p 17)[4]: “For a pure sine-wave input of specified amplitude and frequency, the root-sum-of-squares (rss) of all the harmonic distortion components including their aliases in the spectral output of the analog-to-digital converter. Unless otherwise specified, THD is estimated by the rss of the second through the tenth harmonics, inclusive. THD is often expressed as a decibel ratio with respect to the root-mean-square amplitude of the output component at the input frequency.”

[SOURCE: Ainslie et al. (2020)[8]]

3.1.6.8 system-filtered sound pressure level

$L_{p,sysw}$

level of the mean-square system-filtered sound pressure (3.1.5.9)

Note 1 to entry: In formula form, $L_{p,sysw} = 10 \log_{10} \frac{\overline{p_{sysw}^2}}{p_0^2}$ dB, where $\overline{p_{sysw}^2}$ is the mean-square system-filtered sound pressure.

Note 2 to entry: System-filtered sound pressure level is expressed in decibels (dB).

Note 3 to entry: The reference value of mean-square system-filtered sound pressure is $p_0^2 / f_0 = 1 \mu\text{Pa}^2 / \text{Hz}$.

[SOURCE: Ainslie et al. (2020)[8]]

3.1.6.9 signal-to-noise level difference

ΔL_{SN}

quantity given by $\Delta L_{SN} = 10 \log_{10} R_{SN}$ dB, where R_{SN} is the signal-to-noise power ratio (3.1.3.10)

Note 1 to entry: Signal-to-noise level difference is expressed in decibels (dB).

3.1.6.10 signal-plus-noise-to-noise level difference

ΔL_{S+NN}

quantity ΔL_{S+NN} in the formula

$$\Delta L_{S+NN} = 10 \log_{10} R_{S+NN} \text{ dB}$$

where R_{S+NN} is the *signal-plus-noise-to-noise power ratio* (3.1.3.11)

Note 1 to entry: Signal-plus-noise-to-noise level difference is expressed in decibels (dB).

Note 2 to entry: When the *signal-to-noise level difference* (3.1.6.9) is greater than 20 dB, there is less than 1 dB difference between signal-plus-noise-to-noise level difference and signal-to-noise level difference. When the *signal-to-noise level difference* (3.1.6.9) is greater than 40 dB, there is less than 0,1 dB difference between signal-plus-noise-to-noise level difference and signal-to-noise level difference.

3.1.6.11 equivalent bandwidth self-noise pressure level

$L_{N,eq}$

quantity $L_{N,eq}$ given by $L_{N,eq} = 10 \log_{10} \frac{p_{N,eq}^2}{p_0^2}$ dB, where $p_{N,eq}$ is the *equivalent bandwidth self-noise pressure* (3.1.2.2) and p_0 is the reference sound pressure

Note 1 to entry: Equivalent bandwidth self-noise pressure level is expressed in decibels (dB).

Note 2 to entry: The reference sound pressure, p_0 , is 1 μPa .

3.2 Concepts and quantities: Data processing

3.2.1 Temporal windows

3.2.1.1 temporal observation window TOW

interval of time within which a quantity representing a single sample of a distribution is obtained by combining multiple sound pressure values

Note 1 to entry: Examples of quantity representing a single sample include root-mean-square sound pressure, peak sound pressure and sound pressure kurtosis.

Note 2 to entry: The duration of the temporal observation window is denoted t_{OW} .

[SOURCE: adapted from Ainslie et al. (2020)^[8]]

3.2.1.2 temporal analysis window TAW

interval of time during which statistics are calculated over multiple samples, with each sample being evaluated for a specified *temporal observation window* (3.2.1.1)

Note 1 to entry: An example is a 24 h distribution of root-mean-square sound pressure calculated using a *temporal observation window* (3.2.1.1) of 1 min. In this example, the temporal analysis window is 24 h and the temporal observation window is 1 min.

Note 2 to entry: The duration of the temporal analysis window is denoted t_{AW} .

Note 3 to entry: The duration of the temporal analysis window is greater than or equal to the *temporal observation window* (3.2.1.1).

[SOURCE: adapted from Ainslie et al. (2020)^[8]]

3.2.2 Discrete Fourier transform terms

3.2.2.1

Fourier series representation

representation of a periodic discrete time signal $x(n)$ having N samples of the form:

$$x(n) = \sum_{m=0}^{N-1} c_m e^{\frac{i2\pi nm}{N}}$$

where c_m are complex coefficients that define the amplitude and phase for each of the N complex exponential terms in the summand

Note 1 to entry: this form is restricted to periodic signals and to periodic approximations of aperiodic signals.

3.2.2.2

discrete Fourier transform

DFT

discrete transform in time and frequency, based on the discrete Fourier series, used to obtain a spectral estimation of N_{DFT} uniformly time-spaced samples of a signal observed over a finite duration

$$X(m) = \frac{1}{N_{\text{DFT}}} \sum_{n=0}^{N_{\text{DFT}}-1} x(n) e^{\frac{-i2\pi nm}{N_{\text{DFT}}}}$$

where $X(m)$ is equal to c_m in the *Fourier series representation* (3.2.2.1) of $x(n)$

Note 1 to entry: The fast Fourier transform is a computationally efficient way to calculate this sum, although commercial packages differ on how the sum is scaled.

Note 2 to entry: Further information on the discrete Fourier transform is provided in Informative [Annex B](#).

Note 3 to entry: The discrete Fourier transform does not alter the dimension of the operand. If the input $x(n)$ is in unit U, the output $X(m)$ is in the same unit, U.

Note 4 to entry: The discrete Fourier transform assumes a uniform *sampling rate* (3.1.4.1).

Note 5 to entry: The discrete Fourier transform is a computation that allows one to obtain the spectral representation of a signal sampled in time.

Note 6 to entry: See also Oppenheim et al (2001)^[26].

3.2.2.3

DFT data window

subset of time series data that is selected for data processing

Note 1 to entry: An example of data processing is a *DFT* (3.2.2.2).

Note 2 to entry: The data window excludes any zero padding.

3.2.2.4

DFT zero padding

zero padding

appending or prepending zeros to an existing series of values prior to carrying out a *DFT* (3.2.2.2)

Note 1 to entry: The purpose of zero padding is either to achieve a desired *DFT frequency step* (3.2.2.13), or to achieve a desired DFT length (e.g., a power of 2).

3.2.2.5**data window size** N_w

number of samples in a data window

Note 1 to entry: The data window size excludes any *zero padding* (3.2.2.4).**3.2.2.6****data window duration** t_w

duration of a data window

Note 1 to entry: Data window duration is expressed in seconds (s).

Note 2 to entry: The data window duration excludes the duration of any *zero padding* (3.2.2.4).Note 3 to entry: The data window duration is equal to the duration of the smallest possible *temporal observation window* (3.2.1.1).**3.2.2.7****DFT window**expanded data window comprising the data window and all appended or prepended zeros associated with *zero padding* (3.2.2.4)**3.2.2.8****DFT window size** N_{DFT} number of samples in a *DFT window* (3.2.2.7)Note 1 to entry: The DFT window size is greater than or equal to the *sampling rate* (3.1.4.1) times the duration of the *temporal observation window* (3.2.1.1): $N_{\text{DFT}} \geq f_s \cdot t_{\text{OW}}$.Note 2 to entry: The DFT window size includes any *zero padding* (3.2.2.4). The DFT window size is the data window size (3.2.2.5) plus the number of zeros associated with *zero padding* (3.2.2.4).**3.2.2.9****DFT window duration** Δt duration of a *DFT data window* (3.2.2.3)

Note 1 to entry: DFT window duration is expressed in seconds (s).

Note 2 to entry: The DFT window duration is equal to the sum of the *data window duration* (3.2.2.6) and the zero-padding duration.Note 3 to entry: The DFT window duration is equal to the product of the *DFT window size* (3.2.2.8) and the *data sampling interval* (3.2.2.12).Note 4 to entry: The DFT window duration can be chosen to achieve a desired *DFT frequency step* (3.2.2.13) δf using the formula $\Delta t = 1 / \delta f$.Note 5 to entry: The DFT window duration can be chosen to achieve a desired DFT size N_{DFT} (e.g. an integer power of 2) using the formula $\Delta t = 1 / (N_{\text{DFT}} \delta t)$, where δt is the sampling interval.**3.2.2.10****DFT window function** $w(n)$

pre-defined mathematical function that is multiplied with the data in a time window to improve some characteristics of the frequency description

Note 1 to entry: If a window function is used, the windowed DFT is computed as:

$$Y(m) = \frac{1}{N_{\text{DFT}}} \sum_{n=0}^{N_{\text{DFT}}-1} w(n)x(n)e^{\frac{-i2\pi nm}{N_{\text{DFT}}}}.$$

Note 2 to entry: DFT window functions are used to isolate a specific portion of a signal for analysis, with the goal of reducing spectral leakage by tapering the signal at the edges. *DFT windows* (3.2.2.7) have desirable properties that influence how energy is concentrated within the frequency bins^[14], and their application results in apodization (see ISO 19702). For example, the Hann window dampens the sides of the sliced time signal to minimise spectral leakage, which would otherwise introduce an error in the resulting spectrum. The selection of DFT window function depends on the characteristics of the signal being analysed and the analysis objectives, such as prioritising *frequency resolution* (3.2.2.13) or dynamic range, since different windows offer different trade-offs between these factors.

Note 3 to entry: If *zero padding* (3.2.2.4) is applied, the *DFT window duration* (3.2.2.9) exceeds the *data window duration* (3.2.2.6). In this case, the window function is multiplied with the data in the data window, not with all data in the *DFT window* (3.2.2.7).

Note 4 to entry: The DFT window function $w(n)$ is power normalized and is scaled by N_{DFT} / N_w , according to [Formula \(6\)](#).

Note 5 to entry: The DFT window function $w(n)$ is nonzero only for $0 \leq n \leq N_w \leq N_{\text{DFT}}$. See [Formula \(6\)](#).

Note 6 to entry: See [Annex B](#) for an example involving a Hann window.

[SOURCE: ISO 2041:2018, 3.5.18, modified — "DFT" and the symbol " $w(n)$ " were added in the term.]

3.2.2.11

DFT power spectrum

$Q_{\text{ms}}(m)$

mean-square power due to the windowed Fourier series component at frequency $f_m = m \cdot \delta f$ given by

$$Q_{\text{ms}}(m) = \begin{cases} Y(m)^2, & m=0 \vee m = \frac{N_{\text{DFT}}}{2} \\ 2 \cdot |Y(m)|^2, & 1 \leq m \leq \frac{N_{\text{DFT}}}{2} - 1 \end{cases}$$

where $Y(m)$ is the windowed DFT

3.2.2.12

data sampling interval

δt

time between two successive data samples

Note 1 to entry: In formula form, $\delta t = \Delta t / N_{\text{DFT}}$, or equivalently $\delta t = 1 / f_s$.

Note 2 to entry: Data sampling interval is expressed in seconds (s).

3.2.2.13

frequency resolution

DFT frequency step

δf

difference in frequency between successive Fourier coefficients

Note 1 to entry: In formula form, $\delta f = 1 / \Delta t$, where Δt is the DFT data duration.

Note 2 to entry: Frequency resolution is expressed in hertz (Hz).

3.2.2.14

DFT time advance

t_A

how far in time the data window is advanced between successive *DFTs* (3.2.2.2)

Note 1 to entry: DFT time advance is expressed in seconds (s).