Package ‘seewave’

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ZipData no

Description Functions for analysing, manipulating, displaying, editing and synthesizing time waves (particularly sound). This package processes time analysis (oscillograms and envelopes), spectral content, resonance quality factor, entropy, cross correlation and autocorrelation, zero-crossing, dominant frequency, analytic signal, frequency coherence, 2D and 3D spectrograms and many other analyses. See Sueur et al. (2008) <doi:10.1080/09524622.2008.9753600> and Sueur (2018) <doi:10.1007/978-3-319-77647-7>.

License GPL (>= 2)

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NeedsCompilation no

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R topics documented:

`ACI` *Acoustic Complexity Index*

Description

This function computes the Acoustic Complexity Index (ACI) as described in Pieretti *et al.* (2011)

Usage

```
ACI(wave, f, channel = 1, wl = 512, ovlp = 0,  wn = "hamming", flim = NULL, nbwindows = 1)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`wl`	window length for the analysis (even number of points) (by default = 512).
`ovlp`	overlap between two successive windows (in %).
`wn`	window name, see `ftwindow` (by default `"hanning"`).
`flim`	a numeric vector of length 2 to select a frequency band (in kHz).
`nbwindows`	a numeric vector of length 1 specifying the number of windows (by default 1, ie a single window including the complete `wave` object.

Details

The function computes first a short-term Fourier transform and then the ACI index.
The function returns only the ACI total, ACI tot in Pieretti *et al.* (2010).
See the references for details on computation.

Value

A vector of length 1 returning the ACI total.

Note

Values returned were checked with the results provided by the add-on Soundscapemeter for the software Wavesurfer https://www.speech.kth.se/wavesurfer/.

Author(s)

Laurent Lellouch, improved by Amandine Gasc and Morgane Papin

References

Pieretti N, Farina A, Morri FD (2011) A new methodology to infer the singing activity of an avian community: the Acoustic Complexity Index (ACI). *Ecological Indicators*, 11, 868-873.
Farina A, Pieretti N, Piccioli L (2011) The soundscape methodology for long-term bird monitoring: a Mediterranean Europe case-study. *Ecological Informatics*, 6, 354-363.

See Also

spectro, specflux

Examples

```
data(tico)
ACI(tico)
## dividing the sound sample into 4 windows of equal duration
ACI(tico, nbwindows=4)
## selection of a frequency band
ACI(tico, flim=c(2,6))
```

---

acoustat *Statistics on time and frequency STFT contours*

Description

This function returns statistics based on STFT time and frequency contours.

Usage

```
acoustat(wave, f, channel = 1, wl = 512, ovlp = 0, wn = "hanning",
tlim = NULL, flim = NULL,
aggregate = sum, fraction = 90,
plot = TRUE, type = "l", ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`wl`	window length for the analysis (even number of points) (by default = 512).
`ovlp`	overlap between two successive windows (in %).
`wn`	window name, see `ftwindow` (by default `"hanning"`).
`tlim`	modifications of the time limits of the analysis (in s).
`flim`	modifications of the frequency limits of the analysis (in kHz).
`aggregate`	a character vector of length 1 specifying the function to be applied on the rows (time) and columns (frequency) of the STFT matrix. By default set to `sum`.
`fraction`	a numeric vector of length 1, specifying a particular fraction of the contours amplitude to be captured by the initial and terminal percentile values (in %). See details.
`plot`	a logical, if `TRUE` a two-frame plot is returned with the time and frequency contours and percentiles displayed.
`type`	if `plot` is `TRUE`, type of plot that should be drawn. See `plot` for details (by default "l" for lines).
`...`	other `plot` graphical parameters.

Details

The principle of `acoustat` is as follows:

1. Compute the short-term Fourier transform (STFT) with usual parameters (`wl` for window length, `ovlp` for overlap of successive windows, and `wn` for the name of window shape).
2. This results in a time * frequency matrix.
3. Compute an aggregation function (specified with the argument `aggregate` set by default to `sum`) accross rows and columns of time * frequency matrix.)
4. This results in two components: (*i*) the time contour, and (*ii*) the frequency contour.
5. Each contour is considered as a probability mass function (PMF) and transformed into a cumulated distribution function (CDF).
6. Measures are extracted from each CDF: median (M), initial percentile (P1) value, terminal percentile (P2) value, interpercentile range (IPR). P1, P2 and IPR are defined using a fraction parameter (`fraction`) that sets the percent of the contour amplitude to be captured by the initial and terminal percentile values. A fraction of 50% would result in the familiar quartiles and interquartile range. An energy fraction of 80% would return the 10th and 90th percentile values, and the width of the range in between.

Value

The function returns a list with 10 items:

`time.contour`	the time contour as a two-column matrix, the first colum being time (s) and the second colum being the amplitude probability mass function (no scale).
`freq.contour`	the frequency contour as a two-column matrix, the first colum being frequency (kHz) and the second colum being the amplitude probability mass function (no scale).
`time.P1`	the time initial percentile
`time.M`	the time median
`time.P2`	the time terminal percentile
`time.IPR`	the time interpercentile range
`freq.P1`	the frequency initial percentile
`freq.M`	the frequency median
`freq.P2`	the frequency terminal percentile
`freq.IPR`	the frequency interpercentile range

Note

`acoustat` was originally developped in Matlab language by Kurt Fristrup and XXXX Watkins (1992) .
The R function was kindly checked by Kurt Fristrup.

Author(s)

Jerome Sueur

References

Fristrup, K. M. and Watkins, W. A. 1992. Characterizing acoustic features of marine animal sounds. Woods Hole Oceanographic Institution Technical Report WHOI-92-04.

See Also

`meanspec`, `specprop`

Examples

```
data(tico)
note <- cutw(tico, from=0.5, to=0.9, output="Wave")
## default setting
acoustat(note)
## change the percentile fraction
acoustat(note, fraction=50)
## change the STFT parameters
acoustat(note, wl=1024, ovlp=80)
## change the function to compute the aggregate contours
```

```
## standard deviation instead of sum
acoustat(note, aggregate=sd)
## direct time and frequency selection
acoustat(tico, tlim=c(0.5,0.9), flim=c(3,6))
## some useless graphical changes
acoustat(note, type="o", col="blue")
```

---

addsilw *Add or insert a silence section*

Description

Add or insert a silence section to a time wave.

Usage

```
addsilw(wave, f, channel = 1, at = "end", choose = FALSE, d = NULL,
plot = FALSE, output = "matrix", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
at	position where to add the silence section (in s). Can be also specified as "start", "middle" or "end".
choose	logical, if TRUE the point where silence will be added into wave2 (=at) can be graphically chosen with a cursor.
d	duration of the silence section to add (in s).
plot	logical, if TRUE returns an oscillographic plot of wave with the new silence section (by default TRUE).
output	character string, the class of the object to return, either "matrix", "Wave", "Sample", "audioSample" or "ts".
...	other oscillo graphical parameters.

Value

If plot is FALSE, a new wave is returned. The class of the returned object is set with the argument output.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

oscillo, cutw,deletew, fadew,pastew, mutew,revw, zapsilw `afilter` *Amplitude filter*

Description

This function deletes all signal which amplitude is below a selected threshold.

Usage

```
afilter(wave, f, channel = 1, threshold = 5, plot = TRUE,
listen = FALSE, output = "matrix", ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`threshold`	amplitude threshold (in %).
`plot`	logical, if `TRUE` plots the new oscillogram (by default `TRUE`).
`listen`	if `TRUE` the new sound is played back.
`output`	character string, the class of the object to return, either `"matrix"`, `"Wave"`, `"Sample"`, `"audioSample"` or `"ts"`.
`...`	other `oscillo` graphical parameters.

Details

The `threshold` value is in % relative to the maximal value of `wave`. Signal inferior to this value is clipped.

Value

If `plot` is `FALSE`, a new wave is returned. The class of the returned object is set with the argument `output`.

Note

This function is used as an argument (`threshold`) in the following functions: `autoc`, `csh`, `dfreq`, `timer` and `zc`.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

oscillo

Examples

```
data(orni)
op<-par(mfrow=c(2,1))
afilter(orni,f=22050)
title(main = "threshold level = 5")
afilter(orni,f=22050,threshold=0.5,colwave="blue")
title(main = "threshold level = 0.5")
par(op)
```

---

akamatsu — *Water tank minimum resonant and cutoff frequencies*

Description

This function computes the resonant and cutoff frequencies when recording in a given aquarium according to the criteria explained in Akamatsu et al. (2002)

Usage

```
akamatsu(Lx, Ly, Lz, mode = c(1,1,1),
         c = 148000,  plot = FALSE, xlab = "Frequency (kHz)",
         ylab = "Attenuation distance (cm)", ...)
```

Arguments

Lx	watertank length (in cm).
Ly	watertank width (in cm).
Lz	watertank height (in cm).
mode	mode, see details.
c	sound velocity in cm/s (by default 148000 cm/s in water).
plot	logical, if TRUE plots the attenuation distance in function of frequency.
xlab	title of the x axis if plot is TRUE.
ylab	title of the y axis if plot is TRUE.
...	other plot graphical parameters.

Details

From Akamatsu et al. (2002):

1. Resonant frequency

The calculated resonant frequencies of a rectangular glass tank with the dimension of Lx , Ly , and Lz (in centimeters) can be described by the following equation:

$$f_{lmn}^{rectangular} = \frac{c}{2} \times \sqrt{\left(\frac{l}{L_x}\right)^2 + \left(\frac{m}{L_y}\right)^2 + \left(\frac{n}{L_z}\right)^2}$$

where *c* is the sound velocity (cm/s) and each *l*, *m*, *n* reprents an integer, and the combination of these paramameters designates the 'mode number'. The mode (1, 1, 1) represents the resonance wave of minimum frequency. The mode (2, 1, 1) represents one of the higher order of resonant component and has additional node of the soundpressure level at the middle of the X axis, *i.e.*, Lx/2.

2. Cutoff frequency

The cutoff frequency can be calculated as follows:

$$f_{cutoff}^{rectangular} = \frac{c}{2} \times \sqrt{\left(\frac{1}{L_y}\right)^2 + \left(\frac{1}{L_z}\right)^2}$$

3. Attenuation distance

The theoretical attenuation distance *D* can be expressed in function of the cutoff frequency and the projected frequency following:

$$D^{rectangular}(f) = 2 \times log_{10} \times \frac{c}{4\pi f_{cutoff}^{rectangular}} \times \frac{1}{\sqrt{1 - \left(\frac{f}{f_{cutoff}^{rectangular}}\right)^2}}$$

Value

A list of two items:

`res`	Resonant frequency (in Hz). See `Details`
`cut`	Cut frequency (in Hz). See `Details`

Author(s)

Camille Desjonqueres

References

Akamatsu T, Okumura T, Novarini N, Yan HY (2002) Emprical refinements applicable to the recording of fish sounds in small tanks. *Journal of the Acoustical Society of America*, 112, 3073-3082.

Examples

```
akamatsu(60, 30, 40)
```

---

ama *Amplitude modulation analysis of a time wave*

Description

This function computes the Fourier analysis of a time wave envelope. This allows to detect periodicity, in particular those generated by amplitude modulations.

Usage

```
ama(wave, f, channel = 1, envt = "hil", wl = 512, plot = TRUE, type = "l", ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`envt`	the type of envelope to be used: either "abs" for absolute amplitude envelope or "hil" for Hilbert amplitude envelope.
`wl`	length of the window for the analysis (even number of points, by default = 512).
`plot`	logical, if `TRUE` the spectrum of the envelope (by default `TRUE`).
`type`	if `plot` is `TRUE`, type of plot that should be drawn. See `plot` for details (by default "l" for lines).
`...`	other `meanspec` parameters.

Details

This function is based on `env` and `meanspec`.
The envelope of `wave` is first computed and the spectrum of this envelope is then processed. All `env` and `meanspec` arguments can be set up. Be sure to set up `wl` large enough if you want to detect low amplitude modulation periodicity.

Value

If plot is FALSE, ama returns a numeric vector corresponding to the computed spectrum. If peaks is not NULL, ama returns a list with two elements:

spec	the spectrum computed
peaks	the peaks values (in kHz).

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

env, fma, meanspec

Examples

```
data(orni)
# detection of the main amplitude modulation in a cicada song:
# one with a 0.258 kHz frequency (due to pulses in the echemes)
# one with a 2.369 kHz frequency (fundamental frequency)
ama(orni,f=22050,wl=1024)
# these amplitude modulations can be identify with a cursor:
ama(orni,f=22050,wl=1024,identify=TRUE)
```

AR — *Acoustic Richness index*

Description

This function computes the Acoustic Richness index based on M and Ht indices

Usage

```
AR(..., datatype = "objects", envt = "hil",
msmooth = NULL, ksmooth = NULL, ssmooth = NULL,
pattern = "[wav]$|[WAV]$|[mp3]$")
```

Arguments

...	Wave, WaveMC, audioSample objects if datatype="objects", or a path as a character string to a directory including .wav and/or .mp3 files if datatype="files".
datatype	A character string to specify if inputs are either R objects (datatype="objects", default) or files (datatype="files").
envt	the type of envelope to be returned: either "abs" for absolute amplitude envelope or "hil" for Hilbert (default) amplitude envelope. See env.
msmooth	mean smooth. See env.

`ksmooth`	kernel smooth via kernel. See `env`.
`ssmooth`	sum smooth. See `env`.
`pattern`	an optional regular expression. Only file names which match the regular expression will be returned when `datatype="files"`. By default `.wav` or `.mp3` files. See `dir`.

Details

AR is ranked index based on the rank of the M and Ht indices obtained with the functions `M` and `th` respectively following:

$$AR = \frac{rank(M) \times rank(H_t)}{n^2}$$

with

$$0 \leq AR \leq 1$$

Value

A `data.frame` with three columns (M, Ht, AR) and n columns, with n the number of objects (respectively files) used as input.

Note

As a ranked index, the results returned by AR strongly depends with the set of objects (respectively files) used as input. Comparaison between different data sets may be spurious. Computing AR on a set of a single object does not make any sense but is allowed.

Author(s)

Jerome Sueur and Marion Depraetere

References

Depraetere M, Pavoine S, Jiguet F, Gasc A, Duvail S, Sueur J (2012) Monitoring animal diversity using acoustic indices: implementation in a temperate woodland. *Ecological Indicators*, **13**, 46-54.

See Also

`M`, `th`, `env`

Examples

```
## input as R objects
data(orni)
data(tico)
AR(orni, tico)
## give names to objects if you wish to have them as row names of the returned data.frame
AR(orni=orni, tico=tico)
## input as files stored in the working directory
## file names will be used as row names of the returned data.frame
## Not run:
```

```
require(tuneR)
AR(getwd(), datatype="files")

## End(Not run)
```

---

attenuation — *Generate sound intensity attenuation data*

Description

This function generates dB data following theoretical spherical attenuation of sound.

Usage

```
attenuation(lref, dref = 1, dstop, n, plot = TRUE,
xlab = "Distance (m)", ylab = "dB", type = "l", ...)
```

Arguments

`lref`	reference intensity or pressure level (in dB).
`dref`	reference distance corresponding to `lref` (in m.) (by default = 1).
`dstop`	maximal distance of propagation (in m.).
`n`	number of points generated between `dref` and `dstop`.
`plot`	logical, if `TRUE` plots attenuation against distance of propagation (by default `TRUE`).
`xlab`	title of the x axis.
`ylab`	title of the y axis.
`type`	if `plot` is `TRUE`, type of plot that should be drawn. See `plot` for details (by default "l" for lines).
`...`	other `plot` graphical parameters.

Value

If `plot` is `FALSE` return a numeric vector with the data generated.

Note

Sound attenuation in a free, unbounded medium behaves in accordance with the inverse square law. `attenuation` generates data following this rule from a reference point where sound intensity level (SIL) or sound pressure level (SPL) is known. Such theoretical data can be compared with experimental data collected in a real environment.

Author(s)

Jerome Sueur

References

Hartmann, W. M. 1998 *Signals, sound and sensation*. New York: Springer.

See Also

convSPL, moredB

Examples

```
# theoretical attenuation up to 150 m of a 100 dB/1m sound source
attenuation(lref=100, dref=1, dstop=150, n=200)
```

---

audiomoth *Reading and interpreting Audiomoth file name*

Description

This function reads and decomposes the files names generated by an Audiomoth device, audio digal recorders produced by the society Open Acoustic Devices.

Usage

```
audiomoth(x, tz = "")
```

Arguments

x	a character vector with .wav file names.
tz	a character vector defining a time zone specification. See as.POSIXct

Details

The digital recorder Audiomoth produced by Open Acoustic Devices (https://www.openacousticdevices.info/) generates .wav files which names contains information about the time of recording. The information is encoded in hexadecimal (e.g. "5E9089F0"). The function audiomoth decodes this information so that time of recording can be retrieved in numeric or time format.

Value

The function returns a data.frame with the following columns:

year	year of recording, numeric
month	month of recording, numeric
day	day of recording, numeric
hour	hour of recording, numeric
min	minute of recording, numeric
sec	second of recording, numeric
time	time in POSIX format

Note

For the time zone see the 607 time zone names stored in `OlsonNames`.
The file names of Audiomoth may change with time. There is no guarantee that the function will be updated on time.

Author(s)

Jerome Sueur

References

See Open Acoustic Devices website for details regarding the Audiomoth: https://www.openacousticdevices.info/.

See Also

`audiomoth.rename`, `as.POSIXct`, `OlsonNames`, `songmeter`

Examples

```
## recording done on Friday 10 April 2020 16:54:44 UTC
## computer time zone (local time, Europe, Paris for the test)
audiomoth("5E90A4D4.WAV")
## UTC
audiomoth("5E90A4D4.WAV", tz="UTC")
## GMT (= UTC as UTC and GMT are synonyms)
audiomoth("5E90A4D4.WAV", tz="GMT")
## UTC -2
audiomoth("5E90A4D4.WAV", tz="Etc/GMT-2")
## in Asia, Japan
audiomoth("5E90A4D4.WAV", tz="Japan")
## in South-America, Cayenne
audiomoth("5E90A4D4.WAV", tz="America/Cayenne")
## several files
filenames <- c("5E914ED0.WAV", "5E915128.WAV",
"5E915380.WAV", "5E9155D8.WAV", "5E915830.WAV",
"5E915A88.WAV", "5E915CE0.WAV", "5E915F38.WAV",
"5E916190.WAV", "5E9163E8.WAV")
audiomoth(filenames)
```

`audiomoth.rename` *Rename audiomoth files in a readable format*

Description

This function renames or copies files created with an Audiomoth device in a readable format including the data and time of recording.

Usage

```
audiomoth.rename(dir, overwrite = FALSE, tz = "", prefix = "")
```

Arguments

`dir`	a character vector, path to directory where the .WAV files are stored.
`overwrite`	a logical, to specify if the files should be renamed or copied, if `TRUE` the files are copied, if `FALSE` the files are renamed.
`tz`	a character vector defining a time zone specification. See `as.POSIXct`
`prefix`	a charcter vector for a prefix name to be added at the beginning of the file name.

Details

The format of the new file names follows the format of the SongMeter SM2/SM4 deveices: `PREFIX_YYYYMMDD_HHMMSS.wav`.

Value

1 logical vector indicating which operation succeeded for each of the files attempted.

Note

Be careful if you overwrite the files.

Author(s)

Jerome Sueur

See Also

`audiomoth`, `songmeter`

`autoc` *Short-term autocorrelation of a time wave*

Description

This function returns the fundamental frequency of a harmonic time wave. This is achieved by computing a correlation of the signal with itself after a time delay.

Usage

```
autoc(wave, f, channel = 1, wl = 512, fmin, fmax, threshold = NULL, plot = TRUE,
xlab = "Time (s)", ylab = "Frequency (kHz)", ylim = c(0, f/2000), pb =
FALSE, ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
wl	length of the window for the analysis (even number of points, by default = 512).
fmin	the minimum frequency to detect (in Hz). See details.
fmax	the maximum frequency to detect (in Hz). See details
threshold	amplitude threshold for signal detection (in %).
plot	logical, if TRUE plots the fundamental frequency against time (by default TRUE).
xlab	title of the x-axis.
ylab	title of the y-axis.
ylim	the range of y values.
pb	if TRUE returns a text progress bar in the console.
...	other plot graphical parameters.

Details

'fmin' and 'fmax' can help by reducing computing time but can also produce less accurate results.

Value

When plot is FALSE, autoc returns a two-column matrix, the first column corresponding to time in seconds (*x*-axis) and the second column corresponding to to fundamental frequency in kHz (*y*-axis). NA corresponds to pause sections in wave (see threshold).

Author(s)

Jerome Sueur <sueur@mnhn.fr> and Thierry Aubin <thierry.aubin@u-psud.fr>

References

Hopp, S. L., Owren, M. J. and Evans, C. S. (Eds) 1998. *Animal acoustic communication*. Springer, Berlin, Heidelberg.

See Also

ceps, acf

Examples

```
data(sheep)
# fundamental frequency of a sheep
res <- autoc(sheep, f=8000, threshold=5, fmin=100, fmax=700, plot=FALSE)
spectro(sheep, f=8000, ovlp=75, scale=FALSE)
points(res, pch=20)
legend(0.5, 3.6, "Fundamental frequency", pch=20, bty=0, cex=0.7)
```

beep — *Beep sound*

Description

Generate a simple beep to be used as an alert, for instance at the end of a loop of when ending up a long script.

Usage

```
beep(d = 0.5, f = 8000, cf = 1000)
```

Arguments

d	duration (in s)
f	sampling frequency (in Hz)
cf	carrier frequency (in Hz)

Value

Nothing returned, a pure tone sound is played back. The default duration is 0.5 s and the default frequency is 1000 Hz

Note

The function uses `listen` of `seewave` which calls `play` of `tuneR`. You might need to set up your sound player with `setWavPlayer` of `tuneR`.

Author(s)

Jerome Sueur

Examples

```
## Not run:
# default settings
beep()
# change the duration and the frequency
beep(d=1, cf=880)

## End(Not run)
```

bwfilter *Butterworth frequency filter*

Description

This function is a Butterworth frequency filter that filters out a selected frequency section of of a time wave (low-pass, high-pass, low-stop, high-stop, bandpass or bandstop frequency filter).

Usage

```
bwfilter(wave, f, channel = 1, n = 1, from = NULL, to = NULL,
bandpass = TRUE, listen = FALSE, output = "matrix")
```

Arguments

wave	an R object.
f	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
channel	channel of the R object, by default left channel (1).
n	Order of the filter. See details.
from	start frequency (in Hz) where to apply the filter.
to	end frequency (in Hz) where to apply the filter.
bandpass	if `TRUE` a band-pass filter is applied between `from` and `to`, if not `NULL` a band-stop filter is applied between `from` and `to` (by default `NULL`).
listen	if `TRUE` the new sound is played back.
output	character string, the class of the object to return, either `"matrix"`, `"Wave"`, `"Sample"`, `"audioSample"` or `"ts"`.

Details

The order of the filter determines the value of the roll-off value, that is the dB decrease per octave of the transfer function. A filter of order *n* will have a transfer function with a roll-off value of - *n*6* dB.

Value

A new wave is returned. The class of the returned object is set with the argument `output`.

Note

This function mainly uses the functions `filter()` and `filtfilt()` from the package `signal`

Author(s)

Jerome Sueur, functions `filter()` and `filtfilt()` from the package `signal`.

References

Stoddard, P. K. (1998). Application of filters in bioacoustics. *In*: Hopp, S. L., Owren, M. J. and Evans, C. S. (Eds), *Animal acoustic communication*. Springer, Berlin, Heidelberg,pp. 105-127.

See Also

`ffilter`, `bwfilter`, `preemphasis`, `lfs`, `afilter`

Examples

```
require(signal)
f <- 8000
a <- noisew(f=f, d=1)
## low-pass
# 1st order filter
res <- bwfilter(a, f=f, n=1, to=1500)
# 8th order filter
res <- bwfilter(a, f=f, n=8, to=1500)
## high-pass
res <- bwfilter(a, f=f, from=2500)
## band-pass
res <- bwfilter(a, f=f, from=1000, to=2000)
## band-stop
res <- bwfilter(a, f=f, from=1000, to=2000,bandpass=FALSE)
```

---

ccoh — *Continuous coherence function between two time waves*

Description

This function returns a two-dimension coherence representation between two time waves. The function corresponds to a sliding coherence function along the two signals. This produces a 2-D density plot. An amplitude contour plot can be overlaid.

Usage

```
ccoh(wave1, wave2, f, channel = c(1,1), wl = 512, ovlp = 0, plot = TRUE,
grid = TRUE, scale = TRUE, cont = FALSE,
collevels = seq(0, 1, 0.01), palette = reverse.heat.colors,
contlevels = seq(0, 1, 0.01), colcont = "black",
colbg="white", colgrid = "black",
colaxis = "black", collab="black",
xlab = "Time (s)", ylab = "Frequency (kHz)",
scalelab = "Coherence",
main = NULL,
scalefontlab = 1, scalecexlab =0.75, axisX = TRUE, axisY = TRUE,
flim = NULL, flimd = NULL,
...)
```

Arguments

wave1	a first R object
wave2	a second R object
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R objects, by default left channel (1) for each object.
wl	window length for the analysis (even number of points, by default = 512).
ovlp	overlap between two successive windows (in %).
plot	logical, if TRUE plots the continuous coherence function (by default TRUE).
grid	logical, if TRUE plots a y-axis grid (by default TRUE).
scale	logical, if TRUE plots a dB colour scale on the right side of the plot (by default TRUE).
cont	logical, if TRUE overplots contour lines on the plot (by default FALSE).
collevels	a set of levels which are used to partition the amplitude range of the coherence (should be between 0 and 1.
palette	a color palette function to be used to assign colors in the plot, see Details.
contlevels	a set of levels which are used to partition the amplitude range for contour overplot (in dB).
colcont	colour for cont plotting.
colbg	background colour.
colgrid	colour for grid plotting.
colaxis	color of the axes.
collab	color of the labels.
xlab	label of the time axis.
ylab	label of the frequency axis.
scalelab	label fo the amplitude scale.
main	label of the main title.
scalefontlab	font of the amplitude scale label.
scalecexlab	cex of the amplitude scale label.
axisX	logical, if TRUE plots time X-axis (by default TRUE).
axisY	logical, if TRUE plots frequency Y-axis (by default TRUE).
flim	modifications of the frequency Y-axis limits.
flimd	dynamic modifications of the frequency Y-axis limits. New wl and ovlp arguments are applied to increase time/frequency resolution.
...	other contour and oscillo graphical parameters.

Details

Coherence is a frequency domain function computed to show the degree of a relationship between two signals. The value of the coherence function ranges between zero and one, where a value of zero indicates there is no causal relationship between the signals. A value of one indicates the existence of linear frequency response between the two signals. This can be used, for instance, to compare the input and output signals of a system.
Any colour palette can be used. In particular, it is possible to use other palettes coming with **seewave**: `temp.colors`, `reverse.gray.colors.1`, `reverse.gray.colors.2`, `spectro.colors`, `reverse.terrain.colors`, `reverse.topo.colors`, `reverse.cm.colors` corresponding to the reverse of `terrain.colors`, `topo.colors`, `cm.colors`.
Use `locator` to identify points.

Value

This function returns a list of three items:

`time`	a numeric vector corresponding to the time axis.
`freq`	a numeric vector corresponding to the frequency axis.
`amp`	a numeric matrix corresponding to the coherence. Each column corresponds to a coherence function of length `wl`.

Note

This function is based on `spec.pgram`, `contour` and `filled.contour`. See `spectro` for graphical changes.

Author(s)

Jerome Sueur <`sueur@mnhn.fr`> but this function is mainly based on `spec.pgram` by Martyn Plummer, Adrian Trapletti and B.D. Ripley

See Also

`coh`, `spectro`, `spec.pgram`.

Examples

```
wave1<-synth(d=1,f=4000,cf=500)
wave2<-synth(d=1,f=4000,cf=800)
ccoh(wave1,wave2,f=4000)
```

ceps *Cepstrum or real cepstrum*

Description

This function returns the cepstrum of a time wave allowing fundamental frequency detection.

Usage

```
ceps(wave, f, channel = 1, phase = FALSE, wl = 512, at = NULL, from = NULL, to = NULL,
tidentify = FALSE, fidentify = FALSE, col = "black", cex = 1, plot = TRUE,
qlab = "Quefrency (bottom: s, up: Hz)", alab = "Amplitude",
qlim = NULL, alim = NULL, type = "l", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
phase	if TRUE than the phase is taken into account in the computation of the cepstrum.
wl	if at is not null, length of the window for the analysis (even number of points, by defaults = 512).
at	position where to compute the cepstrum (in s).
from	start position where to compute the cepstrum (in s).
to	end position to compute the cepstrum (in s).
tidentify	to identify time values on the plot with the help of a cursor.
fidentify	to identify frequency values on the plot with the help of a cursor.
col	colour of the cepstrum.
cex	pitch size of the cepstrum.
plot	logical, if TRUE plots the cepstrum.
qlab	title of the quefrency axis (in s).
alab	title of the amplitude axis.
qlim	range of quefrency axis.
alim	range of amplitude axis.
type	if plot is TRUE, type of plot that should be drawn. See plot for details (by default "l" for lines).
...	other plot graphical parameters.

Details

The cepstrum of a time wave is the inverse Fourier transform of the logarithm of the Fourier transform. The cepstrum of a wave s is then calculated as follows:

$$C(s) = Re[FFT^{-1}(\log{(|FFT(s)|)}]$$

The independent variable of a cepstral graph is called the quefrency. The quefrency is a measure of time, though not in the sense of a signal in the time domain. A correspondence with the frequency domain is obtained by simply computing the reverse of the temporal x coordinate. For instance if a peak appears at 0.005 s, this reveals a frequency peak at 200 Hz (=1/0.005). This explain the two scales plotted when `plot` is `TRUE`.
If `at`, `from` or `to` are `FALSE` then `ceps` computes the cepstrum of the whole signal.
When using `tidentify` or `tidentify`, press 'stop' tools bar button to return values in the console.

Value

When `plot` is `FALSE`, `ceps` returns the cesptral profile as a two-column matrix, the first column corresponding to quefrency (*x*-axis) and the second corresponding to amplitude (*y*-axis).

Warning

The argument `peaks` is no more available (version > 1.5.6). See the function `fpeaks` for peak(s) detection.

Note

Cepstral analysis is mainly used in speech processing. This analysis allows to extract the fundamental frequency, see the examples.
This function is based on `fft`.

Author(s)

Jerome Sueur `<sueur@mnhn.fr>`

References

Oppenheim, A.V. and Schafer, R.W. 2004. From frequency to quefrency: a history of the cepstrum. *Signal Processing Magazine IEEE*, 21: 95-106.

See Also

`cepstro`, `fund`, `autoc`

Examples

```
data(sheep)
par(mfrow=c(2,1))
# phase not taken into account
ceps(sheep,f=8000,at=0.4,wl=1024)
# phase taken into account
ceps(sheep,f=8000,at=0.4,wl=1024, phase=TRUE)
```

---

cepstro *2D-cepstrogram of a time wave*

Description

This function returns a two-dimension cepstrographic representation of a time wave. The function corresponds to a short-term cepstral transform. An amplitude contour plot can be overlaid.

Usage

```
cepstro(wave, f, channel = 1, wl = 512, ovlp = 0, plot = TRUE, grid = TRUE,
scale = TRUE, cont = FALSE, collevels = seq(0, 1, 0.01),
palette = reverse.heat.colors, contlevels = seq(0, 1, 0.01),
colcont = "black", colbg="white", colgrid = "black",
colaxis = "black", collab = "black",
xlab = "Time (s)", ylab = "Quefrency (ms)",
scalelab = "Amplitude", main = NULL, scalefontlab = 1, scalecexlab = 0.75,
axisX = TRUE, axisY = TRUE, tlim = NULL, qlim = NULL, ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`wl`	if `at` is not null, length of the window for the analysis (even number of points, by defaults = 512).
`ovlp`	overlap between two successive windows (in %).
`plot`	logical, if `TRUE` plots the cepstrogram (by default `TRUE`).
`grid`	logical, if `TRUE` plots a y-axis grid (by default `TRUE`).
`scale`	logical, if `TRUE` plots a dB colour scale on the right side of the cesptrogram (by default `TRUE`).
`cont`	logical, if `TRUE` overplots contour lines on the cepstrogram (by default `FALSE`).
`collevels`	a set of levels which are used to partition the amplitude range of the cepstrogram (in dB).
`palette`	a color palette function to be used to assign colors in the plot.

contlevels	a set of levels which are used to partition the amplitude range for contour overplot (in dB).
colcont	colour for cont plotting.
colbg	background colour.
colgrid	colour for grid plotting.
colaxis	color of the axes.
collab	color of the labels.
xlab	label of the time axis.
ylab	label of the quefrency axis.
main	label of the main title.
scalelab	amplitude scale label.
scalefontlab	font of the amplitude scale label.
scalecexlab	cex of the amplitude scale label.
axisX	if TRUE plots time X-axis (by default TRUE).
axisY	if TRUE plots frequency Y-axis (by default TRUE).
tlim	modifications of the time X-axis limits.
qlim	modifications of the quefrency Y-axis limits (in ms).
...	other contour graphical parameters.

Details

It is unfortunately not possible to turn the *y*-axis to a frequency scale.
See spectro for the use of the graphical arguments.

Value

This function returns a list of three items:

time	a numeric vector corresponding to the time axis.
freq	a numeric vector corresponding to the quefrency axis.
amp	a numeric matrix corresponding to the the successive cepstral profiles computed along time.

Note

This function is based on ceps.

Author(s)

Jerome Sueur <sueur@mnhn.fr>.

References

Oppenheim, A.V. and Schafer, R.W. 2004. From frequency to quefrency: a history of the cepstrum. *Signal Processing Magazine IEEE*, 21: 95-106.

See Also

ceps, fund, autoc

Examples

```
data(sheep)
sheepc <- cutw(sheep, f=8000, from = 0.19, to = 2.3)
cepstro(sheepc,f=8000)
```

---

coh — *Coherence between two time waves*

Description

This function returns the frequency coherence between two time waves.

Usage

```
coh(wave1, wave2, f, channel=c(1,1), plot =TRUE, xlab = "Frequency (kHz)",
ylab = "Coherence", xlim = c(0,f/2000), type = "l",...)
```

Arguments

`wave1`	a first R object.
`wave2`	a second R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R objects, by default left channel (1) for each object.
`plot`	logical, if `TRUE` plots the continuous coherence function (by default `TRUE`).
`xlab`	title of the frequency X-axis.
`ylab`	title of the coherence Y-axis.
`xlim`	range of frequency X-axis.
`type`	if `plot` is `TRUE`, type of plot that should be drawn. See `plot` for details (by default "l" for lines).
`...`	other `plot` graphical parameters.

Details

Coherence is a frequency domain function computed to show the degree of a relationship between two signals. The value of the coherence function ranges between zero and one, where a value of zero indicates there is no causal relationship between the signals. A value of one indicates the existence of linear frequency response between the two signals. This can be used, for instance, to compare the input and output signals of a system.

Value

When `plot` is `FALSE`, this `coh` returns a two-column matrix, the first column being the frequency axis in kHz (*x*-axis) and the second column being the coherence (*y*-axis).

Note

This function is based on `spec.pgram`.

Author(s)

Jerome Sueur `<sueur@mnhn.fr>` but this function is based on `spec.pgram` by Martyn Plummer, Adrian Trapletti and B.D. Ripley.

See Also

`ccoh`, `spectro`, `spec.pgram`.

Examples

```
wave1<-synth(d=1,f=4000,cf=500)
wave2<-synth(d=1,f=4000,cf=800)
coh(wave1,wave2,f=4000)
```

---

combfilter *Comb filter*

Description

This function processes a feedforward comb filter and plots a spectrogram of the filtered wave associated with the frequency response of the filter.

Usage

```
combfilter(wave, f, channel = 1, alpha, K, units = c("samples", "seconds"),
plot = FALSE, output = "matrix", ...)
```

Arguments

`wave`	an R object
`f`	sampling frequency (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`alpha`	a numeric vector of length 1 for the scaling factor. See Details.
`K`	a numeric vector of lenght 1 for the delay length, in `units`. See Details.
`units`	units in which `K` is given, the default is `'samples'` but can be set to `'seconds'`.
`plot`	a logical, if `TRUE` plots the spectrogram of the filtered `wave` and the frequency response of the comb filter.

output	character string, the class of the object to return, either 'matrix', 'Wave', 'Sample', 'audioSample' or 'ts'.
...	other arguments to be passed to spectro except scale and osc that are set by default to FALSE.

Details

A comb filter consists in adding a delayed version of a signal to itself resulting in constructive and destructive interference. The feedforward version of a comb filter can be written following:

$$y(n) = x(n) + \alpha \times x(n - K)$$

where alpha is the scaling factor and K the delay length. The frequency response of the filter is obtained with:

$$H(f) = \sqrt{(1 + \alpha^2) + 2 \times \cos(\omega K)}$$

The frequency response is periodic. The depth of the cycles is controlled with alpha and the number of cycles with K.

Value

A new wave is returned. The class of the returned object is set with the argument output.

Note

Setting K to high values may generate unwanted results.
The feedback form of the combfilter is not implemented yet.

Author(s)

Jerome Sueur

See Also

combfilter, fir, squarefilter, drawfilter, ffilter, bwfilter

Examples

```
## Not run:
f <- 44100
## chirp
s1 <- synth(f=f, cf=1, d=2, fm=c(0,0,f/2,0,0), out="Wave")
combfilter(s1, alpha=1, K=50, plot=TRUE)
## harmonic sound
s2 <- synth(f=f, d=2, cf=600, harmonics=rep(1, 35), output="Wave")
combfilter(s2, alpha=1, K=10, plot=TRUE)
## noise, units in seconds
s3 <- noisew(d=2, f=44100, out="Wave")
combfilter(s3, alpha=0.5, K=1e-4, units="seconds", plot=TRUE)

## End(Not run)
```

convSPL *Convert sound pressure level in other units*

Description

This function converts sound pressure level (in dB) in sound power (Watt), intensity (Watt/m2) and pressure (Pa). By default, these conversions are applied to air-borne sound.

Usage

```
convSPL(x, d = 1, Iref = 10^-12, pref = 2*10^-5)
```

Arguments

`x`	a numeric vector or a matrix describind SPL values (in dB).
`d`	the distance from the sound source where SPL values have been measured (in meter) (by default = 1m)
`Iref`	reference intensity (in Watt/m2) (by default = 10^-12)
`pref`	reference pressure (in Pa) (by default = 2*10^-5)

Value

`convSPL` returns a list containing three components:

`P`	data converted in sound power (in Watt).
`I`	data converted in sound intensity (in Watt/m2).
`p`	data converted in sound pressure (in Pa).

Note

`Iref` and `pref` correspond to a 1 kHz sound in air.

Author(s)

Jerome Sueur <`sueur@mnhn.fr`>

References

Hartmann, W. M. 1998 *Signals, sound and sensation*. New York: Springer.

See Also

`moredB`, `dBweight`, `attenuation`

Examples

```
# conversion of two SPL measurements taken at 0.5 m from the source
convSPL(c(80,85),d=0.5)
```

corenv *Cross-correlation between two time wave envelopes*

Description

This function tests the similarity between two time wave envelopes by returning their maximal correlation and the time shift related to it.

Usage

```
corenv(wave1, wave2, f, channel=c(1,1), envt="hil", msmooth = NULL, ksmooth = NULL,
ssmooth = NULL, plot = TRUE, plotval = TRUE,
method = "spearman", col = "black", colval = "red",
cexval = 1, fontval = 1, xlab = "Time (s)",
ylab = "Coefficient of correlation (r)", type = "l", pb = FALSE, ...)
```

Arguments

wave1	a first R object.
wave2	a second R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R objects, by default left channel (1) for each object.
envt	the type of envelope to be used: either "abs" for absolute amplitude envelope or "hil" for Hilbert amplitude envelope. See env.
msmooth	a vector of length 2 to smooth the amplitude envelope with a mean sliding window. The first component is the window length (in number of points). The second component is the overlap between successive windows (in %). See env.
ksmooth	kernel smooth via kernel. See env.
ssmooth	sum smooth. See env.
plot	logical, if TRUE plots r values against frequency shift (by default TRUE).
plotval	logical, if TRUE adds to the plot maximum r value and frequency offset (by default TRUE).
method	a character string indicating which correlation coefficient is to be computed ("pearson", "spearman", or "kendall") (see cor).
col	colour of r values.
colval	colour of r max and frequency offset values.
cexval	character size of r max and frequency offset values.
fontval	font of r max and frequency offset values.
xlab	title of the frequency axis.
ylab	title of the r axis.

type	if plot is TRUE, type of plot that should be drawn. See plot for details (by default "l" for lines).
pb	if TRUE returns a text progress bar in the console.
...	other plot graphical parameters.

Details

Successive correlations between the envelopes of wave1 and wave2 are computed when regularly sliding forward and backward wave2 along wave1.
The maximal correlation is obtained at a particular shift (time offset). This shift may be positive or negative.
The higher smooth is set up, the faster will be the computation but less precise the results will be.
The corresponding p value, obtained with cor.test, is plotted. Inverting wave1 and wave2 may give slight different results.

Value

If plot is FALSE, corenv returns a list containing four components:

r	a two-column matrix, the first colum corresponding to the time shift (frequency x-axis) and the second column corresponding to the successive r correlation values between env1 and env2 (correlation y-axis).
rmax	the maximum correlation value between x and y.
p	the p value corresponding to rmax.
t	the time offset corresponding to rmax.

Author(s)

Jerome Sueur

See Also

env,corspec,covspectro, cor,cor.test.

Examples

```
## Not run:
data(orni)
# cross-correlation between two echemes of a cicada song
wave1<-cutw(orni,f=22050,from=0.3,to=0.4,plot=FALSE)
wave2<-cutw(orni,f=22050,from=0.58,to=0.68,plot=FALSE)
corenv(wave1,wave2,f=22050)

## End(Not run)
```

corspec *Cross-correlation between two frequency spectra*

Description

This function tests the similarity between two frequency spectra by returning their maximal correlation and the frequency shift related to it.

Usage

```
corspec(spec1, spec2, f = NULL, mel = FALSE, plot = TRUE, plotval = TRUE,
method = "spearman", col = "black", colval = "red",
cexval = 1, fontval = 1, xlab = NULL,
ylab = "Coefficient of correlation (r)", type="l",...)
```

Arguments

`spec1`	a first data set resulting of a spectral analysis obtained with `spec` or `meanspec` (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
`spec2`	a first data set resulting of a spectral analysis obtained with `spec` or `meanspec` (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
`f`	sampling frequency of waves used to obtain `spec1` and `spec2` (in Hz). Not necessary if `spec1` and/or `spec2` is a two columns matrix obtained with `spec` or `meanspec`.
`mel`	a logical, if `TRUE` the (htk-)mel scale is used.
`plot`	logical, if `TRUE` plots r values against frequency shift (by default `TRUE`).
`plotval`	logical, if `TRUE` adds to the plot maximum r value and frequency offset (by default `TRUE`).
`method`	a character string indicating which correlation coefficient is to be computed ("pearson", "spearman", or "kendall") (see `cor`).
`col`	colour of r values.
`colval`	colour of r max and frequency offset values.
`cexval`	character size of r max and frequency offset values.
`fontval`	font of r max and frequency offset values.
`xlab`	title of the frequency axis.
`ylab`	title of the r axis.
`type`	if `plot` is `TRUE`, type of plot that should be drawn. See `plot` for details (by default "l" for lines).
`...`	other `plot` graphical parameters.

Details

It is important not to have data in dB.
Successive correlations between spec1 and spec2 are computed when regularly shifting spec2 towards lower or higher frequencies.
The maximal correlation is obtained at a particular shift (frequency offset). This shift may be positive or negative.
The corresponding p value, obtained with cor.test, is plotted.
Inverting spec1 and spec2 may give slight different results, see examples.

Value

If plot is FALSE, corspec returns a list containing four components:

r	a two-column matrix, the first colum corresponding to the frequency shift (frequency x-axis) and the second column corresponding to the successive r correlation values between spec1 and spec2 (correlation y-axis).
rmax	the maximum correlation value between spec1 and spec2.
p	the p value corresponding to rmax.
f	the frequency offset corresponding to rmax.

Author(s)

Jerome Sueur, improved by Laurent Lellouch

References

Hopp, S. L., Owren, M. J. and Evans, C. S. (Eds) 1998. *Animal acoustic communication*. Springer, Berlin, Heidelberg.

See Also

spec, meanspec, corspec, covspectro, cor, cor.test.

Examples

```
## Not run: data(tico)
## compare the two first notes spectra
a<-spec(tico,f=22050,wl=512,at=0.2,plot=FALSE)
c<-spec(tico,f=22050,wl=512,at=1.1,plot=FALSE)
op<-par(mfrow=c(2,1), mar=c(4.5,4,3,1))
spec(tico,f=22050,at=0.2,col="blue")
par(new=TRUE)
spec(tico,f=22050,at=1.1,col="green")
legend(x=8,y=0.5,c("Note A", "Note C"),lty=1,col=c("blue","green"),bty="o")
par(mar=c(5,4,2,1))
corspec(a,c, ylim=c(-0.25,0.8),xaxs="i",yaxs="i",las=1)
par(op)
## different correlation methods give different results...
op<-par(mfrow=c(3,1))
corspec(a,c,xaxs="i",las=1, ylim=c(-0.25,0.8))
```

```
title("spearmann correlation (by default)")
corspec(a,c,xaxs="i",las=1,ylim=c(0,1),method="pearson")
title("pearson correlation")
corspec(a,c,xaxs="i",las=1,ylim=c(-0.23,0.5),method="kendall")
title("kendall correlation")
par(op)
## inverting x and y does not give exactly similar results
op<-par(mfrow=c(2,1),mar=c(2,4,3,1))
corspec(a,c)
corspec(c,a)
par(op)
## mel scale
require(tuneR)
data(orni)
orni.mel <- melfcc(orni, nbands = 256, dcttype = "t3", fbtype = "htkmel", spec_out=TRUE)
orni.mel.mean <- apply(orni.mel$aspectrum, MARGIN=2, FUN=mean)
tico.mel <- melfcc(tico, nbands = 256, dcttype = "t3", fbtype = "htkmel", spec_out=TRUE)
tico.mel.mean <- apply(tico.mel$aspectrum, MARGIN=2, FUN=mean)
corspec(orni.mel.mean, tico.mel.mean, f=22050, mel=TRUE, plot=TRUE)

## End(Not run)
```

`covspectro` *Covariance between two spectrograms*

Description

This function tests the similarity between two spectrograms by returning their maximal covariance and the time shift related to it.

Usage

```
covspectro(wave1, wave2, f, channel = c(1,1), wl = 512, wn = "hanning", n,
plot = TRUE, plotval = TRUE,
method = "spearman", col = "black", colval = "red", cexval = 1,
fontval = 1, xlab = "Time (s)",
ylab = "Normalised covariance (cov)", type = "l", pb = FALSE, ...)
```

Arguments

`wave1`	a first R object.
`wave2`	a second R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R objects, by default left channel (1) for each object.
`wl`	length of the window for the analysis (even number of points, by default = 512).
`wn`	window name, see `ftwindow` (by default `"hanning"`).

n	number of covariances computed between wave1 and wave2 when sliding wave2 along wave1.
plot	logical, if TRUE plots r values against frequency shift (by default TRUE).
plotval	logical, if TRUE adds to the plot maximum R value and frequency offset (by default TRUE).
method	a character string indicating which correlation coefficient is to be computed ("pearson", "spearman", or "kendall") (see cor).
col	colour of r values.
colval	colour of r max and frequency offset values.
cexval	character size of r max and frequency offset values.
fontval	font of r max and frequency offset values.
xlab	title of the frequency axis.
ylab	title of the r axis.
type	if plot is TRUE, type of plot that should be drawn. See plot for details (by default "l" for lines).
pb	if TRUE returns a text progress bar in the console.
...	other plot graphical parameters.

Details

Successive covariances between the spectrogram of wave1 and the spectrogram of wave2 are computed when regularly sliding forward and backward wave2 along wave1.
The maximal covariance is obtained at a particular shift (time offset). This shift may be positive or negative.
n sets in how many steps wave2 will be slided along wave1. Time process can be then decreased by setting low n value.
Inverting wave1 and wave2 may give slight different results.

Value

If plot is FALSE, covspectro returns a list containing three components:

cov	the successive covariance values between wave1 and wave2.
covmax	the maximum covariance between wave1 and wave2.
t	the time offset corresponding to cov.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

References

Hopp, S. L., Owren, M. J. and Evans, C. S. (Eds) 1998. *Animal acoustic communication*. Springer, Berlin, Heidelberg.

See Also

corspec, corenv, spectro, cor,

Examples

```
# covariance between two notes of a birdsong
## Not run:
data(tico)
note1<-cutw(tico, f=22050, from=0.5, to=0.9)
note2<-cutw(tico, f=22050, from=0.9, to=1.3)
covspectro(note1,note2,f=22050,n=37)

## End(Not run)
```

---

crest *Crest factor and visualization*

Description

This function returns the crest factor and localizes the different crest(s).

Usage

```
crest(wave, f, channel = 1, plot = FALSE, col = 2, cex = 3, symbol = "*", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
plot	if TRUE plots the oscillogram of wave and indicates the location of the crest(s)
col	color of the symbol indicating the localisation of the crest(s)
cex	symbol magnification
symbol	symbol indicating the localisation of the crest(s)
...	other

Details

The crest factor of a time series s is calculated according to:

$$C = \frac{max(s)}{rms(s)}$$

with rms the root-mean-square (see rms).

Value

The function returns a list of three items

`C`	crest factor
`val`	value of the crest(s)
`loc`	location of the crest(s)

Note

There might be several crests (maxima) along the time wave but there is a single crest factor.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

References

Hartmann, W. M. 1998 *Signals, sound and sensation*. New York: Springer.

See Also

`oscillo`, `rms`

Examples

```
data(tico)
crest(tico, f=22050)
# see the crest location and change the default graphical parameters
crest(tico, f=22050, plot=TRUE, sym="-")
```

`csh` *Continuous spectral entropy*

Description

This function computes the continuous spectral entropy (H) of a time wave.

Usage

```
csh(wave, f, channel = 1, wl = 512, wn = "hanning", ovlp = 0,
fftw = FALSE, threshold = NULL,
plot = TRUE, xlab = "Times (s)", ylab = "Spectral Entropy",
ylim = c(0, 1.1), type = "l", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
wl	if at is not null, length of the window for the analysis (even number of points, by default = 512).
wn	window name, see ftwindow (by default "hanning").
ovlp	overlap between two successive windows (in %).
fftw	if TRUE calls the function FFT of the library fftw. See Notes of the spectro.
threshold	amplitude threshold for signal detection (in %).
plot	logical, if TRUE plots the spectral entropy against time (by default TRUE).
xlab	title of the x axis.
ylab	title of the y axis.
ylim	the range of y values.
type	if plot is TRUE, type of plot that should be drawn. See plot for details (by default "l" for lines).
...	other plot graphical parameters.

Details

See sh for computing method.

Value

When plot is FALSE, csh returns a two-column matrix, the first column being time in seconds (*x*-axis) and the second column being the spectral entropy (*y*-axis) computed along time.
NA corresponds to pause sections in wave (see threshold).

Note

The spectral entropy of a noisy signal will tend towards 1 whereas the spectral entropy of a pure tone signal will tend towards 0.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

References

Toh, A. M., Togneri, R. & Nordholm, S. 2005 Spectral entropy as speech features for speech recognition. *Proceedings of PEECS*, pp. 60-65.

See Also

sh, th

Examples

```
data(orni)
csh(orni,f=22050,wl=512,ovlp=50)
# using the threshold argument can lead to some edge effets
# here sh=1 at the end of echemes
csh(orni,f=22050,wl=512,ovlp=50,threshold=5)
```

---

cutspec *Cut a frequency spectrum*

Description

This function can be used to select (cut) a specific part of a frequency spectrum.

Usage

```
cutspec(spec, f = NULL, flim, mel = FALSE, norm = FALSE, PMF = FALSE)
```

Arguments

`spec`	a vector or a two-column matrix set resulting of a spectral analysis. This can be the value obtained with `spec` or `meanspec`.
`f`	sampling frequency of `spec` (in Hz).
`flim`	a vector of length 2 to specify the new frequency range (in kHz).
`mel`	a logical, if `TRUE` the (htk-)mel scale is used.
`norm`	a logical, if `TRUE` the spectrum returned is normalised between 0 and 1.
`PMF`	a logical, if `TRUE` the spectrum returned is a probability mass function.

Value

A new spectrum is returned. The class of the returned object is the one of the input object (`spec`)

Note

The sampling frequency `f` is not necessary if `spec` has been obtained with either `spec` or `meanspec`. This function can be used before calling analysis function like `sh` or `sfm`. See examples.

Author(s)

Jerome Sueur, improved by Laurent Lellouch

See Also

`spec`, `meanspec`

Examples

```
data(orni)
a <- meanspec(orni,f=22050,plot=FALSE)
b <- cutspec(a,flim=c(4,8))
## quick check with a plot
plot(b,type="l")
## effects on spectral entropy
sfm(a)
sfm(b)
## mel scale
require(tuneR)
mel <- melfcc(orni, nbands = 256, dcttype = "t3", fbtype = "htkmel", spec_out=TRUE)
melspec.mean <- apply(mel$aspectrum, MARGIN=2, FUN=mean)
c <- cutspec(melspec.mean, f=22050, flim=c(4000,8000), mel=TRUE)
```

---

cutw *Cut a section of a time wave*

Description

This function selects and cuts a section of data describing a time wave. Original and cut sections can be plotted as oscillograms for comparison.

Usage

```
cutw(wave, f, channel=1, from = NULL, to = NULL, choose = FALSE,
plot = FALSE, marks = TRUE, output="matrix", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
from	start mark (in s).
to	end mark (in s).
choose	logical, if TRUE start (=from) and end (=to) points can be graphically chosen with a cursor on the oscillogram.
plot	logical, if TRUE returns an oscillographic plot of original and cut sections (by default FALSE).
marks	logical, if TRUE shows the start and end mark on the plot (by default TRUE).
output	character string, the class of the object to return, either "matrix", "Wave", "Sample", "audioSample" or "ts".
...	other oscillo graphical parameters.

Details

If `plot` is `TRUE` returns a two-frame plot with both original and cut sections.

Value

If `plot` is `FALSE`, a new wave is returned. The class of the returned object is set with the argument `output`.

Author(s)

Jerome Sueur

See Also

`oscillo`, `addsilw`,`deletew`, `fadew`,`mutew`,`pastew`,`revw`, `zapsilw`

Examples

```
# a 0.4 s section in a bird song
data(tico)
a<-cutw(tico,f=22050,from=0.5,to=0.9)
oscillo(a,22050)
# a direct way to see what has been cut
cutw(tico,f=22050,from=0.5,to=0.9,plot=TRUE)
```

---

dBscale *dB colour scale for a spectrogram display*

Description

This function displays a vertical or horizontal dB colour scale to be used with `spectro` plots.

Usage

```
dBscale(collevels, palette = spectro.colors, side = 4,
textlab = "Amplitude\n(dB)", cexlab = 0.75,
fontlab = 1, collab = "black", colaxis = "black",...)
```

Arguments

`collevels`	a set of levels which are used to partition the amplitude range of the spectrogram (in dB).
`palette`	a color palette function to be used to assign colors in the plot, see `note`.
`side`	side of the axis.
`textlab`	text of the label.
`cexlab`	character size of the label.

`fontlab` font of the label.

`collab` colour of the label.

`colaxis` colour of the axis.

`...` other `axis` arguments.

Note

This function, based on `filled.contour` by Ross Ihaka, is not supposed to be used by itself but as a legend of `spectro`.
Any colour palette can be used. In particular, it is possible to use other palettes coming with **seewave**: `rev.gray.colors.1`, `rev.gray.colors.2`, `rev.heat.colors`, `rev.terrain.colors`, `rev.topo.colors`, `rev.cm.colors` corresponding to the reverse of `heat.colors`, `terrain.colors`, `topo.colors`, `cm.colors`.

Author(s)

Jerome Sueur <sueur@mnhn.fr> and Caroline Simonis <csimonis@mnhn.fr>.

See Also

`spectro`.

Examples

```
data(pellucens)
# place the scale on the left and not on the right as spectro() does
def.par <- par(no.readonly = TRUE)
layout(matrix(c(1, 2), nc = 2), widths = c(1, 5))
par(mar=c(5,3,4,2))
dBscale(collevels=seq(-30,0,1),side=2)
par(mar=c(5,4,4,2))
spectro(pellucens, f=22050,wl=512,scale=FALSE)
par(def.par)
# place the scale on the top and not on the right as spectro() does
def.par <- par(no.readonly = TRUE)
layout(matrix(c(0,1,2,2), nc = 2, byrow=TRUE),widths=c(1,2),heights=(c(1,5.5)))
par(mar=c(0.5,3,4,2))
dBscale(collevels=seq(-30,0,1), textlab = "",side=3)
mtext("Amplitude (dB)",side=2,line = 1,at=0.6,cex=0.75)
par(mar=c(5,4,0.5,2))
spectro(pellucens, f=22050,wl=512,scale=FALSE)
par(def.par)
```

dBweight *dB weightings*

Description

This function returns the four most common dB weightings.

Usage

```
dBweight(f, dBref = NULL)
```

Arguments

`f`	frequency (in Hz).
`dBref`	dB reference level (by default `NULL`).

Details

By default, the function returns four weightings. When `dBref` is not `NULL` then the function returns the conversion from a dB reference level to four dB weighting levels.

Value

`dBweight` returns a list of five items corresponding to five dB weightings.

`A`	dB (A)
`B`	dB (B)
`C`	dB (C)
`D`	dB (D)
`ITU`	dB ITU-R 468

Note

The transfer equations used here come from Wipipedia but they were originally coming from the appendix of an international standard on the design performance of sound level meters IEC 651:1979 (Neil Glenister, pers. com.).

Author(s)

Jerome Sueur <sueur@mnhn.fr>, Zev Ross, and Andrey Anikin

References

https://en.wikipedia.org/wiki/A-weighting, https://en.wikipedia.org/wiki/ITU-R_468_noise_weighting

See Also

convSPL, moredB

Examples

```
# weight for a 50 Hz frequency
dBweight(f=50)
# A weight for the 1/3 Octave centre frequencies.
dBweight(f=c(20,25,31.5,40,50,63,80,100,125,160,200,250,
315,400,500,630,800,1000,1500,
1600,2000,2500,3150,4000,5000,
6300,8000,10000,12500,16000,20000))$A
# correction for a 50 Hz sound emitted at 100 dB
dBweight(f=50, dB=100)
# weighting curves plot
f <- seq(10,20000,by=10)
par(las=1)
plot(f, dBweight(f)$A, type="n", log="x",
xlim=c(10,10^5),ylim=c(-80,20),xlab="",ylab="",xaxt="n",yaxt="n")
abline(v=c(seq(10,100,by=10),seq(100,1000,by=100),
seq(1000,10000,by=1000),seq(10000,100000,by=10000),
c(100,1000,10000,100000)),col="lightgrey",lty=2)
abline(v=c(100,1000,10000,100000),col="grey")
abline(h=seq(-80, 20, 20),col="grey")
par(new=TRUE)
plot(f, dBweight(f)$A, type="l", log="x",
xlab="Frequency (Hz)", ylab="dB",lwd=2, col="blue", xlim=c(10,10^5),ylim=c(-80,20))
title(main="Acoustic weighting curves (10 Hz - 20 kHz)")
lines(x=f, y=dBweight(f)$B, col="green",lwd=2)
lines(x=f, y=dBweight(f)$C, col="red",lwd=2)
lines(x=f, y=dBweight(f)$D, col="black",lwd=2)
legend("bottomright",legend=c("dB(A)","dB(B)","dB(C)","dB(D)"),
lwd=2,col=c("blue","green","red","black"),bty="o",bg="white")
```

---

deletew *Delete a section of a time wave*

Description

This function selects and delete a section of data describing a time wave. Original section and section after deletion can be plotted as oscillograms for comparison.

Usage

```
deletew(wave, f, channel = 1, from = NULL, to = NULL, choose = FALSE, plot = FALSE,
marks = TRUE, output = "matrix", ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`from`	start position (in s).
`to`	end position (in s).
`choose`	logical, if `TRUE` start (=`from`) and end (=`to`) points can be graphically chosen with a cursor on the oscillogram.
`plot`	logical, if `TRUE` returns an oscillographic plot of original and cut sections (by default `FALSE`).
`marks`	logical, if `TRUE` shows the start and end mark on the plot (by default `TRUE`).
`output`	character string, the class of the object to return, either "matrix", "Wave", "Sample", "audioSample" or "ts".
`...`	other `oscillo` graphical parameters.

Details

If `plot` is `TRUE` returns a two-frame plot with both original and resulting sections.

Value

If `plot` is `FALSE`, a new wave is returned. The class of the returned object is set with the argument `output`.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

`oscillo`, `addsilw`,`cutw`, `fadew`, `mutew`, `pastew`, `revw`, `zapsilw`

Examples

```
# deletion a 0.4 s section in a bird song
data(tico)
a<-deletew(tico,f=22050,from=0.5,to=0.9)
oscillo(a,22050)
# a direct way to see what has been cut
deletew(tico,f=22050,from=0.5,to=0.9,plot=TRUE)
```

dfreq *Dominant frequency of a time wave*

Description

This function gives the dominant frequency (i. e. the frequency of highest amplitude) of a time wave.

Usage

```
dfreq(wave, f, channel = 1, wl = 512, wn = "hanning", ovlp = 0, fftw= FALSE, at =
NULL, tlim = NULL, threshold = NULL, bandpass = NULL, clip = NULL,
plot = TRUE, xlab = "Times (s)", ylab = "Frequency (kHz)",
ylim = c(0, f/2000), ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`wl`	length of the window for the analysis (even number of points, by default = 512).
`wn`	window name, see `ftwindow` (by default `"hanning"`).
`ovlp`	overlap between two successive analysis windows (in %).
`fftw`	if `TRUE` calls the function `FFT` of the library `fftw`. See Notes of the `spectro`.
`at`	time position where the dominant frequency has to be computed (in s.).
`tlim`	modifications of the time X-axis limits.
`threshold`	amplitude threshold for signal detection (in %).
`bandpass`	a numeric vector of length two, giving the lower and upper limits of a frequency bandpass filter (in Hz).
`clip`	a numeric value to select dominant frequency values according to their amplitude in reference to a maximal value of 1 for the whole signal (has to be >0 & < 1).
`plot`	logical, if `TRUE` plots the dominant frequency against time (by default `TRUE`).
`xlab`	title of the x axis.
`ylab`	title of the y axis.
`ylim`	the range of y values.
`...`	other `plot` graphical parameters.

Value

When `plot` is `FALSE`, `dfreq` returns a two-column matrix, the first column corresponding to time in seconds (*x*-axis) and the second column corresponding to to dominant frequency in kHz (*y*-axis). NA corresponds to pause sections in `wave` (see `threshold`).

Note

This function is based on fft.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

spec, meanspec,spectro.

Examples

```
data(tico)
f <- 22050
# default
dfreq(tico,f)
# using the amplitude threshold and changing the graphical output
dfreq(tico, f, ovlp=50,threshold=5, type="l", col=2)
# using 'at' argument for specific positions along the time axis
dfreq(tico, f, at=c(0.25, 0.75, 1.2, 1.6))
dfreq(tico, f, at=seq(0.5, 1.4, by=0.005), threshold=5)
# a specific number of measures on a single note
dfreq(tico, f, at=seq(0.5, 0.9, len=100), threshold=5, xlim=c(0.5,0.9))
# overlap on spectrogram
# and use of 'clip' argument to better track the dominant frequency
# in noisy conditions
op <- par()
ticon <- tico@left/max(tico@left) + noisew(d=length(tico@left)/f, f)
spectro(ticon, f)
res <- dfreq(ticon, f, clip=0.3, plot=FALSE)
points(res, col=2, pch =13)
par(op)
```

---

diffcumspec *Difference between two cumulative frequency spectra*

Description

This function compares two distributions (e.g. two frequency spectra) by computing the difference between two cumulative frequency spectra

Usage

```
diffcumspec(spec1, spec2, f = NULL, mel = FALSE,
plot = FALSE, type = "l", lty = c(1, 2), col = c(2, 4, 8),
flab = NULL, alab = "Cumulated amplitude",
flim = NULL, alim = NULL,
title = TRUE, legend = TRUE, ...)
```

Arguments

`spec1`	any distribution, especially a spectrum obtained with `spec` or `meanspec` (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
`spec2`	any distribution, especially a spectrum obtained with `spec` or `meanspec` (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
`f`	sampling frequency of waves used to obtain `spec1` and `spec2` (in Hz). Not necessary if `spec1` and/or `spec2` is a two columns matrix obtained with `spec` or `meanspec`.
`mel`	a logical, if `TRUE` the (htk-)mel scale is used.
`plot`	logical, if `TRUE` plots both cumulative spectra and their distance.
`type`	if `plot` is `TRUE`, type of plot that should be drawn. See `plot` for details (by default "l" for lines).
`col`	a vector of length 3 for the colour of `spec1`, `spec2`, and the difference between each of them.
`lty`	a vector of length 2 for the line type of `spec1` and `spec2` if `type`="l".
`flab`	title of the frequency axis.
`alab`	title of the amplitude axis.
`flim`	the range of frequency values.
`alim`	range of amplitude axis.
`title`	logical, if `TRUE`, adds a title with D and F values.
`legend`	logical, if `TRUE` adds a legend to the plot.
`...`	other `plot` graphical parameters.

Details

Both spectra are transformed into cumulative distribution functions (CDF).
Spectral difference is then computed according to:

$$D_{cf}(x,y) = \frac{\sum_{i=1}^{n} |X_i - Y_i|}{n}, with with X and Y the spectrum CDFs, and D \in [0,1].$$

Value

A numeric vector of length 1 returning the difference between the two spectra. No unit.

Note

This metric is sensitive not only to the spectral overlap between but also to the mean frequential distance between the different frequency peaks.

Author(s)

Laurent Lellouch, Jerome Sueur

References

Lellouch L, Pavoine S, Jiguet F, Glotin H, Sueur J (2014) Monitoring temporal change of bird communities with dissimilarity acoustic indices. *Methods in Ecology and Evolution*, in press.

See Also

kl.dist, ks.dist, simspec, diffspec, logspec.dist, itakura.dist

Examples

```
## Hz scale
data(tico)
data(orni)
orni.hz <- meanspec(orni, plot=FALSE)
tico.hz <- meanspec(tico, plot=FALSE)
diffcumspec(orni.hz, tico.hz, plot=TRUE)
## mel scale
require(tuneR)
orni.mel <- melfcc(orni, nbands = 256, dcttype = "t3", fbtype = "htkmel", spec_out=TRUE)
orni.mel.mean <- apply(orni.mel$aspectrum, MARGIN=2, FUN=mean)
tico.mel <- melfcc(tico, nbands = 256, dcttype = "t3", fbtype = "htkmel", spec_out=TRUE)
tico.mel.mean <- apply(tico.mel$aspectrum, MARGIN=2, FUN=mean)
diffcumspec(orni.mel.mean, tico.mel.mean, f=22050, mel=TRUE, plot=TRUE)
```

---

diffenv — *Difference between two amplitude envelopes*

Description

This function estimates the surface difference between two amplitude envelopes.

Usage

```
diffenv(wave1, wave2, f, channel = c(1,1), envt = "hil", msmooth = NULL, ksmooth = NULL,
plot = FALSE, lty1 = 1, lty2 = 2, col1 = 2, col2 = 4, cold = 8,
xlab = "Time (s)", ylab = "Amplitude", ylim = NULL, legend = TRUE, ...)
```

Arguments

`wave1`	a first R object.
`wave2`	a second R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R objects, by default left channel (1) for each object.
`envt`	the type of envelope to be used: either "abs" for absolute amplitude envelope or "hil" for Hilbert amplitude envelope. See env.

msmooth	a vector of length 2 to smooth the amplitude envelope with a mean sliding window. The first component is the window length (in number of points). The second component is the overlap between successive windows (in %). See env.
ksmooth	kernel smooth via kernel. See env.
plot	logical, if TRUE plots both envelopes and their surface difference (by default FALSE).
lty1	line type of the first envelope (envelope of wave1).
lty2	line type of the second envelope (envelope of wave2).
col1	colour of the first envelope (envelope of wave1).
col2	colour of the second envelope (envelope of wave2).
cold	colour of the surface difference.
xlab	title of the time axis.
ylab	title of the amplitude axis.
ylim	range of amplitude axis.
legend	logical, if TRUE adds a legend to the plot.
...	other plot graphical parameters.

Details

D is a Manhattan distance (l1 norm).
Envelopes of both waves are first transformed as probability mass functions (PMF).
Envelope difference is then computed according to:

$$D = \frac{\sum |env1 - env2|}{2}, withD \in [0, 1].$$

Value

The difference is returned. This value is without unit. When plot is TRUE, both envelopes and their difference surface are plotted on the same graph.

Note

This method can be used as a relative distance estimation between different envelopes.

Author(s)

Jerome Sueur <sueur@mnhn.fr>.

References

Sueur, J., Pavoine, S., Hamerlynck, O. & Duvail, S. (2008) - Rapid acoustic survey for biodiversity appraisal. *PLoS ONE*, 3(12): e4065.

See Also

env, corenv, diffspec, diffwave

Examples

```
data(tico) ; tico <- tico@left
data(orni) ; orni <- orni@left
# selection in tico of two waves with similar duration
tico2<-tico[1:length(orni)]
diffenv(tico2,orni,f=22050,plot=TRUE)
# smoothing the envelope gives a better graph but slightly changes the result
diffenv(tico2,orni,f=22050,msmooth=c(20,0),plot=TRUE)
```

---

diffspec — *Difference between two frequency spectra*

Description

This function estimates the surface difference between two frequency spectra.

Usage

```
diffspec(spec1, spec2, f = NULL, mel = FALSE,
plot = FALSE, type="l",
lty=c(1, 2), col =c(2, 4, 8),
flab = NULL, alab = "Amplitude",
flim = NULL, alim = NULL, title = TRUE, legend = TRUE, ...)
```

Arguments

`spec1`	a first data set resulting of a spectral analysis obtained with `spec` or `meanspec` (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
`spec2`	a first data set resulting of a spectral analysis obtained with `spec` or `meanspec` (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
`f`	sampling frequency of waves used to obtain `spec1` and `spec2` (in Hz). Not necessary if `spec1` and/or `spec2` is a two-column matrix obtained with `spec` or `meanspec`.
`mel`	a logical, if `TRUE` the (htk-)mel scale is used.
`plot`	logical, if `TRUE` plots both spectra and their surface difference (by default `FALSE`).
`type`	if `plot` is `TRUE`, type of plot that should be drawn. See `plot` for details (by default "l" for lines).
`lty`	a vector of length 2 for the line type of `spec1` and `spec2` if `type="l"`.
`col`	a vector of length 3 for the colour of `spec1`, `spec2`, and the surface difference between each of them.
`flab`	title of the frequency axis.
`alab`	title of the amplitude axis.

`flim`	the range of frequency values.
`alim`	range of amplitude axis.
`title`	logical, if `TRUE`, adds a title with D value.
`legend`	logical, if `TRUE` adds a legend to the plot.
`...`	other `plot` graphical parameters.

Details

D is a Manhattan distance (l1 norm).
Both spectra are first transformed as probability mass functions (PMF).
Spectral difference is then computed according to:

$$D = \frac{\sum |spec1 - spec2|}{2}, withD \in [0,1].$$

, with 0 < D < 1.

Value

The difference is returned. This value is without unit. When `plot` is `TRUE`, both spectra and their difference surface are plotted on the same graph.

Note

This method can be used as a relative distance estimation between different spectra.
The dB value obtained can be very different from the one visually estimated when looking at the graph (`plot=TRUE`).

Author(s)

Jerome Sueur, Sandrine Pavoine and Laurent Lellouch

References

Sueur, J., Pavoine, S., Hamerlynck, O. and Duvail, S. (2008). Rapid acoustic survey for biodiversity appraisal. *PLoS One*, 3(12): e4065.

See Also

`spec`, `meanspec`, `corspec`, `simspec`, `diffcumspec`, `diffenv`, `kl.dist`, `ks.dist`, `logspec.dist`, `itakura.dist`

Examples

```
a <- noisew(f=8000,d=1)
b <- synth(f=8000,d=1,cf=2000)
c <- synth(f=8000,d=1,cf=1000)
d <- noisew(f=8000,d=1)
speca <- spec(a,f=8000,wl=512,at=0.5,plot=FALSE)
specb <- spec(b,f=8000,wl=512,at=0.5,plot=FALSE)
```

```
specc <- spec(c,f=8000,wl=512,at=0.5,plot=FALSE)
specd <- spec(d,f=8000,wl=512,at=0.5,plot=FALSE)
diffspec(speca,speca,f=8000)
#[1] 0 => similar spectra of course !
diffspec(speca,specb)
diffspec(speca,specc,plot=TRUE)
diffspec(specb,specc,plot=TRUE)
diffspec(speca,specd,plot=TRUE)
## mel scale
require(tuneR)
data(orni)
data(tico)
orni.mel <- melfcc(orni, nbands = 256, dcttype = "t3", fbtype = "htkmel", spec_out=TRUE)
orni.mel.mean <- apply(orni.mel$aspectrum, MARGIN=2, FUN=mean)
tico.mel <- melfcc(tico, nbands = 256, dcttype = "t3", fbtype = "htkmel", spec_out=TRUE)
tico.mel.mean <- apply(tico.mel$aspectrum, MARGIN=2, FUN=mean)
diffspec(orni.mel.mean, tico.mel.mean, f=22050, mel=TRUE, plot=TRUE)
```

---

diffwave *Difference between two time waves*

Description

This function estimates the difference between two waves by computing the product between envelope surface difference and frequency surface difference.

Usage

```
diffwave(wave1, wave2, f, channel = c(1,1), wl = 512, envt = "hil",
msmooth = NULL, ksmooth = NULL)
```

Arguments

`wave1`	a first R object.
`wave2`	a second R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R objects, by default left channel (1) for each object.
`wl`	window length for spectral analysis (even number of points).
`envt`	the type of envelope to be used: either "abs" for absolute amplitude envelope or "hil" for Hilbert amplitude envelope. See `env`.
`msmooth`	a vector of length 2 to smooth the amplitude envelope with a mean sliding window. The first component is the window length (in number of points). The second component is the overlap between successive windows (in %). See `env`.
`ksmooth`	kernel smooth via `kernel`. See `env`.

Details

D is a Manhattan distance (l1 norm).
This function computes the product between the values obtained with `diffspec` and `diffenv` functions.
This then gives a global (time and frequency) estimation of dissimilarity.
The frequency mean spectrum and the amplitude envelope needed for computing respectively `diffspec` and `diffenv` are automatically generated. They can be controlled through `wl`, `msmooth` and `ksmooth` arguments respectively.
See examples below and examples in `diffspec` and `diffenv` for implications on the results.

Value

A single value varying between 0 and 1 is returned. The value has no unit.

Note

This method can be used as a relative distance estimation between different waves.

Author(s)

Jerome Sueur <`sueur@mnhn.fr`>

References

Sueur, J., Pavoine, S., Hamerlynck, O. & Duvail, S. (2008) - Rapid acoustic survey for biodiversity appraisal. *PLoS ONE*, 3(12): e4065.

See Also

`diffspec`, `diffenv`

Examples

```
data(tico) ; tico <- tico@left
data(orni) ; orni <- orni@left
# selection in tico to have two waves of similar duration (length)
tico <- tico[1:length(orni)]
diffwave(tico,orni,f=22050)
# changing the frequency parameter (wl)
diffwave(tico,orni,f=22050,wl=1024)
# changing the temporal parameter (msmooth)
diffwave(tico,orni,f=22050,msmooth=c(20,0))
```

`discrets` *Time series discretisation*

Description

This function transforms a numeric (time) series into a sequence of symbols

Usage

```
discrets(x, symb = 5, collapse = TRUE, plateau=1)
```

Arguments

`x`	a `vector`, a `matrix` (first column), an object of class `ts`, `Sample` (left channel), or `Wave` (left channel).
`symb`	the number of symbols used for the discretisation, can be set to 3 or 5 only.
`collapse`	logical, if `TRUE`, the symbols are pasted in a character string of length 1.
`plateau`	a numeric vector of length 1 taking the values `1` or `2` only. See details.

Details

The function partitions the numeric (time) series into a sequence of finite number of symbols. These symbols result of the comparaison of each series value with its temporal neighbours.
They are two discretisations available:
when `symb` is set to 3, each value will be replaced by either:
- *I* if the series is *I*ncreasing,
- *D* if the series is *D*ecreasing,
- *F* if the series remains *F*lat,
when `symb` is set to 5, each value will be replaced by either:
- *I* if the series is *I*ncreasing,
- *D* if the series is *D*ecreasing,
- *F* if the series remains *F*lat,
- *P* if the series shows a *P*eak,
- *T* if the series shows a *T*rough.

The argument `plateau` can be used to control the way a plateau is encoded. A plateau is an elevated flat region that can be either considered a 'flat peak' encoded as *PF...FP* (`plateau = 1`) or as an increase, a flat region and a decrease encoded as *IF...FD* (`plateau = 1`. The default value (`plateau = 1`) refers to Cazelles *et al.* (2004).

Value

A character string of length 1 if `collapse` is `TRUE`. Otherwise, a character string of length *n-2* if `symbol=5` (the first and last values cannot be replaced with a symbol) or *n-1* if `symbol=3` (the first value cannot be replaced with a symbol.)

Author(s)

Jerome Sueur, improved by Laurent Lellouch

References

Cazelles, B. 2004 Symbolic dynamics for identifying similarity between rhythms of ecological time series. *Ecology Letters*, **7**: 755-763.

See Also

symba

Examples

```
# a random variable
discrets(rnorm(30))
discrets(rnorm(30),symb=3)
# a frequency spectrum
data(tico)
spec1<-spec(tico,f=22050,at=0.2,plot=FALSE)
discrets(spec1[,2])
```

---

drawenv *Draw the amplitude envelope of a time wave*

Description

This function lets the user modifying the amplitude envelope of a time wave by drawing it with the graphics device

Usage

```
drawenv(wave, f, channel = 1, n = 20, plot = FALSE, listen = FALSE, output = "matrix")
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
n	the maximum number of points to draw the new envelope. Valid values start at 1.
plot	if TRUE returns the oscillogram of the new time wave (by default FALSE).
listen	if TRUE the new sound is played back.
output	character string, the class of the object to return, either "matrix", "Wave", "Sample", "audioSample" or "ts".

Details

The function first plots an oscillogram view of `wave`.
The user has then to choose points on the positive side of the y-axis (amplitude). The junction of these points will draw a new amplitude envelope.
The order of points along the x-axis (time) is not important but points cannot be cancelled. When this process is finished the new time wave is returned in the console or as an oscillogram in a second graphics device if `plot` is `TRUE`.
The function uses `locator`.

Value

If `plot` is `FALSE`, a new wave is returned. The class of the returned object is set with the argument `output`.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

`setenv`, `env`, `synth`

Examples

```
## Not run:
a<-synth(d=1,f=22050,cf=1000)
# drawenv(a,f=22050,plot=TRUE)
# choose points on the oscillogram view to draw a new enveloppe
# stop (ESC on Windows; right mouse button on Linux)
# check the result on the second graphics device opened thanks to plot=TRUE

## End(Not run)
```

`drawfilter` *Draw the amplitude profile of a frequency filter*

Description

This function lets the user drawing the amplitude profile of a frequency filter.

Usage

```
drawfilter(f, n = 256, continuous = TRUE, discrete = TRUE)
```

Arguments

`f`	a numeric vector of length 1 for the sampling frequency of the object to be filtered (in Hz).
`n`	a numeric vector of length 1 for the length (i.e. number of points) of the filter. By default = 256 to fit with a FIR with wl = 512.
`continuous`	a logical (`TRUE` by default) to draw a continuous filter.
`discrete`	a logical (`TRUE` by default) to draw a discrete filter.

Details

If the same frequency of a discrete filter is selected twice then the sum of the amplitudes of the two selections is used. If both arguments `continuous` and `discrete` are set to `TRUE` and if frequencies selected overlap between the two filters then only the frequencies of the discrete filter are considered.

Value

The function returns a two-column matrix, the first column is the frequency in kHz and the second column is the amplitude of the filter.

Note

This function can be used to prepare bandpass or bandstop custom filters to be used with `fir` and `ffilter`. See examples.

Author(s)

Laurent Lellouch

See Also

`fir`, `squarefilter`, `combfilter`, `ffilter`, `drawenv`

Examples

```
## Not run:
f <- 8000
a <- noisew(f=f, d=1)
## bandpass continuous and discrete
cont.disc <- drawfilter(f=f/2)
a.cont.disc <- fir(a, f=f, custom=cont.disc)
spectro(a.cont.disc, f=f)
## bandpass continuous only
cont <- drawfilter(f=f/2, discrete=FALSE)
a.cont <- fir(a, f=f, custom=cont)
spectro(a.cont, f=f)
## bandstop continuous only
cont.stop <- drawfilter(f=f/2, discrete=FALSE)
a.cont.stop <- fir(a, f=f, custom=cont.stop, bandpass=FALSE)
spectro(a.cont.stop, f=f)
```

```
## bandpass discrete only
disc <- drawfilter(f=f/2, continuous=FALSE)
a.disc <- fir(a, f=f, custom=disc, bandpass=FALSE)
spectro(a.disc, f=f)

## End(Not run)
```

---

duration *Duration of a time wave*

Description

Returns the duration (in second) of a time wave

Usage

```
duration(wave, f, channel=1)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).

Value

A numeric vector of length 1 returning the duration in second.

Author(s)

Jerome Sueur

Examples

```
data(tico)
duration(tico)
```

`dynoscillo` *Dynamic oscillogram*

Description

This graphical function displays a time wave as an windowed oscillogram.

Usage

```
dynoscillo(wave, f, channel = 1, wd = NULL, wl = NULL, wnb = NULL, title = TRUE, ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`wd`	a numerical vector, duration of the window (in seconds)
`wl`	a numerical vector, length of the window (in number of points).
`wnb`	a numerical vector, number of windows (no unit).
`title`	a logical, if `TRUE` displays a title with information regarding window size and number.
`...`	other `plot` graphical parameters.

Details

The arguments `wd`, `wl` and `wn` have to be used isolated, not in conjunction. They basically do the same, ie they set the duration of the zooming window that is slided along the signal. For instance, for a 5 seconds sound with a sampling rate (`f`) at 44.1 kHz, `wl = 4096` is equivalent to `wd = 4096 / 44100 = 0.093 s` and equivalent to `wnb = 5*4096 / 44100 = 53`.

Note

This function requires the package `rpanel`.

Author(s)

Jerome Sueur

See Also

`oscillo`, `oscilloST`, `dynspec`.

Examples

```
## Not run:
require(rpanel)
data(tico)
dynoscillo(tico, wn=4)
## End(Not run)
```

---

dynspec *Dynamic sliding spectrum*

Description

This function plots dynamically a sliding spectrum along a time wave. This basically corresponds to a short-term Fourier transform.

Usage

```
dynspec(wave, f, channel = 1, wl = 512, wn = "hanning", zp = 0,
ovlp = 0, fftw = FALSE, norm = FALSE, dB = NULL,  dBref = NULL, plot = TRUE,
title = TRUE, osc = FALSE,
tlab = "Time (s)", flab = "Frequency (kHz)",
alab = "Amplitude", alim = NULL, flim = c(0, f/2000),
type = "l", from = NULL, to = NULL, envt = NULL,
msmooth = NULL, ksmooth = NULL, colspec = "black",
coltitle = "black", colbg = "white", colline = "black",
colaxis = "black", collab = "black", cexlab = 1,
fontlab = 1, colwave = "black",
coly0 = "lightgrey", colcursor = "red", bty = "l")
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`wl`	if `at` is not null, length of the window for the analysis (even number of points, by defaults = 512).
`wn`	window name, see `ftwindow` (by default `"hanning"`).
`zp`	zero-padding (even number of points), see `Details`.
`ovlp`	overlap between two successive windows (in %).
`fftw`	if `TRUE` calls the function `FFT` of the library `fftw`. See Notes of the `spectro`.
`norm`	logical, if `TRUE` compute a normalised sliding spectrum.
`dB`	a character string specifying the type dB to return: "max0" for a maximum dB value at 0, "A", "B", "C", "D", and "ITU" for common dB weights.

- `dBref` a dB reference value when `dB` is not `NULL`. `NULL` by default but should be set to 2*10e-5 for a 20 microPa reference (SPL).
- `plot` logical, if `TRUE` plots in an ew graphics device the successive spectra sliding along the time wave (by default `TRUE`).
- `title` logical, if `TRUE` adds a title with the time position of the current spectrum along the time wave.
- `osc` logical, if `TRUE` plots an oscillogram beneath the sliding spectrum with a cursor showing the position of the current spectrum (by default `FALSE`).
- `tlab` title of the time axis.
- `flab` title of the frequency axis.
- `alab` title of the amplitude axis.
- `flim` range of frequency axis.
- `alim` range of amplitude axis.
- `type` type of plot that should be drawn for the sliding spectrum. See `plot` for details (by default "l" for lines).
- `from` start mark where to compute the sliding spectrum (in s).
- `to` end mark where to compute the sliding spectrum (in s).
- `envt` the type of envelope to be plooted: either "abs" for absolute amplitude envelope or "hil" for Hilbert amplitude envelope. See `env`.
- `msmooth` when `env` is not `NULL`, a vector of length 2 to smooth the amplitude envelope with a mean sliding window. The first component is the window length (in number of points). The second component is the overlap between successive windows (in %). See `env`.
- `ksmooth` when `env` is not `NULL`, kernel smooth via `kernel`. See `env`.
- `colspec` colour of the sliding spectrum.
- `coltitle` if `title` is `TRUE`, colour of the title.
- `colbg` background colour.
- `colline` colour of axes line.
- `colaxis` colour of the axes.
- `collab` colour of axes title.
- `cexlab` character size for axes title.
- `fontlab` font for axes title.
- `colwave` colour of the oscillogram or of the envelope (only when `osc` is `TRUE`).
- `coly0` colour of the y=0 line (only when `osc` is `TRUE`).
- `colcursor` colour of oscillogram cursor (only when `osc` is `TRUE`).
- `bty` the type of box to be drawn around the oscillogram (only when `osc` is `TRUE`).

Details

Use the slider panel to move along the time wave.
Use the argument `norm` if you wish to have each spectrum normalised, *i.e.* with values between 0 and 1 or maximised to 0 dB when `dB` is `TRUE`.
The function requires the package **rpanel** that is based on the package **tcltk**.

Value

This function returns a list of three items:

- `time` a numeric vector corresponding to the time axis.
- `freq` a numeric vector corresponding to the frequency axis.
- `amp` a numeric matrix corresponding to the amplitude values. Each column is a Fourier transform of length `wl/2`.

Note

This function is very similar to a spectrogram. See the `Details` of `spectro` for some information regarding the short term Fourier transform.

Author(s)

Jerome Sueur and Caroline Simonis

See Also

`spectro`, `spectro3D`, `wf`, `spec`, `dynspectro`, `fft`, `oscillo`, `env`.

Examples

```
## Not run:
data(sheep)
require(rpanel)
dynspec(sheep,f=8000,wl=1024,ovlp=50,osc=TRUE)

## End(Not run)
```

`dynspectro` *Dynamic sliding spectrogramn*

Description

This function plots dynamically a sliding spectrogram along a time wave.

Usage

```
dynspectro(wave, f, channel = 1, slidframe = 10,
wl = 512, wn = "hanning", zp = 0, ovlp = 75,
fftw = FALSE, dB = TRUE, plot = TRUE,
title = TRUE, osc = FALSE,
tlab = "Time (s)", flab = "Frequency (kHz)", alab = "Amplitude",
from = NULL, to = NULL,
collevels = NULL, palette = spectro.colors,
envt = NULL, msmooth = NULL, ksmooth = NULL,
```

```
coltitle = "black", colbg = "white", colline = "black",
colaxis = "black", collab = "black", cexlab = 1,
fontlab = 1, colwave = "black",
coly0 = "lightgrey", colcursor = "red", bty = "l")
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
slidframe	size of the sliding frame (in percent of the wave duration).
wl	if at is not null, length of the window for the analysis (even number of points, by defaults = 512).
wn	window name, see ftwindow (by default "hanning").
zp	zero-padding (even number of points), see Details.
ovlp	overlap between two successive windows (in %).
fftw	if TRUE calls the function FFT of the library fftw. See Notes of the spectro.
dB	a logical, if TRUE then uses dB values
plot	logical, if TRUE plots in an ew graphics device the successive spectrograms sliding along the time wave (by default TRUE).
title	logical, if TRUE adds a title with the time position of the current spectrogram along the time wave.
osc	logical, if TRUE plots an oscillogram beneath the sliding spectrogram with a cursor showing the position of the current spectrum (by default FALSE).
tlab	title of the time axis.
flab	title of the frequency axis.
alab	title of the amplitude axis.
from	start mark where to compute the sliding spectrogram (in s).
to	end mark where to compute the sliding spectrogram (in s).
collevels	a set of levels which are used to partition the amplitude range of the spectrogram.
palette	a color palette function to be used to assign colors in the plot.
envt	the type of envelope to be plooted: either "abs" for absolute amplitude envelope or "hil" for Hilbert amplitude envelope. See env.
msmooth	when env is not NULL, a vector of length 2 to smooth the amplitude envelope with a mean sliding window. The first component is the window length (in number of points). The second component is the overlap between successive windows (in %). See env.
ksmooth	when env is not NULL, kernel smooth via kernel. See env.
coltitle	if title is TRUE, colour of the title.
colbg	background colour.

`colline`	colour of axes line.
`colaxis`	colour of the axes.
`collab`	colour of axes title.
`cexlab`	character size for axes title.
`fontlab`	font for axes title.
`colwave`	colour of the oscillogram or of the envelope (only when `osc` is `TRUE`).
`coly0`	colour of the y=0 line (only when `osc` is `TRUE`).
`colcursor`	colour of oscillogram cursor (only when `osc` is `TRUE`).
`bty`	the type of box to be drawn around the oscillogram (only when `osc` is `TRUE`).

Details

Use the slider panel to move along the time wave.
The function requires the package **rpanel** that is based on the package **tcltk**.
The function is mainly written for inspecting long sounds.
The function is based on `image` for fast display when `spectro` is based on `filled.contour`. Displaying the amplitude envelope with the argument `envt` can slow down significantly the display.

Value

This function returns a list of three items:

`time`	a numeric vector corresponding to the time axis.
`freq`	a numeric vector corresponding to the frequency axis.
`amp`	a numeric matrix corresponding to the amplitude values. Each column is a Fourier transform of length `wl/2`.

Note

This function is very similar to a spectrogram. See the `Details` of `spectro` for some information regarding the short term Fourier transform.

Author(s)

David Pinaud and Jerome Sueur

See Also

`spectro`, `spectro3D`, `wf`, `spec`, `dynspec`, `fft`, `oscillo`, `env`.

Examples

```
## Not run:
data(sheep)
require(rpanel)
dynspectro(sheep, ovlp=95, osc=TRUE)

## End(Not run)
```

echo *Echo generator*

Description

This function generate echoes of a time wave.

Usage

```
echo(wave, f, channel = 1, amp, delay, plot = FALSE,
listen = FALSE, output = "matrix", ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`amp`	a vector describing the relative amplitude of the successive echoes. Each value of the vector should be in [0,1]
`delay`	a vector describing the time delays of the successive echoes from the beginning of `wave` (in s.)
`plot`	logical, if `TRUE` returns an oscillographic plot of the wave modified (by default `FALSE`).
`listen`	if `TRUE` the new sound is played back.
`output`	character string, the class of the object to return, either `"matrix"`, `"Wave"`, `"Sample"`, `"audioSample"` or `"ts"`.
`...`	other `oscillo` graphical parameters.

Details

`amp` and `delay` should strictly have the same length corresponding to the number of desired echoes.

Value

If `plot` is `FALSE`, a new wave is returned. The class of the returned object is set with the argument `output`.

Note

This function is based on a convolution (`convolve`) between the input wave and a pulse echo filter.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

References

Stoddard, P. K. (1998). Application of filters in bioacoustics. *In*: Hopp, S. L., Owren, M. J. and Evans, C. S. (Eds), *Animal acoustic communication*. Springer, Berlin, Heidelberg,pp. 105-127.

See Also

synth

Examples

```
# generation of the input wave
a <- synth(f=11025,d=1,cf=2000,shape="tria",am=c(50,10),fm=c(1000,10,1000,0,0))
# generation of three echoes
# with respectively a relative amplitude of 0.8, 0.4, and 0.2
# and with a delay of 1s, 2s, and 3s  from the beginning of the input wave
aecho <- echo(a,f=11025,amp=c(0.8,0.4,0.2),delay=c(1,2,3))
# another echo with time delays overlapping with the input wave
aecho <- echo(a,f=11025,amp=c(0.4,0.2,0.4),delay=c(0.6,0.8,1.5))
```

---

env — *Amplitude envelope of a time wave*

Description

This function returns the absolute or Hilbert amplitude envelope of a time wave.

Usage

```
env(wave, f, channel = 1, envt = "hil",
msmooth = NULL, ksmooth = NULL, ssmooth = NULL,
asmooth = NULL,
fftw = FALSE, norm = FALSE,
plot = TRUE, k = 1, j = 1, ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
envt	the type of envelope to be returned: either "abs" for absolute amplitude envelope or "hil" for Hilbert amplitude envelope. See Details section.
msmooth	a vector of length 2 to smooth the amplitude envelope with a mean sliding window. The first component is the window length (in number of points). The second component is the overlap between successive windows (in %). See examples.

ksmooth	kernel smooth via kernel. See examples.
ssmooth	length of the sliding window used for a sum smooth.
asmooth	length of the sliding window used for an autocorrelation smooth.
fftw	if TRUE calls the function FFT of the library fftw for faster computation for the Hilbert amplitude envelope (envt="hil") and/or for kernell smoothing (ksmooth).
norm	a logical, if TRUE the amplitude of the envelope is normalised between 0 and 1.
plot	logical, if TRUE returns a plot of wave envelope (by default TRUE).
k	number of horizontal sections when plot is TRUE (by default =1).
j	number of vertical sections when plot is TRUE (by default =1).
...	other oscillo graphical parameters.

Details

When envt is set as "abs", the amplitude envelope returned is the absolute value of wave.
When envt is set as "hil", the amplitude envelope returned is the modulus (Mod) of the analytical signal of wave obtained through the Hilbert transform (hilbert).

Value

Data are returned as one-column matrix when plot is FALSE.

Note

Be aware that smoothing with either msmooth or ksmooth changes the original number of points describing wave.

Author(s)

Jerome Sueur. Implementation of 'fftw' argument by Jean Marchal and Francois Fabianek. Implementation of 'asmooth' by Thibaut Marin-Cudraz.

See Also

oscillo,hilbert

Examples

```
data(tico)
# Hilbert amplitude envelope
env(tico)
# absolute amplitude envelope
env(tico, envt="abs")
# smoothing with a 10 points and 50% overlaping mean sliding window
env(tico, msmooth=c(10,50))
# smoothing kernel
env(tico, ksmooth=kernel("daniell",10))
# sum smooth
env(tico, ssmooth=50)
```

```
# autocorrelation smooth
env(tico, asmooth=50)
# overplot of oscillographic and envelope representations
oscillo(tico)
par(new=TRUE)
env(tico, colwave=2)
```

---

export *Export sound data*

Description

Export sound data as a text file that can be read by a sound player like 'Goldwave'

Usage

```
export(wave, f = NULL, channel = 1, filename = NULL, header=TRUE, ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`filename`	name of the new file. (by default the name of `wave`).
`header`	either a logical or a character vector, if `TRUE` add a header to be read by Goldwave, if `FALSE` does not add any header, if a character vector add the character vector as a header.
`...`	other `write.table` parameters.

Details

Creates a new text file with a header describing the main features of the sound (wave). For instance, for a 2 s sound with a sampling frequency of 8000 Hz, the header will be: [ASCII 8000Hz, Channels: 1, Samples: 160000, Flags: 0]. This type of file can be read by sound players like Goldwave (`http://www.goldwave.com/`).

Author(s)

Jerome Sueur <sueur@mnhn.fr>

Examples

```
a<-synth(f=8000,d=2,cf=2000,plot=FALSE)
export(a,f=8000)
unlink("a.txt")
```

fadew *Fade in and fade out of a time wave*

Description

This function applies a “fade in” and/or a “fade out” to a time wave following a linear, exponential or cosinus-like shape.

Usage

```
fadew(wave, f, channel = 1, din = 0, dout = 0, shape = "linear", plot = FALSE,
listen = FALSE, output = "matrix", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
din	fade in duration.
dout	fade out duration.
shape	fade shape, "linear", "exp" for exponential, "cos" for cosinus-like, (by default "linear").
plot	logical, if TRUE returns an oscillographic plot of the wave modified (by default FALSE).
listen	if TRUE the new sound is played back.
output	character string, the class of the object to return, either "matrix", "Wave", "Sample", "audioSample" or "ts".
...	other oscillo graphical parameters.

Value

If plot is FALSE, a new wave is returned. The class of the returned object is set with the argument output.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

oscillo, addsilw, cutw, deletew,mutew, pastew, revw, zapsilw

Examples

```
a<-noisew(d=5,f=4000)
op<-par(mfrow=c(3,1))
fadew(a,f=4000,din=1,dout=2,plot=TRUE,title="Linear",cexlab=0.8)
fadew(a,f=4000,din=1,dout=2,shape="exp",plot=TRUE,title="Exponential shape",
    colwave="blue",coltitle="blue",cexlab=0.8)
fadew(a,f=4000,din=1,dout=2,shape="cos",plot=TRUE,title="Cosinus-like shape",
    colwave="red",coltitle="red",cexlab=0.8)
par(op)
```

`fbands` *Frequency bands plot*

Description

This graphical function returns a frequency spectrum as a bar plot.

Usage

```
fbands(spec, f = NULL, bands = 10, width = FALSE, mel = FALSE, plot = TRUE,
xlab = NULL, ylab = "Relative amplitude", ...)
```

Arguments

`spec`	a data set resulting of a spectral analysis obtained with `spec` or `meanspec`. Can be in dB.
`f`	sampling frequency of `spec` (in Hz). Not requested if the first column of `spec` contains the frequency axis.
`bands`	a numeric vector. If vector of length 1, then sets the number of bands dividing in equal parts the spectrum. If of length > 1, then takes the values as kHz limits of the bands dividing the spectrum. These bands can be of different size. See details and examples.
`width`	logical, if `TRUE` and that `bands` is an irregular series of values, then the width of the bands will be proportional to the frequency limits defined in `bands`.
`mel`	a logical, if `TRUE` the (htk-)mel scale is used.
`plot`	logical, if `TRUE`, a plot showing the peaks is returned.
`xlab`	label of the x-axis.
`ylab`	label of the y-axis.
`...`	other `plot` graphical parameters.

Details

The function proceeds as follows

- divides the spectrum in bands. The limits of the bands are set with the argument `bands`. There are two options:
  - you set a number of bands with equal size by giving a single value to `bands`. For instance, setting `bands` to a value of 10 will slice the spectrum in 10 equal parts and return 10 local peaks.
  - you set the limits of the bands. This is achieve by giving a numeric vector to `bands`. The limits can follow a regular or irregular series. For instance attributing the vector c(0,2,4,8) will generate the following bands [0,2[, [2,4[, [4,8] kHz. Be aware that the last value should not exceed half the sampling frequency used to obtain the spectrum `spec`.
- uses the function `barplot`.

Value

A two-column matrix, the first column corresponding to the frequency values (x-axis, mean of the bars limits) and the second column corresponding to height values (y-axis) of the bars.

Note

The value below bars is the mean between the corresponding frequency limits.

Author(s)

Jerome Sueur, improved by Laurent Lellouch

See Also

`meanspec`, `spec`, `barplot`.

Examples

```
data(sheep)
spec <- meanspec(sheep, f=8000, plot=FALSE)
# default plot
fbands(spec)
# setting a specific number of bands
fbands(spec, bands=6)
#setting specific regular bands limits
fbands(spec, bands=seq(0,4,by=0.25))
# some plot tuning
op <- par(las=1)
fbands(spec, bands=seq(0,4,by=0.1),
       horiz=TRUE, col=heat.colors(41),
       xlab="", ylab="",
       cex.axis=0.75, cex.names = 0.75,
       axes=FALSE)
par(op)
# showing or not the width of the bands
```

```
oct <- octaves(440,3)/1000
op <- par(mfrow=c(2,1))
fbands(spec, bands=oct, col="blue")
fbands(spec, bands=oct, width = TRUE, col="red")
par(op)
# kind of horizontal zoom
op <- par(mfrow=c(2,1))
fbands(spec, bands=seq(0,4,by=0.2), col=c(rep(1,10),
   rep("orange",5),rep(1,5)), main="all frequency range")
fbands(spec, bands=seq(2,3,by=0.2),
   col="orange", main="a subset or zoom in")
par(op)
# kind of dynamic frequency bands
specs <- dynspec(sheep, f=8000, plot= FALSE)$amp
out <- apply(specs, f=8000, MARGIN=2,
     FUN = fbands, bands = seq(0,4,by=0.2),
     col = 1, ylim=c(0,max(specs)))
# mel scale
require(tuneR)
mel <- melfcc(sheep, nbands = 256, dcttype = "t3", fbtype = "htkmel", spec_out=TRUE)
melspec.mean <- apply(mel$aspectrum, MARGIN=2, FUN=mean)
melspec.mean <- melspec.mean/max(melspec.mean) # [0,1] scaling
fbands(melspec.mean, f=8000, bands=8)
```

`fdoppler` *Doppler effect*

Description

This function computes the altered frequency of a moving source due to the Doppler effect.

Usage

```
fdoppler(f, c = 340, vs, vo = 0, movs = "toward", movo = "toward")
```

Arguments

`f`	original frequency produced by the source (in Hz or kHz)
`c`	speed of sound in meters/second.
`vs`	speed of the source in meters/second.
`vo`	speed of the observer in meters/second. The observer is static by default *i.e.* `vo` = 0
`movs`	movement direction of the source in relation with observer position, either `"toward"` (by default) or `"away"`.
`movo`	movement direction of the observer in relation with the source position, either `"toward"` (by default, but be aware that the observer is static by default) or `"away"`.

Details

The altered frequency *f'* is computed according to:

$$f' = f \times \frac{c \pm v_o}{c \pm v_s}$$

with *f* = original frequency produced by the source (in Hz or kHz),
vs = speed of the source,
vo = speed of the observer.

Value

The altered frequency is returned in a vector.

Note

You can use `wasp` to have exact values of `c`. See examples.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

`wasp`

Examples

```
# a 400 Hz source moving toward or away from the observer at 85 m/s
fdoppler(f=400,vs=85)
# [1] 533.3333
fdoppler(f=400,vs=85,movs="away")
# [1] 320
# use wasp() if you wish to have exact sound speed at a specific temperature
fdoppler(f=wasp(f=400,t=25)$c, vs=85)
# [1] 461.8667
# Doppler effect at different source speeds
f<-seq(1,10,by=1); lf<-length(f)
v<-seq(10,300,by=20); lv<-length(v)
res<-matrix(numeric(lf*lv),ncol=lv)
for(i in 1:lv) res[,i]<-fdoppler(f=f,vs=v[i])
op<-par(bg="lightgrey")
matplot(x=f,y=res,type="l",lty=1,las=1,col= spectro.colors(lv),
xlab="Source frequency (kHz)", ylab="Altered frequency (kHz)")
legend("topleft",legend=paste(as.character(v),"m/s"),
lty=1,col= spectro.colors(lv))
title(main="Doppler effect at different source speeds")
par(op)
```

ffilter *Frequency filter*

Description

This function filters out a selected frequency section of a time wave (low-pass, high-pass, low-stop, high-stop, bandpass or bandstop frequency filter).

Usage

```
ffilter(wave, f, channel = 1, from = NULL, to = NULL, bandpass = TRUE,
custom = NULL, wl = 1024, ovlp = 75, wn = "hanning", fftw = FALSE,
rescale=FALSE, listen=FALSE, output="matrix")
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
from	start frequency (in Hz) where to apply the filter.
to	end frequency (in Hz) where to apply the filter.
bandpass	if TRUE a band-pass filter is applied between from and to, if FALSE a band-stop filter is applied between from and to (by default TRUE).
custom	a vector describing the frequency response of a custom filter. This can be manually generated or obtained with spec and meanspec. The length of the vector should be half the length of wl. See examples.
wl	window length for the analysis (even number of points).
ovlp	overlap between successive FFT windows (in %).
wn	window name, see ftwindow (by default "hanning").
fftw	if TRUE calls the function FFT of the library fftw. See Notes of the spectro.
rescale	a logical, if TRUE then the sample values of new wave (output) are rescaled according to the sample values of wave (input).
listen	a logical, if TRUE the new sound is played back.
output	character string, the class of the object to return, either "matrix", "Wave", "audioSample" or "ts".

Details

A short-term Fourier transform is first applied to the signal (see spectro), then the frequency filter is applied and the new signal is eventually generated using the reverse of the Fourier Transform (istft).
There is therefore neither temporal modifications nor amplitude modifications.

Value

If plot is FALSE, a new wave is returned. The class of the returned object is set with the argument output.

Author(s)

Jerome Sueur

See Also

afilter,lfs,fir, preemphasis, combfilter, bwfilter

Examples

```
a<-noisew(f=8000,d=1)
# low-pass
b<-ffilter(a,f=8000,to=1500)
spectro(b,f=8000,wl=512)
# high-pass
c<-ffilter(a,f=8000,from=2500)
spectro(c,f=8000,wl=512)
# band-pass
d<-ffilter(a,f=8000,from=1000,to=2000)
spectro(d,f=8000,wl=512)
# band-stop
e<-ffilter(a,f=8000,from=1500,to=2500,bandpass=FALSE)
spectro(e,f=8000,wl=512)
# custom
myfilter1<-rep(c(rep(0,64),rep(1,64)),4)
g<-ffilter(a,f=8000,custom=myfilter1)
spectro(g,f=8000)
```

---

field *Near field and far field limits*

Description

This function helps in knowing whether you are working in the near or far field.

Usage

```
field(f, d)
```

Arguments

f	frequency (Hz)
d	distance from the sound source (m)

Details

Areas very close to the sound source are in the near-field where the contribution of particle velocity to sound energy is greater thant that of sound pressure and where these components are not in phase. Sound propagation properties are also different near or far from the source. It is therefore important to know where the microphone was from the source.
To know this, the product k*d is computed according to:

$$k \times d = \frac{f}{c} \times d$$

with d = distance from the source (m), f = frequency (Hz) and c = sound celerity (m/s).
If k*d is greatly inferior 1 then the microphone is in the near field.
The decision help returned by the function follows the rule:
far field:

$$k \times d > 1$$

between near and far field limits:

$$0.1 \leq k \times d \leq 1$$

near field:

$$k \times d < 0.1$$

.

Value

A list of two values is returned:

kd	the numeric value k*d used to take a decision
d	a character string giving the help decision.

Note

This function works for air-borne sound only.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

Examples

```
# 1 kHz near field at 1 cm from the source
field(f=1000,d=0.01)
# playing with distance from source and sound frequency
op<-par(bg="lightgrey")
D<-seq(0.01,0.5,by=0.01); nD<-length(D)
F<-seq(100,1000,by=25); nF<-length(F)
a<-matrix(numeric(nD*nF),nrow=nD)
for(i in 1:nF) a[,i]<-field(f=F[i],d=D)$kd
matplot(x=D,y=a,type="l",lty=1,col= spectro.colors(nF),
  xlab="Distance from the source (m)", ylab="k*d")
title("Variation of the product k*d with distance and frequency")
```

```
text(x=c(0.4,0.15),y=c(0.02,1), c("Near Field","Far Field"),font=2)
legend(x=0.05,y=1.4,c("100 Hz","1000 Hz"),lty=1,
  col=c(spectro.colors(nF)[1],spectro.colors(nF)[nF]),bg="grey")
abline(h=0.1)
par(op)
```

---

fir *Finite Impulse Response filter*

Description

This function is a FIR filter that filters out a selected frequency section of a time wave (low-pass, high-pass, low-stop, high-stop, bandpass or bandstop frequency filter).

Usage

```
fir(wave, f, channel = 1, from = NULL, to = NULL, bandpass = TRUE, custom = NULL,
wl = 512, wn = "hanning", rescale=FALSE, listen = FALSE, output = "matrix")
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`from`	start frequency (in Hz) where to apply the filter.
`to`	end frequency (in Hz) where to apply the filter.
`bandpass`	if `TRUE` a band-pass filter is applied between `from` and `to`, if not `NULL` a band-stop filter is applied between `from` and `to` (by default `NULL`).
`custom`	a vector describing the frequency response of a custom filter. This can be manually generated or obtained with `spec` and `meanspec`. `wl` is no more required. See examples.
`wl`	window length of the impulse filter (even number of points).
`wn`	window name, see `ftwindow` (by default `"hanning"`).
`rescale`	a logical, if `TRUE` then the sample values of new wave (output) are rescaled according to the sample values of `wave` (input).
`listen`	a logical, if `TRUE` the new sound is played back.
`output`	character string, the class of the object to return, either `"matrix"`, `"Wave"`, `"Sample"`, `"audioSample"` or `"ts"`.

Details

This function is based on the reverse of the Fourier Transform (`fft`) and on a convolution (`convolve`) between the wave to be filtered and the impulse filter.

Value

A new wave is returned. The class of the returned object is set with the argument `output`.

Author(s)

Jerome Sueur

References

Stoddard, P. K. (1998). Application of filters in bioacoustics. *In*: Hopp, S. L., Owren, M. J. and Evans, C. S. (Eds), *Animal acoustic communication*. Springer, Berlin, Heidelberg,pp. 105-127.

See Also

`ffilter`, `bwfilter`, `preemphasis`, `lfs`, `afilter`

Examples

```
a<-noisew(f=8000,d=1)
# low-pass
b<-fir(a,f=8000,to=1500)
spectro(b,f=8000)
# high-pass
c<-fir(a,f=8000,from=2500)
spectro(c,f=8000)
# band-pass
d<-fir(a,f=8000,from=1000,to=2000)
spectro(d,f=8000)
# band-stop
e<-fir(a,f=8000,from=1500,to=2500,bandpass=FALSE)
spectro(e,f=8000)
# custom filter manually generated
myfilter1<-rep(c(rep(0,32),rep(1,32)),4)
g<-fir(a,f=8000,custom=myfilter1)
spectro(g,f=8000)
# custom filter generated using spec()
data(tico)
myfilter2<-spec(tico,f=22050,at=0.7,wl=512,plot=FALSE)
b<-noisew(d=1,f=22050)
h<-fir(b,f=22050,custom=myfilter2)
spectro(h,f=22050)
```

`fma` *Frequency modulation analysis*

Description

This function computes the Fourier analysis of the instantaneous frequency of a time wave. This allows to detect periodicity in frequency modulation.

Usage

```
fma(wave, f, channel = 1, threshold = NULL, plot = TRUE, ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
threshold	amplitude threshold for signal detection (in %).
plot	logical, if TRUE the spectrum of the instantaneous frequency (by default TRUE).
...	other spec parameters.

Details

This function is based on ifreq and spec.
The instantaneous frequency of wave is first computed and the spectrum of this frequency modulation is then processed. All env and spec arguments can be set up.

Value

If plot is FALSE, fma returns a numeric vector corresponding to the computed spectrum. If peaks is not NULL, fma returns a list with two elements:

spec	the spectrum computed
peaks	the peaks values (in kHz).

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

ifreq, hilbert, spec, ama

Examples

```
# a sound with a 1 kHz sinusoid FM
a<-synth(d=1, f=8000, cf=1500, fm=c(1000,1000,0,0,0), output="Wave")
fma(a)
```

fpeaks *Frequency peak detection*

Description

This function searches for peaks of a frequency spectrum.

Usage

```
fpeaks(spec, f = NULL,
nmax = NULL, amp = NULL, freq = NULL, threshold = NULL,
mel =FALSE,
plot = TRUE, title = TRUE,
xlab = NULL, ylab = "Amplitude",
labels = TRUE, digits = 2,
legend = TRUE, collab = "red", ...)
```

Arguments

`spec`	a data set resulting of a spectral analysis obtained with `spec` or `meanspec`. Can be in dB.
`f`	sampling frequency of `spec` (in Hz). Not requested if the first column of `spec` contains the frequency axis.
`nmax`	maximal number of peaks detected. Overrides `amp` and `freq`. See details.
`amp`	amplitude slope parameter, a numeric vector of length 2. Refers to the amplitude slopes of the peak. The first value is the left slope and the second value is the right slope. Only peaks with higher slopes than threshold values will be kept. See details.
`freq`	frequency threshold parameter (in Hz). If the frequency difference of two successive peaks is less than this threshold, then the peak of highest amplitude will be kept only. See details.
`threshold`	amplitude threshold parameter. Only peaks above this threshold will be considered. See details.
`mel`	a logical, if `TRUE` the (htk-)mel scale is used.
`plot`	logical, if `TRUE`, a plot showing the peaks is returned.
`title`	logical, if `TRUE` add the number of peaks detected as a plot title.
`xlab`	label of the x-axis.
`ylab`	label of the y-axis.
`labels`	logical, if `TRUE` peak labels are plotted.
`digits`	if `labels` is `TRUE`, the number of decimal places (`round`) for the peak labels.
`legend`	logical, if `TRUE` a legend returning the different selection parameters (`nmax`, `amp`, `freq`, `threshold`, `threshold`) is added to the plot.
`collab`	labels color.
`...`	other `plot` graphical parameters.

Details

Here are some details regarding the different selection parameters:

- `nmax`: this parameter is to be used if you wish to get a specific number of peaks. The peaks selected are those with the highest slopes. It then does not work in conjunction with the other parameters.
- `freq`: this parameter allows to remove from the selection successive peaks with a small frequency difference. Imagine you have two successive peaks at 1200 Hz and 1210 Hz and at 0.5 and 0.25 in amplitude. If you set `freq` to 50 Hz, then only the first peak will be kept.
- `amp`: this parameter allows to remove from the selection peaks with low slopes. You can make the selection on both slopes or on a single one. Imagine you have an asymetric peak with a 0.01 left slope and a 0.02 right slope. The peak will be discarded for the following settings: both values higher than 0.02 (*e.g.* `amp = c(0.03,0.04)`), the first value higher than 0.01 (*e.g.* `amp = c(0.02,0.001)`), the second value higher than 0.02 (*e.g.* `amp = c(0.001,0.03)`). If you do not want apply the selection on one of the slope use 0. For instance, a selection on the left slope only will be achieved with: `amp = c(0.02,0)`.
- `threshold`: this parameter can be used to do a rough selection on the spectrum. Peaks with an amplitude value (not a slope) lower than this threshold will be automatically discarded. This can be useful when you want to remove peaks of a low-amplitude background noise.

Value

A two-column matrix, the first column corresponding to the frequency values (x-axis) and the second column corresponding to the amplitude values (y-axis) of the peaks.

Note

You can also use `fpeaks` with other kind of spectrum, for instance a cepstral spectrum. See examples.

Author(s)

Jerome Sueur and Amandine Gasc

See Also

`localpeaks`, `meanspec`, `spec`

Examples

```
data(tico)
spec <- meanspec(tico, f=22050, plot=FALSE)
specdB <- meanspec(tico, f=22050, dB="max0", plot=FALSE)
# all peaks
fpeaks(spec)
# 10 highest peaks
fpeaks(spec, nmax=10)
# highest peak (ie dominant frequency)
```

```
fpeaks(spec, nmax=1)
# peaks that are separated by more than 500 Hz
fpeaks(spec, freq=500)
# peaks with a left slope higher than 0.1
fpeaks(spec, amp=c(0.1,0))
# peaks with a right slope higher than 0.1
fpeaks(spec, amp=c(0,0.1))
# peaks with left and right slopes higher than 0.1
fpeaks(spec, amp=c(0.1,0.1))
# peaks above a 0.5 threshold
fpeaks(spec, threshold=0.5)
# peaks of a dB spectrum with peaks showing slopes higher than 3 dB
fpeaks(specdB, amp=c(3,3))
# comparing different parameter settings
meanspec(tico, f=22050)
col <- c("#ff000090","#0000ff75","#00ff00")
cex <- c(2,1.25,1.5)
pch <- c(19,17,4)
title(main="Peak detection \n (spectrum with values between 0 and 1)")
res1 <- fpeaks(spec, plot = FALSE)
res2 <- fpeaks(spec, amp=c(0.02,0.02), plot =FALSE)
res3 <- fpeaks(spec, amp=c(0.02,0.02), freq=200, plot = FALSE)
points(res1, pch=pch[1], col=col[1], cex=cex[1])
points(res2, pch=pch[2], col=col[2], cex=cex[2])
points(res3, pch=pch[3], col=col[3], cex=cex[3])
legend("topright", legend=c("all peaks","amp", "amp & freq"), pch=pch,
pt.cex=cex, col=col, bty="n")
# example with a cepstral spectrum
data(sheep)
res <- ceps(sheep,f=8000,at=0.4,wl=1024,plot=FALSE)
fpeaks(res, nmax=4, xlab="Quefrency (s)")
# melscale
require(tuneR)
mel <- melfcc(sheep, nbands = 256, dcttype = "t3", fbtype = "htkmel", spec_out=TRUE)
melspec.mean <- apply(mel$aspectrum, MARGIN=2, FUN=mean)
melspec.mean <- melspec.mean/max(melspec.mean) # [0,1] scaling
fpeaks(melspec.mean, nmax=4, f=8000, mel=TRUE)
fpeaks(melspec.mean, freq=4, f=8000, mel=TRUE) # freq in Hz!
fpeaks(melspec.mean, threshold=0.3, f=8000, mel=TRUE)
fpeaks(melspec.mean, amp=c(0.1,0.1), f=8000, mel=TRUE)
```

---

ftwindow *Fourier transform windows*

Description

Generates different Fourier Transform windows.

Usage

```
ftwindow(wl, wn = "hamming",
        correction = c("none", "amplitude", "energy"))
```

Arguments

`wl`	window length
`wn`	window name: `bartlett`, `blackman`, `flattop`, `hamming`, `hanning`, or `rectangle` (by default `hamming`).
`correction`	a character vector of length 1 to apply an amplitude ("amplitude") or an energy ("energy") correction to the FT window. By default no correction is applied ("none").

Value

A vector of length `wl`.

Note

Try the example to see windows shape.

Author(s)

Jerome Sueur <`sueur@mnhn.fr`>

References

Harris, F.J., 1978. On the use of windows for harmonic analysis with the discrete Fourier Transform. *Proceedings of the IEEE*, 66(1): 51-83.

See Also

`covspectro`, `dfreq`, `meanspec`, `spec`, `spectro`, `spectro3D`

Examples

```
a<-ftwindow(512)
b<-ftwindow(512,wn="bartlett")
c<-ftwindow(512,wn="blackman")
d<-ftwindow(512,wn="flattop")
e<-ftwindow(512,wn="hanning")
f<-ftwindow(512,wn="rectangle")
all<-cbind(a,b,c,d,e,f)
matplot(all,type="l",col=1:6,lty=1:6)
legend(legend=c("hamming","bartlett","blackman","flattop","hanning","rectangle"),
x=380,y=0.95,col=1:6,lty=1:6,cex=0.75)
```

fund *Fundamental frequency track*

Description

This function estimates the fundamental frequency through a short-term cepstral transform.

Usage

```
fund(wave, f, channel = 1, wl = 512, ovlp = 0, fmax = f/2, threshold = NULL,
at = NULL, from = NULL, to = NULL,
plot = TRUE, xlab = "Time (s)", ylab = "Frequency (kHz)",
ylim = c(0, f/2000), pb = FALSE, ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`wl`	if `at` is not null, length of the window for the analysis (even number of points, by defaults = 512).
`ovlp`	overlap between two successive windows (in %).
`fmax`	the maximum frequency to detect (in Hz).
`threshold`	amplitude threshold for signal detection (in %).
`at`	position where the estimate the fundamental frequency (in s).
`from`	start position where to compute the fundamental frequency (in s).
`to`	end position to compute the fundamental frequency (in s).
`plot`	logical, if `TRUE` plots the fundamental frequency modulations against time (by default `TRUE`).
`xlab`	title of the time axis (s).
`ylab`	title of the frequency axis (Hz).
`ylim`	the range of frequency values.
`pb`	if `TRUE` returns a text progress bar in the console.
`...`	other `plot` graphical parameters.

Value

When `plot` is `FALSE`, `fund` returns a two-column matrix, the first column corresponding to time in seconds (*x*-axis) and the second column corresponding to to fundamental frequency in kHz (*y*-axis). NA corresponds to pause sections in `wave` (see `threshold`).
No plot is produced when using `at`.

Note

This function is based on ceps.

Author(s)

Jerome Sueur <sueur@mnhn.fr>.

References

Oppenheim, A.V. and Schafer, R.W. 2004. From frequency to quefrency: a history of the cepstrum. *Signal Processing Magazine IEEE*, 21: 95-106.

See Also

cepstro, ceps, autoc

Examples

```
data(sheep)
# estimate the fundamental frequency at a single position
fund(sheep, f=8000, fmax=300, at=1, plot=FALSE)
# track the fundamental frequency along time
fund(sheep,f=8000,fmax=300,type="l")
# with 50% overlap between successive sliding windows, time zoom and
# amplitude filter (threshold)
fund(sheep,f=8000,fmax=300,type="b",ovlp=50,threshold=5,ylim=c(0,1),cex=0.5)
# overlaid on a spectrogram
spectro(sheep,f=8000,ovlp=75,zp=16,scale=FALSE,palette=reverse.gray.colors.2)
par(new=TRUE)
fund(sheep,f=8000,fmax=300,type="p",pch=24,ann=FALSE,
  xaxs="i",yaxs="i",col="black",bg="red",threshold=6)
```

---

gammatone *Gammatone filter*

Description

Generate gammatone filter in the time domain (impulse response).

Usage

```
gammatone(f, d, cfreq, n = 4, a = 1, p = 0, output = "matrix")
```

Arguments

`f`	sampling frequency (in Hz).
`d`	duration (in s).
`cfreq`	center frequency (in Hz).
`n`	filter order (no unit).
`a`	amplitude (linear scale, no unit).
`p`	initial phase (in radians).
`output`	character string, the class of the object to return, either `"matrix"`, `"Wave"`, `"Sample"`, `"audioSample"` or `"ts"`.

Details

The gammatone function in the time domain (impulse response) is obtained with:

$$g(t) = a \times t^{n-1} \times e^{-2\pi\beta t} \times \cos(2\pi cft + \phi)$$

with *a* the amplitude, *t* time, *n* the filter order, *cf* the center frequency, ϕ the initial phase.
The parameter β is the equivalent rectangular bandwidth (ERB) bandwidth which varies according to the center frequency cf following:

$$\beta = 24.7 \times (4.37 \times \frac{cf}{1000} + 1)$$

Value

A wave is returned. The class of the returned object is set with the argument `output`.

Note

Use the FFT based function, as `spec` or `meanspec`, to get the filter in the frequency domain. See examples.

Author(s)

Jerome Sueur

References

Holdsworth J, Nimmo-Smith I, Patterson R, Rice P (1988) Implementing a gammatone filter bank. Annex C of the SVOS Final Report: Part A: The Auditory Filterbank, 1, 1-5.

See Also

`melfilterbank`

Examples

```
## gammatone filter in the time domain (impulse response)
f <- 44100
d <- 0.05
res <- gammatone(f=f, d=d, cfreq=440, n=4)
## time display
oscillo(res, f=f)
## frequency display
spec(res, f=f)
## generate and plot a bank of 32 filters from 500 to 10000 Hz
n <- 32
cfreq <- round(seq(500, 10000, length.out=n))
res <- matrix(NA, nrow=f*d/2, ncol=n)
for(i in 1:n){
    res[,i] <- spec(gammatone(f=f, d=d, cfreq=cfreq[i]), f=f, dB="max0", plot=FALSE)[,2]
}
x <- seq(0,f/2,length.out=nrow(res))/1000
plot(x=x, y=res[,1],
     xlim=c(0,14), ylim=c(-60,0),
     type="l", col=2, las=1,
     xlab="Frequency (kHz)", ylab="Relative amplitude (dB)")
for(i in 2:n) lines(x, res[,i], col=2)
## use the frequency domain to filter a white noise input
## here around the center frequency 2000 Hz
res <- gammatone(f=f, d=d, cfreq=2000, n=4)
gspec <- spec(res, f=f, plot=FALSE)[,2]
nw <- noisew(f=44100, d=1)
nwfilt <- fir(nw, f=44100, wl=length(gspec)*2, custom=gspec)
spectro(nwfilt, f=f)
```

---

ggspectro — *2D-spectrogram of a time wave using ggplot2*

Description

This function returns a ggplot object to draw a spectrogram with the package ggplot2. This is an alternative to spectro.

Usage

```
ggspectro(wave, f, tlab = "Time (s)",
flab = "Frequency (kHz)", alab = "Amplitude\n(dB)\n", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.

tlab	label of the time axis.
flab	label of the frequency axis.
alab	label of the amplitude axis.
...	other non-graphical arguments to be passed to spectro (wl, ovlp etc).

Details

This function return the fist layer (data and aesthetic mapping) of a ggplot2 plot.
See the example section to understand how to build a spectrogram and consult ggplot2 help to get what you exactly need.
There is no way to plot the oscillogram as spectro does.

Value

A ggpot layer.

Note

This function requires **ggplot2** package.

Author(s)

Jerome Sueur

References

Wickham H (2009) – *ggplot2: elegant graphics for data analysis*. UseR! Springer.

See Also

spectro, spectro3D, dynspec

Examples

```
## Not run:
require(ggplot2)
## first layer
v <- ggspectro(tico, ovlp=50)
summary(v)
## using geom_tile ##
v + geom_tile(aes(fill = amplitude)) + stat_contour()
## coordinates flip (interest?)
v + geom_tile(aes(fill = amplitude)) + stat_contour() + coord_flip()
## using stat_contour ##
# default (not nice at all)
v + stat_contour(geom="polygon", aes(fill=..level..))
# set up to 30 color levels with the argument bins
(vv <- v + stat_contour(geom="polygon", aes(fill=..level..), bins=30))
# change the limits of amplitude and NA values as transparent
vv + scale_fill_continuous(name="Amplitude\n(dB)\n", limits=c(-30,0), na.value="transparent")
```

```
# Black-and-white theme
(vv + scale_fill_continuous(name="Amplitude\n(dB)\n", limits=c(-30,0),
  na.value="transparent", low="white", high="black") + theme_bw())
# Other colour scale (close to spectro() default output)
v + stat_contour(geom="polygon", aes(fill=..level..), bins=30)
  + scale_fill_gradientn(name="Amplitude\n(dB)\n", limits=c(-30,0),
  na.value="transparent", colours = spectro.colors(30))

## End(Not run)
```

H *Total entropy*

Description

This function estimates the total entropy of a time wave.

Usage

```
H(wave, f, channel = 1, wl = 512, envt="hil", msmooth = NULL, ksmooth = NULL)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`wl`	window length for spectral entropy analysis (even number of points). See `sh`.
`envt`	the type of envelope to be used: either "abs" for absolute amplitude envelope or "hil" for Hilbert amplitude envelope. See `env`.
`msmooth`	a vector of length 2 to smooth the amplitude envelope with a mean sliding window. The first component is the window length (in number of points). The second component is the overlap between successive windows (in %). See `env`.
`ksmooth`	kernel smooth via `kernel`. See `env`.

Details

This function computes the product between the values obtained with `sh` and `th` functions.
This then gives a global (time and frequency) estimation of signal entropy.
The frequency mean spectrum and the amplitude envelope needed for computing respectively `sh` and `th` are automatically generated. They can be controlled through `wl` and `smooth` arguments respectively. See examples below and examples in `sh` and `th` for implications on the results.

Value

A single value varying between 0 and 1 is returned. The value has no unit.

Note

The entropy of a noisy signal will tend towards 1 whereas the entropy of a pure tone signal will tend towards 0.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

References

Sueur, J., Pavoine, S., Hamerlynck, O. & Duvail, S. (2008) - Rapid acoustic survey for biodiversity appraisal. *PLoS ONE*, 3(12): e4065.

See Also

`sh`, `th`, `csh`

Examples

```
data(orni)
H(orni,f=22050)
# changing the spectral parameter (wl)
H(orni,f=22050,wl=1024)
# changing the temporal parameter (msmooth)
H(orni,f=22050,msmooth=c(20,0))
```

`hilbert` *Hilbert transform and analytic signal*

Description

This function returns the analytic signal of a time wave through Hilbert transform.

Usage

```
hilbert(wave, f, channel = 1, fftw = FALSE)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`fftw`	if `TRUE` calls the function `FFT` of the library `fftw` for faster computation. See Notes of the function `spectro`.

Details

The analytic signal is useful to get the amplitude envelope (see argument henv of oscillo and the instantaneous phase or frequency (see ifreq) of a time wave.

Value

hilbert returns the analytic signal as a complex matrix. The imaginary part of this matrix is the Hilbert transform.

Note

To get the Hilbert component only, use Im(Hilbert(wave)).

Author(s)

Jonathan Lees <jonathan.lees@unc.edu>. Implementation of 'fftw' argument by Jean Marchal and Francois Fabianek.

References

Mbu Nyamsi, R. G., Aubin, T. & Bremond, J. C. 1994 On the extraction of some time dependent parameters of an acoustic signal by means of the analytic signal concept. Its application to animal sound study. *Bioacoustics*, 5: 187-203.

See Also

ifreq

Examples

```
a<-synth(f=8000, d=1, cf=1000)
aa<-hilbert(a, f=8000)
```

---

ifreq *Instantaneous frequency*

Description

This function returns the instantaneous frequency (and/or phase) of a time wave through the computation of the analytic signal (Hilbert transform).

Usage

```
ifreq(wave, f, channel = 1, phase = FALSE, threshold = NULL,
plot = TRUE, xlab = "Time (s)", ylab = NULL,
ylim = NULL, type = "l", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
phase	if TRUE and plot is also TRUE plots the instantaneous phase instead of the instantaneous frequency.
threshold	amplitude threshold for signal detection (in %).
plot	logical, if TRUE plots the instantaneous frequency or phase against time (by default TRUE).
xlab	title of the x axis.
ylab	title of the y axis.
ylim	the range of y values.
type	if plot is TRUE, type of plot that should be drawn. See plot for details (by default "l" for lines).
...	other plot graphical parameters.

Details

The instantaneous phase is the argument of the analytic signal obtained throught the Hilbert transform.
The instantaneous phase is then unwrapped and derived against time to get the instantaneous frequency.
There may be some edge effects at both start and end of the time wave.

Value

If plot is FALSE, ifreq returns a list of two components:

f	a two-column matrix, the first column corresponding to time in seconds (x-axis) and the second column corresponding to instantaneous frequency in kHz (y-axis).
p	a two-column matrix, the first column corresponding to time in seconds (x-axis) and the second column corresponding to wrapped instantaneous phase in radians (y-axis).

Note

This function is based on the analytic signal obtained with the Hilbert transform (see hilbert).
The function requires the package **signal**.
The matrix describing the instantaneous phase has one more row than the one describing the instantaneous frequency.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

References

Mbu Nyamsi, R. G., Aubin, T. & Bremond, J. C. 1994 On the extraction of some time dependent parameters of an acoustic signal by means of the analytic signal concept. Its application to animal sound study. *Bioacoustics*, 5: 187-203.

See Also

hilbert, zc

Examples

```
# generate a sound with sine and linear frequency modulations
a<-synth(d=1, f=8000, cf=1500, fm=c(200,10,1000,0,0))
# plot on a single graphical device the instantaneous frequency and phase
op<-par(mfrow=c(2,1))
ifreq(a,f=8000,main="Instantaneous frequency")
ifreq(a,f=8000,phase=TRUE,main="Instantaneous phase")
par(op)
```

---

istft *Inverse of the short-term Fourier transform*

Description

This function returns a wave object from a complex STFT matrix by computing the inverse of the short-term Fourier transform (STFT)

Usage

```
istft(stft, f, wl, ovlp=75, wn="hanning", output = "matrix")
```

Arguments

stft	a complex matrix resulting of a short-term Fourier transform.
f	sampling frequency of the original wave object (in Hz)
wl	FFT window length for the analysis (even number of points).
ovlp	overlap between successive FFT windows (in %, by default 75%, see the Details section).
wn	character string specifying the FFT window name, see ftwindow (by default "hanning").
output	character string, the class of the object to return, either "matrix", "Wave", "audioSample" or "ts".

Details

The function is based on the inverse of the FFT (see fft) and on the overlap add (OLA) method.
The overlap percentage must satisfy the Perfect Reconstruction OLA-constraint. For the most windows, this constraint is:

$$ovlp = 100 \times (1 - \frac{1}{4 \times n}),$$

with n being a positive integer.
A default value is set to 75%. We suggest not to change it.

Value

A new wave is returned. The class of the returned object is set with the argument `output`.

Note

The `stft` input data must be complex.
This function is used by ffilter, lfs to respectively filter in frequency and shift in frequency a sound.
The function can be used to reconstruct or modify a sound. See examples.

Author(s)

Original Matlab code by Hristo Zhivomirov (Technical University of Varna, Bulgaria), translated and adapted to R by Jerome Sueur

See Also

spectro, ffilter, lfs

Examples

```
## Not run:
# STFT and iSTFT parameters
wl <- 1024
ovlp <- 75
# reconstruction of the tico sound from the stft complex data matrix
data(tico)
data <- spectro(tico, wl=wl, ovlp=ovlp, plot=FALSE, norm=FALSE, dB=NULL, complex=TRUE)$amp
res <- istft(data, ovlp=ovlp, wn="hanning", wl=wl, f=22050, out="Wave")
spectro(res)
# a strange frequency filter
n <- noisew(d=1, f=44100)
data <- spectro(n, f=44100, wl=wl, ovlp=ovlp, plot=FALSE, norm=FALSE, dB=NULL, complex=TRUE)$amp
data[64:192, 6:24] <- 0
nfilt <- istft(data, f=8000, wl=wl, ovlp=ovlp, output="Wave")
spectro(nfilt, wl=wl, ovlp=ovlp)

## End(Not run)
```

`itakura.dist` *Itakuro-Saito distance*

Description

Compare two distributions (e.g. two frequency spectra) by computing the Itakuro-Saito distance

Usage

```
itakura.dist(spec1, spec2, scale=FALSE)
```

Arguments

`spec1`	any distribution, especially a spectrum obtained with `spec` or `meanspec` (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
`spec2`	any distribution, especially a spectrum obtained with `spec` or `meanspec` (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
`scale`	a logical, if `TRUE` the distance is scaled by dividing the distance by the length of `spec1` (or `spec2`).

Details

The Itakura-Saito (I-S) distance is a non-symmetric measure of the difference between two probability distributions. It is here adapted for frequency spectra. The distance is asymmetric, ie computing the I-S distance between spec1 and spec2 is not the same as computing it between spec2 and spec1. A symmetry can be obtained by calculating the mean between the two directions.
The distance is obtained following:

$$D_{I-S}(spec1\|spec2) = \sum \frac{spec1}{spec2} - log(\frac{spec1}{spec2}) - 1$$

Value

The function returns a list of three items:

`D1`	The I-S distance of 'spec2' with respect to 'spec1' (*i.e.* D(spec1 || spec2))
`D2`	The I-S distance of 'spec1' with respect to 'spec2' (*i.e.* D(spec2 || spec1))
`D`	The symmetric distance (*i.e.* D = 0.5*(D1+D2))

If `scale = TRUE` the distance is divided by the length of `spec1` (or `spec2`).

Note

The function works for both Hz and (htk-)mel scales.

Author(s)

Jerome Sueur, improved by Laurent Lellouch

See Also

kl.dist, ks.dist, logspec.dist, simspec, diffspec

Examples

```
# Comparison of two spectra
data(tico)
tico1 <- spec(tico, at=0.65, plot=FALSE)
tico2 <- spec(tico, at=1.1, plot=FALSE)
itakura.dist(tico1, tico2)
itakura.dist(tico1, tico2, scale=TRUE)
```

---

kl.dist *Kullback-Leibler distance*

Description

Compare two distributions (e.g. two frequency spectra) by computing the Kullback-Leibler distance

Usage

```
kl.dist(spec1, spec2, base = 2)
```

Arguments

spec1	any distribution, especially a spectrum obtained with spec or meanspec (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
spec2	any distribution, especially a spectrum obtained with spec or meanspec (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
base	the logarithm base used to compute the distance. See log.

Details

The Kullback-Leibler distance or relative entropy is a non-symmetric measure of the difference between two probability distributions. It is here adapted for frequency spectra. The distance is asymmetric, ie computing the K-L distance between spec1 and spec2 is not the same as computing it between spec2 and spec1. A symmetry can be obtained by calculating the mean between the two directions.
The distance is obtained following:

$$D_{K-L}(spec1\|spec2) = \sum spec1 \times log(\frac{spec1}{spec2})$$

Value

The function returns a list of three items:

D1	The K-L distance of 'spec2' with respect to 'spec1' (*i.e.* D(spec1 || spec2))
D2	The K-L distance of 'spec1' with respect to 'spec2' (*i.e.* D(spec2 || spec1))
D	The symmetric K-L distance (*i.e.* D = 0.5*(D1+D2))

Note

The base of the logarithm can be changed using the argument `base`. When sets to base 2, the information is measured in units of bits. When sets to base *e*, the information is measured in nats. The function works for both Hz and (htk-)mel scales.

Author(s)

Jerome Sueur, improved by Laurent Lellouch

References

Kullback, S., Leibler, R.A. (1951). On information and sufficiency. *Annals of Mathematical Statistics*, 22: 79-86

See Also

`ks.dist`, `logspec.dist`, `simspec`, `diffspec`

Examples

```
# Comparison of two spectra
data(tico)
tico1 <- spec(tico, at=0.65, plot=FALSE)
tico2 <- spec(tico, at=1.1, plot=FALSE)
kl.dist(tico1, tico2)    # log2 (binary logarithm)
kl.dist(tico1, tico2, base=exp(1))  # ln (natural logarithm)
```

---

ks.dist *Kolmogorov-Smirnov distance*

Description

This function compares two distributions (e.g. two frequency spectra) by computing the Kolmogorov-Smirnov distance

Usage

```
ks.dist(spec1, spec2, f = NULL, mel = FALSE,
plot = FALSE, type = "l",
lty = c(1, 2), col = c(2, 4),
flab = NULL, alab = "Cumulated amplitude",
flim = NULL, alim = NULL,
title = TRUE, legend = TRUE, ...)
```

Arguments

`spec1`	any distribution, especially a spectrum obtained with `spec` or `meanspec` (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
`spec2`	any distribution, especially a spectrum obtained with `spec` or `meanspec` (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
`f`	sampling frequency of waves used to obtain `spec1` and `spec2` (in Hz). Not necessary if `spec1` and/or `spec2` is a two columns matrix obtained with `spec` or `meanspec`.
`mel`	a logical, if `TRUE` the (htk-)mel scale is used.
`plot`	logical, if `TRUE` plots both cumulated spectra and their maximal distance (*i.e.* the K-S distance.)
`type`	if `plot` is `TRUE`, type of plot that should be drawn. See `plot` for details (by default "l" for lines).
`lty`	a vector of length 2 for the line type of `spec1` and `spec2` if `type`="l".
`col`	a vector of length 2 for the colour of `spec1` and `spec2`.
`flab`	title of the frequency axis.
`alab`	title of the amplitude axis.
`flim`	the range of frequency values.
`alim`	range of amplitude axis.
`title`	logical, if `TRUE`, adds a title with D and F values.
`legend`	logical, if `TRUE` adds a legend to the plot.
`...`	other `plot` graphical parameters.

Details

The Kolmogorov distance is the maximal distance between the cumulated spectra. The function returns this distance and the corresponding frequency. This is an adaptation of the statistic computed by the non-parametric Kolmogorov-Smirnov test (see `ks.test`).

Value

The function returns a list of two items

`D`	the Kolomogorov-Smirnov distance
`F`	the frequency (in KHz) where the Kolmogorov-Smirnov distance was found

Note

There is no p-value associated to the K-S distance.
If no frequency is provided, only the distance D.

Author(s)

Jerome Sueur, improved by Laurent Lellouch

See Also

kl.dist, simspec, diffspec, logspec.dist, diffcumspec, itakura.dist

Examples

```
## Comparison of two spectra and plot of the cumulated spectra with the K-S distance
data(tico)
tico1 <- spec(tico, at=0.65, plot=FALSE)
tico2 <- spec(tico, at=1.1, plot=FALSE)
ks.dist(tico1, tico2, plot=TRUE)
## mel scale
require(tuneR)
data(orni)
orni.mel <- melfcc(orni, nbands = 256, dcttype = "t3", fbtype = "htkmel", spec_out=TRUE)
orni.mel.mean <- apply(orni.mel$aspectrum, MARGIN=2, FUN=mean)
tico.mel <- melfcc(tico, nbands = 256, dcttype = "t3", fbtype = "htkmel", spec_out=TRUE)
tico.mel.mean <- apply(tico.mel$aspectrum, MARGIN=2, FUN=mean)
ks.dist(orni.mel.mean, tico.mel.mean, f=22050, mel=TRUE, plot=TRUE)
```

---

lfs *Linear Frequency Shift*

Description

This function linearly shifts all the frequency content of a time wave.

Usage

```
lfs(wave, f, channel = 1, shift, wl = 1024, ovlp = 75,
wn = "hanning", fftw = FALSE, output = "matrix")
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
shift	positive or negative frequency shift to apply (in Hz).

wl	window length for the analysis (even number of points, by default = 1024).
ovlp	overlap between successive FFT windows (in %, by default 75%).
wn	window name, see ftwindow (by default "hanning").
fftw	if TRUE calls the function FFT of the library fftw. See Notes of the spectro.
output	character string, the class of the object to return, either "matrix", "Wave", "Sample", "audioSample" or "ts".

Details

A short-term Fourier transform is first applied to the signal (see spectro), then the frequency shift is applied and the new signal is eventually generated using the reverse of the Fourier Transform (istft).
There is therefore neither temporal modifications nor amplitude modifications.

Value

If plot is FALSE, a new wave is returned. The class of the returned object is set with the argument output.

Author(s)

Jerome Sueur <sueur@mnhn.fr> and Thierry Aubin <thierry.aubin@u-psud.fr>

References

Hopp, S. L., Owren, M. J. and Evans, C. S. (Eds) 1998. *Animal acoustic communication*. Springer, Berlin, Heidelberg.

See Also

ffilter, spectro

Examples

```
data(orni)
a<-lfs(orni,f=22050,shift=1000)
spectro(a,f=22050)
# to be compared with the original signal
spectro(orni,f=22050)
```

listen *Play a sound wave*

Description

Play a sound wave

Usage

```
listen(wave, f, channel=1, from = NULL, to = NULL, choose = FALSE)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
from	start of play (in s).
to	end of play (in s).
choose	logical, if TRUE start (=from) and end (=to) points can be graphically chosen with a cursor on the oscillogram.

Note

This function is based on play but allows to read one-colum matrix, data.frame, time-series and Sample objects.

Author(s)

Jerome Sueur <sueur@mnhn.fr> but the original play function is by Uwe Ligges (package **tuneR**).

See Also

play

Examples

```
## NOT RUN
# data(tico)
# listen(tico,f=22050)
# listen(tico,f=22050,from=0.5,to=1.5)
# listen(noise(d=1,f=8000,Wave=TRUE))
## change f to play the sound a different speed
# data(sheep)
## normal
# listen(sheep,f=8000)
## two times faster
```

```
# listen(sheep,f=8000*2)
## two times slower
# listen(sheep,f=8000/2)
```

`localpeaks` *Local maximum frequency peak detection*

Description

This functions searches for local peaks of a frequency spectrum

Usage

```
localpeaks(spec, f = NULL, bands = 10, mel = FALSE, plot = TRUE,
xlab = NULL, ylab = "Amplitude", labels = TRUE, ...)
```

Arguments

`spec`	a data set resulting of a spectral analysis obtained with `spec` or `meanspec`. Can be in dB.
`f`	sampling frequency of `spec` (in Hz). Not requested if the first column of `spec` contains the frequency axis.
`bands`	a numeric vector. If vector of length 1, then sets the number of bands dividing in equal parts the spectrum. If of length > 1, then takes the values as kHz limits of the bands dividing the spectrum. These bands can be of different size. See details and examples.
`mel`	a logical, if `TRUE` the (htk-)mel scale is used.
`plot`	logical, if `TRUE`, a plot showing the peaks is returned.
`xlab`	label of the x-axis.
`ylab`	label of the y-axis.
`labels`	logical, if `TRUE` peak labels are plotted.
`...`	other `plot` graphical parameters.

Details

The function proceed as follows

- divides the spectrum in bands. The limits of the bands are set with the argument `bands`. There are two options:
  - you set a number of bands with equal size by giving a single value to `bands`. For instance, setting `bands` to a value of 10 will slice the spectrum in 10 equal parts and return 10 local peaks.
  - you set the limits of the bands. This is achieve by giving a numeric vector to `bands`. The limits can follow a regular or irregular series. For instance attributing the vector c(0,2,4,8) will generate the following bands [0,2[, [2,4[, [4,8] kHz. Be aware that the last value should not exceed half the sampling frequency used to obtain the spectrum `spec`.
- uses the function `fpeaks` with the argument `nmax` set to 1.

Value

A two-column matrix, the first column corresponding to the frequency values (x-axis) and the second column corresponding to the amplitude values (y-axis) of the peaks.

Author(s)

Jerome Sueur

See Also

fpeaks, meanspec, spec

Examples

```
data(sheep)
spec <- meanspec(sheep, f=8000)
# a specific number of bands with all the same size
localpeaks(spec, bands=5)
# bands directly specified  with a regular sequence
localpeaks(spec, bands=seq(0,8/2,by=0.5))
# bands directly specified  with an irregular sequence
localpeaks(spec, bands=c(0,0.5,1,1.5,3,4))
# Amaj octave bands, note that there is no peak detection
# in the higher part of the spectrum as sequence stops at 3520 Hz
localpeaks(spec, bands=octaves(440, below=3, above=3)/1000)
# melscale
require(tuneR)
mel <- melfcc(sheep, nbands = 256, dcttype = "t3", fbtype = "htkmel", spec_out=TRUE)
melspec.mean <- apply(mel$aspectrum, MARGIN=2, FUN=mean)
melspec.mean <- melspec.mean/max(melspec.mean) # [0,1] scaling
localpeaks(melspec.mean, f=8000, bands=8)
```

---

logspec.dist   *Log-spectral distance*

Description

Compare two distributions (e.g. two frequency spectra) by computing the log-spectral distance

Usage

```
logspec.dist(spec1, spec2, scale=FALSE)
```

Arguments

- `spec1` any distribution, especially a spectrum obtained with `spec` or `meanspec` (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
- `spec2` any distribution, especially a spectrum obtained with `spec` or `meanspec` (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
- `scale` a logical, if `TRUE` the distance is scaled by dividing by the square-root of the length of `spec1` (or `spec2`).

Details

The distance is computed according to:

$$D_{LS}(spec1\|spec2) = D_{LS}(spec2\|spec1) = \sqrt{\sum 10 \times log_{10}(\frac{spec1}{spec2})^2}$$

If `scale = TRUE` the distance is divided by the length of `spec1` (or `spec2`).

Value

A numeric vector of length 1 returning the D distance.

Note

The function works for both Hz and (htk-)mel scales.

Note

The distance is symmetric.

Author(s)

Jerome Sueur, improved by Laurent Lellouch

See Also

`ks.dist`, `kl.dist`, `itakura.dist`, `simspec`, `diffspec`

Examples

```
# Comparison of two spectra
data(tico)
tico1 <- spec(tico, at=0.65, plot=FALSE)
tico2 <- spec(tico, at=1.1, plot=FALSE)
logspec.dist(tico1, tico2)
logspec.dist(tico1, tico2, scale=TRUE)
```

lts *Long-term spectrogram*

Description

A spectrogram computed over several survey files obtained with a Wildlife Acoustics SongMeter recorder

Usage

```
lts(dir, f, wl = 512,
wn = "hanning", ovlp = 0, rmoffset = TRUE, FUN = mean, col = spectro.colors(30),
fftw = FALSE, norm = FALSE, verbose = TRUE,
tlab = "Time", ntann = NULL, flab = "Frequency (kHz)",
recorder = c("songmeter", "audiomoth"), plot = TRUE, ...)
```

Arguments

dir	a character vector, the path to the directory where the .wav files are stored or directly the names of the .wav files to be processed.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in the .wav files contained in dir.
wl	window length for the analysis (even number of points) (by default = 512).
wn	window name, see ftwindow (by default "hanning").
ovlp	overlap between two successive windows (in %).
rmoffset	a logical to sepcify whether DC offset should be removed. By default TRUE.
FUN	the function to apply to compute the successive frequency spectra, by default mean for a mean spectrum but could be other as median or var for a median spectrum or variance spectrum.
col	a list of colors or the color palette with a number of colors
fftw	if TRUE calls the function FFT of the library fftw. See Notes.
norm	a logical, to specify if each mean spectrum should be normalised between 0 and 1 (default FALSE) before to concatenate the image.
verbose	a logical, if TRUE (default) the file number and name processed are displayed in the console.
tlab	label of the time axis.
ntann	a numeric of length 1, the number of axis annotations (all annotations by default).
flab	label of the frequency axis.
recorder	the type of automatic recorder used, either a Wildlife SongMeter or a Open Audio deveices Audiomoth.
plot	logical, if TRUE plots the spectrogram (by default TRUE).
...	other image graphical parameters.

Details

The function reads each .wav file and computes its mean spectrum with `meanspec`. The successive mean spectra are then concatenated into a single image with the function `image`. The parameters `wl`, `ovlp`, and `wn` are those of the function `meanspec`.

Value

This function returns a list of three items:

`time`	a numeric vector corresponding to the time axis.
`freq`	a numeric vector corresponding to the frequency axis.
`amp`	a numeric or a complex matrix corresponding to the amplitude values. Each column is a Fourier transform of length `wl/2`.

Author(s)

Jerome Sueur

See Also

`spectro`, `meanspec`, `image`, `spectro3D`, `ggspectro`, `songmeter`, `audiomoth`

Examples

```
## Not run:
## if 'dir' contains a set of files recorded with a Wildlife Acoustics
# songmeter recorder then a direct way to obtain
# the spectrogram of all .wav files is
dir <- "pathway-to-directory-containing-wav-files"
lts(dir)
# to normalise each mean spectrum
lts(dir, norm=TRUE)
# to change the STFT parameters used to obtain each mean spectrum
lts(dir, wl=1024, wn="hamming", ovlp=50)
# to change the colors and the number of time labels and to make it quiet
lts(dir, col=cm.colors(20), ntann=10, verbose=FALSE)
## direct use of files names stored in the working directory
files <- c("S4A09154_20190213_150000.wav", "S4A09154_20190213_153000.wav",
"S4A09154_20190213_160000.wav", "S4A09154_20190213_163000.wav",
"S4A09154_20190213_170000.wav", "S4A09154_20190213_173000.wav",
"S4A09154_20190213_180000.wav", "S4A09154_20190213_183000.wav",
"S4A09154_20190213_190000.wav", "S4A09154_20190213_193000.wav")
lts(files)

## End(Not run)
```

M *Median of the amplitude envelope*

Description

This function computes an acoustic index based on the median of the amplitude envelope.

Usage

```
M(wave, f, channel = 1, envt = "hil", plot = FALSE, ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
envt	the type of envelope to be used: either "abs" for absolute amplitude envelope or "hil" for Hilbert amplitude envelope. See env.
plot	logical, if TRUE returns a plot of the amplitude envelope of wave (by default FALSE).
...	other env parameters, in particular smoothing parameters. See env.

Details

This amplitude index M is computed according to:

$$M = \bar{A}(t) \times 2^{1-depth}$$

with

$$0 \leq M \leq 1$$

where A(t) is the amplitude envelope and depth is the signal digitization depth in number of bits.

Value

A numeric vector of length 1 between 0 and 1, without unit.

Author(s)

Jerome Sueur and Marion Depraetere

References

Depraetere M, Pavoine S, Jiguet F, Gasc A, Duvail S, Sueur J (2012) Monitoring animal diversity using acoustic indices: implementation in a temperate woodland. *Ecological Indicators*, **13**, 46-54.

See Also

env, AR

Examples

```
data(tico)
M(tico)
# smoothing the amplitude may change slightly the result
M(tico, msmooth=c(500,50), plot=TRUE)
```

---

meandB *Mean of dB values*

Description

This function calculates the mean of dB values

Usage

```
meandB(x, level="IL")
```

Arguments

x	a numeric vector or a numeric matrix.
level	intensity level ("IL") or sound pressure level ("SPL")

Details

The mean of dB values is not linear. See examples.

Value

A numeric vector of length 1 is returned.

Author(s)

Jerome Sueur and Zev Ross

References

Hartmann, W. M. 1998 *Signals, sound and sensation*. New York: Springer.

See Also

sddB, moredB, convSPL, dBweight

Examples

```
meandB(c(89,90,95))
```

meanspec *Mean frequency spectrum of a time wave*

Description

This function returns the mean frequency spectrum (i.e. the mean relative amplitude of the frequency distribution) of a time wave. Results can be expressed either in absolute or dB data.

Usage

```
meanspec(wave, f, channel = 1, wl = 512, wn = "hanning", ovlp = 0, fftw = FALSE,
norm = TRUE, PSD = FALSE, PMF = FALSE, FUN = mean, correction = "none", dB = NULL,
dBref = NULL, from = NULL, to = NULL, identify = FALSE,
col = "black", cex = 1, plot = 1, flab = "Frequency (kHz)",
alab = "Amplitude", flim = NULL, alim = NULL, type ="l", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
wl	length of the window for the analysis (even number of points, by default = 512).
wn	window name, see ftwindow (by default "hanning").
ovlp	overlap between two successive analysis windows (in %).
fftw	if TRUE calls the function FFT of the library fftw. See Notes of spectro.
norm	if TRUE the mean spectrum is normalised (i. e. scaled) by its maximum.
PSD	if TRUE return Power Spectra Density, *i. e.* the square of the spectra.
PMF	if TRUE return Probability Mass Function, *i. e.* the probability distribution of frequencie.
FUN	the function to apply on the rows of the STFT matrix, by default mean for a mean spectrum but could be other as median or var for a median spectrum or variance spectrum.
correction	a character vector of length 1 to apply an amplitude ("amplitude") or an energy ("energy") correction to the FT window. This argument is useful only when one wish to obtain absolute values that is when norm=FALSE and PMF=FALSE. By default no correction is applied ("none").
dB	a character string specifying the type dB to return: "max0" for a maximum dB value at 0, "A", "B", "C", "D", and "ITU" for common dB weights.
dBref	a dB reference value when dB is not NULL. NULL by default but should be set to 2*10e-5 for a 20 microPa reference (SPL).
from	start mark where to compute the spectrum (in s).

to	end mark where to compute the spectrum (in s).
identify	to identify frequency and amplitude values on the plot with the help of a cursor.
col	colour of the spectrum.
cex	pitch size.
plot	if 1 returns frequency on x-axis, if 2 returns frequency on y-axis, (by default 1).
flab	title of the frequency axis.
alab	title of the amplitude axis.
flim	range of frequency axis (in kHz).
alim	range of amplitude axis.
type	if plot is TRUE, type of plot that should be drawn. See plot for details (by default "l" for lines).
...	other plot graphical parameters.

Details

See examples of spec. This function is based on fft.

Value

If plot is FALSE, meanspec returns a two columns matrix, the first column corresponding to the frequency axis, the second column corresponding to the amplitude axis.
If identify is TRUE, spec returns a list with two elements:

freq	the frequency of the points chosen on the spectrum
amp	the relative amplitude of the points chosen on the spectrum

Warning

The argument peaks is no more available (version > 1.5.6). See the function fpeaks for peak(s) detection.

Note

The argument fftw can be used to try to speed up process time. When set to TRUE, the Fourier transform is computed through the function FFT of the package fftw. This pacakge is a wrapper around the fastest Fourier transform of the free C subroutine library FFTW (http://www.fftw.org/). FFT should be then installed on your OS.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

spec,fpeaks, localpeaks, dynspec, corspec, diffspec, simspec, fft.

Examples

```
data(orni)
# compute the mean spectrum of the whole time wave
meanspec(orni,f=22050)
# compute the mean spectrum of a time wave section (from 0.32 s to 0.39 s)
meanspec(orni,f=22050,from=0.32,to=0.39)
# different window lengths
op<-par(mfrow=c(3,1))
meanspec(orni,f=22050,wl=256)
title("wl=256")
meanspec(orni,f=22050,wl=1024)
title("wl=1024")
meanspec(orni,f=22050,wl=4096)
title("wl=4096")
par(op)
# different overlap values (almost no effects here...)
op<-par(mfrow=c(3,1))
meanspec(orni,f=22050)
title("ovlp=0")
meanspec(orni,f=22050,ovlp=50)
title("ovlp=50")
meanspec(orni,f=22050,ovlp=95)
title("ovlp=95")
par(op)
# use of flim to zoom in
op<-par(mfrow=c(2,1))
meanspec(orni,f=22050)
title("zoom in")
meanspec(orni,f=22050,wl=512,flim=c(4,6))
par(op)
# comparaison of spectrum and mean spectrum
op<-par(mfrow=c(2,1))
spec(orni,f=22050)
title("spec()")
meanspec(orni,f=22050)
title("meanspec()")
par(op)
# log scale on frequency axis
meanspec(orni, f=22050, log="x")
# median spectrum
meanspec(orni,f=22050, FUN=median)
# variance spectrum
meanspec(orni,f=22050, FUN=var)
```

`mel` *Hertz / Mel conversion*

Description

This function converts Hertz data in Mel data.

Usage

```
mel(x, inverse = FALSE)
```

Arguments

`x`	a value in Hertz (or in Mel if `inverse` is `TRUE`)
`inverse`	logical, if `TRUE` converts the Mel data in Hertz data.

Details

Hertz to mel conversion is computed according to:

$$m = 1127.01048 \times \log\left(1 + \left(\frac{f}{700}\right)\right)$$

with *m* in Mel and *f* in Hertz.

Mel to Hertz conversion (when `inverse` is `TRUE`) is therefore computed according to:

$$f = 700 \times \left(e^{\frac{m}{1127.01048}} - 1\right)$$

with *f* in Hertz and *m* in Mel.

Value

A corresponding **R** object is returned.

Note

The Mel scale is a perceptual scale of pitches judged by listeners to be equal in distance from one another. The name Mel comes from the word melody to indicate that the scale is based on pitch comparisons. The reference point between this scale and normal frequency measurement is defined by equating a 1000 Hz tone, 40 dB above the listener's threshold, with a pitch of 1000 mels.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

References

Stevens, S. S., Volkman, J. and Newman, E. B. 1937. A scale for the measurement of psychological magnitude pitch. *Journal of the Acoustical Society of America*, 8: 185-190.

See Also

`melfilterbank`

Examples

```
x<-seq(0,10000,by=50)
y<-mel(x)
plot(x,y,type="l",xlab = "f (hertz)", ylab = "f (mel)",
  main = "Mel scale", col="red")
```

melfilterbank *Mel-filter bank for MFCC computation*

Description

This functions returns graphically and numerically the Mel-filters used to compute MFCC.

Usage

```
melfilterbank(f = 44100, wl = 1024,
minfreq = 0, maxfreq = f/2, m = 20,
palette, alpha = 0.5, plot = FALSE)
```

Arguments

`f`	sammpling frequency (in Hz).
`wl`	the Fourier window length (in number of samples).
`minfreq`	the minimum (or lower) frequency of the filter bank (in Hz).
`maxfreq`	the maximum (or upper) frequency of the filter bank (in Hz).
`m`	the total number of filters.
`palette`	an optional colour palette if `plot` is `TRUE`.
`alpha`	alpha-transparency when a colour palette is used.
`plot`	if `TRUE` all filters are displayed in a single plot.

Value

A list of 3 items:

`central.freq`	the kHz central frequencies of the filters,
`freq`	the kHz frequency scale,
`amp`	the amplitude of the filters, scaled between 0 and 1.

Note

These triangular filters are used for computing MFCCs.

Author(s)

Jerome Sueur

References

Sharan RV & Moir TJ (2016) Applications and advancements in automatic sound recognition. *Neurocomputing*.

See Also

mel, gammatone

Examples

```
## default values
melfilterbank(plot=TRUE)
## with color surfaces
melfilterbank(palette=cm.colors, plot=TRUE)
## values changed
res <- melfilterbank(f=16000, wl=512, minfreq=300, plot=TRUE)
## plot the 1st filter only
plot(res$freq, res$amp[,1], type="l", xlab="Frequency (kHz)", ylab="Amplitude")
## plot the last filter only
plot(res$freq, res$amp[,ncol(res$amp)], type="l", xlab="Frequency (kHz)", ylab="Amplitude")
## get the kHz central frequencies of the succesive filters
res$central.freq
```

---

micsens *Microphone sensitivity and conversion*

Description

This function converts microphone sensitivity from mV/Pa to dB.

Usage

```
micsens(x, sref = 1, inverse = FALSE)
```

Arguments

`x`	a measured sensitivity in mV/Pa (or in dB if `inverse` is `TRUE`)
`sref`	the sensitivity reference (by default equals to 1 V/Pa)
`inverse`	logical, if `TRUE`, the inverse conversion from dB to mV/Pa is computed.

Details

The sensitivity *S* in dB is calculated according to:

$$S_{dB} = 20 \times log_{10}(\frac{s}{s_{ref}})$$

with *s* the measured sensitivity in mv/Pa and *sref* the reference sensitivity (by default 1 mV/Pa).

Value

A numeric value in dB *re* 1V/Pa with default settings, in mV/Pa if `inverse` is set to `FALSE`.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

convSPL

Examples

```
# conversion of a sensitivity of 2 mV/Pa
micsens(2)
# conversion of a sensitivity of -54 dB re 1V/Pa
micsens(-54,inverse=TRUE)
```

---

moredB *Addition of dB values*

Description

This functions calculates the sum of dB values

Usage

```
moredB(x, level="IL")
```

Arguments

x	a numeric vector or numeric matrix.
level	intensity level ("IL") or sound pressure level ("SPL")

Details

The addition of dB values is not linear. See examples.

Value

A numeric vector of length 1.

Author(s)

Jerome Sueur

References

Hartmann, W. M. 1998 *Signals, sound and sensation*. New York: Springer.

See Also

meandB, sddB, convSPL, dBweight

Examples

```
# two sources of 60 dB give an intensity level of 63 dB
moredB(c(60,60))
# addition of three sources
moredB(c(89,90,95))
```

`mutew` *Replace time wave data by 0 values*

Description

This functions replaces a time wave or a section of a time wave by 0 values. For a time wave describing a sound, this corresponds in muting the sound or a section of it.

Usage

```
mutew(wave, f, channel = 1, from = NULL, to = NULL, choose = FALSE, plot = TRUE,
output = "matrix", ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`from`	start of the silence section (in s).
`to`	end of the silence section (in s).
`choose`	logical, if `TRUE` start (=`from`) and end (=`to`) points can be graphically chosen with a cursor on the oscillogram.
`plot`	logical, if `TRUE` returns an oscillographic plot of `wave` with the new silence section (by default `TRUE`).
`output`	character string, the class of the object to return, either `"matrix"`, `"Wave"`, `"Sample"`, `"audioSample"` or `"ts"`.
`...`	other `oscillo` graphical parameters.

Details

By default, `from` and `from` are `NULL`, this results in completely muting `wave`.

Value

If `plot` is `FALSE`, a new wave is returned. The class of the returned object is set with the argument `output`.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

oscillo, addsilw, cutw, deletew,fadew, pastew, revw, zapsilw

Examples

```
data(tico)
mutew(tico,f=22050,from=0.5,to=0.9)
```

NDSI *Normalized Difference Soundscape Index*

Description

This function computes the Normalized Difference Soundscape Index as described by Kasten et al. (2012).

Usage

```
NDSI(x, anthropophony = 1, biophony = 2:8, max = FALSE)
```

Arguments

x	a two-column numeric matrix computed with soundscapespec.
anthropophony	a numeric vector defining the frequency band(s) of the anthropophony (in kHz).
biophony	a numeric vector defining the frequency band(s) of the biophony (in kHz).
max	a logical, if TRUE then defines the biophony as the maximum - not the sum - of the 2 and 8 kHz frequency bands

Details

NDSI aims at estimating the level of anthropogenic disturbance on the soundscape by computing the ratio of human-generated (anthropophony) to biological (biophony) acoustic components found in field collected sound samples. In terms of frequency, the anthropophony is defined as the [1-2[kHz frequency bin and the biophony as the [2-8[kHz frequency bins of a soundscape frequency spectrum (see soundscapespec).

NDSI is computed according to:

$$NDSI = \frac{(biophony - anthropophony)}{(biophony + anthropophony)}$$

NDSI varies between -1 and +1, where +1 indicates a signal containing no anthropophony.

Value

A numeric vector of length 1 giving the NDSI value.

Author(s)

Jerome Sueur

References

Kasten, E.P., Gage, S.H., Fox, J. & Joo, W. (2012). The remote environmental assessment laboratory's acoustic library: an archive for studying soundscape ecology. *Ecological Informatics*, 12, 50-67.

See Also

`soundscapespec`, `SAX`, `NDSI`

Examples

```
## Note that 'tico' is not a soundscape recording...
data(tico)
spec <- soundscapespec(tico, plot=FALSE)
NDSI(spec)
NDSI(spec, max=TRUE)
```

---

noisew — *Generate noise*

Description

This function generates noise.

Usage

```
noisew(f, d, type="unif", listen = FALSE, output = "matrix")
```

Arguments

`f`	sampling frequency of the signal to be generated (in Hz)
`d`	duration of the signal to be generated.
`type`	a character string to specify the type of noise, either "unif" or "gaussian".
`listen`	if `TRUE` the new sound is played back.
`output`	character string, the class of the object to return, either `"matrix"`, `"Wave"`, `"Sample"`, `"audioSample"` or `"ts"`.

Details

Uniform noise is generated using runif and gaussian noise is based on rnorm

Value

A new wave is returned. The class of the returned object is set with the argument output.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

synth, pulse

Examples

```
# add noise to a synthetic signal
a<-noisew(d=1,f=8000)
b<-synth(f=8000,d=1,cf=2000,plot=FALSE)
c<-a+b
spectro(c,f=8000)
```

---

notefreq *Frequency of a muscical note*

Description

This function computes the frequency of a musical note (Equal temperament)

Usage

```
notefreq(note, ref = 440, octave = 3)
```

Arguments

note	a numerical or a character vector. See Note.
ref	a numerical vector of length 1 for the reference frequency.
octave	a numerical vector of length for the octave number.

Details

The frequency is computed according to:

$$f = ref \times 2^{octave-3+\frac{note-10}{12}}$$

with:
ref = reference frequency,
octave = octave number, and
note = rank of the note along the scale.

Value

The frequency in Hz is returned.

Note

The note can be given in two ways. The first solution is to give the rank of the note along the scale (e.g. rank 10 for A) or to give its names in characters with the following notation: C, D, E, F, G, A, B.

Author(s)

Jerome Sueur

See Also

octaves

Examples

```
# Some notes frequency (use apply-like functions when dealing with character strings)
sapply(c("C", "A", "Gb"), notefreq)

# C major scale plot
n <- 1:12
freq <- notefreq(n)
names <- c("C", "C#", "D", "D#", "E", "F", "F#", "G", "G#", "A", "A#", "B")
plot(n, freq, pch=19, cex=1.5,
     xlab = "Note name",
     ylab = "Frequency (Hz)",
     xaxt="n", las=1, main="Third octave")
axis(side=1, at=n, labels=names)
abline(h=freq, col="lightgrey")

# C major scale sound
f <- 2000 # sampling rate
s <- NULL
for (i in 1:length(freq))
  {
    tmp <- synth(d=0.5, f=f, cf=freq[i])
    s <- pastew(s, tmp, at="start", f)
  }
spectro(s, f, ovlp=75)
```

`octaves` *Octave values*

Description

This functions returns the frequency values of the octaves below and above a specific frequency

Usage

```
octaves(x, below = 3, above = 3)
```

Arguments

`x`	a numeric vector, frequency of the note in Hz or kHz.
`below`	the number of octaves below `x`.
`above`	the number of octaves above `x`.

Value

A numeric vector with the octave series in frequency (Hz or kHz depending on `x` unit).

Author(s)

Jerome Sueur

See Also

`notefreq`

Examples

```
names <- c("C","D","E","F","G","A","B")
values <- c(261.63, 293.66, 329.64, 349.23, 392, 440, 493.88)
res <- sapply(values, FUN=octaves)/1000
op <- par(las=1,mfrow=c(2,1))
par(mar=c(0,4,1,1))
matplot(x=1:7, y=res, t="o", pch=names, xlab="",
    ylab="Frequency (kHz) [linear scale]", col=rainbow(7), xaxt="n")
par(mar=c(4.5,4,0,1))
matplot(x=1:7, y=res, t="o", pch=names, xlab="Octave",
    ylab="Frequency (kHz) [log scale]", col=rainbow(7), ylog=TRUE, log="y")
par(op)
```

`orni` *Song of the cicada Cicada orni*

Description

Recording of a calling song section of the Mediterranean cicada *Cicada orni*.

Usage

```
data(orni)
```

Format

A Wave object.

Details

Duration = 0.719 s. Sampling frequency = 22050 Hz.

Source

Recording by Jerome Sueur.

Examples

```
data(orni)
oscillo(orni,f=22050)
```

---

oscillo *Show a time wave as an oscillogram*

Description

This graphical function displays a time wave as an oscillogram in a single or multi-frame plot. The envelope of the wave can also be shown.

Usage

```
oscillo(wave, f, channel = 1, from = NULL, to = NULL, fastdisp = FALSE,
scroll = NULL, zoom = FALSE, k=1, j=1, cex,
labels = TRUE,  tlab = "Time (s)", alab = "Amplitude",
byrow = TRUE, identify = FALSE, nidentify = NULL,
plot = TRUE, colwave = "black",
coltitle = "black", cextitle = 1.2, fonttitle = 2,
collab = "black", cexlab = 1, fontlab = 1,
colline = "black",
colaxis = "black", cexaxis = 1, fontaxis = 1,
coly0 = "lightgrey",
tcl = 0.5, title = FALSE, xaxt="s", yaxt="n", type="l", bty = "l")
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
from	start of the oscillogram (in s).
to	end of the oscillogram (in s).
fastdisp	faster graphic display for long wave. The oscillogram is displayed/saved faster in the graphic device/ graphic file when set to TRUE, with a cost on graphic resolution.

`scroll`	a numeric of length 1 allowing to move along the time wave using a slider panel. This numeric corresponds to the number of successive windows dividing the time wave.
`zoom`	time zoom in with start and end points chosen on the oscillogram with a cursor.
`k`	number of horizontal sections (by default =1).
`j`	number of vertical sections (by default =1).
`cex`	Pitch size if `type = "p"`.
`labels`	if `TRUE` plots time and amplitude labels (by default `TRUE`).
`tlab`	Label of time axis.
`alab`	Label of amplitude axis.
`byrow`	logical, if `TRUE`, the sections are filled by rows, otherwise the sections are filled by colmuns (by default `TRUE`).
`identify`	returns the time and amplitude coordinates of points chosen with a cursor on the oscillogram.
`nidentify`	a numeric vector of length 1, specifies the number of points to identified on `wave` if `identify` is `TRUE`.
`plot`	logical, if `TRUE` returns an oscillographic or envelope plot of `wave`(by default `TRUE`).
`colwave`	colour of the oscillogram or of the envelope.
`coltitle`	if `title` is `TRUE`, colour of the title.
`cextitle`	character size for the title.
`fonttitle`	font for the title.
`cexlab`	character size for axes labels.
`fontlab`	font for axes labels.
`collab`	colour of axes labels.
`colline`	colour of axes line.
`colaxis`	colour of the axis annotation.
`fontaxis`	font of axis annotation.
`cexaxis`	magnification for axis annotation.
`coly0`	colour of the y=0 line.
`tcl`	length of tick marks.
`title`	`TRUE` to add a title with information on `wave` duration and `f`, `FALSE` to live it blank, or a character string to add any desired title.
`xaxt`	equivalent to `xaxt` of `par` (by default `="s"`).
`yaxt`	equivalent to `yaxt` of `par` (by default `="n"`).
`type`	type of plot, by default `"l"`. Use `"n"` for no plot.
`bty`	the type of box to be drawn around the oscillogram.

Value

Data are returned as one-column matrix if plot is FALSE. identify returns a two-column matrix with the time and amplitude coordinates of points successively chosen on the oscillogram.

Note

zoom is similar to but more visual than from and/or to. zoom and identify do work with a single-frame window only (*i. e.* with k = 1 and j = 1).
Press 'Stop' button of the tools bar after choosing the appropriate points on the oscillogram.

Author(s)

Jerome Sueur <sueur@mnhn.fr> and Caroline Simonis <csimonis@mnhn.fr>.

See Also

dynoscillo, oscilloST, cutw, pastew, timer

Examples

```
data(tico)
# a simple oscillogram of a bird song
oscillo(tico)
# zoom in
op<-par(mfrow=c(4,1),mar=c(4.5,4,2,2))
oscillo(tico,22050,cexlab=0.75)
oscillo(tico,22050,from=0.5,to=0.9,cexlab=0.75)
oscillo(tico,22050,from=0.65,to=0.75,cexlab=0.75)
oscillo(tico,22050,from=0.68,to=0.70,cexlab=0.75)
par(op)
# the same divided in four lines
oscillo(tico,f=22050,k=4,j=1)
# the same divided in different numbers of lines and columns
oscillo(tico,f=22050,k=4,j=4)
oscillo(tico,f=22050,k=2,j=2,byrow=TRUE)
oscillo(tico,f=22050,k=2,j=2,byrow=FALSE)
# overplot of oscillographic and envelope representations
oscillo(tico,f=22050)
par(new=TRUE)
env(tico,f=22050,colwave=2)
# full colour modifications in a two-frame oscillogram
op<-par(bg="grey")
oscillo(tico,f=22050,k=4,j=1,title=TRUE,colwave="black",
    coltitle="yellow",collab="red",colline="white",
    colaxis="blue",coly0="grey50")
par(op)
# change the title
data(orni)
oscillo(orni,f=22050,title="The song of a famous cicada")
# move along the signal using scroll
```

```
## Not run:
require(rpanel)
oscillo(tico,f=22050,scroll=8)
## End(Not run)
```

---

oscilloST *Show a stereo time wave as oscillograms*

Description

This graphical function displays a stereo (2 channels) time wave as an oscillogram in a two-frame plot. The envelope of the wave can also be shown.

Usage

```
oscilloST(wave1, wave2 = NULL, f, from = NULL, to = NULL,
fastdisp = FALSE,
identify = FALSE, plot = TRUE, colwave1 = "black",
colwave2 = "blue", coltitle = "black",
collab = "black", cexlab = 1, fontlab = 1, colaxis = "black",
cexaxis = 1, coly01 = "grey47", coly02 = "black", title = FALSE,
bty = "l")
```

Arguments

wave1	a first R object.
wave2	a second R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
from	start of the oscillogram (in s).
to	end of the oscillogram (in s).
fastdisp	faster graphic display for long wave. The stereo oscillogram is displayed/saved faster in the graphic device/ graphic file when set to TRUE, with a cost on the graphic resolution.
identify	returns the time coordinate of points chosen with a cursor on the bottom oscillogram.
plot	logical, if TRUE returns an oscillographic or envelope plot of wave(by default TRUE).
colwave1	colour of the oscillogram or of the envelope of wave1.
colwave2	colour of the oscillogram or of the envelope of wave2.
coltitle	if title is TRUE, colour of the title.
collab	colour of axes title.
cexlab	character size for axes title.

fontlab	font for axes title.
colaxis	colour of the axes
cexaxis	mangification for axes annotation.
coly01	colour of the y=0 line of wave1.
coly02	colour of the y=0 line of wave1.
title	logical, if TRUE plots the title with information on time and f (by default FALSE).
bty	the type of box to be drawn around the oscillogram.

Value

Data are returned as two-column matrix if plot is FALSE. identify returns a numeric object with the time coordinate of points successively chosen on the bottom oscillogram.

Author(s)

Jerome Sueur and Caroline Simonis.

See Also

oscillo, dynoscillo

Examples

```
a<-synth(f=8000,d=1,cf=2000,am=c(50,10),plot=FALSE)
b<-synth(f=8000,d=1,cf=1000,fm=c(0,0,2000,0,0),plot=FALSE)
oscilloST(a,b,f=8000)
```

---

pastew *Paste a time wave to another one*

Description

This function pastes a first time wave to a second one. The time wave to be pasted, the time wave to be completed and the resulting time wave can be displayed in a three-frame oscillographic plot.

Usage

```
pastew(wave1, wave2, f, channel = c(1,1), at = "end",
join = FALSE, tjunction = 0,
choose = FALSE, plot = FALSE,
marks = TRUE, output = "matrix", ...)
```

Arguments

wave1	a first R object.
wave2	a second R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R objects, by default left channel (1) for each object.
at	wave2 position in seconds where wave1 will be pasted into. Can be also specified as "start", "middle" or "end".
join	if TRUE the two waves will be pasted and jointed by removing the last point of wave2. See examples.
tjunction	a numeric vector to remove clicks at the junction of 'wave1' and 'wave2'. The value specifies the duration in seconds where the real vales will be replaced by a linear interpolation. This duration should be a few milliseconds.
choose	logical, if TRUE the point where wave1 will be pasted into wave2 (=at) can be graphically chosen with a cursor.
plot	logical, if TRUE returns an oscillographic plot of wave1, wave2 and wave1 + wave2 (by default FALSE).
marks	logical, if TRUE shows where wave1 has been pasted (by default TRUE).
output	character string, the class of the object to return, either "matrix", "Wave", "Sample", "audioSample" or "ts".
...	other oscillo graphical parameters.

Details

If plot is TRUE returns a two-frame plot with three waves:
(1) the wave to be pasted (wave1),
(2) the wave to be completed (wave2),
(3) the resulting wave.

Value

If plot is FALSE, a new wave is returned. The class of the returned object is set with the argument output.

Author(s)

Jerome Sueur, improved by Laurent Lellouch

See Also

oscillo, addsilw, cutw, deletew, fadew, mutew , revw, repw, timelapse , zapsilw

Examples

```
data(tico)
# double a data set describing a bird song
a<-pastew(tico,tico,f=22050)
oscillo(a,f=22050)
# a direct way to see what has been pasted
pastew(tico,tico,f=22050,plot=TRUE)
# cut a section and then paste it at the beginning
a<-cutw(tico, f=22050, from=0.5, to=0.9)
pastew(a,tico,f=22050,at="start",plot=TRUE)
# or paste it at a specific location
pastew(a,tico,f=22050,at=1.4,plot=TRUE)
# setting the argument 'join' to TRUE might be useful
# to smooth pasting when some phase problem occur
# generate two sine waves
a <- synth(cf=50, f=400, d=0.1)
b <- synth(cf=100, f=400, d=0.1)
# paste it with 'join' turned to FALSE
# there is a click at the junction between the two waves
pastew(a, b, f=400, plot=TRUE)
# that can be removed by setting 'join' to TRUE
pastew(a, b, f=400, join=TRUE, plot=TRUE)
# or by using the argument 'tjunction'
pastew(a, b, f=400, tjunction=0.01, plot=TRUE)
```

---

peewit *Song of the bird Vanellus vanellus*

Description

Recording of a song emitted by a peewit (lapwing) male *Vanellus vanellus*

Usage

```
data(peewit)
```

Format

A Wave object.

Details

Duration = 0.706 s. Sampling frequency = 22050 hz.

Source

Recording by Thierry Aubin.

Examples

```
data(peewit)
oscillo(peewit,f=22050)
```

---

pellucens *Calling song of the tree cricket Oecanthus pellucens*

Description

Recording of a calling song section emitted by the European tree cricket *Oecanthus pellucens*.

Usage

```
data(pellucens)
```

Format

A Wave object.

Details

Duration = 3.309 s. Sampling frequency = 11025 hz.

Source

Recording by Jerome Sueur.

Examples

```
data(pellucens)
oscillo(pellucens,f=11025)
```

---

phaseplot *Phase-phase 2D or 3D plot of a time wave*

Description

This function returns a 2D or 3D representation of a time wave according to its first, second and possibly third derivatives.

Usage

```
phaseplot(wave, f, channel = 1, dim = 3, plot = TRUE, type = "l",
xlab = "1st derivative",
ylab = "2nd derivative",
zlab = "3rd derivative", ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`dim`	a vector of lenght 1, the number of dimensions of the plot. Can be either 2 or 3.
`plot`	logical, if `TRUE` plots phase-phase plot (by default `TRUE`).
`type`	type of plot that should be drawn. See `plot` for details (by default "l" for lines).
`xlab`	title of the x axis.
`ylab`	title of the y axis.
`zlab`	title of the z axis.
`...`	other `plot` or `plot3d` graphical parameters of the package **rgl**.

Value

If `plot` is `FALSE` then a 2 or 3 column matrix is returned. The position of the column is related to the order of the derivative (*i. e.* first colum = first derivative.

Note

Phase-phase plot can be used to test non-linearity.

Author(s)

Jerome Sueur

References

For use of such plots see: Rice AN, Land BR, Bass AH (2011) - Nonlinear acoustic complexity in a fish 'two-voice' system. *Proceedings of the Royal Society B*, in press.

See Also

`phaseplot2`

Examples

```
## Not run:
require(rgl)
data(tico)
phaseplot(tico)

## End(Not run)
s <- synth(d=0.05, f=44100, cf=440, out="Wave")
n <- noisew(d=0.05, f=44100, out="Wave")
par(mfrow=c(2,1))
phaseplot(s, dim=2)
phaseplot(n, dim=2)
```

phaseplot2 *Phase-phase 2D plot of a time wave*

Description

This functions returns a 2D representation of a time wave against a delayed version of itself.

Usage

```
phaseplot2(wave, f, channel = 1, tau = 1, type = "l",
xlab = "x(t)",
ylab = paste("x(t+", tau, ")", sep = ""), ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
tau	the time delay to apply in number of samples.
type	type of plot that should be drawn. See plot for details (by default "l" for lines).
xlab	title of the x axis.
ylab	title of the y axis.
...	other plot parameters.

Details

The principle consists in displaying in a single x-y graph the original time wave with a delayed version of itself. The delay is controlled with the argument tau that needs to be specified in number of samples. The conversion of tau in second is obtained by calculating tau/f, with f the sampling frequency.

Value

Nothing is returned except an x-y plot.

Note

Phase-phase plot can be used to test non-linearity.

Author(s)

Jerome Sueur

References

Kantz H, Schreiber T (2003) *Non linear time series analysis*. Cambridge University Press.

See Also

phaseplot

Examples

```
s <- synth(d=0.05, f=44100, cf=440, out="Wave")
n <- noisew(d=0.05, f=44100, out="Wave")
par(mfrow=c(2,1))
phaseplot2(s)
phaseplot2(n)
```

---

playlist *Play a list of sound files*

Description

This function works as a playlist, ie it plays back a list of sound files.

Usage

```
playlist(directory, sample = FALSE, loop = 1)
```

Arguments

`directory`	a character vector indicating the path to the directory where sound files to played are saved.
`sample`	a logical, if `TRUE` the order of sounds files to be played back is shuffled.
`loop`	a numeric vector of length 1, number of loops.

Details

The success of using this function depends on the wave player in use. This works particularly well with SoX under Linux. The type of files (.mp3, .wav, .ogg etc) depends on the wave player as well)

Value

None. Listen and enjoy!

Note

The function is mainly based on `play`

Author(s)

Jérôme Sueur

See Also

play, listen

Examples

```
## Not run:
playlist("MyMusic", sample = TRUE, loop=2)

## End(Not run)
```

---

preemphasis *Pre-emphasis speech filter*

Description

A pre-emphasis frequency filter for speech

Usage

```
preemphasis(wave, f, channel = 1, alpha = 0.9,
plot = FALSE, output = "matrix", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
alpha	time constant, see Details.
plot	a logical, if TRUE plots the spectrogram of the filtered wave and the frequency response of the comb filter.
output	character string, the class of the object to return, either 'matrix', 'Wave', 'Sample', 'audioSample' or 'ts'.
...	other arguments to be passed to spectro except scale and osc that are set by default to FALSE.

Details

The function applies a pre-emphasis filter usually applied in speech analysis. The filter is a kind of high-pass frequency filter that amplifies the high-frequency content of the sample. The filter is defined with:

$$y(n) = x(n) - \alpha \times x(n-1)$$

where alpha is a time constant usually set between 0.9 and 1.

The frequency response of the filter is obtained with:

$$H(f) = 1 + a^2 - 2 \times \alpha \times \cos(2 \times \pi \times f/f_s)$$

Value

A new wave is returned. The class of the returned object is set with the argument `output`.

Author(s)

Jerome Sueur

See Also

`bwfilter`, `combfilter`, `ffilter`, `fir`,`lfs`, `afilter`

Examples

```
data(sheep)
fc <- 150
f <- sheep@samp.rate
alpha <- exp(-2*pi*fc/f)
res <- preemphasis(sheep, alpha=alpha, output="Wave")
```

---

pulsew *Generate rectangle pulse*

Description

This function generates a rectangle pulse.

Usage

```
pulsew(dbefore, dpulse, dafter, f, plot = FALSE, output = "matrix", ...)
```

Arguments

`dbefore`	duration of the silent period before the pulse
`dpulse`	duration of the pulse to generate
`dafter`	duration of silent period after the pulse
`f`	sampling frequency of the signal to be generated (in Hz)
`plot`	logical, if `TRUE` returns an oscillographic plot of the pulse generated (by default `FALSE`).
`output`	character string, the class of the object to return, either `"matrix"`, `"Wave"`, `"Sample"`, `"audioSample"` or `"ts"`.
`...`	other `plot` parameters.

Value

If `plot` is `FALSE`, a new wave is returned. The class of the returned object is set with the argument `output`.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

synth, noisew

Examples

```
pulsew(dbefore=0.5,dpulse=0.1,dafter=0.3,f=8000,plot=TRUE)
```

Q *Resonance quality factor of a frequency spectrum*

Description

This function estimates the frequency pureness of a time wave by returning the resonant quality factor Q at a specific dB level.

Usage

```
Q(spec, f = NULL, level = -3, mel = FALSE, plot = TRUE, colval = "red",
cexval = 1, fontval = 1, flab = NULL,
alab = "Relative amplitude (dB)", type = "l", ...)
```

Arguments

spec	a data set resulting of a spectral analysis obtained with spec, or meanspec (in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
f	sampling frequency of the wave used to obtain spec (in Hz). Not necessary if spec is a two columns matrix obtained with spec or meanspec.
level	frequency bandwidth set by an amplitude value relative to spectrum (in dB).
mel	a logical, if TRUE the (htk-)mel scale is used.
plot	logical, if TRUE returns the spectrum with Q plotted (by default TRUE).
colval	colour of plotting Q.
cexval	character size of plotting Q.
fontval	font of plotting Q.
flab	title of the frequency axis.
alab	title of the amplitude axis.
type	if plot is TRUE, type of plot that should be drawn. See plot for details (by default "l" for lines).
...	other plot graphical parameters.

Details

A high Q value indicates a highly resonant system.

Value

A list is returned with the following four items:

`Q`	a numeric vector of length 1 returning the Q factor (no units)
`dfreq`	a numeric vector of length 1 the dominant frequency (kHz)
`fmin`	a numeric vector of length 1 returning the minimum frequency of the -dB level bandwidth (kHz)
`fmax`	a numeric vector of length 1 returning the minimum frequency of the -dB level bandwidth (kHz)
`bwd`	a numeric vector of length 1 returning the bandwidth, i. e. `fmax-fmin` (kHz)

Note

This function is based on an linear interpolation of the spectrum so that the result should be considered as an estimation, not an exact measure.

Author(s)

Jerome Sueur, improved by Laurent Lellouch

See Also

`spec`, `meanspec`, `corspec`, `fft`.

Examples

```
# bird song
data(tico)
t<-spec(tico,f=22050,at=1.1,plot=FALSE,dB="max0")
op<-par(mfrow=c(2,1),las=1)
Q(t,type="l")
Q(t,type="l",xlim=c(3.8,4.2),ylim=c(-60,0))
title("zoom in")
par(op)
# cricket, changing the dB level
data(pellucens)
p<-spec(pellucens,f=11025,at=0.5,plot=FALSE,dB="max0")
op<-par(mfrow=c(3,1))
Q(p,type="l",xlim=c(1.8,2.6),ylim=c(-70,0))
title("level = - 3 (default value)",col.main="red")
Q(p,type="l",level=-6,
    xlim=c(1.8,2.6),ylim=c(-70,0),colval="blue")
title("level = - 6",col.main="blue")
Q(p,type="l",level=-9,
    xlim=c(1.8,2.6),ylim=c(-70,0),colval="green")
title("level = - 9",col.main="green")
par(op)
```

`read.audacity` *Audacity audio markers import*

Description

Read audio markers as exported by Audacity.

Usage

```
read.audacity(file, format)
```

Arguments

`file`	A .txt file produced by Audacity when exporting time or time x frequency markers.
`format`	The format of the file name that will appear in the value, that is in the first column of the data frame returned. if `"dir"` then the full path to the file is returned, if `"base"` only the base name of the file is returned.

Details

Audacity opens the possibility to annotate sound files with a marker channel. These markers can be exported as .txt files. The function `read.audacity` import such .txt files whether they contain time markers or time x frequency markers.

Value

A `data.frame`. The size of the `data.frame` differs whether the .txt file contains time markers or time x frequency markers.
For time markers, the `data.frame` contains 4 columns:

1. `file` returning the name of the input file either with the full path or with the base name only (see argument `format`),
2. `label` the text label,
3. `t1` the start time in seconds,
4. `t2`the end time in seconds.

For time x frequency markers, the `data.frame` contains 6 columns:

1. `file` returning the name of the input file either with the full path or with the base name only (see argument `format`),
2. `label` the text label,
3. `t1` the start time in seconds,
4. `t2` the end time in seconds,
5. `f1` the lower frequency in Hz,
6. `f2` the upper frequency in Hz.

Author(s)

Jerome Sueur

References

Audacity is a free software distributed under the terms of the GNU General Public License.
Web site: https://www.audacityteam.org/

See Also

write.audacity

Examples

```
## Not run:
## If 'markers.txt' is an export of Audacity markers
x <- read.audacity("markers.txt")

## End(Not run)
```

---

repw *Repeat a time wave*

Description

This function repeats a time wave

Usage

```
repw(wave, f, channel = 1, times = 2, join = FALSE, plot = FALSE, output= "matrix", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
times	a numeric of length 1 describing the number of times the wave has to be repeated.
join	if TRUE the last point of wave will be removed for smoothing junction between repetitions. See examples.
plot	logical, if TRUE plots the repeated time wave.
output	character string, the class of the object to return, either "matrix", "Wave", "Sample", "audioSample" or "ts".
...	other oscillo graphical parameters.

Value

If plot is FALSE, a new wave is returned. The class of the returned object is set with the argument output.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

oscillo, addsilw, cutw, deletew, fadew, mutew, pastew, revw, zapsilw

Examples

```
data(tico)
repw(tico,f=22050,plot=TRUE)
# use 'join' for smooth pasting
par(mfrow=c(2,1))
a <- synth(cf=50, f=400, d=0.1)
repw(a, f=400, plot=TRUE)
title(main="join is FALSE")
points(x=0.1, y=0, cex=2, col=2)
repw(a, f=400, join=TRUE, plot=TRUE)
title(main="join is TRUE")
points(x=0.1, y=0, cex=2, col=2)
```

---

resamp *Resample a time wave*

Description

This function resamples (down- or over-samples) a time wave. This corresponds to a sampling frequency change.

Usage

```
resamp(wave, f, g, channel = 1, output="matrix")
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
g	new sampling frequency of wave (in Hz).
channel	channel of the R object, by default left channel (1).
output	character string, the class of the object to return, either "matrix", "Wave", "Sample", "audioSample" or "ts".

Value

If `plot` is `FALSE`, a new wave is returned. The class of the returned object is set with the argument `output`.

Note

Resampling might change frequency properties of the time wave.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

Examples

```
data(peewit)
# downsampling
a<-resamp(peewit,f=22050,g=11025)
# oversampling
b<-resamp(peewit,f=22050,g=44100)
```

---

revw *Time reverse of a time wave*

Description

Reverse the wave along the time axis.

Usage

```
revw(wave, f, channel = 1, env = TRUE, ifreq = TRUE,
plot = FALSE, output = "matrix", ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`env`	logical, if `TRUE` the amplitude envelope is reversed.
`ifreq`	logical, if `TRUE` the instantaneous frequency is reversed.
`plot`	logical, if `TRUE` returns an oscillographic plot of the reversed wave (by default `FALSE`).
`output`	character string, the class of the object to return, either `"matrix"`, `"Wave"`, `"Sample"`, `"audioSample"` or `"ts"`.
`...`	other `oscillo` graphical parameters.

Details

If `plot` is `TRUE` returns an oscillogram of the reversed wave. The amplitude and the instantaneous frequency can be independently reversed thanks to the arguments `env` and `ifreq`. See the examples.

Value

If `plot` is `FALSE`, a new wave is returned. The class of the returned object is set with the argument `output`.

Author(s)

Jerome Sueur `<sueur@mnhn.fr>`

References

Beeman, K. 1998. Digital signal analysis, editing and synthesis *in* Hopp, S. L., Owren, M. J. and Evans, C. S. (Eds) 1998. *Animal acoustic communication*, pp. 59-103. Springer, Berlin, Heidelberg.

See Also

`oscillo`, `addsilw`, `deletew`, `fadew`, `pastew`, `mutew`

Examples

```
data(tico)
# simple reverse
revw(tico,f=22050,plot=TRUE)
# envelope reverse only
revw(tico,f=22050,ifreq=FALSE, plot=TRUE)
# instantaneous frequency reverse only
revw(tico,f=22050,env=FALSE, plot=TRUE)
```

---

rmam — *Remove the amplitude modulations of a time wave*

Description

This functions removes the amplitude modulation of a time wave through the Hilbert amplitude envelope.

Usage

```
rmam(wave, f, channel = 1, plot = FALSE, listen = FALSE, output = "matrix", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
plot	logical, if TRUE returns an oscillographic plot of the nwe time wave (by default FALSE).
listen	if TRUE the new sound is played back.
output	character string, the class of the object to return, either "matrix", "Wave", "Sample", "audioSample" or "ts".
...	other oscillo graphical parameters.

Details

The new time wave is obtained by dividing the original time wave by its Hilbert amplitude envelope.

Value

If plot is FALSE, a new wave is returned. The class of the returned object is set with the argument output.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

References

Mbu Nyamsi, R. G., Aubin, T. & Bremond, J. C. 1994 On the extraction of some time dependent parameters of an acoustic signal by means of the analytic signal concept. Its application to animal sound study. *Bioacoustics*, 5: 187-203.

See Also

hilbert.

Examples

```
# generate a new sound with amplitude modulation
a<-synth(f=8000, d=1, cf=1500, am=c(50,10))
# remove the amplitude modulation and plot the result
rmam(a,f=8000,plot=TRUE)
```

rmnoise *Remove noise*

Description

This function removes background noise by smoothing

Usage

```
rmnoise(wave, f, channel = 1, output = "matrix", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
output	character string, the class of the object to return, either "matrix", "Wave", "Sample", "audioSample" or "ts".
...	other smooth.spline arguments.

Details

This function is based on smooth.spline. You can use the arguments of the later to modify the smoothing.

Value

A new wave is returned. The class of the returned object is set with the argument output.

Note

Low frequency noise might not be removed out properly.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

afilter, noisew

Examples

```
# synthesis of a 440 Hz sound with background noise
n <- noisew(d=1,f=8000)
s <- synth(d=1,f=8000,cf=440)
ns <- n+s
# remove noise (but low frequency content still there)
a <- rmnoise(ns,f=8000)
```

`rmoffset` *Remove the offset of a time wave*

Description

This function removes the offset of a time wave.

Usage

```
rmoffset(wave, f, channel = 1, FUN = mean, plot = FALSE, output = "matrix", ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`FUN`	a function used to apply the offset correction. See Details.
`plot`	logical, if `TRUE` returns an oscillographic plot of the wave after removing the offset (by default `FALSE`).
`output`	character string, the class of the object to return, either `"matrix"`, `"Wave"`, `"Sample"`, `"audioSample"` or `"ts"`.
`...`	other `oscillo` graphical parameters.

Value

The offset is removed by substracting the wave by its mean (argument `FUN`). But other function can be used. For instance, it can be more approriate to use the median to remove the offtset and transients. See Examples.
If `plot` is `FALSE`, a new wave is returned. The class of the returned object is set with the argument `output`.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

`oscillo`

Examples

```
data(tico)
# artifically generates an offset
tico2<-tico+0.1
# see the wave with an offset
oscillo(tico2, f=22050)
# remove the offset with the mean (by default)
rmoffset(tico2, f=22050, plot=TRUE)
# remove the offset with the median
rmoffset(tico2, f=22050, FUN=median, plot=TRUE)
```

---

rms *Root Mean Square*

Description

This function computes the root mean square or quadratic mean.

Usage

```
rms(x, ...)
```

Arguments

x	an R object
...	further arguments passed to mean

Details

The Root Mean Square or quadratic mean is computed according to:

$$RMS = \sqrt{\frac{1}{n} \times \sum_{i=1}^{N} {x_i}^2}$$

Value

A numeric vector of length 1

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

mean

Examples

```
# simple rms
rms(1:10)
# rms of a normalized envelope
data(sheep)
env <- env(sheep, f=8000)
rms(env)
```

`roughness` *Roughness or total curvature*

Description

This function computes the roughness or total curvature of a curve, i.e. of a time wave or of a spectrum

Usage

```
roughness(x, std = FALSE)
```

Arguments

`x`	a vector
`std`	a logical, if set to `TRUE` then `x` is standardized by its maximum.

Details

Roughness or total curvature is the integrated squared second derivative :

$$roughness = \int [D^2 x(t)]^2 \, \mathrm{d}t$$

.

Value

A vector of length 1.

Note

The value has not unit.

Author(s)

Jerome Sueur

References

Ramsay JO, Silverman BW (2005) *Functional data analysis*. Springer, Berlin.

See Also

rugo, rms, sh, th, H.

Examples

```
data(tico)
spec <- meanspec(tico, plot=FALSE)[,2]
roughness(spec)
```

---

rugo *Rugosity of a time wave*

Description

This function computes the rugosity of a time wave or time series

Usage

```
rugo(x, ...)
```

Arguments

x	a vector
...	other mean parameters.

Details

The formula has been slightly modified from Mezquida & Martinez (2009: 826) to fit with the classical definition of the root-mean-square (see rms).
The rugosity is then computed as following:

$$rugo = \sqrt{\sum_{i=1}^{n-1} \frac{(x_{i+1} - x_i)^2}{n}}$$

for a vector x of length n.

Value

A vector of length 1.

Note

The rugosity of a noisy signal will tend to be higher than that of a pure tone signal, all other things being equal.

Author(s)

Jerome Sueur

References

Mezquida DA, Martinez JL (2009) - Platform for bee-hives monitoring based on sound analysis. A perpetual warehouse for swarm's daily activity. *Spanish Journal of Agricultural Research* **7**, 824-828.

See Also

roughness, rms, sh, th, H.

Examples

```
data(tico) ; tico <-tico@left
# rugosity of the original recording normalised
rugo(tico/max(tico))
# synthesis of white noise with the same duration as tico
noise <- noisew(d=length(tico)/22050, f=22050)
# tico is normalised to get similar amplitude with the noise
tico.norm <- tico/max(tico)
# addition of noise to tico
tico.noisy <- tico.norm + 0.5*noise
# new rugosity (higher) on normalised signal
rugo(tico.noisy/max(tico.noisy))
```

---

savewav *Save a .wav file*

Description

Save sound data as .wav file

Usage

```
savewav(wave, f, channel = 1, filename = NULL, rescale = NULL, ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
filename	name of the new file. (by default the name of wave).
rescale	a numeric vector of length 2 giving the lower (negative value) and upper (positive value) amplitude limits of the .wav file to be exported.
...	other arguments to be passed to writeWave.

Details

This function uses three functions from the package **tuneR**: `Wave`, `normalize` and `writeWave`.

Note

The file automatically owerwrites an existing file with the same name.
The amplitude (volume) of the .wav file is normalized by defaults but can be changed with the argument `rescale`. See examples

Author(s)

Jerome Sueur <`sueur@mnhn.fr`>, Ethan C. Brown for the argument 'rescale'

See Also

`export`.

Examples

```
require(tuneR)
a<-synth(f=8000,d=2,cf=2000,plot=FALSE)
# the name of the file is automatically the name of the object
# here: "a.wav"
savewav(a,f=22050)
unlink("a.wav")
# if you wish to change the name, use the 'file' argument
savewav(a,f=22050,file="b.wav")
unlink("b.wav")
# if you wish to change the amplitude of the file, use the argument 'rescale'
# this will turn down the volume of a 16 bit sound
# which amplitude was originally ranging between -2^15 and +2^15
savewav(a, f=22050, file="c.wav", rescale=c(-1500,1500))
unlink("c.wav")
```

`SAX` *Symbolic Aggregate approXimation*

Description

This function converts a numeric times seris into a series of letters with a specific length and alphabet.

Usage

```
SAX(x, alphabet_size, PAA_number,
breakpoints = "gaussian", collapse = NULL)
```

Arguments

`x`	a numeric vector.
`alphabet_size`	a numeric vector of length 1 setting the size of the alphabet.
`PAA_number`	a numeric vector of length 1 setting the number of elements (subsequences) of the Piecewise Aggregate Approximation (PAA).
`breakpoints`	either a character vector ("gaussian", "quantiles") or a numeric vector specifying the sorted values of the breakpoints along the distribution of `x`. See details and examples.
`collapse`	a character vector of length 1, specifying the way to collapse the output letters, see `paste`. By default letters are returned separated.

Details

The SAX method has been developed to reduce the dimensionality of a numerical series into a short chain of characters. SAX follows a two-step process: (1) Piecewise Aggregate Approximation (PAA) and (2) conversion a PAA sequence into a series of letters.

PAA consists in a Z-normalisation, a segmentation of the series of length n into w segments, and the computation of each segment average.

The conversion of the PAA into a series of letters is achieved by attributing with equiprobability each value of the PAA to a letter in reference to a Gaussian distribution. This process therefore assumes that the distribution of the numeric series `x` follows a Gaussian distribution. To relax the constraints of normality we here added the possibility to directly work on the quantiles of the original data distribution or to specify particular breakpoints along the distribution of `x`. See the examples.

Value

A character vector of length (when `collapse` is `NULL`) or number of character (when `collapse` is not `NULL`) corresponding to `PAA_number` argument.

Note

SAX has been used recently to search similar times series in a soundcape data base (Kasten et al., 2012).

Author(s)

Laurent Lellouch. An improvement added by Pavel Senin.

References

Kasten, E.P., Gage, S.H., Fox, J. & Joo, W. (2012). The remote environmental assessment laboratory's acoustic library: an archive for studying soundscape ecology. *Ecological Informatics*, 12, 50 - 67.

Lin, J., Keogh, E., Lonardi, S., Chiu, B., June (2003). A symbolic representation of time series with implications for streaming algorithms. Proceedings of the 8th ACM SIGMOD Workshop on Research Issues in Data Mining and Knowledge Discovery. San Diego, California, USA.

See Also

discrets, symba, soundscapespec

Examples

```
data(tico)
spec <- soundscapespec(tico, plot=FALSE)[,2]
SAX(spec, alphabet = 5, PAA = 10)

# change breakpoints
SAX(spec,  alphabet = 5, PAA = 10, breakpoints="quantiles")
SAX(spec,  alphabet = 5, PAA = 10, breakpoints=c(0, 0.5, 0.75, 1))
SAX(spec,  alphabet = 5, PAA = 10, breakpoints=c(0, 0.33, 0.66, 1))

# different output formats
SAX(spec,  alphabet = 5, PAA = 10, collapse="")
SAX(spec,  alphabet = 5, PAA = 10, collapse="-")
```

---

sddB *Standard deviation of dB values*

Description

This function estimates the standard deviation of dB values

Usage

```
sddB(x, level = "IL")
```

Arguments

x	a numeric vector.
level	intensity level ("IL") or sound pressure level ("SPL")

Details

The standard deviation of dB values is not linear. The function is an estimation not an exact computation which is not possible.

Value

A numeric vector of length 1.

Author(s)

Jérôme Sueur

References

Wikipedia, https://en.wikipedia.org/wiki/Propagation_of_uncertainty

See Also

meandB, moredB, convSPL, dBweight

Examples

```
sddB(c(89,90,95))
sddB(c(89,90,95), level="SPL")
```

---

seedata *A quick look at quantitative data*

Description

See quantitative data at a glance

Usage

```
seedata(data, na.rm = FALSE, col = "grey")
```

Arguments

data	a numeric vector describing quantitative data.
na.rm	logical, if TRUE removes NA.
col	main color.

Details

The red curves depict the corresponding Normal law (same mean and sd as data).

Value

A multi-plot graphic is returned.

Author(s)

Caroline Simonis <csimonis@mnhn.fr> and Jerome Sueur <sueur@mnhn.fr>.

Examples

```
seedata(rnorm(1000))
```

seewave	*Sound analysis and synthesis*

Description

seewave provides functions for analysing, manipulating, displaying, editing and synthesizing time waves (particularly sound). This package processes in particular time analysis (oscillograms and envelopes), spectral content, resonance quality factor, entropy, cross correlation and autocorrelation, zero-crossing, frequency coherence, dominant frequency, analytic signal, 2D and 3D spectrograms.

Details

Package:	seewave
Type:	Package
Version:	2.2.0
Date:	2022-03-04
License:	GPL version 2 or newer
Contributors :	Pierre Aumond, Ethan C. Brown, Guillaume Corbeau, Camille Desjonqueres, Marion Depraetere, Francois Fabianek, Amandine Gasc, Eric Kasten, Laurent Lellouch, Stefanie LaZerte, Jonathan Lees, Jean Marchal, Thibaut Marin-Cudraz, Andre Mikulec, Sandrine Pavoine, David Pinaud, Luis J. Villanueva-Rivera Zev Ross, Carl G. Witthoft, Hristo Zhivomirov
Acknowledgments:	Marianna Anichini, Andrey Anikin, Michel Baylac, Charlotte Cure, Denis Dupeyron, Kurt Fristrup, Arnold Fertin, Sylvain Haupert, Kurt Hornik, Yannick Jadoul, Emiliano A. Laca, Uwe Ligges, Morgane Papin, Emmanuel Paradis, Daniel Ridley-Ellis, Brian Ripley, Jesse Ross, Zev Ross, Pavel Senin, David Savage, Arvind Sowmyan, Simon Urbanek Maria A. Wis, George Zhang
Webpage:	`https://rug.mnhn.fr/seewave/`
Discussion group :	`https://groups.google.com/g/seewave`
Source reference:	Sueur J, Aubin T, Simonis C (2008) - seewave: a free modular tool for sound analysis and synthesis. Bioacoustics, 18: 213-226.
Book:	Sueur J (2018) - Sound analysis and synthesis with R. Springer.

Author(s)

Jerome Sueur <sueur@mnhn.fr>
Thierry Aubin
Caroline Simonis
Maintainer: Jerome Sueur <sueur@mnhn.fr>

---

setenv *Set the amplitude envelope of a time wave to another one*

Description

This function sets the amplitude envelope of a time wave to another one

Usage

```
setenv(wave1, wave2, f, channel = c(1,1), envt="hil", msmooth = NULL, ksmooth = NULL,
plot = FALSE, listen = FALSE, output = "matrix", ...)
```

Arguments

wave1	a first R object.
wave2	a second R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R objects, by default left channel (1) for each object.
envt	the type of envelope to be used for wave2: either "abs" for absolute amplitude envelope or "hil" for Hilbert amplitude envelope. See env.
msmooth	a vector of length 2 to smooth the amplitude envelope of wave2 with a mean sliding window. The first component is the window length (in number of points). The second component is the overlap between successive windows (in %). See env.
ksmooth	kernel smooth via kernel to apply to the amplitude envelope ofwave2. See env.
plot	if TRUE returns the oscillogram of the new time wave (by default FALSE).
listen	if TRUE the new sound is played back.
output	character string, the class of the object to return, either "matrix", "Wave", "Sample", "audioSample" or "ts".
...	other oscillo graphical parameters.

Details

wave1 and wave2 can have different duration (length)
Smoothing the envelope with smooth or ksmooth can significantly change the value returned.

Value

If `plot` is `FALSE`, a new wave is returned. The class of the returned object is set with the argument `output`.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

`drawenv`, `env`, `synth`

Examples

```
data(tico)
a<-synth(d=1,f=22050,cf=1000)
# apply 'tico' ammplitude envelope to 'a' that has a square amplitude envelope
setenv(a,tico,f=22050,plot=TRUE)
# the same but with smoothing the envelope
setenv(a,tico,f=22050,ksmooth=kernel("daniell",50),plot=TRUE)
```

`sfm` *Spectral Flatness Measure*

Description

This function estimates the flatness of a frequency spectrum.

Usage

```
sfm(spec)
```

Arguments

`spec`	a data set resulting of a spectral analysis obtained with `spec` or `meanspec` (not in dB).

Details

SFM is calculated as the ratio between the geometric mean and the arithmetic mean :

$$F = N \times \frac{\sqrt[N]{\prod_{i=1}^{N} y_i}}{\sum_{i=1}^{N} y_i}$$

with:
y = relative amplitude of the *i* frequency,
and *N* = number of frequencies.

Value

A single value varying between 0 and 1 is returned. The value has no unit.

Note

The SFM of a noisy signal will tend towards 1 whereas the SFM of a pure tone signal will tend towards 0.
See sh for another measure of signal noisiness/pureness.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

sh, csh

Examples

```
a<-synth(f=8000,d=1,cf=2000,plot=FALSE)
speca<-spec(a,f=8000,at=0.5,plot=FALSE)
sfm(speca)
# [1] 0
b<-noisew(d=1,f=8000)
specb<-spec(b,f=8000,at=0.5,plot=FALSE)
sfm(specb)
# [1] 0.8233202
```

---

sh *Shannon and Renyi spectral entropy*

Description

This function computes the Shannon or Renyi entropy of a frequency spectrum

Usage

```
sh(spec, alpha = "shannon")
```

Arguments

spec	a data set resulting of a spectral analysis obtained with spec or meanspec (not in dB).
alpha	a character string, by default "shannon" to compute Shannon entropy, "simpson" to compute Simpson entropy otherwise a numeric vector of length 1 with a value superior to 0 but different to 1 to compute Renyi entropy. See the examples.

Details

. Shannon spectral entropy is calculated according to:

$$S = -\frac{\sum_{i=1}^{N} y_i log_2(y_i)}{log_2(N)}$$

. Simpson or Gini-Simpson spectral entropy (or index) is computed according to:

$$GS = 1 - \sum_{i=1}^{N} y_i^2$$

. Renyi spectral entropy of order alpha is calucalted according to:

$$R = \frac{1}{1-\alpha} \times log_2(\sum_{i=1}^{N} y_i^{\alpha})$$

with

$$\alpha \geq 0$$

$$\alpha \neq 1$$

y = relative amplitude of the *i* frequency,

$$\sum_{i=1}^{N} y_i = 1$$

and *N* = number of frequencies.

Value

A numeric vector of length 1 is returned. The value has no unit.

Note

The Shannon entropy scaled between 0 and 1 is also known as Pielou's evenness index

Note

The Shannon spectral entropy of a noisy signal will tend towards 1 whereas the Shannon spectral entropy of a pure tone signal will tend towards 0. See Han *et al*. for details regarding the Renyi entropy.

Author(s)

Jerome Sueur and Laurent Lellouch

References

Han, NC, Muniandy SV, Dayou J (2011) Acoustic classification of Australian anurans based on hybrid spectral-entropy approach. *Applied Acoustics*.

Nunes RR, Almeida de MP, Sleigh JW (2004) - Spectral entropy: a new method for anesthetic adequacy. *Revista Brasileira de Anestesiologia*, **54**, 413-422.

Renyi A (1961) - On measures of information and entropy. Proceedings of the 4th Berkeley Symposium on Mathematics, Statistics and Probability 1960. pp. 547-561.

Simpson EH (1949) - Measurement of diversity. *Nature*, **163**, 688.

See Also

csh,th, H, sfm

Examples

```
a<-synth(f=8000,d=1,cf=2000,plot=FALSE)
speca<-spec(a,f=8000,at=0.5,plot=FALSE)
## Shannon spectral entropy
sh(speca)
# [1] 0.2336412
b<-noisew(d=1,f=8000)
specb<-spec(b,f=8000,at=0.5,plot=FALSE)
sh(specb)
# close to 1
## Renyi spectral entropy
sh(speca, alpha=2)
sh(speca, alpha=3)
```

---

sheep *Sheep bleat*

Description

Recording of a sheep bleat.

Usage

```
data(sheep)
```

Format

A Wave object.

Details

Duration = 2.47 s. Sampling frequency = 8000 hz.

Source

Recording by Frederic Sebe.

Examples

```
data(sheep)
oscillo(sheep,f=8000)
```

---

simspec *Similarity between two frequency spectra*

Description

This function estimates the similarity between two frequency spectra.

Usage

```
simspec(spec1, spec2, f = NULL, mel = FALSE,
norm = FALSE, PMF = FALSE,
plot = FALSE, type = "l",
lty =c(1, 2, 3), col = c(2, 4, 1),
flab = NULL, alab = "Amplitude (percentage)",
flim = NULL, alim = NULL,
title = TRUE, legend = TRUE, ...)
```

Arguments

spec1	a first data set resulting of a spectral analysis obtained with spec or meanspec (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
spec2	a first data set resulting of a spectral analysis obtained with spec or meanspec (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
f	sampling frequency of waves used to obtain spec1 and spec2 (in Hz). Not necessary if spec1 and/or spec2 is a two columns matrix obtained with spec or meanspec.
mel	a logical, if TRUE the (htk-)mel scale is used.
norm	a logical, if TRUE spec1 and spec2 are normalised (scaled) between 0 and 1.
PMF	a logical, if TRUE spec1 and spec2 are transformed into probability mass functions.
plot	logical, if TRUE plots both spectra and similarity function (by default FALSE).

type	if plot is TRUE, type of plot that should be drawn. See plot for details (by default "l" for lines).
lty	a vector of length 3 for the line type of spec1, spec2 and of the similarity function if type="l".
col	a vector of length 3 for the colour of spec1, spec2, and the similarity function.
flab	title of the frequency axis.
alab	title of the amplitude axis.
flim	the range of frequency values.
alim	range of amplitude axis.
title	logical, if TRUE, adds a title with S value.
legend	logical, if TRUE adds a legend to the plot.
...	other plot graphical parameters.

Details

Spectra similarity is assessed according to:

$$S = \frac{100/N}{\times}\sum_{i=1}^{N}\frac{\min spec1(i), spec2(i)}{\max spec1(i), spec2(i)}$$

with S in %.

Value

The similarity index is returned. This value is in %.
When plot is TRUE, both spectra and the similarity function are plotted on the same graph. The similarity index is the mean of this function.

Author(s)

Jerome Sueur, improved by Laurent Lellouch

References

Deecke, V. B. and Janik, V. M. 2006. Automated categorization of bioacoustic signals: avoiding perceptual pitfalls. *Journal of the Acoustical Society of America*, 119: 645-653.

See Also

spec, meanspec, corspec, diffspec, diffenv, kl.dist, ks.dist, logspec.dist, itakura.dist

Examples

```
a<-noisew(f=8000,d=1)
b<-synth(f=8000,d=1,cf=2000)
c<-synth(f=8000,d=1,cf=1000)
d<-noisew(f=8000,d=1)
speca<-spec(a,f=8000,at=0.5,plot=FALSE)
specb<-spec(b,f=8000,at=0.5,plot=FALSE)
specc<-spec(c,f=8000,at=0.5,plot=FALSE)
specd<-spec(d,f=8000,at=0.5,plot=FALSE)
simspec(speca,speca)
simspec(speca,specb)
simspec(speca,specc,plot=TRUE)
simspec(specb,specc,plot=TRUE)
#[1] 12.05652
simspec(speca,specd,plot=TRUE)
## mel scale
require(tuneR)
data(orni)
data(tico)
orni.mel <- melfcc(orni, nbands = 256, dcttype = "t3", fbtype = "htkmel", spec_out=TRUE)
orni.mel.mean <- apply(orni.mel$aspectrum, MARGIN=2, FUN=mean)
tico.mel <- melfcc(tico, nbands = 256, dcttype = "t3", fbtype = "htkmel", spec_out=TRUE)
tico.mel.mean <- apply(tico.mel$aspectrum, MARGIN=2, FUN=mean)
simspec(orni.mel.mean, tico.mel.mean, f=22050, mel=TRUE, plot=TRUE)
```

---

smoothw — *A function to tentativily smooth a time wave*

Description

This function tries to smooth with a sum sliding window a time wave, and then to remove residual noise.

Usage

```
smoothw(wave, f, channel = 1, wl, padding=TRUE, output = "matrix")
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`wl`	window length in number of points (samples).
`padding`	a logical, if `TRUE` add 0 values at the start and end of the file to match `wave` length (duration).
`output`	character string, the class of the object to return, either `"matrix"`, `"Wave"`, `"Sample"`, `"audioSample"` or `"ts"`.

Details

A window slides along the signal and sums up the sample amplitude values. Zero values are added at the end of the wave to keep wave length (duration).

Value

A new wave is returned. The class of the returned object is set with the argument `output`. If `padding` is `TRUE`, the new wave starts and ends up with `0` values to match the size of `wave`.

Warning

This function should be used with care as this kind of filter may change the frequency content of the sound. See the examples section for an illustration.

Author(s)

Jerome Sueur

See Also

`fir`, `filter`

Examples

```
# An example to show that smoothw() may change
# the frequency content of your sound
data(orni)
orni2 <- smoothw(orni, wl=2, out="Wave")
orni10 <- smoothw(orni, wl=10, out="Wave")
orni50 <- smoothw(orni, wl=50, out="Wave")
orni100 <- smoothw(orni, wl=100, out="Wave")
meanspec(orni)
lines(meanspec(orni2, plot=FALSE), col=2)
lines(meanspec(orni10, plot=FALSE), col=3)
lines(meanspec(orni50, plot=FALSE), col=4)
lines(meanspec(orni100, plot=FALSE), col=5)
legend("topright", col=1:5, lty=1, legend=c("original","wl=2","wl=10","wl=50","wl=100"))
```

---

songmeter *Reading and interpreting SongMeter file name*

Description

This function reads and decomposes the files names generated by a SongMeter device, audio digal recorders produced by the society Wildlife Acoustics.

Usage

```
songmeter(x)
```

Arguments

x	a character vector with file names, either `.wac` or `.wav`

Details

The digital recorder SongMeter (either SM2, SM3, or SM4 device model) produced by the society 'Wildlife Acoustics' (https://www.wildlifeacoustics.com/) generates '.wav' files which names include useful information. Here are the character format of the files:

- **SM2 or SM4**: `PREFIX_YYYYMMDD_HHMMSS.wav`
- **SM3**:
  - *without geolocalisation* `PREFIX_XXX_YYYYMMDD_HHMMSS.wav`
  - *with geolocalisation* `PREFIX_XXX_YYYYMMDD$HHMMSS.wav`

with:

- `PREFIX`: prefix set when programming the SongMeter
- `XXX`: microphone information
- `YYYY`: year
- `MM`: month
- `DD`: day
- `HH`: hour
- `MM`: month
- `SS`: minute

This information is read and decomposed by the function songmeter().
Please note that the function does not read the content of audio file but the name of the file.

Value

The function returns a `data.frame` with the following columns:

model	device model, either "SM2/SM4" or "SM3"
prefix	prefix of the file, specifying for instance to recording site
mic	microphone information specifying if the recording is mono left channel ("monoL"), mono right ("monoR") or stereo ("stereo"). This works for SM3 only, NA for SM2
year	year of recording, numeric
month	month of recording, numeric
day	day of recording, numeric
hour	hour of recording, numeric
min	minute of recording, numeric
sec	second of recording, numeric
time	time in `POSIX` format
geo	logical, `TRUE` if the device was GPS synchronized

Note

The file names of Songmeters may change with time. There is no guarantee that the function will be updated on time.

Author(s)

Jerome Sueur

References

See Wildlife Acoustics website for details regarding the SongMeters 2, 3 and 4: https://www.wildlifeacoustics.com/

See Also

songmeterdiag, audiomoth, strptime for the POSIX time format.

Examples

```
file1 <- "MNHN_20141225_234500.wav"      # SM2 file
file2 <- "CNRS_0+1_20130824_153000.wav"  # SM3 file without geolocalisation
file3 <- "PARIS_-0-_20150410$195550.wav" # SM3 file with geolocalisation
file4 <- "MNHN_20141225_234500.txt"      # not a .wav or a .wac file
file5 <- "myfile.wav"                    # not a Wildlife Acoustics filename
files <- c(file1, file2, file3, file4, file5)
songmeter(files)
```

---

songmeterdiag   *Songmeter file diagnostics and diagram*

Description

This function looks for files generated by a SongMeter device (audio digal recorders produced by the society Wildlife Acoustics) and checks for possible missing or small files according to a predefined recording schedule.

Usage

```
songmeterdiag(dir, start, end, frequency,
pch.exi = 1, pch.mis = 19,
col.exi = 1, col.mis = 2,
cex.exi = NULL, cex.mis = 0.5,
limits = FALSE, output="file", plot = FALSE)
```

Arguments

`dir`	a character vector, path to directory(ies) where the .wav files are stored. Typically a "Data" folder as generated by SongMeter devices.
`start`	a character vector, start date/time of the recording schedule as programmed on the SongMeter device, must be in the format "year-month-day hour:minute:second".
`end`	a character vector, end date/time of the recording schedule as programmed on the SongMeter device, must be in the format "year-month-day hour:minute:second".
`frequency`	a numeric vector, frequency of the recording schedule expressed in minute.
`pch.exi`	symbol for plotting the existing file(s).
`pch.mis`	symbol for plotting the missing file(s)
`col.exi`	colour of the symbol for plotting the existing file(s).
`col.mis`	colour of the symbol for plotting the missing file(s).
`cex.exi`	size of the symbol for plotting the existing file(s), by default `NULL` so that the size of the symbol corresponds to the size of the .wav file in Mb divided by the average size of all .wav files found in the directory. If not NA then symbol size as in `plot`.
`cex.mis`	size of the symbol for plotting the missing file(s).
`limits`	a logical, if `TRUE` adds to the plot the limits (start and end date/time) of the recording schedule as programmed on the SongMeter device.
`output`	a character vector of length 1, either "file" or "time" to get the file name or the time slot in POSIXct format respectively.
`plot`	a logical, if `TRUE` plots a time plot indicating the existing and missing files (by default `TRUE`).

Details

The function works for a single or several directories so that the operation of several SongMeters can be compared visually. This function should be helpful to check quickly how the devices worked.

Value

A character vector with the names of the missing files.

Note

The file names of Songmeters may change with time. There is no guarantee that the function will be perfectly updated.

Author(s)

Jerome Sueur and Sylvain Haupert

References

See Wildlife Acoustics website for details regarding the SongMeters 2, 3 and 4: `https://www.wildlifeacoustics.com/`

See Also

songmeter

Examples

```
## Not run:
##################
# simulated data
##################
# a recording schedule programmed on four SongMeters SM4
# named "S4A03895", "S4A03998", "S4A03536", and "S4A04430"
# starting the 1st of January 2019 at 00:00:00
# and stopping the 31st January 2019 at 23:30:00
# with a recording frequency of 30 minutes
# all directories stored in a single directory named "project"
# recorder names
recorders <- c("S4A03895", "S4A03998", "S4A03536", "S4A04430")
n <- length(recorders)
# schedule as programmed on the devices
format <- "
start <- strptime("20190101_000000", format)
end <- strptime("20190131_233000", format)
schedule <- seq(from=start, to=end, by=30*60)
schedule <- paste(format(schedule, "
# directories and files
dir.create("project")
for(i in 1:n) {
dir.create(paste("project", recorders[i], sep="/"))
}
for(i in 1:n) {
file.create(paste("project", recorders[i],
paste(recorders[i], each=schedule, sep="_"), sep="/"))
}
# removing some files to simulate missing files
dirs <- paste("project", recorders, sep="/")
file.remove(paste(dirs[1], dir(dirs[1])[200:500], sep="/"))

######################
# use of the function
######################
# directories where the .wav files are stored (as above)
dirs <- paste("project", recorders, sep="/")
# function call with a plot, cex.exi is here specify because we deal
# with ghost files (the .wav file are not truly created)
res <- songmeterdiag(dirs,
                   start="2019-01-01 00:00:00", end="2019-01-31 23:30:00", frequency=30,
                     cex.exi=1,  plot=TRUE)
# clear out
unlink("project", recursive=TRUE)

## End(Not run)
```

soundscapespec *Soundscape frequency spectrum of a time wave*

Description

This function returns a kHz binned spectrum as described by Kasten et al. (2012) for the description of a soundscape.

Usage

```
soundscapespec(wave, f, channel = 1, wl = 1024, wn = "hamming", ovlp = 50,
plot = TRUE, xlab = "Frequency (kHz)", ylim = c(0, 1), ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`wl`	length of the window for the analysis (even number of points, by default = 1024).
`wn`	window name, see `ftwindow` (by default `"hamming"`).
`ovlp`	overlap between two successive analysis windows (in %), by default = 50%.
`plot`	if `TRUE` returns a barplot.
`xlab`	title of the barplot x axis.
`ylim`	range of the barplot y axis.
`...`	other `barplot` graphical parameters.

Details

The soundscape frequency spectrum is based on the computation of a spectrogram power spectral density using Welch'smethod (Welch & June, 1967). Parameters used in Kasten et al. (2012) were a Hamming window of 1024 samples with 50% of overlap and are used here as default values.

Value

A two-column numeric matrix, the first column returning the frequency (kHz) bands and the second column returning the power value within each frequency band.
A barplot is returned when `plot` is `TRUE`.

Author(s)

Jerome Sueur and Eric Kasten

References

Kasten, E.P., Gage, S.H., Fox, J. & Joo, W. (2012). The remote environmental assessment laboratory's acoustic library: an archive for studying soundscape ecology. *Ecological Informatics*, 12, 50-67.

Welch, P.D., June (1967). The use of the fast Fourier transform for the estimation of power spectra: a method based on time-averaging over short, modified periodograms. *IEEE Transactions on Audio and Electroacoustics*, 15: 70-73.

See Also

spec, meanspec, SAX, NDSI

Examples

```
## Note that 'tico' is not a soundscape recording...
data(tico)
soundscapespec(tico, plot=TRUE, col="darkgreen")
```

---

sox *Calls SoX*

Description

This function calls SoX, the Swiss Army knife of sound processing programs.

Usage

```
sox(command, exename = NULL, path2exe = NULL, option = NULL,
shQuote_type = NULL)
```

Arguments

command	the SoX command to invoke.
exename	a character string specifying the name of the SoX binary file. If NULL, the default name "sox" will be used for Linux OS.
path2exe	a character string giving the path to the SoX binary file g
option	option to be passed to the SoX command
shQuote_type	type of shell quotes ("cmd" or "cmd2", for Windows OS; "sh" or "csh" Unix OS)

Details

See the documentation of SoX for proper use.

Note

Sox must be installed to use this function but not to install the package seewave. As mentioned on the SoX webpage, the primary development platform is Linux. Using SoX with Windows from R might not be straightforward. In particular, it is advisable to pay attention to file path and exe name.

Author(s)

Jerome Sueur, Stefanie LaZerte, Andre Mikulec

References

https://en.wikipedia.org/wiki/SoX

Examples

```
## Not run:
##########
## data ##
##########
## Generate a simple sound file at 440 Hz
s <- synth(cf=440, f=8000, d=1, fm=c(0,0,1000,0,0), output="Wave")
savewav(s, file="mysound.wav")
##############
## Linux OS ##
##############
## Play the file
sox("mysound.wav", exename="play")
## Slow down  the  audio  tempo  (but  not  its  pitch)
sox("mysound.wav myslowsound.wav tempo 0.5")
## Cut the file
sox("myslowsound.wav myslowcutsound.wav trim 0.25 0.75")
################
## Windows OS ##
################
## path with simple slash
path <- "C:/Program Files (x86)/sox-14-4-2"
## or path with double backslash
## path <- "C:\Program Files (x86)\sox-14-4-2"
sox("mysound.wav", path2exe=path, option="-t waveaudio")
## with the option directly passed to the command
sox("mysound.wav -t waveaudio", path2exe=path)
## Slow down  the  audio  tempo  (but  not  its  pitch)
sox("mysound.wav myslowsound.wav tempo 0.5", path2exe=path)
## Cut the file
sox("myslowsound.wav myslowcutsound.wav trim 0.25 0.75", path2exe=path)
###########
## clean ##
###########
file.remove("mysound.wav", "myslowsound.wav", "myslowcutsound.wav")

## End(Not run)
```

---

spec *Frequency spectrum of a time wave*

Description

This function returns the frequency spectrum (*i.e.* the relative amplitude of the frequency content) of a time wave. Results can be obtained either as absolute or dB data.

Usage

```
spec(wave, f, channel = 1, wl = 512, wn = "hanning", fftw = FALSE, norm = TRUE,
scaled = FALSE, PSD = FALSE, PMF = FALSE, correction="none", dB = NULL, dBref = NULL,
at = NULL, from = NULL, to = NULL,
identify = FALSE, col = "black", cex = 1,
plot = 1, flab = "Frequency (kHz)",
alab = "Amplitude", flim = NULL,
alim = NULL, type="l",...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`wl`	if `at` is not null, length of the window for the analysis (by default = 512).
`wn`	window name, see `ftwindow` (by default `"hanning"`).
`fftw`	if `TRUE` calls the function `FFT` of the library `fftw` for faster computation. See Notes of the function `spectro`.
`norm`	if `TRUE` the spectrum is normalised by its maximum.
`scaled`	if `TRUE` the spectrum is scaled by the length of the FFT.
`PSD`	if `TRUE` return Power Spectrum Density, *i. e.* the square of the spectrum.
`PMF`	if `TRUE` return Probability Mass Function, *i. e.* the probability distribution of frequencies.
`correction`	a character vector of length 1 to apply an amplitude ("amplitude") or an energy ("energy") correction to the FT window. This argument is useful only when one wish to obtain absolute values that is when `norm=FALSE`, `scaled=FALSE`, and `PMF=FALSE`. By default no correction is applied ("none").
`dB`	a character string specifying the type dB to return: "max0" for a maximum dB value at 0, "A", "B", "C", "D", and "ITU" for common dB weights.
`dBref`	a dB reference value when `dB` is not `NULL`. `NULL` by default but should be set to 2*10e-5 for a 20 microPa reference (SPL).
`at`	position where to compute the spectrum (in s).
`from`	start mark where to compute the spectrum (in s).
`to`	end mark where to compute the spectrum (in s).
`identify`	to identify frequency and amplitude values on the plot with the help of a cursor.
`col`	colour of the spectrum.
`cex`	pitch size of the spectrum.

plot	if 1 returns frequency on x-axis, if 2 returns frequency on y-axis, (by default 1).
flab	title of the frequency axis.
alab	title of the amplitude axis.
flim	range of frequency axis.
alim	range of amplitude axis.
type	if plot is TRUE, type of plot that should be drawn. See plot for details (by default "l" for lines).
...	other plot graphical parameters.

Details

If at, from or to are FALSE then spec computes the spectrum of the whole signal.

Value

This function returns a two-column matrix, the first column corresponding to the frequency axis, the second column corresponding to the amplitude axis.
If identify is TRUE, spec returns a list with two elements:

freq	the frequency of the points chosen on the spectrum
amp	the relative amplitude of the points chosen on the spectrum

Warning

The argument peaks is no more available (version > 1.5.6). See the function fpeaks for peak(s) detection.

Note

This function is based on fft.

Author(s)

Jerome Sueur

See Also

meanspec, fpeaks, localpeaks, dynspec, corspec, fft.

Examples

```
data(tico)
# spectrum of the whole signal, in absolute or dB amplitude,
# horizontaly or vertically
op<-par(mfrow=c(2,2))
spec(tico,f=22050)
spec(tico,f=22050,col="red",plot=2)
spec(tico,f=22050,dB="max0",col="blue")
spec(tico,f=22050,dB="max0",col="green",plot=2)
par(op)
# an indirect way to compare spectra
a<-spec(tico,f=22050,wl=512,at=0.2,plot=FALSE)
b<-spec(tico,f=22050,wl=512,at=0.7,plot=FALSE)
c<-spec(tico,f=22050,wl=512,at=1.1,plot=FALSE)
d<-spec(tico,f=22050,wl=512,at=1.6,plot=FALSE)
all<-cbind(a[,2],b[,2],c[,2],d[,2])
matplot(x=a[,1],y=all,yaxt="n",
    xlab="Frequency (kHz)",ylab="Amplitude",xaxs="i",type="l")
legend(8,0.8,c("Note A","Note B", "Note C", "Note D"),bty="o",
    lty=c(1:4),col=c(1:4))
# spectrum from a particular position to another one
op<-par(mfrow=c(2,1))
oscillo(tico,f=22050)
abline(v=c(0.5,0.9),col="red",lty=2)
spec(tico,f=22050,wl=512,from=0.5,to=0.9,col="red")
title("Spectrum of the note B")
par(op)
# spectrum and spectrogram
data(orni)
orni1<-cutw(orni,f=22050,from=0.32,to=0.39)
layout(matrix(c(1,2),nc=2),widths=c(3,1))
par(mar=c(5,4,3,0.5))
spectro(orni1,f=22050,wl=128,zp=8,ovlp=85,scale=FALSE)
par(mar=c(5,1,3,0.5))
spec(orni1,f=22050,col="red",plot=2,flab="",yaxt="n")
```

---

specflux *Spectral flux*

Description

Compute spectral flux

Usage

```
specflux(wave, f, channel = 1,
         wl = 512, ovlp = 0, wn = "rectangle", flim = NULL,
         norm = FALSE,  p = 2,
         plot = TRUE, xlab = "Times (s)", ylab = "Flux", type = "l", ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`wl`	window length for the analysis (even number of points) (by default = 512).
`ovlp`	overlap between two successive windows (in %).
`wn`	window name, see `ftwindow` (by default `"rectangle"`).
`flim`	a numeric vector of length 2 to select a frequency band (in kHz).
`norm`	if is `TRUE` then the normalised spectra are used. The spectra are normalised by their sum.
`p`	the norm type, by default = 2.
`plot`	logical, if `TRUE` the spectral flux is displayed against time (s) (by default `TRUE`).
`xlab`	title of the x axis.
`ylab`	title of the y axis.
`type`	if `plot` is `TRUE`, type of plot that should be drawn. See `plot` for details (by default "l" for lines).
`...`	other `plot` parameters.

Details

The spectral flux (F) is the sum of the time (t) derivative of the columns – that is the successive spectra – (s) of the normalized short-term Fourier transform (z).
F is then computed according to:

$$F = (\sum |s(t+1) - s(t)|^p)^{\frac{1}{p}}$$

Value

When `plot` is `FALSE`, `specflux` returns a two-column matrix, the first column being time in seconds (*x*-axis) and the second column being the spectral flux (*y*-axis) computed along time.

Note

The sum of the successive spectral flux values could be used as an ecoacoustic index, quite close to the acoustic complexity index (`ACI`). See examples.

Author(s)

Jérôme Sueur

References

Scheirer E, Slaney M (1997). Construction and evaluation of a robust multifeature speech/music discriminator. *IEEE International Conference on Acoustics, Speech, and Signal Processing*, 2, 1221-1224.

See Also

spectro, ACI

Examples

```
## default use
data(tico)
specflux(tico)
## norm 1
specflux(tico, p = 1)
## frequency limit between 2 and 4 kHz
specflux(tico, flim = c(2,4))
## index computation
sum(specflux(tico, plot=FALSE)[,2])
```

---

specprop *Spectral properties*

Description

This function returns a list of statistical properties of a frequency spectrum.

Usage

```
specprop(spec, f=NULL,
str = FALSE, flim=NULL, mel=FALSE,
plot = FALSE, type = "l", xlab=NULL, ylab = NULL,
col.mode = 2, col.quartiles = 4, ...)
```

Arguments

spec	a data set resulting of a spectral analysis obtained with spec or meanspec (not in dB).
f	sampling frequency of spec (in Hz).
str	logical, if TRUE returns the results in a structured table.
flim	a vector of length 2 to specifgy the frequency limits of the analysis (in kHz)
mel	a logical, if TRUE the (htk-)mel scale is used.
plot	if 1 returns the spectrum , if 2 returns the cumulative spectrum, both of them with the first quartile, the third quartile, the median and the mode plotted (by default FALSE).
type	if plot is TRUE, type of plot that should be drawn. See plot for details (by default "l" for lines).
xlab	label of the x axis.
ylab	label of the y axis.
col.mode	colour of the mode segments (by default blue).
col.quartiles	colour of the quartiles segments (by default red).
...	other arguments to be passed to plot

Details

The spectrum is converted in a probability mass function (PMF).
If a selected value has to be selected with $, the argument `str` has to be set to `FALSE`.

Value

A list of 15 values is returned

`mean`	mean frequency (see `mean`)
`sd`	standard deviation of the mean (see `sd`)
`sem`	standard error of the mean
`median`	median frequency (see `median`)
`mode`	mode frequency, *i.e.* the dominant frequency
`Q25`	first quartile (see `quantile`)
`Q75`	third quartile (see `quantile`)
`IQR`	interquartile range (see `IQR`)
`cent`	centroid, see `note`
`skewness`	skewness, a measure of asymmetry, see `note`
`kurtosis`	kurtosis, a measure of peakedness, see `note`
`sfm`	spectral flatness measure (see `sfm`)
`sh`	spectral entropy (see `sh`)
`prec`	frequency precision of the spectrum

Note

Centroid is computed according to:

$$C = \sum_{i=1}^{N} x_i \times y_i$$

with:
x = frequencies, y = relative amplitude of the i frequency,
N = number of frequencies.

Skewness is computed according to:

$$S = \frac{\sum_{i=1}^{N}(x_i - \bar{x})^3}{N-1} \times \frac{1}{\sigma^3}$$

.
S < 0 when the spectrum is skewed to left,
S = 0 when the spectrum is symetric,
S > 0 when the spectrum is skewed to right.
Spectrum asymmetry increases with |S|.

Kurtosis is computed according to:

$$K = \frac{\sum_{i=1}^{N}(x_i - \bar{x})^4}{N-1} \times \frac{1}{\sigma^4}$$

.

K < 3 when the spectrum is platikurtic, *i.e.* it has fewer items at the center and at the tails than the normal curve but has more items in the shoulders,
K = 3 when the spectrum shows a normal shape,
K > 3 when the spectrum is leptokurtic, *i.e.* it has more items near the center and at the tails, with fewer items in the shoulders relative to normal distribution with the same mean and variance.

Author(s)

Jerome Sueur and Caroline Simonis, and a patch by Jesse Ross (Dec. 2012)

Examples

```
data(orni)
a<-meanspec(orni,f=22050,plot=FALSE)
specprop(a,f=22050)
# to get a single measure of the list
specprop(a,f=22050)$mode
# to get the results structured
specprop(a,f=22050,str=TRUE)
# to limit the analysis between 4 and 6 kHz
specprop(a,f=22050,flim=c(4,6),str=TRUE)
# plots
specprop(a,f=22050,plot=1)
specprop(a,f=22050,plot=2)
# (htk-)mel scale
require(tuneR)
mel <- melfcc(orni, nbands = 256, dcttype = "t3", fbtype = "htkmel", spec_out=TRUE)
melspec.mean <- apply(mel$aspectrum, MARGIN=2, FUN=mean)
specprop(melspec.mean, f=22050, mel=TRUE)
# be aware that flim is always given in kHz even if mel=TRUE
specprop(melspec.mean, f=22050, flim=c(4,6), mel=TRUE, plot=TRUE)
```

`spectro` *2D-spectrogram of a time wave*

Description

This function returns a two-dimension spectrographic representation of a time wave. The function corresponds to short-term Fourier transform. An amplitude contour plot can be overlaid.

Usage

```
spectro(wave, f, channel = 1, wl = 512, wn = "hanning", zp = 0,
ovlp = 0, noisereduction = NULL, fastdisp = FALSE,
complex = FALSE, norm = TRUE, correction="none",
fftw = FALSE, dB = "max0", dBref = NULL, plot = TRUE,
flog = FALSE, grid = TRUE, osc = FALSE, scale = TRUE, cont = FALSE,
collevels = NULL, palette = spectro.colors,
contlevels = NULL, colcont = "black",
colbg = "white", colgrid = "black",
colaxis = "black", collab="black",
cexlab = 1, cexaxis = 1,
tlab = "Time (s)",
flab = "Frequency (kHz)",
alab = "Amplitude",
scalelab = "Amplitude\n(dB)",
main = NULL,
scalefontlab = 1, scalecexlab =0.75,
axisX = TRUE, axisY = TRUE, tlim = NULL, trel = TRUE,
flim = NULL, flimd = NULL,
widths = c(6,1), heights = c(3,1),
oma = rep(0,4),
listen=FALSE,
...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`wl`	window length for the analysis (even number of points) (by default = 512).
`wn`	window name, see `ftwindow` (by default `"hanning"`).
`zp`	zero-padding (even number of points), see `Details`.
`ovlp`	overlap between two successive windows (in %).
`noisereduction`	a numeric vector of length 1, if `1` a noise reduction is applied along the rows of the spectrogram, if `2` a noise reduction applied along the columns. See `Details`.
`fastdisp`	faster graphic display for long `wave`. The spectrogram/oscillogram is displayed/saved faster in the graphic device/ graphic file when set to `TRUE`, with a cost on graphical resolution.
`complex`	if `TRUE` the STFT will be returned as complex numbers.
`norm`	if `TRUE` the STFT is normalised (i. e. scaled) by its maximum.
`correction`	a character vector of length 1 to apply an amplitude ("amplitude") or an energy ("energy") correction to each FT window. This argument is useful only when one wish to obtain absolute values that is when `norm=FALSE`. By default no correction is applied ("none").

`fftw`	if `TRUE` calls the function `FFT` of the library `fftw`. See Notes.
`dB`	a character string specifying the type dB to return: "max0" (default) for a maximum dB value at 0, "A", "B", "C", "D", and "ITU" for common dB weights. If set to `NULL`, then a linear scale is used.
`dBref`	a dB reference value. `NULL` by default but should be set to 2*10e-5 for a 20 microPa reference.
`plot`	logical, if `TRUE` plots the spectrogram (by default `TRUE`).
`flog`	a logical to plot the frequency on a logarithmic scale.
`grid`	logical, if `TRUE` plots a y-axis grid (by default `TRUE`).
`osc`	logical, if `TRUE` plots an oscillogram beneath the spectrogram (by default `FALSE`).
`scale`	logical, if `TRUE` plots a dB colour scale on the right side of the spectrogram (by default `TRUE`).
`cont`	logical, if `TRUE` overplots contour lines on the spectrogram (by default `FALSE`).
`collevels`	a set of levels which are used to partition the amplitude range of the spectrogram (in dB).
`palette`	a color palette function to be used to assign colors in the plot, see `Details`.
`contlevels`	a set of levels which are used to partition the amplitude range for contour overplot (in dB).
`colcont`	colour for `cont` plotting.
`colbg`	background colour.
`colgrid`	colour for `grid` plotting.
`colaxis`	color of the axes.
`collab`	color of the labels.
`cexlab`	size of the labels.
`cexaxis`	size of the axes.
`tlab`	label of the time axis.
`flab`	label of the frequency axis.
`alab`	label of the amplitude axis.
`scalelab`	amplitude scale label.
`main`	label of the main title.
`scalefontlab`	font of the amplitude scale label.
`scalecexlab`	cex of the amplitude scale label.
`axisX`	logical, if `TRUE` plots time X-axis (by default `TRUE`).
`axisY`	logical, if `TRUE` plots frequency Y-axis (by default `TRUE`).
`tlim`	modifications of the time X-axis limits.
`trel`	time X-axis with a relative scale when `tlim` is not null, *i.e.* relative to `wave`.
`flim`	modifications of the frequency Y-axis limits (in kHz).
`flimd`	dynamic modifications of the frequency Y-axis limits. New `wl` and `ovlp` arguments are applied to increase time/frequency resolution.

widths	a vector of length 2 to control the relative widths of columns on the device when scale is TRUE.
heights	a vector of length 2 to control the relative heights of rows on the device when osc is TRUE.
oma	a vector of length 4 to control the size of outer margins when either scale or osc is TRUE.
listen	if TRUE the sound is played back (by default FALSE).
...	other contour and oscillo graphical parameters.

Details

Following Heisenberg uncertainty principle, the short-term Fourier transform cannot be precised in both time and frequency. The temporal and frequency precisions of the function are actually dependent of the wl value. Choosing a high wl value will increase the frequency resolution but reduce the temporal one, and *vice versa*. The frequency precision is obtained by calculating the ratio f/wl, and the temporal precision is obtained by calculating the reverse ratio wl/f. This problem can be reduced in some way with zp that adds 0 values on both sides of the analysis window. This increases frequency resolution without altering time resolution.
Any colour palette can be used. In particular, it is possible to use other palettes coming with **seewave**: temp.colors, reverse.gray.colors.1, reverse.gray.colors.2, reverse.heat.colors, reverse.terrain.colors, reverse.topo.colors, reverse.cm.colors corresponding to the reverse of heat.colors, terrain.colors, topo.colors, cm.colors.
Use locator to identify points. The noise reduction using the argument noisereduction is an image filter, not a signal filter. The principle consists in subtracting each spectrogram row or column by its median. Noise reduction alters energy conservation, it should then be used for visual display only.

Value

This function returns a list of three items:

time	a numeric vector corresponding to the time axis.
freq	a numeric vector corresponding to the frequency axis.
amp	a numeric or a complex matrix corresponding to the amplitude values. Each column is a Fourier transform of length wl/2.

Note

The argument fftw can be used to try to speed up process time. When set to TRUE, the Fourier transform is computed through the function FFT of the package fftw. This pacakge is a wrapper around the fastest Fourier transform of the free C subroutine library FFTW (http://www.fftw.org/). FFT should be then installed on your OS.

Note

This function is based on fft, contour and filled.contour

Author(s)

Jerome Sueur and Caroline Simonis.

References

Hopp, S. L., Owren, M. J. and Evans, C. S. (Eds) 1998. *Animal acoustic communication*. Springer, Berlin, Heidelberg.

See Also

ggspectro, spectro3D, lts, dynspec, wf, oscillo, dBscale, fft.

Examples

```
## Not run:
data(tico)
data(pellucens)
# simple plots
spectro(tico,f=22050)
spectro(tico,f=22050,osc=TRUE)
spectro(tico,f=22050,scale=FALSE)
spectro(tico,f=22050,osc=TRUE,scale=FALSE)
# change the dB scale by setting a different dB reference value (20microPa)
spectro(tico,f=22050, dBref=2*10e-5)
# unnormalised spectrogram with a linear amplitude scale
spectro(tico, dB=NULL, norm=FALSE, scale=FALSE)
# manipulating wl
op<-par(mfrow=c(2,2))
spectro(tico,f=22050,wl=256,scale=FALSE)
title("wl = 256")
spectro(tico,f=22050,wl=512,scale=FALSE)
title("wl = 512")
spectro(tico,f=22050,wl=1024,scale=FALSE)
title("wl = 1024")
spectro(tico,f=22050,wl=4096,scale=FALSE)
title("wl = 4096")
par(op)
# vertical zoom using flim
spectro(tico,f=22050, flim=c(2,6))
spectro(tico,f=22050, flimd=c(2,6))
# a full plot
pellu2<-cutw(pellucens,f=22050,from=1,plot=FALSE)
spectro(pellu2,f=22050,ovlp=85,zp=16,osc=TRUE,
    cont=TRUE,contlevels=seq(-30,0,20),colcont="red",
    lwd=1.5,lty=2,palette=reverse.terrain.colors)
# black and white spectrogram
spectro(pellu2,f=22050,ovlp=85,zp=16,
    palette=reverse.gray.colors.1)
# colour modifications
data(sheep)
spectro(sheep,f=8000,palette=temp.colors,collevels=seq(-115,0,1))
spectro(pellu2,f=22050,ovlp=85,zp=16,
```

```
palette=reverse.cm.colors,osc=TRUE,colwave="orchid1")
spectro(pellu2,f=22050,ovlp=85,zp=16,osc=TRUE,palette=reverse.heat.colors,
colbg="black",colgrid="white", colwave="white",colaxis="white",collab="white")

## End(Not run)
```

---

spectro3D *3D-spectrogram of a time wave*

Description

This function returns a three-dimension spectrographic representation of a time wave. The function corresponds to short-term Fourier transform.

Usage

```
spectro3D(wave, f, channel = 1, wl = 512, wn = "hanning", zp = 0,
ovlp = 0, noisereduction = FALSE, norm = TRUE, correction = "none", fftw = FALSE,
dB = "max0", dBref = NULL, plot = TRUE,
magt = 10, magf = 10, maga = 2,
palette = reverse.terrain.colors)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`wl`	length of the window for the analysis (even number of points).
`wn`	window name, see `ftwindow` (by default `"hanning"`).
`zp`	zero-padding (even number of points), see `Details`.
`ovlp`	overlap between two successive windows (in %).
`noisereduction`	a logical, if `TRUE` a noise reduction is applied.
`norm`	if `TRUE` the STFT is normalised (i. e. scaled) by its maximum.
`correction`	a character vector of length 1 to apply an amplitude ("amplitude") or an energy ("energy") correction to the FT window. This argument is useful only when one wish to obtain absolute values that is when `norm=FALSE`, `scaled=FALSE`, and `PMF=FALSE`. By default no correction is applied ("none").
`fftw`	if `TRUE` calls the function `FFT` of the library `fftw`. See Notes of the `spectro`.
`dB`	a character string specifying the type dB to return: "max0" for a maximum dB value at 0, "A", "B", "C", "D", and "ITU" for common dB weights.
`dBref`	a dB reference value when `dB` is `TRUE`. `NULL` by default but should be set to 2*10e-5 for a 20 microPa reference.

plot	logical, if TRUE plots the spectrogram (by default TRUE).
magt	magnification of the time axis.
magf	magnification of the frequency axis.
maga	magnification of the amplitude axis.
palette	a color palette function to be used to assign colors in the plot, see Details.

Details

Following Heisenberg uncertainty principle, the short-term Fourier transform cannot be precised in both time and frequency. The temporal and frequency precisions of the function are actually dependent of the wl value. Choosing a high wl value will increase the frequency resolution but reduce the temporal one, and *vice versa*. The frequency precision is obtained by calculating the ratio f/wl, and the temporal precision is obtained by calculating the reverse ratio wl/f. This problem can be reduced in some way with zp that adds 0 values on both sides of the analysis window. This increases frequency resolution without altering time resolution.
Any colour palette can be used. In particular, it is possible to use other palettes coming with **seewave**: reverse.gray.colors.1, reverse.gray.colors.2, spectro.colors, temp.colors, reverse.heat.colors, reverse.cm.colors, reverse.topo.colors, corresponding to the reverse of heat.colors,topo.colors, cm.colors.
Use magt, magf and maga to resize the plot.

Value

This function returns a list of three items:

time	a numeric vector corresponding to the time axis.
freq	a numeric vector corresponding to the frequency axis.
amp	a numeric matrix corresponding to the amplitude values. Each column is a Fourier transform of length wl/2.

Note

This function requires **rgl** and is based on fft. See examples of spectro for analysis arguments (wl,zp, ovlp).

Author(s)

Jerome Sueur <sueur@mnhn.fr> and Caroline Simonis <csimonis@mnhn.fr>.

See Also

spectro, ggspectro, lts, dynspec, wf, fft.

Examples

```
## Not run:
require(rgl)
data(tico)
spectro3D(tico,f=22050,wl=512,ovlp=75,zp=16,maga=4,palette=reverse.terrain.colors)
# linear amplitude scale without a normisation of the STFT matrix
# time and frequency scales need to be dramatically amplified
spectro3D(tico, norm=FALSE, dB=NULL, magt=100000, magf=100000)

## End(Not run)
```

`squarefilter` *Frequency square filter*

Description

This function prepares the amplitude profile of a square frequency filter.

Usage

```
squarefilter(f, from = NULL, to = NULL, bandpass = TRUE, wl = 1024)
```

Arguments

`f`	a numeric vector of length 1 for the sampling frequency of the object to be filtered (in Hz).
`from`	a numeric vector for the start frequencies (in Hz) where to apply the filter.
`to`	a numeric vector of the end frequencies (in Hz) where to apply the filter.
`bandpass`	if `TRUE` a band-pass filter is prepared between start and end frequencies (arguments `from` and `to`), if `FALSE` a bandstop filter is prepared.
`wl`	window length of the impulse filter (even number of points).

Value

The function returns a two-column matrix, the first column is the frequency in kHz and the second column is the amplitude of the filter (frequency response of the filter).

Note

This function can be used to prepare bandpass or bandstop filters to be used with `fir` and `ffilter`. See examples.

Author(s)

Laurent Lellouch

See Also

fir, drawfilter, ffilter, combfilter, bwfilter

Examples

```
f <- 44100
a <- noisew(f = f, d = 1)
p <- squarefilter(f, from = c(100, 1000, 4000), to = c(500, 3000, 8000))
plot(p, type="l")
h <- fir(a, f = f, custom = p, wl = 1024, output = 'Wave')
spectro(h)
```

---

symba *Symbol analysis of a numeric (time) series*

Description

This function analyses one or two sequences of symbols from numeric (time) series.

Usage

```
symba(x, y = NULL, symb = 5, collapse = TRUE, entropy = "abs",
plot = FALSE, type = "l", lty1 = 1, lty2 = 2, col1 = 2, col2 = 4,
cex1 = 0.75, cex2= 0.75, xlab = "index", ylab = "Amplitude", legend=TRUE, ...)
```

Arguments

x	a first R object.
y	a second R object
symb	the number of symbols used for the discretisation, can be set to 3 or 5 only.
collapse	logical, if TRUE, the symbols are pasted in a character string of length 1.
entropy	either "abs" for an absolute value or "rel" for a relative value, i. e. between 0 and 1.
plot	logical, if TRUE plots the series x (and y) and the respective symbols.
type	if plot is TRUE, type of plot that should be drawn. See plot for details (by default "l" for lines).
lty1	line type of the object x if type="l".
lty2	line type of the object y if type="l".
col1	colour of the object x.
col2	colour of the object y.
cex1	character size of x symbols.
cex2	character size of y symbols.
xlab	title of the x axis.
ylab	title of the y axis.
legend	logical, if TRUE and if y is not NULL adds a legend to the plot.
...	other plot graphical parameters.

Details

The analysis consists in transforming the series into a sequence of symbols (see the function discrets) and in computing the absolute frequency of each symbol within the sequence.
The entropy (*H*) is then calculated using the symbol frequencies. Using the argument `entropy`, the entropy can be expressed along an absolute scale or as a relative value varying between 0 and 1.
If two numeric (time) series are provided (`x` and `y`) the absolute symbol frequencies and entropy of each series is returned. Besides the mutual information (*I*) is estimated according to:

$$I = H_x + H_y - Hxy$$

with *Hx* the entropy of `x` symbol series, *Hy* the entropy of `y` symbol series, and *Hxy*$ the joint entropy of `x` and `y` symbol series.

Value

If `y` is `NULL` a list of three items is returned (s1, freq1, h1).
If `y` is not `NULL`, a list of 6 items is returned (s1, freq1, h1, s2, freq2, h2, I):

`s1`	the sequence of symbols of `x`,
`freq1`	the relative frequency of each `x` symbol,
`h1`	the entropy of `x` symbol sequence,
`s2`	the sequence of symbols of `y`,
`freq2`	the relative frequency of each `y` symbol,
`h2`	the entropy of `y` symbol sequence,
`I`	the mutual information between `x` and `y`.

Note

It might be useful to round the values of the input series (see examples).
The mutual information (*I*) should increase with the similarity between the series to compare (`x` and `y`).

Author(s)

Jerome Sueur <sueur@mnhn.fr>

References

Cazelles, B. 2004 Symbolic dynamics for identifying similarity between rhythms of ecological time series. *Ecology Letters*, 7: 755-763.

See Also

discrets, SAX

Examples

```
# analysis of a frequency spectrum
data(tico)
spec1<-spec(tico,f=22050,at=0.2,plot=FALSE)
symba(spec1[,2],plot=TRUE)
# it might be better to round the values
symba(round(spec1[,2],2),plot=TRUE)
# in that case the symbol entropy is close to the spectral entropy
symba(round(spec1[,2],2),entrop="rel")$h1
sh(spec1)
# to compare two frequency spectra
spec2<-spec(tico,f=22050,wl=512,at=1.1,plot=FALSE)
symba(round(spec1[,2],2),round(spec2[,2],2),plot=TRUE)
```

---

synth *Synthesis of time wave (additive model)*

Description

This functions synthesizes pure or harmonic tone sound with amplitude modulation (am) and/or frequency modulation (fm).

Usage

```
synth(f, d, cf, a = 1, signal = "sine", shape = NULL, p = 0,
am = c(0, 0, 0), fm = c(0, 0, 0, 0, 0), harmonics = 1,
plot = FALSE, listen = FALSE, output = "matrix",...)
```

Arguments

f	sampling frequency (in Hz).
d	duration (in s).
cf	carrier frequency (in Hz).
a	amplitude (linear scale, relative when adding different waves).
signal	a character vector specifying the shape of the signal, see `details`.
shape	modification of the whole amplitude shape of the wave, see `details`.
p	initial phase (in radians).
am	a numeric vector of length 3 describing amplitude modulation parameters, see `details`.
fm	a numeric vector of length 5 describing frequency modulation parameters, see `details`.
harmonics	a numeric specifying the number and the relative amplitude of harmonics, see `details`.
plot	if `TRUE` returns the spectrogram of the synthezised sound (by default `FALSE`).

listen	if TRUE the new sound is played back.
output	character string, the class of the object to return, either "matrix", "Wave", "Sample", "audioSample" or "ts".
...	other spectro graphical parameters.

Details

- signal is a character vector of length 1 that specifies the function used to synthesize the signal. There are three options:
  1. "sine": for a sinus function,
  2. "tria": for a triangle function,
  3. "square": for a square function,
  4. "saw": for a square function.
- shape is a character vector of length 1 that allows to modify the whole amplitude shape of the wave. There are four options:
  1. "incr": linear increase
  2. "decr": linear decrease
  3. "sine": sinusoid-like shape
  4. "tria": triangular shape
- am is a numeric vector of length 3 including:
  1. the amplitude modulation depth (in %)
  2. the frequency of the amplitude modulation (in Hz),
  3. the phase of the amplitude modulation (in radian).
- fm is a numeric vector of length 5 including:
  1. the maximum excursion of a sinusoidal frequency modulation (in Hz),
  2. the frequency of a sinusoidal frequency modulation (in Hz),
  3. the maximum excursion of a linear frequency modulation (in Hz).
  4. the phase of the frequency modulation (in radian).
  5. the maximum excursion of an exponential frequency modulation (in Hz).
- harmonics is a numeric vector that controls the number and the relative amplitude of harmonics synthesized.
  By default harmonics = 1 meaning that a pure tone made of a single harmonic (fundamental) will be produced.
  To produce harmonics, the length of harmonics has to be greater than 1. The length of harmonics will set the number of harmonics, including the first one (fundamental). The value of each element of harmonics specify the relative ampltiude of each harmonic. The first value must equal to 1.
  Here are some examples:
  - harmonics = c(1,0.5,0.25) will produce a sound with three harmonics (fundamental + 2 harmonics), the second harmonic having an amplitude half the fundamental ampltiude and the second harmonic an amplitude a quarter of the fundamental amplitude.
  - harmonics = c(1,0,0.25) will produce a sound with two harmonics (fundamental + 1 harmonic) the second harmonic having a null relative amplitude.
  - harmonics = rep(1,4) will produce a sound with four harmonics (fundamental + 3 harmonics) of equal amplitude.

Value

If `plot` is `FALSE`, a new wave is returned. The class of the returned object is set with the argument `output`.

Author(s)

Jerome Sueur and Laurent Lellouch.

References

Hartmann, W. M. 1998 *Signals, sound and sensation*. New York: Springer.

See Also

`synth2`, `noisew`, `pulse`, `echo`

Examples

```
## You can use plot=TRUE and spectro() options
## to directly 'see' the new-built sounds
f <- 8000 # sampling frequency
d <- 1    # duration (1 s)
cf <- 440 # carrier frequecy (440 Hz, i.e. flat A tone)
# pure sinusoidal tone
s <- synth(f=f,d=d,cf=cf)
# pure triangular tone
s <- synth(f=f,d=d,cf=cf, signal="tria")
# pure tone with triangle overall shape
s <- synth(f=f,d=d,cf=cf,shape="tria")
# pure tones with am
s <- synth(f=f,d=d,cf=cf,am=c(50,10))
# pure tones with am
# and phase shift of pi radian (180 degrees)
s <- synth(f=f,d=d,cf=cf,am=c(50,10,pi))
# pure tone with +1000 Hz linear fm
s <- synth(f=f,d=d,cf=cf,fm=c(0,0,1000,0,0))
# pure tone with sinusoidal fm
# (maximum excursion of 250 Hz, frequency of 10 Hz)
s <- synth(f=f,d=d,cf=cf,fm=c(250,10,0,0,0))
# pure tone with sinusoidal fm
# (maximum excursion of 250 Hz, frequency of 10 Hz,
# phase shift of pi radian (180 degrees))
s <- synth(f=f,d=d,cf=cf,fm=c(250,10,0, pi,0))
# pure tone with sinusoidal am
# (maximum excursion of 250 Hz, frequency of 10 Hz)
# and linear fm (maximum excursion of 500 Hz)
s <- synth(f=f,d=d,cf=cf,fm=c(250,10,500,0,0))
# the same with am
s <- synth(f=f,d=d,cf=cf,am=c(50,10), fm=c(250,10,250,0,0))
# the same with am and a triangular overall shape
s <- synth(f=f,d=d,cf=cf,shape="tria",am=c(50,10), fm=c(250,10,250,0,0))
# an harmonic sound
```

```
s <- synth(f=f,d=d,cf=cf, harmonics=c(1, 0.5, 0.25))
# a clarinet-like sound
clarinet <- c(1, 0, 0.5, 0, 0.14, 0, 0.5, 0, 0.12, 0, 0.17)
s <- synth(f=f, d=d, cf = 235.5, harmonics=clarinet)
# inharmonic FM sound built 'manually'
fm <- c(250,5,0,0,0)
F1<-synth(f=f,d=d,cf=cf,fm=fm)
F2<-synth(f=f,d=d,a=0.8,cf=cf*2,fm=fm)
F3<-synth(f=f,d=d,a=0.6,cf=cf*3.5,fm=fm)
F4<-synth(f=f,d=d,a=0.4,cf=cf*6,fm=fm)
final1<-F1+F2+F3+F4
spectro(final1,f=f,wl=512,ovlp=75,scale=FALSE)
```

---

synth2 *Synthesis of time wave (tonal model)*

Description

This functions synthesizes pure tone sound based on an amplitude envelope and an instantaneous frequency contour. The function can also be used to modify a reference sound.

Usage

```
synth2(env = NULL, ifreq, f, plot = FALSE, listen = FALSE, output = "matrix", ...)
```

Arguments

env	a numeric vector describing the amplitude envelope (i.e. the amplitude modulation). By default NULL, generating a squared enveloppe.
ifreq	a numeric vector describing the instantaneous frequency (in Hz).
f	a numeric vector for the sampling frequency (in Hz)
plot	if TRUE returns the spectrogram of the synthezised sound (by default FALSE).
listen	if TRUE the new sound is played back.
output	character string, the class of the object to return, either "matrix", "Wave", "Sample", "audioSample" or "ts".
...	other spectro graphical parameters.

Details

env and ifreq must have exactly the same length.
The amplitude envelope can be obtained with the Hilbert envelope (function env) and the instantaneous frequency can be obtained with the Hilbert transform (function ifreq). This opens a great variety of signal modidications as shown in the example section.

Value

If plot is FALSE, a new wave is returned. The class of the returned object is set with the argument output.

Author(s)

Jérôme Sueur and Laurent Lellouch

References

Beeman, K. 1998 Digital signal analysis, editing and synthesis, in *Animal acoustic communication* edited by Hopp SL, Owren MJ, Evans CS, Springer, 59-103.

See Also

synth2, noisew, pulse, echo

Examples

```
## You can use plot=TRUE and spectro() options
## to directly 'see' the new-built sounds
## MODIFICATION OF A REFERENCE SIGNAL
data(tico)
env.tico <- env(tico, f=22050, plot=FALSE)
ifreq.tico <- ifreq(tico, f=22050, plot=FALSE)$f[,2]
# recover the original signal
s <- synth2(env=env.tico, ifreq=ifreq.tico*1000, f=22050)
# original signal with instantaneous frequency reversed
s <- synth2(env=env.tico, ifreq=rev(ifreq.tico)*1000, f=22050)
# original signal with a +1000 Hz linear frequency shift
s <- synth2(env=env.tico, ifreq=ifreq.tico*1000+1000, f=22050)
# original signal with instantaneous frequency multiplied by 2
s <- synth2(env=env.tico, ifreq=ifreq.tico*1000*2, f=22050)
# original signal with a linear  instantaneous frequency at 2000 Hz
s <- synth2(env=env.tico, ifreq=rep(2000, times=length(tico@left)), f=22050)

## DE NOVO SYNTHESIS
# instantaneous frequency increasing by step of 500 Hz
s <- synth2(ifreq=rep(c(500,1000,1500,2000,2500,3000,3500,4000), each=2000), f=16000)
# square function of the instantaenous frequency
s <- synth2(ifreq=500+seq(-50,50, length.out=8000)^2, f=8000)
# linear increase of the amplitude envelope
s <- synth2(env=seq(0,1,length=8000), ifreq=rep(2000,8000), f=8000)
# square-root increase of the amplitude envelope
s <- synth2(env=sqrt(seq(0,1,length=8000)), ifreq=rep(2000,8000), f=8000)
# square-root increase and decrease of the amplitude envelope
s <- synth2(env=c(sqrt(seq(0,1,length=4000)), sqrt(seq(1,0,length=4000))),
    ifreq=rep(2000,8000), f=8000)
# amplitude envelope and instantaneous frequency following a normal density shape
norm <- rep(dnorm(-4000:3999, sd=1000), 2)
s <- synth2(env=norm, ifreq=500+(norm/max(norm))*1000, f=8000)
```

TFSD *normalized Time and Frequency Second Derivative*

Description

This function computes the normalized Time and Frequency Second Derivative as described by Aumond et al. (2017).

Usage

```
TFSD(wave, f, channel = 1, ovlp = 0,  wn = "hamming", flim = c(2,6), nbwindows = 1)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
ovlp	overlap between two successive windows (in %).
wn	window name, see ftwindow (by default "hanning").
flim	a numeric vector of length 2 to select a frequency band (in kHz). Cannot be NULL.
nbwindows	a numeric vector of length 1 specifying the number of windows (by default 1, ie a single window including the complete wave object.

Details

The TFSD aims at estimating the time of presence of avian or human vocalizations within a sound environment. It calculates the variation in time and frequency of a signal around frequencies of interest, normalized by the spectral time variation of a signal as a whole.

Warning, this index was initially developed to work from a third octave spectrogram with a time sampling of 125 ms.

TFSD is computed according to formulation in reference.

The higher the TFSD varies between 0 and 1, the greater the temporal presence of avian or human vocalizations. With the default configuration, a TFSD > 0.3 indicates a very important presence time of the vocalizations in the signal. The TFSD is always greater than 0.

Value

A numeric vector of length nbwindows giving the TFSD values.

Author(s)

Pierre Aumond, Guillaume Corbeau

References

Aumond, P., Can, A., De Coensel, B., Botteldooren, D., Ribeiro, C., & Lavandier, C. (2017). Modeling soundscape pleasantness using perceptual assessments and acoustic measurements along paths in urban context. *Acta Acustica united with Acustica*, 12, 50-67.

Gontier, F., Lavandier, C., Aumond, P., Lagrange, M., & Petiot, J. F. (2019). Estimation of the perceived time of presence of sources in urban acoustic environments using deep learning techniques. *Acta Acustica united with Acustica*, 105(6), 1053-1066.

See Also

ACI, NDSI

Examples

```
## Note that 'tico' is not a soundscape recording...
data(tico)
TFSD(tico)
## dividing the sound sample into 4 windows of equal duration
TFSD(tico, nbwindows=4)
## selection of a frequency band
TFSD(tico, flim=c(2,6))
```

---

th — *Temporal entropy*

Description

Compute the entropy of a temporal envelope.

Usage

```
th(env, breaks)
```

Arguments

env	a data set resulting of an envelope obtained using env
breaks	'breaks' argument of hist to compute the entropy on the distribution obtained with an histogram.

Details

Temporal entropy is calculated according to:

$$S = -\frac{\sum_{i=1}^{N} y_i log_2(y_i)}{log_2(N)}$$

with:
y = relative amplitude of the i envelope point,
and

$$\sum_{i=1}^{N} y_i = 1$$

and N = number of envelope points.

Value

A single value varying between 0 and 1 is returned. The value has no unit.

Note

The temporal entropy of a noisy signal with many amplitude modulations will tend towards 1 whereas the temporal entropy of quiet signal will tend towards 0.
Note, however, that a sustained sound with an almost flat envelope will also show a very high temporal entropy except if you compute the entropy on the distribution obtained with the histogram. See examples.

Author(s)

Jerome Sueur, George Zhan for the idea and implementation of the argument `breaks`.

See Also

`sh`, `csh`, `H`

Examples

```
# Temporal entropy of a cicada song
data(orni)
envorni<-env(orni,f=22050,plot=FALSE)
th(envorni)
# Smoothing the envelope might slightly change the result.
envorniS<-env(orni,f=22050,smooth=c(50,0),plot=FALSE)
th(envorniS)
# If we mute a part of the cicada song, the temporal entropy decreases
orni2<-mutew(orni,f=22050,from=0.3,to=0.55,plot=FALSE)
envorni2<-env(orni2,f=22050,plot=FALSE)
th(envorni2)
# The temporal entropy of noise tends towards 1
a<-noisew(d=1,f=8000)
enva<-env(a,f=8000,plot=FALSE)
th(enva)
```

```
# But be aware that the temporal entropy
# of a sustained sound also tends towards 1
b<-synth(f=8000,d=1,cf=2000,plot=FALSE)
envb<-env(b,f=8000,plot=FALSE)
th(envb)
# except if you use the distribution of the histogram
th(envb, breaks="Sturges")
```

---

tico *Song of the bird Zonotrichia capensis*

Description

Recording of a song emitted by a male of the neotropical sparrow *Zonotrichia capensis*.

Usage

```
data(tico)
```

Format

A Wave object.

Details

Duration = 1.795 s. Sampling frequency = 22050 hz.

Source

Recording by Thierry Aubin.

Examples

```
data(tico)
oscillo(tico,f=22050)
```

timelapse *Time lapse*

Description

Append successive input sounds into a single output sound

Usage

```
timelapse(dir, from = 1, to = Inf,
units = c("samples", "seconds", "minutes", "hours"), verbose = TRUE)
```

Arguments

`dir`	a character vector, the path to the directory where the .wav files are stored or directly the names of the .wav files to be appended.
`from`	where to start reading the input files, in `units`. See `readWave` of the package `tuneR`.
`to`	where to stop reading, in `units`. See `readWave` of the package `tuneR`.
`units`	time units in which `from` and `to` is given, the default is `"samples"`, but can be set to time intervals such as `"seconds"`. See `readWave` of the package `tuneR`.
`verbose`	a logical, if `TRUE` (default) the file number and name processed are displayed in the console.

Details

The function takes the `.wav` files which names are provided in the argument `dir` and append (paste) them successively so that a single object is obtained. This can be used to produce sound time lapse based on a series of ordered files as those produced by an automatic recorder (e.g. SongMeter of the society 'Wildlife Acoustics').
Only a section of each file can be extracted by using the arguments `from` and `to`. The function is based on `readWave` and `bind` of the package `tuneR`.

Value

A `Wave` object, a class defined in the package `tuneR`.

Note

The characteristics (sampling rate, number of bits, stereo/mono) of the output object are those of the .wav file.
The files should be alphabatically ordered according to time to ensure a proper time lapse.
You should use either `savewav` or `writeWave` to save the results as a `.wav` file.

Author(s)

Jérôme Sueur

See Also

pastew

Examples

```
## Not run:
## if 'dir' contains a set of files recorded with a Wildlife Acoustics
# songmeter recorder then a direct way to obtain
# the spectrogram of all .wav files is
dir <- "pathway-to-directory-containing-wav-files"
res <- timelapse(dir)
# to extract a selection of each file (here a section starting
# at 10 s and ending at 12 s)
res <- timelapse(dir, from=10, to=12, unit="seconds")

## End(Not run)
```

---

timer *Time measurements of a time wave*

Description

This function computes and shows the duration of signal periods, pause periods and their ratio.

Usage

```
timer(wave, f, channel = 1, threshold = 5, dmin = NULL, envt="abs",
power = 1, msmooth = NULL, ksmooth = NULL,
ssmooth = NULL, asmooth=NULL, tlim = NULL, plot = TRUE, plotthreshold = TRUE,
col = "black", colval = "red",
xlab = "Time (s)", ylab = "Amplitude", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
threshold	amplitude threshold for signal detection (in %), or alternatively a function to be applied on the waveform scaled between 0 and 1. See examples.
dmin	time threshold (minimum duration) for signal detection (in s).
envt	the type of envelope to be used: either "abs" for absolute amplitude envelope or "hil" for Hilbert amplitude envelope. See env.
power	a power factor applied to the amplitude envelope. Increasing power will reduce low amplitude modulations and increase high amplide modulations. This can be used to reduce background noise (by default equals to 1, *i.e.* no change.

msmooth	a vector of length 2 to smooth the amplitude envelope with a mean sliding window. The first component is the window length (in number of points). The second component is the overlap between successive windows (in %). See env.
ksmooth	kernel smooth for the amplitude enveloppe via kernel. See env.
ssmooth	sum smooth for the amplitude enveloppe. See env.
asmooth	autocorrelation smooth for the amplitude enveloppe. See env.
tlim	modifications of the time X-axis limits.
plot	logical, if TRUE plots the envelope and the measurements (by default TRUE).
plotthreshold	logical, if TRUE plots the threshold as an horizontal line on the graph (by default TRUE).
col	colour of the envelope.
colval	colour of plotted measurements.
xlab	title of the x-axis.
ylab	title of the y-axis.
...	other plot graphical parameters.

Value

A list containing seven items:

s	duration of signal period(s) in seconds
p	duration of pause period(s) in seconds
r	ratio between the signal and silence periods(s)
positions	a list containing four elements:
s.start	start position(s) of signal period(s)
s.end	end position(s) of signal period(s)
first	whether the first event detected is a pause or a signal

Warning

Setting to high values to msmooth or ssmooth might return inaccurate results. Double check your results if so.

Author(s)

Jerome Sueur

See Also

env, cutw, pastew.

TKEO *Teager-Kaiser energy tracking operator*

Description

This function computes the Teager-Kaiser energy operator.

Usage

```
TKEO(wave, f, channel = 1, m = 1, M = 1, plot = TRUE,
xlab = "Time (s)", ylab = "Energy",
type = "l", bty = "l", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
m	a numeric vector of length 1 for the exponent parameter. See details.
M	a numeric vector of length 1 for the lag parameter. See details.
plot	logical, if TRUE returns a plot of the TK energy along time (by default TRUE).
xlab	Label of time x-axis.
ylab	Label of energy y-axis.
type	if plot is TRUE, type of plot that should be drawn. See plot for details (by default "l" for lines).
bty	the type of box to be drawn around the energy plot.
...	other plot graphical parameters.

Details

The discrete version of the Teager-Kaiser operator is computed according to:

$$y_n = x_n^{2/m} - (x_{n-M} \times x_{n+M})^{1/m}$$

,
with *m* the exponent parameter and *M* the lag parameter which both are usually equal to 1 for a conventional operator.
The Teaser-Kaiser operator can be used to track amplitude modulations (AM) and/or frequency modulations (FM).
See examples.

Value

This function returns a two-column matrix, the first column is time and the second column includes the successive energy values.
m/2 NA values are added at the start and end of the vector.

Author(s)

Jerome Sueur

References

Kvedalen, E. (2003). *Signal processing using the Teager Energy Operator and other nonlinear operators*. University of Oslo, Department of Informatics, PhD Thesis, x + 100 p.

See Also

env, ifreq.

Examples

```
op <- par(mfrow=c(2,1))

## sinusoid AM
s1 <- synth(f=8000, d=0.1, cf=200, am=c(100,10), output="Wave")
oscillo(s1)
TKEO(s1)
## linear AM decrease
s2 <- synth(f=8000, d=0.1, cf=200, shape="decr", output="Wave")
oscillo(s2)
TKEO(s2)
## sinusoid FM
s3 <- synth(f=8000, d=0.1, cf=200, fm=c(150,50,0,0,0), output="Wave")
oscillo(s3)
TKEO(s3)
## linear FM increase
s4 <- synth(f=8000, d=0.1, cf=200, fm=c(0,0,600,0,0), output="Wave")
oscillo(s4)
TKEO(s4)
## AM and FM
s5 <- synth(f=8000, d=0.1, cf=200, am=c(100,10), fm=c(150,50,0,0,0), output="Wave")
oscillo(s5)
TKEO(s5)
par(op)
```

`wasp` *WAve length and SPeed of sound*

Description

This function returns the wavelength and the speed of sound of a given frequency in air, fresh-water or sea-water.

Usage

```
wasp(f, t = 20, c = NULL, s = NULL, d = NULL, medium = "air")
```

Arguments

`f`	frequency (Hz).
`t`	temperature (degree Celsius).
`c`	celerity (m/s) if a wavelength is to be found at a particular speed of sound.
`s`	salinity (parts per thousand) when `medium` is `"sea"`.
`d`	depth (m) when `medium` is `"sea"`.
`medium`	medium for sound propagation, either "air", "fresh" for fresh, or pure, water, "sea" for sea water.

Details

Speed of sound in air is computed according to:

$$c = 331.4 + 0.6 \times t$$

Speed of sound in fresh-water is computed according to Marczak equation:

$$c = 1.402385.10^3 + 5.038813 \times t - 5.799136.10^{-2} \times t^2$$
$$+3.287156.10^{-4} \times t^3 - 1.398845.10^{-6} \times t^4$$
$$+2.787860.10^{-9} \times t^5$$

with t = temperature in degrees Celsius; range of validity: 0-95 degrees Celcius at atmospheric pressure.

Speed of sound in sea-water is computed according to Mackenzie equation:

$$c = 1448.96 + 4.591 \times t - 5.304.10^{-2} \times t^2$$
$$+2.374.10^{-4} \times t^3 + 1.34 \times (s - 35) + 1.63.10^{-2} \times d$$
$$+1.675.10^{-7} \times d^2 - 1.025.10^{-2} \times t \times (s - 35)$$

$$-7.139.10^{-13} \times t \times d^3$$

with t = temperature in degrees Celsius; s = salinity in parts per thousand; d = depth in meters; range of validity: temperature 2 to 30 degrees Celcius, salinity 25 to 40 parts per thousand, depth 0 to 8000 m.

Wavelength is obtained following:

$$\lambda = \frac{c}{f}$$

with c = speed of sound in meters/second; f = frequency in Hertz.

Value

A list of two values is returned:

`l`	wavelength in meters
`c`	speed of sound in meters/second.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

References

http://resource.npl.co.uk

Examples

```
# wavelength (m) of a 2000 Hz air-borne sound at 20 degrees Celsius
wasp(f=2000)$l
# [1] 0.1717

# sound speed in sea at 0 and -500 m
# for a respective temperature of 22 degrees Celcius and 11 degrees Celcius
wasp(f=1000,s=30,d=c(0,500),t=c(22,11),medium="sea")$c
# [1] 1521.246 1495.414

# wavelength (m) of a 1000 Hz sound in a medium unspecified where c = 1497 m/s
wasp(f=1000,c=1497)$l
# [1] 1.497

# variation of wavelength according to frequency and air temperature
op<-par(bg="lightgrey")
a<-seq(1000,20000,by=100) ; na<-length(a)
b<-seq(-20,40,by=10) ; nb<-length(b)
res<-matrix(numeric(na*nb),nrow=na)
for(i in 1:nb) res[,i]<-wasp(a,t=b[i])$l
matplot(x=a,y=res,type="l",lty=1,col= spectro.colors(nb),
  xlab="Frequency (Hz)",ylab="Wavelength (m)")
title("Wavelength of air-borne sound at different temperatures (deg. C)")
```

```
legend(x=15000,y=0.3,c("-20","-10","0","10","20","30","40"),
  lty=1,col= spectro.colors(nb),bg="grey")
par(op)
```

wav2flac — *wav-flac file conversion*

Description

This function converts .wav files into .flac files and reversely

Usage

```
wav2flac(file, reverse = FALSE, overwrite = FALSE,
exename = NULL, path2exe = NULL)
```

Arguments

`file`	the .wav or .flac file to convert.
`reverse`	logical, if `TRUE` converts a .flac file into a .wav file.
`overwrite`	logical, if `TRUE` overwrites the file to convert.
`exename`	a character string specifying the name of the FLAC binary file. If `NULL`, the dedault name "flac" will be used for Linux OS and "flac.exe" for Windows OS.
`path2exe`	a character string giving the path to the FLAC binary file. If `NULL`, the dedault path "c:/Program Files/FLAC/" will be used for Windows OS.

Details

The function runs FLAC. FLAC has then to be installed first, if not the function will not work.

Value

A new file is created.

Note

FLAC must be installed to use this function but not to install the package `seewave`. Free Lossless Audio Codec (FLAC) is a file format by Josh Coalson for lossless audio data compression. FLAC reduces bandwidth and storage requirements without sacrificing the integrity of the audio source. Audio sources encoded to FLAC are typically reduced in size 40 to 50 percent.

Author(s)

Luis J. Villanueva-Rivera

See Also

`savewav`

Examples

```
## Not run:
# synthesis of a 1kHz sound
a<-synth(d=10,f=8000,cf=1000)
# save it as a .wav file in the default working directory
savewav(a,f=8000)
# compress it to FLAC format and overwrite on the file a.wav
wav2flac("a.wav", overwrite=TRUE)
# back to .wav format
wav2flac("a.flac", reverse=TRUE)
# remove the files
unlink(c("a.wav","a.flac"))

## End(Not run)
```

---

wf *Waterfall display*

Description

This function returns a waterfall display of a short-term Fourier transform or of any matrix.

Usage

```
wf(wave, f, channel = 1, wl = 512, zp = 0, ovlp = 0, fftw= FALSE, dB = "max0",
dBref = NULL, wn = "hanning", x = NULL,
hoff = 1, voff = 1, col = heat.colors,
xlab = "Frequency (kHz)", ylab = "Amplitude (dB)",
xaxis = TRUE, yaxis = TRUE,
density = NULL, border = NULL, lines = FALSE, lwd=NULL, ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
wl	window length for the analysis (even number of points). (by default = 512)
zp	zero-padding (even number of points), see Details.
ovlp	overlap between two successive windows (in %).
fftw	if TRUE calls the function FFT of the library fftw. See Notes of the spectro.
dB	a character string specifying the type dB to return: "max0" for a maximum dB value at 0, "A", "B", "C", "D", and "ITU" for common dB weights.
dBref	a dB reference value when dB is TRUE. NULL by default but should be set to 2*10e-5 for a 20 microPa reference.

wn	window name, see ftwindow (by default "hanning").
x	a matrix if wave is not provided.
hoff	horizontal 'offset' which shifts actual x-values slightly per row for visibility. Fractional parts will be removed.
voff	vertical 'offset' which separates traces.
col	a color or a color palette function to be used to assign colors in the plot
xlab	title of the frequency x-axis.
ylab	title of the amplitude y-axis.
xaxis	a logical, if TRUE adds the frequency x-axis according to f.
yaxis	a logical, if TRUE adds the amplitude y-axis according.
density	argument of polygon: the density of shading lines, in lines per inch. The default value of 'NULL' means that no shading lines are drawn. A zero value of 'density' means no shading nor filling whereas negative values (and 'NA') suppress shading (and so allow color filling).
border	argument of polygon: the color to draw the border. The default, 'NULL', means to use 'par("fg")'. Use 'border = NA' to omit borders.
lines	a logical, if TRUE plots lines instead of surfaces (polygons).
lwd	line width.
...	other graphical arguments to passed to plot

Details

Data input can be either a time wave (wave) or a matrix (x). In that case, if xaxis is set to TRUE the x-axis will follow the row index. To change it, turn xaxis to FALSE and use axis afterwards. See examples.

Note

The function is well adapted to display short-term Fourier transform. However, any matrix can be called using the argument x instead of wave.

Author(s)

Carl G. Witthoft and Jerome Sueur <sueur@mnhn.fr>

See Also

spectro, spectro3D, dynspec

Examples

```
data(tico)
wf(tico,f=22050)
# changing the display parameters
jet.colors <- colorRampPalette(c("blue", "green"))
wf(tico,f=22050, hoff=0, voff=2, col=jet.colors, border = NA)
```

```
# matrix input instead of a time wave and transparent lines display
m <- numeric()
for(i in seq(-pi,pi,len=40)) {m <- cbind(m,10*(sin(seq(0,2*pi,len=100)+i)))}
wf(x=m, lines=TRUE, col="#0000FF50",xlab="Time", ylab="Amplitude",
main="waterfall display")
```

`write.audacity` *Audacity audio markers export*

Description

Write audio markers to be imported by Audacity.

Usage

```
write.audacity(x, filename)
```

Arguments

`x`	a data frame with the three or five colums, see details.
`filename`	name of the `.txt` file. (by default the name of `x`).

Details

The input `x` object should be a data frame with two or three columns depending on whether the markers include frequency limits or not :

- time limits only:
  1. text label of each marker,
  2. time marker of the beginning of each marker,
  3. time marker of the end of each marker.
- time and frequency limits:
  1. text label of each marker,
  2. time marker of the beginning of each marker,
  3. time marker of the end of each marker,
  4. lower frequency limit of each marker,
  5. higher frequency limit of each marker.

Value

A `.txt` file is generated to be imported as a markers in Audacity.

Note

Naming the columns of `x` is not necessary.

Author(s)

Jerome Sueur

References

Audacity is a free software distributed under the terms of the GNU General Public License.
Web site: https://www.audacityteam.org/

See Also

read.audacity

Examples

```
## 3 markers, time only
t1 <- c(9.2, 16.2, 24.4)
t2 <- c(11.7, 18.7, 26.9)
label <- c("a", "b", "c")
df <- data.frame(label, t1, t2)
write.audacity(df, filename="test-time.txt")
## 3 markers, time and frequency
t1 <- c(9.4, 15.2, 24.9)
t2 <- c(10.54, 16.6, 26.1)
f1 <- c(1703.4, 3406.8, 1608.8)
f2 <- c(7476.2, 8517.2, 5110.3)
label <- c("a", "b", "c")
dff <- data.frame(label, t1, t2, f1, f2)
write.audacity(dff, filename="test-time-frequency.txt")
## delete files
unlink(c("test-time.txt", "test-time-frequency.txt"))
```

---

zapsilw *Zap silence periods of a time wave*

Description

This function simply deletes the silence periods of a time wave.

Usage

```
zapsilw(wave, f, channel = 1, threshold = 5, plot = TRUE, output = "matrix", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).

threshold	amplitude threshold (in %) between silence and signal.
plot	logical, if TRUE plots the orginal and the new oscillograms (by default TRUE).
output	character string, the class of the object to return, either "matrix", "Wave", "Sample", "audioSample" or "ts".
...	other oscillo graphical parameters.

Value

If plot is FALSE, a new wave is returned. The class of the returned object is set with the argument output.

Note

Use the argument threshold to set the level of silence. See the examples.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

afilter, oscillo

Examples

```
data(orni)
zapsilw(orni,f=22050,colwave="red")
# setting the threshold value
zapsilw(orni,f=22050,threshold=1)
```

---

zc *Instantaneous frequency of a time wave by zero-crossing*

Description

This function measures the period of a full oscillating cycle.

Usage

```
zc(wave, f, channel = 1,  plot = TRUE, interpol = 1, threshold = NULL,
xlab = "Time (s)", ylab = "Frequency (kHz)", ylim = c(0, f/2000),
warning = TRUE, ...)
```

Arguments

`wave`	an R object.
`f`	sampling frequency of `wave` (in Hz). Does not need to be specified if embedded in `wave`.
`channel`	channel of the R object, by default left channel (1).
`plot`	logical, if `TRUE` plots the dominant frequency along the time wave(by default `TRUE`).
`interpol`	a numeric vector of length 1, interpolation factor.
`threshold`	amplitude threshold for signal detection (in %).
`xlab`	title of the x axis.
`ylab`	title of the y axis.
`ylim`	the range of y values.
`warning`	a logial to specify if warning message should be displayed or not when `interpol` is > 100.
`...`	other `plot` graphical parameters.

Details

If `plot` is `FALSE`, `zc` returns a vector of numeric data with the instantaneous frequency.

Value

If `plot` is `FALSE`, `zc` returns a two-column matrix, the first column corresponding to time in seconds (x-axis) and the second column corresponding to the instantaneous frequency of the time wave in kHz (y-axis).
'NA's correspond either to pause periods (e. g. detected applying `threshold`) or sections of the time wave not crossing the zero line. To remove 'NA's with `na.omit` allows to get only instantaneous frequency values but discards information about pause sections.

Note

`interpol` adds points to the time wave by linear interpolation (through `approx`). This increases measurement precision but as well time process. Type argument of `plot` cannot be set to "l".

Author(s)

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References

Hopp, S. L., Owren, M. J. and Evans, C. S. (Eds) 1998. *Animal acoustic communication.* Springer, Berlin, Heidelberg.

See Also

`zc`, `ifreq`

Examples

```
data(pellucens)
pellu1 <- cutw(pellucens,f=22050,from=0,to=1,plot=FALSE)
# without interpolation
zc(pellu1,f=22050,threshold=5,pch=20)
# with interpolation
zc(pellu1,f=22050,threshold=5,interpol=20,pch=20)
# a way to plot with a line and to filter low frequencies
pellu2 <- zc(pellu1,f=22050,threshold=5,interpol=20,plot=FALSE)
pellu3 <- na.omit(pellu2[,2])
pellu4 <- pellu3[pellu3>3]
plot(x=seq(0,nrow(pellu1)/22050,length.out=length(pellu4)),
    y=pellu4,type="l",xlab="Time(s)",ylab="Frequency(kHz)")
```

---

zcr *Zero-crossing rate*

Description

This functions computes the zero-crossing rate of a time function, i. e. the average number the sign of a time wave changes.

Usage

```
zcr(wave, f, channel = 1, wl = 512, ovlp = 0, plot = TRUE, type = "o", xlab =
"Time (s)", ylab = "Zero crossing rate", ...)
```

Arguments

wave	an R object.
f	sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.
channel	channel of the R object, by default left channel (1).
wl	length of the window for the analysis (even number of points, by default = 512). If NULL the zero-crossing rate is computed of the complete signal.
ovlp	overlap between two successive analysis windows (in %) if wl is not NULL.
plot	a logical, if TRUE plots a the zero-crossing rate results along time.
type	if plot is TRUE, type of plot that should be drawn. See plot for details (by default "l" for lines).
xlab	if plot is TRUE, label of the x axis.
ylab	if plot is TRUE, label of the y axis.
...	other plot graphical parameters.

Details

The zero-crossing rate is computed according to:

$$zcr = \frac{1}{2 \times N} \sum_{t=0}^{N-1} |sgn[x(t+1)] - sgn[x(t)]|$$

with:
N the length of the signal *x*
and where:

$$sgn[x(t)] = 1$$

if

$$x(t) \geq 0$$

and

$$sgn[x(t)] = -1$$

if

$$x(t) < 0$$

Value

The are two possibilities:

1. a numeric vector of length 1 if `wl` is `NULL`,
2. a numeric two-column matrix is returned with the first columnn being time (s) and the second colum being the zero-crossing rate (no scale) if `wl` is not `NULL`.

Note

The are two possibilities:

1. if `wl` is `NULL` then the zero-crossing rate is computed for the complete signal.
2. if `wl` is not `NULL` the the zero-crossing rate is computed for for a window sliding along the time wave.

The ZCR is supposed to help in detection of voiced/unvoiced sound sections.

Author(s)

Jerome Sueur

References

https://en.wikipedia.org/wiki/Zero-crossing_rate

See Also

`zc`

Examples

```
data(tico)
## a single value for the complete signal, no plot
zcr(tico, wl=NULL)
## a series of values computed for a sliding window of 512 samples, plot
zcr(tico)
```

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