Package ‘seewave’

March 4, 2022

Type Package
Title Sound Analysis and Synthesis
Version 2.2.0
Date 2022-03-04
Maintainer Jerome Sueur <sueur@mnhn.fr>
Encoding UTF-8
SystemRequirements LIBSNDFILE
Imports graphics, grDevices, stats, utils, tuneR, methods
Suggests audio, fftw, ggplot2, rgl, rpanel, phonTools, signal
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)
URL https://rug.mnhn.fr/seewave/
NeedsCompilation no
Author Jerome Sueur [aut, cre],
   Thierry Aubin [aut],
   Caroline Simonis [aut],
   Laurent Lellouch [ctr],
   Pierre Aumond [ctr],
   Ethan C. Brown [ctr],
   Guillaume Corbeau [ctr],
   Marion Depraetere [ctr],
   Camille Desjonquères [ctr],
   François Fabianek [ctr],
   Amandine Gasc [ctr],
Eric Kasten [ctr],
Jonathan Lees [ctr],
Jean Marchal [ctr],
Andre Mikulec [ctr],
Sandrine Pavoine [ctr],
David Pinaud [ctr],
Alicia Stotz [ctr],
Luis J. Villanueva-Rivera [ctr],
Zev Ross [ctr],
Carl G. Witthoft [ctr],
Hristo Zhivomirov [ctr]

Repository CRAN
Date/Publication 2022-03-04 11:40:02 UTC

R topics documented:

ACI ............................................................... 5
acoustat .......................................................... 6
addsilw .......................................................... 9
afilter ........................................................... 10
akamatsu ......................................................... 11
ama .............................................................. 13
AR ............................................................... 14
attenuation ........................................................ 16
audiomoth ....................................................... 17
audiomoth.rename ............................................. 18
autoc ........................................................... 19
beep ............................................................. 21
bwfilter .......................................................... 22
ccoh ............................................................. 23
ceps ............................................................ 26
cepsstro ........................................................ 28
coh ............................................................... 30
combfilter ....................................................... 31
convSPL ........................................................ 33
corenv .......................................................... 34
corspec .......................................................... 36
covspectrum ..................................................... 38
crest ............................................................ 40
csh ............................................................... 41
cutspec .......................................................... 43
cutw ............................................................. 44
dBscale .......................................................... 45
dBweight ........................................................ 47
deletew .......................................................... 48
dfreq ............................................................. 50
diffcumspec ..................................................... 51
R topics documented:

diffenv .................................................. 53
diffspec .................................................. 55
diffwave .................................................. 57
discrets ................................................... 59
drawenv ................................................... 60
drawfilter ................................................ 61
duration .................................................... 63
dynoscillo ................................................ 64
dynspec .................................................... 65
dynspectrum .............................................. 67
echo ......................................................... 70
evt .......................................................... 71
export ....................................................... 73
fadem ......................................................... 74
fbands ....................................................... 75
fdoppler .................................................... 77
filter ......................................................... 79
field ........................................................ 80
fir ............................................................ 82
fma ........................................................... 83
fpeaks ....................................................... 85
ftwindow ................................................... 87
fund .......................................................... 89
gammatone ................................................ 90
gspectro .................................................... 92
H ............................................................... 94
hilbert ....................................................... 95
ifreq ........................................................ 96
istft ........................................................ 98
itakura.dist .............................................. 100
kl.dist ....................................................... 101
ks.dist ...................................................... 102
lf$s .......................................................... 104
listen ......................................................... 106
localpeaks ............................................... 107
logspec.dist .............................................. 108
lts ........................................................... 110
M ............................................................. 112
meandB ..................................................... 113
meanspec .................................................. 114
mel .......................................................... 116
melfilterbank ............................................ 118
micsens .................................................... 119
moredB ...................................................... 120
mutew ....................................................... 121
NDSI ........................................................ 122
noisew ...................................................... 123
notefreq ................................................... 124
R topics documented:

octaves .......................................................... 125
orni ................................................................. 126
oscillo .............................................................. 127
oscilloST ......................................................... 130
pastew ............................................................. 131
peewit .............................................................. 133
pellucens .......................................................... 134
phaseplot ........................................................ 134
phaseplot2 ...................................................... 136
playlist .......................................................... 137
preemphasism ................................................... 138
pulsew ............................................................. 139
Q ................................................................. 140
read.audacity ................................................. 142
repw .............................................................. 143
resamp ........................................................... 144
revw .............................................................. 145
rmam .............................................................. 146
rmnoise ........................................................... 148
rmoffset .......................................................... 149
rms ............................................................... 150
roughness ....................................................... 151
rugo .............................................................. 152
savewav .......................................................... 153
SAX ............................................................... 154
sddB .............................................................. 156
seedata ........................................................... 157
seewave .......................................................... 158
setenv ............................................................ 159
sfm ............................................................... 160
sh ................................................................. 161
sheep ............................................................. 163
simspec ........................................................... 164
smoothw .......................................................... 166
songmeter ....................................................... 167
songmeterdiag .................................................. 169
soundscapespec ................................................ 172
sox ............................................................... 173
spec ............................................................. 174
specflux .......................................................... 177
specprop .......................................................... 179
spectro ............................................................ 181
spectro3D ........................................................ 186
squarefilter ..................................................... 188
symba ........................................................... 189
synth ............................................................. 191
synth2 ............................................................ 194
TFSD ............................................................. 196
**Description**

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

**Usage**

```r
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 "hamming").
- `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, i.e., 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.
acoustat

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


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`) across 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 cumulative 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 column being time (s) and the second column being the amplitude probability mass function (no scale).
- `freq.contour` the frequency contour as a two-column matrix, the first column being frequency (kHz) and the second column 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 developed in Matlab language by Kurt Frisurup and XXXX Watkins (1992).
The R function was kindly checked by Kurt Frisurup.

Author(s)

Jerome Sueur

References


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`, `rew`, `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>
akamatsu

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:

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

where \( c \) is the sound velocity (cm/s) and each \( l, m, n \) represents an integer, and the combination of these parameters 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 sound pressure level at the middle of the X axis, \( i.e., \ L_x/2 \).

2. Cutoff frequency

The cutoff frequency can be calculated as follows:

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

3. Attenuation distance

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

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

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


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.
AR

kernel smooth via kernel. See env.

sum smooth. See env.

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:

\[ AR = \frac{\text{rank}(M) \times \text{rank}(H_t)}{n^2} \]

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


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**

```r
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

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 digital 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:

<table>
<thead>
<tr>
<th>Column</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>year</td>
<td>year of recording, numeric</td>
</tr>
<tr>
<td>month</td>
<td>month of recording, numeric</td>
</tr>
<tr>
<td>day</td>
<td>day of recording, numeric</td>
</tr>
<tr>
<td>hour</td>
<td>hour of recording, numeric</td>
</tr>
<tr>
<td>min</td>
<td>minute of recording, numeric</td>
</tr>
<tr>
<td>sec</td>
<td>second of recording, numeric</td>
</tr>
<tr>
<td>time</td>
<td>time in POSIX format</td>
</tr>
</tbody>
</table>
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)

Description

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

```r
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 character 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 devices: `PREFIX_YYYYMMDD_HHMMSS.wav`.

Value

A 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

```r
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


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 or when ending up a long script.

**Usage**

```r
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**

```r
## 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 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 \(-n\times6\) 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

See Also
ffilter, bwfilter, preemphasis, lfs, afilter

Examples
```r
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 over-plot (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

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

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

Usage

```r
ceps(wave, f, channel = 1, phase = FALSE, wl = 512, at = NULL, from = NULL, to = NULL,
  tid = FALSE, fid = FALSE, col = "black", cex = 1, plot = TRUE,
  qlab = "Freqency (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 then 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).
- `tid`: to identify time values on the plot with the help of a cursor.
- `fid`: 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:

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

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


See Also

cestro, fund, autoc
Examples

```r
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)
```

Description

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

Usage

```r
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 over-plot (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

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

```r
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

```r
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 length 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.
## combfilter

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

```r
## 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**

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

**Arguments**

- `x`: a numeric vector or a matrix describing 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 &lt;sueur@mnhn.fr&gt;

**References**


**See Also**

- `moredB`, `dBweight`, `attenuation`

**Examples**

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

```r
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 column 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

cor.spec, cov.spectro, cor.cor.test.

Examples

```r
## Not run:
data(orni)
# cross-correlation between two schemes 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)
```
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 \texttt{cor.test}, is plotted. Inverting spec1 and spec2 may give slight different results, see examples.

Value

If \texttt{plot} is \texttt{FALSE}, \texttt{corspec} returns a list containing four components:

- \( r \) a two-column matrix, the first column 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).
- \( r_{\text{max}} \) the maximum correlation value between spec1 and spec2.
- \( p \) the p value corresponding to \( r_{\text{max}} \).
- \( f \) the frequency offset corresponding to \( r_{\text{max}} \).

Author(s)

Jerome Sueur, improved by Laurent Lellouch

References


See Also

\texttt{spec}, \texttt{meanspec}, \texttt{corspec}, \texttt{covspectro}, \texttt{cor}, \texttt{cor.test}.

Examples

```r
## 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))
par(new=TRUE)
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))
```
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").
**covspectro**

- **n**  
  \( n \) sets in how many steps \( \text{wave2} \) will be slided along \( \text{wave1} \). Time process can be then decreased by setting low \( n \) value. Inverting \( \text{wave1} \) and \( \text{wave2} \) may give slight different results.

- **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 \( \text{wave1} \) and the spectrogram of \( \text{wave2} \) are computed when regularly sliding forward and backward \( \text{wave2} \) along \( \text{wave1} \). The maximal covariance is obtained at a particular shift (time offset). This shift may be positive or negative.

- If plot is FALSE, covspectro returns a list containing three components:
  - **cov**  
    the successive covariance values between \( \text{wave1} \) and \( \text{wave2} \).
  - **covmax**  
    the maximum covariance between \( \text{wave1} \) and \( \text{wave2} \).
  - **t**  
    the time offset corresponding to \( \text{cov} \).

**Author(s)**

Jerome Sueur <sueur@mnhn.fr>

**References**

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)

Details

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

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

with rms the root-mean-square (see \texttt{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


See Also

oscillo, rms

Examples

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

Description

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

Usage

```r
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


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)

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.
dBscale

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 seeawave: 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)
Description

This function returns the four most common dB weightings.

Usage

```r
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 conversation 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 Wikipediea 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

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**

```r
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 dominant frequency in kHz (y-axis). NA corresponds to pause sections in `wave` (see `threshold`).
Note
This function is based on \texttt{fft}.

Author(s)
Jerome Sueur \texttt{<sueur@mnhn.fr>}

See Also
\texttt{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

\textit{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:

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

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


See Also

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

Examples

```r
## 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)
oriri.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

```r
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:

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

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


See Also

`env`, `corenv`, `diffspec`, `diffwave`
Examples

```r
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

\textit{Difference between two frequency spectra}

Description

This function estimates the surface difference between two frequency spectra.

Usage

```r
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 \texttt{spec} or \texttt{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 \texttt{spec} or \texttt{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 \texttt{spec} or \texttt{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 \texttt{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:

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

, 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


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)
```r
specc <- spec(c,f=8000,wl=512,at=0.5,plot=FALSE)
specd <- spec(d,f=8000,wl=512,at=0.5,plot=FALSE)
diffspect(speca,specb,f=8000)
# [1] 0 => similar spectra of course !
diffspect(speca,specb)
diffspect(specb,specb,plot=TRUE)
diffspect(specb,specc,plot=TRUE)
diffspect(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)
diffspect(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**

```r
diffwave(wave1, wave2, f, channel = c(1,1), wl = 512, envt = "hil", mssmooth = 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`.
- `mssmooth`: 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`.

---

The code snippet demonstrates the use of the `diffwave` function to compare two wave objects, `specc` and `specd`, and illustrates the calculation of mel scales using the `melfcc` function from the `tuneR` package. The `diffspec` function is also used to compare the spectra of different objects, and `apply` is used to calculate the mean of mel scales for two different sets of objects, `orni` and `tico`. The final `diffspec` call compares the mel scale means of `orni.mel.mean` and `tico.mel.mean` using a sampling frequency of 22050 Hz, with mel scale and plot parameters set to TRUE.
**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**


**See Also**

`diffspec`, `diffenv`

**Examples**

```r
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**

```r
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 comparison 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 Increasing,
- `D` if the series is Decreasing,
- `F` if the series remains Flat,

When `symb` is set to 5, each value will be replaced by either:
- `I` if the series is Increasing,
- `D` if the series is Decreasing,
- `F` if the series remains Flat,
- `P` if the series shows a Peak,
- `T` if the series shows a Trough.

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

See Also
symba

Examples

```r
# 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

```r
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

```r
## 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

```r
## 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**

```r
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**

```r
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 wnb 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, \( w_l = 4096 \) is equivalent to \( w_d = \frac{4096}{44100} = 0.093 \) s and equivalent to \( w_nb = \frac{5 \times 4096}{44100} = 53 \).

Note
This function requires the package rpanel.

Author(s)
Jerome Sueur

See Also
oscillo, oscilloST, dynspec.
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 plotted: 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

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

dynspectro Dynamic sliding spectrogram

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).
w1 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").
zk 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 slid-
ing 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 plotted: 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)
Description

This function generates echoes of a time wave.

Usage

```r
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


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,
assmooth = 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 kernel 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)


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% overlapping 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)

---

**Description**

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

**Usage**

```r
extport(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**

```r
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

```r
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)
```

Description

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

Usage

```r
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.
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 achieved 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)
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**

```r
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
- **mofs**: 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),
\( v_s = \) speed of the source,
\( v_o = \) speed of the observer.

Value

The altered frequency is returned in a vector.

Note

You can use \texttt{wasp} to have exact values of \( c \). See examples.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

\texttt{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)
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

```r
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.
field

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 than 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 $k \times d$ 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 $k \times d$ 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 $k \times d$ 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")
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

```r
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


See Also

ffilter, bwfilter, preemphasis, lfs, afilter

Examples

```r
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 asymmetric 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 to 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)
fftwindow

**Description**

Generates different Fourier Transform windows.
ftwindow

Usage

```r
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


See Also

covspectro, dfreq, meanspec, spec, spectro, spectro3D

Examples

```r
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")
alldata<-cbind(a,b,c,d,e,f)
matplot(alldata,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)
```
**Fundamental frequency track**

**Description**

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

**Usage**

```r
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


See Also

cepstro, ceps, autoc

Examples

```r
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

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

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:

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

with \(a\) the amplitude, \(t\) time, \(n\) the filter order, \(cf\) the center frequency, \(\phi\) the initial phase.

The parameter \(\beta\) 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


See Also

`melfilterbank`
Examples

## gammatone filter in the time domain (impulse response)

```r
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)
```

Description

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

Usage

```r
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`.
ggspectro

    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 ggplot layer.

Note

This function requires ggplot2 package.

Author(s)

Jerome Sueur

References


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**

\[ H(wave, f, \text{channel} = 1, \text{wl} = 512, \text{envt} = "hil", \text{msmooth} = \text{NULL}, \text{ksmooth} = \text{NULL}) \]

**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

See Also
sh, th, csh

Examples
```r
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`.
ifreq

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


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 through 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


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)
w FT window length for the analysis (even number of points).
overl 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
```r
## 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

\textit{Itakuro-Saito distance}

\textbf{Description}

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

\textbf{Usage}

\begin{verbatim}
itakura.dist(spec1, spec2, scale=FALSE)
\end{verbatim}

\textbf{Arguments}

\begin{itemize}
  \item \texttt{spec1} \hspace{1cm} any distribution, especially a spectrum obtained with \texttt{spec} or \texttt{meanspec} (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
  \item \texttt{spec2} \hspace{1cm} any distribution, especially a spectrum obtained with \texttt{spec} or \texttt{meanspec} (not in dB). This can be either a two-column matrix (col1 = frequency, col2 = amplitude) or a vector (amplitude).
  \item \texttt{scale} \hspace{1cm} a logical, if \texttt{TRUE} the distance is scaled by dividing the distance by the length of \texttt{spec1} (or \texttt{spec2}).
\end{itemize}

\textbf{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, i.e. computing the I-S distance between \texttt{spec1} and \texttt{spec2} is not the same as computing it between \texttt{spec2} and \texttt{spec1}. A symmetry can be obtained by calculating the mean between the two directions. The distance is obtained following:

\begin{equation}
D_{I-S}(\texttt{spec1}\|\texttt{spec2}) = \sum \frac{\texttt{spec1}}{\texttt{spec2}} - \log \left( \frac{\texttt{spec1}}{\texttt{spec2}} \right) - 1
\end{equation}

\textbf{Value}

The function returns a list of three items:

\begin{itemize}
  \item \texttt{D1} \hspace{1cm} The I-S distance of 'spec2' with respect to 'spec1' (\textit{i.e.} \(D(\texttt{spec1} \| \texttt{spec2})\))
  \item \texttt{D2} \hspace{1cm} The I-S distance of 'spec1' with respect to 'spec2' (\textit{i.e.} \(D(\texttt{spec2} \| \texttt{spec1})\))
  \item \texttt{D} \hspace{1cm} The symmetric distance (\textit{i.e.} \(D = 0.5*(\texttt{D1} + \texttt{D2})\))
\end{itemize}

If \texttt{scale} = \texttt{TRUE} the distance is divided by the length of \texttt{spec1} (or \texttt{spec2}).

\textbf{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, i.e. 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:

\[ D_{K-L}(spec1||spec2) = \sum spec1 \times \log \left( \frac{spec1}{spec2} \right) \]
Value

The function returns a list of three items:

- $D_1$: The K-L distance of 'spec2' with respect to 'spec1' (i.e. $D(spec1 \| spec2)$)
- $D_2$: The K-L distance of 'spec1' with respect to 'spec2' (i.e. $D(spec2 \| spec1)$)
- $D$: The symmetric K-L distance (i.e. $D = 0.5(D_1 + D_2)$)

Note

The base of the logarithm can be changed using the argument `base`. When set to base 2, the information is measured in units of bits. When set 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


See Also

ks.dist, logspec.dist, simspec, diffspec

Examples

```r
# 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)
```

Description

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

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, diffspect, logspect.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

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**

```r
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-column 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**

```r
## 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
```
localpeaks

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.
**logspec.dist**

**Log-spectral distance**

**Description**

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

**Usage**

```r
call = 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:
\[
D_{LS}(\text{spec1}||\text{spec2}) = D_{LS}(\text{spec2}||\text{spec1}) = \sqrt{\sum 10 \times \log_{10}(\frac{\text{spec1}}{\text{spec2}})^2}
\]

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**

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

**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 specify 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 devices 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

```r
## 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)
```
Median of the amplitude envelope

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:

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

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

meandB

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

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 frequencies.

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).
**meanspec**

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 package 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,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:

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

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


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

### Usage

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

### Arguments

- **f**: sampling 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

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 $s_{ref}$ the reference sensitivity (by default 1 mV/Pa).

Value

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

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 function 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


See Also

`soundscapespec, SAX, NDSI`

Examples

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

Description

This function generates noise.

Usage

```r
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 \texttt{runif} and gaussian noise is based on \texttt{rnorm}.

Value

A new wave is returned. The class of the returned object is set with the argument \texttt{output}.

Author(s)

Jerome Sueur <sueur@mnhn.fr>

See Also

\texttt{synth}, \texttt{pulse}

Examples

\begin{verbatim}
# 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)
\end{verbatim}

Description

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

Usage

\begin{verbatim}
notefreq(note, ref = 440, octave = 3)
\end{verbatim}

Arguments

\begin{itemize}
\item \texttt{note} \hspace{1cm} a numerical or a character vector. See Note.
\item \texttt{ref} \hspace{1cm} a numerical vector of length 1 for the reference frequency.
\item \texttt{octave} \hspace{1cm} a numerical vector of length for the octave number.
\end{itemize}

Details

The frequency is computed according to:

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

with:
\begin{itemize}
\item \texttt{ref} = reference frequency,
\item \texttt{octave} = octave number, and
\item \texttt{note} = rank of the note along the scale.
\end{itemize}
octaves

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)
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)
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

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

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

```r
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 columns (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.

cexlabel character size for axes labels.

fontxlabel 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 leave 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.
oscillo

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
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.
paste\texttt{w}

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

\textbf{Value}

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

\textbf{Author(s)}

Jerome Sueur and Caroline Simonis.

\textbf{See Also}

\texttt{oscillo, dynoscillo}

\textbf{Examples}

\begin{verbatim}
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)
\end{verbatim}

\begin{verbatim}
\begin{tabular}{ll}
\texttt{paste\texttt{w}} & \texttt{Paste a time wave to another one} \\
\end{tabular}
\end{verbatim}

\textbf{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.

\textbf{Usage}

\begin{verbatim}
paste\texttt{w}(\texttt{wave1}, \texttt{wave2}, \texttt{f}, \texttt{channel} = \texttt{c(1,1)}, \texttt{at} = "end",
join = \texttt{FALSE}, \texttt{tj}unction = \texttt{0},
\texttt{choose} = \texttt{FALSE}, \texttt{plot} = \texttt{FALSE},
\texttt{marks} = \texttt{TRUE}, \texttt{output} = "\texttt{matrix}" , . . . )
\end{verbatim}
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 values 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)

pelucens

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(pelucens)

Format

A Wave object.

Details

Duration = 3.309 s. Sampling frequency = 11025 hz.

Source

Recording by Jerome Sueur.

Examples

data(pelucens)
oscillo(pelucens,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 length 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 column = first derivative).

**Note**

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

**Author(s)**

Jerome Sueur

**References**


**See Also**

`phaseplot2`

**Examples**

```r
## 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

\textit{Phase-phase 2D plot of a time wave}

\section*{Description}
This function returns a 2D representation of a time wave against a delayed version of itself.

\section*{Usage}
\begin{verbatim}
phaseplot2(wave, f, channel = 1, tau = 1, type = "l",
xlab = "x(t)",
ylab = paste("x(t+\tau)", sep = ""), ...)
\end{verbatim}

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

\section*{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 \texttt{tau} that needs to be specified in number of samples. The conversion of \texttt{tau} in second is obtained by calculating $\texttt{tau}/\texttt{f}$, with \texttt{f} the sampling frequency.

\section*{Value}
Nothing is returned except an x-y plot.

\section*{Note}
Phase-phase plot can be used to test non-linearity.

\section*{Author(s)}
Jerome Sueur

\section*{References}
playlist

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, i.e., 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 be played
  are saved.

sample
  a logical, if TRUE the order of sound 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

```r
## 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

```r
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:

\[ H(f) = 1 + a^2 - 2 \times \alpha \times \cos(2 \times \pi \times f / f_s) \]
**Description**

This function generates a rectangle pulse.

**Usage**

```r
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`. 

---

**Examples**

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

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

```r
# 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**

```r
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.
**$\text{repw}$**

**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/](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)
```

$\text{repw}$  
**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**
a 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

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>wave</td>
<td>an R object.</td>
</tr>
<tr>
<td>f</td>
<td>sampling frequency of wave (in Hz). Does not need to be specified if embedded in wave.</td>
</tr>
<tr>
<td>channel</td>
<td>channel of the R object, by default left channel (1).</td>
</tr>
<tr>
<td>env</td>
<td>logical, if TRUE the amplitude envelope is reversed.</td>
</tr>
<tr>
<td>ifreq</td>
<td>logical, if TRUE the instantaneous frequency is reversed.</td>
</tr>
<tr>
<td>plot</td>
<td>logical, if TRUE returns an oscillographic plot of the reversed wave (by default FALSE).</td>
</tr>
<tr>
<td>output</td>
<td>character string, the class of the object to return, either &quot;matrix&quot;, &quot;Wave&quot;, &quot;Sample&quot;, &quot;audioSample&quot; or &quot;ts&quot;.</td>
</tr>
<tr>
<td>...</td>
<td>other oscillo graphical parameters.</td>
</tr>
</tbody>
</table>
rmam

Remove the amplitude modulations of a time wave

Description

This function 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 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

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


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)
Description

This function removes background noise by smoothing

Usage

```r
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 offset 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

```r
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:

\[
\text{roughness} = \int [D^2 x(t)]^2 dt
\]

Value

A vector of length 1.

Note

The value has not unit.

Author(s)

Jerome Sueur

References

See Also

* rugo, rms, sh, th, H.

Examples

```r
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:

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

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.
savewav

Author(s)
Jerome Sueur

References

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 overwrites 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

```r
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

\begin{verbatim}
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 \texttt{paste}. By default letters are returned separated.
\end{verbatim}

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 Approxima-
tion (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 \texttt{collapse} is \texttt{NULL}) or number of character (when \texttt{collapse} is
not \texttt{NULL}) corresponding to \texttt{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


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.
seedata

Author(s)
Jérôme Sueur

References

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))
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, Emilian 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
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 of wave2. 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' amplitude 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**

```r
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:

\[
F = N \times \sqrt[N]{\prod_{i=1}^{N} y_i} \div \sum_{i=1}^{N} y_i
\]

with:

- \( y \) = relative amplitude of the \( i \) frequency,
- \( N \) = 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)
specia<-spec(a,f=8000,at=0.5,plot=FALSE)
sfm(specia)
  # [1] 0
b<-noisew(d=1,f=8000)
spectb<-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:

\[ S = -\sum_{i=1}^{N} y_i \log_2(y_i) \]

\[ \log_2(N) \]

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 calculated according to:

\[ R = \frac{1}{1 - \alpha} \times \log_2\left(\sum_{i=1}^{N} y_i^\alpha\right) \]

with

\[ \alpha \geq 0 \]

\[ \alpha \neq 1 \]

\[ y = \text{relative amplitude of the } i \text{ frequency,} \]

\[ \sum_{i=1}^{N} y_i = 1 \]

and \( N = \text{number of frequencies.} \)

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
sheep

References


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(specb, alpha=2)
sh(specb, 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

```r
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:

\[ S = \frac{100/N}{\sum_{i=1}^{N} \frac{\min \text{spec1}(i), \text{spec2}(i)}{\max \text{spec1}(i), \text{spec2}(i)}} \]

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


See Also

spec, meanspec, corspec, diffspect, diffenv, kl.dist, ks.dist, logspec.dist, itakura.dist
Examples

```r
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)
spec'a <- spec(a,f=8000,at=0.5,plot=FALSE)
spec'b <- spec(b,f=8000,at=0.5,plot=FALSE)
spec'c <- spec(c,f=8000,at=0.5,plot=FALSE)
spec'd <- spec(d,f=8000,at=0.5,plot=FALSE)
simspec(spec'a,spec'a)
simspec(spec'a,spec'b)
simspec(spec'a,spec'c,plot=TRUE)
simspec(spec'b,spec'c,plot=TRUE)
# [1] 12.05652
simspec(spec'a,spec'd,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).
w         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

```r
# 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

```r
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/](https://www.wildlifeacoustics.com/)

### See Also

`songmeterdiag`, `audiomoth`, `strptime` for the POSIX time format.

### Examples

```r
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)
```

### 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

```r
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/](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
start <- strptime("2019-01-01 00:00:00", format)
end <- strptime("2019-01-31 23:30:00", 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)
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’s method (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


See Also

*spec, meanspec, SAX, NDSI*

Examples

```r
## 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

```r
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
g
- `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

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

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).
w1 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.
spec

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,
# horizontally 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:

$$F = \left( \sum |s(t+1) - s(t)|^p \right)^{\frac{1}{p}}$$

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

specprop

See Also

spectro, ACI

Examples

```r
## 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])
```

Description

This function returns a list of statistical properties of a frequency spectrum.

Usage

```r
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 symmetric,
- \(S > 0\) 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").
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>fftw</code></td>
<td>if TRUE calls the function FFT of the library fftw. See Notes.</td>
</tr>
<tr>
<td><code>dB</code></td>
<td>a character string specifying the type dB to return: &quot;max0&quot; (default) for a maximum dB value at 0, &quot;A&quot;, &quot;B&quot;, &quot;C&quot;, &quot;D&quot;, and &quot;ITU&quot; for common dB weights. If set to NULL, then a linear scale is used.</td>
</tr>
<tr>
<td><code>dBref</code></td>
<td>a dB reference value. NULL by default but should be set to 2*10e-5 for a 20 microPa reference.</td>
</tr>
<tr>
<td><code>plot</code></td>
<td>logical, if TRUE plots the spectrogram (by default TRUE).</td>
</tr>
<tr>
<td><code>flog</code></td>
<td>a logical to plot the frequency on a logarithmic scale.</td>
</tr>
<tr>
<td><code>grid</code></td>
<td>logical, if TRUE plots a y-axis grid (by default TRUE).</td>
</tr>
<tr>
<td><code>osc</code></td>
<td>logical, if TRUE plots an oscillogram beneath the spectrogram (by default FALSE).</td>
</tr>
<tr>
<td><code>scale</code></td>
<td>logical, if TRUE plots a dB colour scale on the right side of the spectrogram (by default TRUE).</td>
</tr>
<tr>
<td><code>cont</code></td>
<td>logical, if TRUE overplots contour lines on the spectrogram (by default FALSE).</td>
</tr>
<tr>
<td><code>collevels</code></td>
<td>a set of levels which are used to partition the amplitude range of the spectrogram (in dB).</td>
</tr>
<tr>
<td><code>palette</code></td>
<td>a color palette function to be used to assign colors in the plot, see Details.</td>
</tr>
<tr>
<td><code>contlevels</code></td>
<td>a set of levels which are used to partition the amplitude range for contour over-plot (in dB).</td>
</tr>
<tr>
<td><code>colcont</code></td>
<td>colour for cont plotting.</td>
</tr>
<tr>
<td><code>colbg</code></td>
<td>background colour.</td>
</tr>
<tr>
<td><code>colgrid</code></td>
<td>colour for grid plotting.</td>
</tr>
<tr>
<td><code>colaxis</code></td>
<td>color of the axes.</td>
</tr>
<tr>
<td><code>collab</code></td>
<td>color of the labels.</td>
</tr>
<tr>
<td><code>cexlab</code></td>
<td>size of the labels.</td>
</tr>
<tr>
<td><code>cexaxis</code></td>
<td>size of the axes.</td>
</tr>
<tr>
<td><code>tlab</code></td>
<td>label of the time axis.</td>
</tr>
<tr>
<td><code>flab</code></td>
<td>label of the frequency axis.</td>
</tr>
<tr>
<td><code>alab</code></td>
<td>label of the amplitude axis.</td>
</tr>
<tr>
<td><code>scalelab</code></td>
<td>amplitude scale label.</td>
</tr>
<tr>
<td><code>main</code></td>
<td>label of the main title.</td>
</tr>
<tr>
<td><code>scalefontlab</code></td>
<td>font of the amplitude scale label.</td>
</tr>
<tr>
<td><code>scalecexlab</code></td>
<td>cex of the amplitude scale label.</td>
</tr>
<tr>
<td><code>axisX</code></td>
<td>logical, if TRUE plots time X-axis (by default TRUE).</td>
</tr>
<tr>
<td><code>axisY</code></td>
<td>logical, if TRUE plots frequency Y-axis (by default TRUE).</td>
</tr>
<tr>
<td><code>tlim</code></td>
<td>modifications of the time X-axis limits.</td>
</tr>
<tr>
<td><code>trel</code></td>
<td>time X-axis with a relative scale when tlim is not null, i.e. relative to wave.</td>
</tr>
<tr>
<td><code>flim</code></td>
<td>modifications of the frequency Y-axis limits (in kHz).</td>
</tr>
<tr>
<td><code>flimd</code></td>
<td>dynamic modifications of the frequency Y-axis limits. New <code>wl</code> and <code>ovlp</code> arguments are applied to increase time/frequency resolution.</td>
</tr>
</tbody>
</table>
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 w1 value. Choosing a high w1 value will increase the frequency resolution but reduce
the temporal one, and vice versa. The frequency precision is obtained by calculating the ratio f/w1,
and the temporal precision is obtained by calculating the reverse ratio w1/f. This problem can be
reduced in some way withzp 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 see-
wave: 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 re-
verse 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 w1/2.

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 package 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

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 (20 microPa)
spectro(tico,f=22050, dBref=2*10^-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,
### 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**

```r
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 \( \text{wl} \) value. Choosing a high \( \text{wl} \) value will increase the frequency resolution but reduce the temporal one, and vice versa. The frequency precision is obtained by calculating the ratio \( f/\text{wl} \), and the temporal precision is obtained by calculating the reverse ratio \( \text{wl}/f \). This problem can be reduced in some way with \( \text{zp} \) 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 \( \text{wl}/2 \).

Note

This function requires \texttt{rgl} and is based on \texttt{fft}. See examples of \texttt{spectro} for analysis arguments (\( \text{wl}, \text{zp}, \text{ovlp} \)).

Author(s)

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

See Also

\texttt{spectro, ggspectro, lts, dynspec, wf, fft}. 
Examples

```r
## 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
symba

See Also

fir, drawfilter, ffilter, combfilter, bwfilter

Examples

f <- 44100
a <- noise(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:

$$I = H_x + H_y - H_{xy}$$

with $H_x$ the entropy of $x$ symbol series, $H_y$ the entropy of $y$ symbol series, and $H_{xy}$ 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


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 function synthesizes pure or harmonic tone sound with amplitude modulation (am) and/or frequency modulation (fm).

Usage

```r
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 synthesized 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 amplitude 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 amplitude
    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 har-
    monics) 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


See Also

`synth2`, `noisew`, `pulse`, `echo`

Examples

```r
## 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 frequency (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 function 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

```r
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 envelope.
- `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 synthesized 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 modifications 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


See Also

synth2, noisew, pulse, echo

Examples

```r
## 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 instantaneous 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)
```
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(2,5), 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, i.e., 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.
th

**Author(s)**

Pierre Aumond, Guillaume Corbeau

**References**


**See Also**

ACI, NDSI

**Examples**

```r
## 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**

```r
th(env, breaks)
```

**Arguments**

<table>
<thead>
<tr>
<th>env</th>
<th>a data set resulting of an envelope obtained using env</th>
</tr>
</thead>
<tbody>
<tr>
<td>breaks</td>
<td>'breaks' argument of hist to compute the entropy on the distribution obtained with an histogram.</td>
</tr>
</tbody>
</table>
Temporal entropy is calculated according to:

\[ S = -\sum_{i=1}^{N} y_i \log_2(y_i) / \log_2(N) \]

with:

\( y = \) relative amplitude of the \( i \) envelope point,

and

\[ \sum_{i=1}^{N} y_i = 1 \]

and \( N = \) number of envelope points.

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

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.

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)
environi<-env(orni,f=22050,plot=FALSE)
th(environi)
# Smoothing the envelope might slightly change the result.
environiS<-env(orni,f=22050,smooth=c(50,0),plot=FALSE)
th(environiS)
# 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)
environi2<-env(orni2,f=22050,plot=FALSE)
th(environi2)
# 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**

```r
data(tico)
```

**Format**

A Wave object.

**Details**

Duration = 1.795 s. Sampling frequency = 22050 hz.

**Source**

Recording by Thierry Aubin.

**Examples**

```r
data(tico)
oscillo(tico,f=22050)
```
timelapse

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 alphabetically 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

```r
## 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

```r
timer(wave, f, channel = 1, threshold = 5, dmin = NULL, envt="abs",
power = 1, msmooth = NULL, ksmooth = NULL,
sssmooth = NULL, assmooth=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 amplitude 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:

\[ y_n = \frac{x_n^{2/m}}{m} - \left( x_{n-M} \times x_{n+M} \right)^{1/m} \]

with \( m \) the exponent parameter and \( M \) the lag parameter which both are usually equal to 1 for a conventional operator.

The Teager-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


See Also

eenv, ifreq.

Examples

```r
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)
```
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 Celsius at atmospheric pressure.

Speed of sound in sea-water is computed according to Mackenzie equation:

\[
 c = 1448.96 + 4.591 \times t - 5.304.10^{-2} \times t^2 \\
 + 2.374.10^{-4} \times t^3 + 1.34 \times (s - 35) + 1.63.10^{-2} \times d \\
 + 1.675.10^{-7} \times d^2 - 1.025.10^{-2} \times t \times (s - 35)
\]
The wavelength of air-borne sound is calculated as

\[-7.139 \times 10^{-13} \times t \times d^3\]

with \( t \) = temperature in degrees Celsius; \( s \) = salinity in parts per thousand; \( d \) = depth in meters; range of validity: temperature 2 to 30 degrees Celsius, 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:

1. wavelength in meters
2. speed of sound in meters/second.

**Author(s)**

Jerome Sueur <sueur@mnhn.fr>

**References**

http://resource.npl.co.uk

**Examples**

```r
# 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 Celsius and 11 degrees Celsius
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=spec.pal(nb),
       xlab="Frequency (Hz)",ylab="Wavelength (m)")
title("Wavelength of air-borne sound at different temperatures (deg. C)")
```
wav2flac

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 default 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 default 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)
```

### 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.
window name, see `ftwindow` (by default "hanning").

a matrix if `wave` is not provided.

horizontal 'offset' which shifts actual x-values slightly per row for visibility. Fractional parts will be removed.

vertical 'offset' which separates traces.

a color or a color palette function to be used to assign colors in the plot

title of the frequency x-axis.

title of the amplitude y-axis.

a logical, if `TRUE` adds the frequency x-axis according to `f`.

a logical, if `TRUE` adds the amplitude y-axis according.

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).

argument of `polygon`: the color to draw the border. The default, 'NULL', means to use 'par("fg")'. Use 'border = NA' to omit borders.

a logical, if `TRUE` plots lines instead of surfaces (polygons).

line width.

other graphical arguments to passed to `plot`

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.

The function is well adapted to display short-term Fourier transform. However, any matrix can be called using the argument `x` instead of `wave`.

Carl G. Witthoft and Jerome Sueur <sueur@mnhn.fr>

`spectro`, `spectro3D`, `dynspec`

```r
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)```
write.audacity

# 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)))}
w(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 columns, 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 original 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, ...)

zc

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)

Jerome Sueur <sueur@mnhn.fr>, Caroline Simonis and Thierry Aubin

References


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 function 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
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 the zero-crossing rate 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:

\[
zcr = \frac{1}{2 \times N} \sum_{t=0}^{N-1} |\text{sgn}(x(t+1)) - \text{sgn}(x(t))|
\]

with:

- \(N\) the length of the signal \(x\)
- and where:

\[
\text{sgn}(x(t)) = 1 \quad \text{if} \quad x(t) \geq 0
\]

and

\[
\text{sgn}(x(t)) = -1 \quad \text{if} \quad x(t) < 0
\]

Value

There are two possibilities:

1. a numeric vector of length 1 if \(w_l\) is NULL,
2. a numeric two-column matrix is returned with the first column being time (s) and the second column being the zero-crossing rate (no scale) if \(w_l\) is not NULL.

Note

There are two possibilities:

1. if \(w_l\) is NULL then the zero-crossing rate is computed for the complete signal.
2. if \(w_l\) is not NULL then the zero-crossing rate is computed 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


See Also

zc
Examples

```r
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)
```
Index

* **IO**
  - export, 73
  - ftwindow, 87
  - savewav, 153
  - sox, 173
  - wav2flac, 207
* **datagen**
  - drawenv, 60
  - echo, 70
  - noise, 123
  - pulse, 139
  - setenv, 159
  - synth, 191
  - synth2, 194
* **datasets**
  - orni, 126
  - peewit, 133
  - pellucens, 134
  - sheep, 163
  - tico, 199
* **data**
  - attenuation, 16
  - audiomoth, 17
  - audiomoth.rename, 18
  - read.audacity, 142
  - songmeter, 167
  - songmeterdiag, 169
  - write.audacity, 210
* **distribution**
  - diffcumspec, 51
  - itakura.dist, 100
  - kl.dist, 101
  - ks.dist, 102
  - logspec.dist, 108
* **dplot**
  - ama, 13
  - autoc, 19
  - cooh, 23
  - ceps, 26
  - cepstro, 28
  - coh, 30
  - corenv, 34
  - corspec, 36
  - covspectro, 38
  - cutw, 44
  - dBscale, 45
  - deletew, 48
  - dfreq, 50
  - diffenv, 53
  - diffspec, 55
  - dynoscillo, 64
  - dynspec, 65
  - dynspectro, 67
  - env, 71
  - fadew, 74
  - fbands, 75
  - fma, 83
  - fpeaks, 85
  - fund, 89
  - ggspectro, 92
  - ifreq, 96
  - localpeaks, 107
  - lts, 110
  - meanspec, 114
  - mutew, 121
  - oscillo, 127
  - oscilloST, 130
  - pastew, 131
  - phaseplot, 134
  - phaseplot2, 136
  - Q, 140
  - repw, 143
  - resamp, 144
  - revw, 145
  - rmoffset, 149
  - seadata, 157
  - seewave, 158
  - simspec, 164

217
soundscapespec, 172
spec, 174
specprop, 179
spectro, 181
spectro3D, 186
wf, 208
zc, 212
* filter
  afilter, 10
  bwfilter, 22
  combfilter, 31
drawfilter, 61
  ffilter, 79
  fir, 82
  preemphasis, 138
  squarefilter, 188
* index
  AR, 14
  M, 112
* input
  audiomoth, 17
  audiomoth.rename, 18
  read.audacity, 142
  songmeter, 167
  songmeterdiag, 169
  write.audacity, 210
* maths
  notefreq, 124
  octaves, 125
* math
  convSPL, 33
dBweight, 47
dfodpller, 77
gammatone, 90
meandB, 113
mel, 116
melfilterbank, 118
micsens, 119
moredB, 120
sddB, 156
wasp, 205
* ts
  ACI, 5
  acoustat, 6
  afilter, 10
  akamatsu, 11
  ama, 13
  AR, 14
  autoc, 19
  beep, 21
  bwfilter, 22
cceps, 23
ccepstro, 26
coh, 30
  combfilter, 31
corenv, 34
corspec, 36
covsens, 38
crest, 40
csh, 41
cutspec, 43
cutw, 44
  dBscale, 45
dBweight, 47
deltew, 48
dffreq, 50
diffcumspec, 51
diffenv, 53
diff,w, 55
diffwave, 57
discerts, 59
drawenv, 60
drawfilter, 61
duration, 63
dynoscillo, 64
dynsens, 65
dynspectro, 67
echo, 70
eve, 71
fadew, 74
fbands, 75
ffilter, 79
field, 80
fir, 82
fma, 83
fpeaks, 85
ftwindow, 87
fund, 89
gammatone, 90
ggsspectro, 92
H, 94
hilbert, 95
ifreq, 96
istft, 98
itakura.dist, 100
INDEX

kl.dist, 101
ks.dist, 102
lfs, 104
listen, 106
localpeaks, 107
logspec.dist, 108
lts, 110
M, 112
meanspec, 114
melfilterbank, 118
mutew, 121
NDSI, 122
noisew, 123
oscillo, 127
oscilloST, 130
pastew, 131
phaseplot, 134
phaseplot2, 136
playlist, 137
preemphasis, 138
pulsew, 139
Q, 140
repw, 143
resamp, 144
revw, 145
rmam, 146
rmnoise, 148
rmoffset, 149
rms, 150
roughness, 151
rugo, 152
SAX, 154
seewave, 158
setenv, 159
sfm, 160
sh, 161
simspec, 164
smoothw, 166
soundscape.spec, 172
spec, 174
specflux, 177
specprop, 179
spectro, 181
spectro3D, 186
squarefilter, 188
symba, 189
synth, 191
synth2, 194
TFSD, 196
th, 197
timelapse, 200
TKEO, 203
wf, 208
zapsilw, 211
zc, 212
zc, 214
acf, 20
ACI, 5, 178, 179, 197
acoustat, 6
addsilw, 9, 45, 49, 74, 122, 132, 144, 146
afilter, 10, 23, 80, 83, 139, 148, 212
akamatsu, 11
ama, 13, 84
approx, 213
AR, 14, 113
as.POSIXct, 18
attenuation, 16, 33
audiomoth, 17, 19, 111, 169
audiomoth.rename, 18, 18
autoc, 10, 19, 27, 30, 90
axis, 46, 209
barplot, 76, 172
beep, 21
bwfilter, 22, 23, 32, 80, 83, 139, 189
ccoh, 23, 31
ceps, 20, 26, 29, 30, 90
cepstrogram, 27, 28, 90
coh, 25, 30
combfilter, 31, 32, 62, 80, 139, 189
covcontour, 24, 25, 29, 184
covfilter, 70, 82
covSPL, 17, 33, 48, 113, 120, 157
cor, 34–37, 39, 40
cor.test, 35, 37
corenv, 34, 40, 54
corspec, 35, 36, 37, 40, 56, 115, 141, 165, 176
covspec, 35, 37, 38, 88
crest, 40
csh, 10, 41, 95, 161, 163, 198
cutspec, 43
cutw, 9, 44, 49, 74, 122, 129, 132, 144, 202
dBscale, 45, 185
dBweight, 33, 47, 113, 120, 157
Index

afew, 9, 45, 48, 74, 122, 132, 144, 146
dfreq. 10, 50, 88
diffcumspec, 51, 56, 104
diffenv, 53, 56, 58, 165
diffspec, 53, 54, 55, 58, 101, 102, 104, 109, 115, 165
diffwave, 54, 57
dir, 15
discrets, 59, 156, 190
drawenv, 60, 62, 160
drawfilter, 32, 61, 189
duration, 63
dynosillo, 64, 129, 131
dynspec, 64, 65, 69, 93, 115, 176, 185, 187, 209
dynspectro, 67, 67
echo, 70, 193, 195
export, 73, 154
fadem, 9, 45, 49, 74, 122, 132, 144, 146
fbands, 75
ffilter, 23, 32, 62, 79, 83, 99, 105, 139, 188, 189
fft, 27, 51, 67, 69, 82, 99, 115, 141, 176, 184, 185, 187
field, 80
filled contour, 25, 46, 69, 184
filter, 167
fir, 32, 62, 80, 82, 139, 167, 188, 189
fma, 14, 83
fpeaks, 27, 85, 107, 108, 115, 176
ftwindow, 5, 7, 38, 42, 50, 65, 68, 79, 82, 87, 98, 105, 110, 114, 172, 175, 178, 182, 186, 196, 209
fund, 27, 30, 89

gammatone, 90, 119
ggspectro, 92, 111, 185, 187
H, 94, 152, 153, 163, 198
hilbert, 72, 84, 95, 97, 98, 147
hist, 197
ifreq, 84, 96, 96, 194, 204, 213

image, 69, 110, 111
IQR, 180
istft, 79, 98, 105
itakura.dist, 53, 56, 100, 104, 109, 165

kernel, 34, 54, 57, 66, 68, 72, 94, 159, 202
kl.dist, 53, 56, 101, 104, 109, 165
ks.dist, 53, 56, 101, 102, 102, 109, 165
ks.test, 103

lfs, 23, 80, 83, 99, 104, 139
listen, 21, 106, 138
localpeaks, 86, 107, 115, 176
locator, 25, 61, 184
log, 101
logspectro, 53, 56, 101, 102, 104, 108, 165
lts, 110, 185, 187

M, 15, 112
mean, 150, 152, 180
meandB, 113, 157
median, 180
melp, 116, 119
melfilterbank, 91, 117, 118
micsens, 119
Mod, 72
moredB, 17, 33, 48, 113, 120, 157
mutew, 9, 45, 49, 74, 121, 132, 144, 146

na.omit, 213
NDSI, 122, 123, 173, 197
noisew, 123, 140, 148, 193, 195
normalize, 154
notefreq, 124, 126

octaves, 125, 125
OlsonNames, 18
orni, 126
oscillo, 9–11, 24, 41, 44, 45, 49, 64, 67, 69, 70, 72, 74, 96, 121, 122, 127, 131, 132, 143–147, 149, 159, 184, 185, 212
oscilloST, 64, 129, 130

par, 128
INDEX

paste, 155
pasteW, 9, 45, 49, 74, 122, 129, 131, 144, 146, 201, 202
peewit, 133
pellucens, 134
phaseplot, 134, 137
phaseplot2, 135, 136
play, 106, 137, 138
playlist, 137
polygon, 209
preemphasis, 23, 80, 83, 138
pulse, 124, 193, 195
pulsew, 139
Q, 140
quantile, 180
read.audacity, 142, 211
repw, 132, 143
resamp, 144
rew, 9, 45, 49, 74, 122, 132, 144, 145
rmam, 146
rmnoise, 148
rmoffset, 149
rms, 40, 41, 150, 152, 153
rnorm, 124
roughness, 151, 153
round, 85
rugw, 152, 152
runif, 124
savewav, 153, 207
SAX, 123, 154, 173, 190
sd, 180
sddB, 113, 156
seedata, 157
seeWave, 158
seewave-package (seeWave), 158
setenv, 61, 159
sfm, 43, 160, 163, 180
sh, 42, 43, 94, 95, 152, 153, 161, 161, 180, 198
sheep, 163
simspec, 53, 56, 101, 102, 104, 109, 115, 164
smooth.spline, 148
smoothw, 166
songmeter, 18, 19, 111, 167, 171
songmeterdiag, 169, 169
soundscapespec, 122, 123, 156, 172
sox, 173
spec.pgram, 25, 31
specflux, 6, 177
specprop, 8, 179
spectro3D, 67, 69, 88, 93, 111, 185, 186, 209
squarefilter, 32, 62, 188
strptime, 169
symba, 60, 156, 189
synth, 61, 71, 124, 140, 160, 191
synth2, 193, 194, 195
TFSD, 196
th, 15, 42, 94, 95, 152, 153, 163, 197
tico, 199
timelapse, 132, 200
timer, 10, 129, 201
TKEO, 203
tkeo(TKEO), 203
wasp, 78, 205
wav2flac, 207
Wave, 59, 154
wf, 67, 69, 185, 187, 208
write.audacity, 143, 210
write.table, 73
writeWave, 153, 154
zapsilw, 9, 45, 49, 74, 122, 132, 144, 211
zc, 10, 98, 212, 213, 215
zcr, 214