QWTB documentation

Implemented algorithms

QWTB version 0.1

https://qwtb.github.io/qwtb/

Contents

Introduction

This document gives overview of the algorithms implemented in toolbox QWTB.

Toolbox was realized within the EMRP-Project SIB59 Q-Wave. The EMRP is jointly funded by the EMRP par- ticipating countries within EURAMET and the European Union.

4PSWF – Four Parameter Sine Wave Fitting

Description

Id: 4PSWF

Name: Four Parameter Sine Wave Fit

Description: Fits a sine wave to the recorded data by means of least squares fitting using 4 parameter (frequency, amplitude, phase and offset) model. An estimate of signal frequency is required. Due to non-linear characteristic, converge is not always achieved. When run in Matlab, function 'lsqnonlin' in Optimization toolbox is used. When run in GNU Octave, function 'leasqr' in GNU Octave Forge package optim is used.

Citation:

Remarks: If Time series |t| is not supplied, wrapper will calculate |t| from sampling frequency |fs| or if not supplied, sampling time |Ts| is used to calculate |t|.

License: MIT License

Provides GUF: no

Provides MCM: no

Input Quantities

Required: `t` or `fs` or `Ts`, `y`

Descriptions:

`Ts`	– Sampling time
`fs`	– Sampling frequency

t – Time series
y – Sampled values

Output Quantities:

A – Amplitude of main signal component
O – Offset of signal
f – Frequency of main signal component
ph – Phase of main signal component

Example

Four parameter sine wave fitting

Example for algorithm 4PSWF.

4PSWF is an algorithm for estimating the frequency, amplitude, and phase of the sine waveform. The algorithm use least squares method. Algorithm requires good estimate of frequency.

Contents

- Generate sample data
- Call algorithm
- Display results

Generate sample data

Two quantities are prepared: t and y, representing 1 second of sinus waveform of nominal frequency 1 kHz, nominal amplitude 1 V, nominal phase 1 rad and offset 1 V sampled at sampling frequency 10 kHz.

```
DI = [];
Anom = 2; fnom = 100; phnom = 1; Onom = 0.2;
DI.t.v = [0:1/1e4:1-1/1e4];
DI.y.v = Anom*sin(2*pi*fnom*DI.t.v + phnom) + Onom;
```

Lets make an estimate of frequency 0.2 percent higher than nominal value:

```
DI.fest.v = 100.2;
```

Call algorithm

Use QWTB to apply algorithm 4PSWF to data DI.

```
CS.verbose = 1;
DO = qwtb('4PSWF', DI, CS);
```

```
QWTB: no uncertainty calculation
Fitting started

Local minimum found.

Optimization completed because the size of the
   gradient is less than
the default value of the function tolerance.



Fitting finished
```

Display results

Results is the amplitude, frequency and phase of sampled waveform.

```
A = DO.A.v
f = DO.f.v
ph = DO.ph.v
O = DO.O.v
```

```

A =
      2.0000


f =

     100


ph =

      1.0000


O =

      0.2000
```

Errors of estimation in parts per milion:

```
Aerrppm = (DO.A.v - Anom)/Anom .* 1e6
ferrppm = (DO.f.v - fnom)/fnom .* 1e6
pherrppm = (DO.ph.v - phnom)/phnom .* 1e6
Oerrppm = (DO.O.v - Onom)/Onom .* 1e6
```

```
Aerrppm =

     4.8894e-07


ferrppm =

       0


pherrppm =

     4.8850e-09
```

```
Oerrppm =

   -1.1102e-08
```

ADEV – Allan Deviation

Description

Id: ADEV

Name: Allan Deviation

Description: Compute the Allan deviation for a set of time-domain frequency data.

Citation: D.W. Allan, "The Statistics of Atomic Frequency Standards", Proc. IEEE, Vol. 54, No. 2, pp. 221-230, Feb. 1966. Implementation by M. A. Hopcroft, mhopeng@gmail.com, Matlab Central, online: `http://www.mathworks.com/matlabcentral/fileexchange/13246-allan` Test data by W. J. Riley, "The Calculation of Time Domain Frequency Stability", online: `http://www.wriley.com/paper1ht.htm`

Remarks: If sampling frequency |fs| is not supplied, wrapper will calculate |fs| from sampling time |Ts| or if not supplied, first two elements of time series |t| are used to calculate |Ts|. If observation time(s) |tau| is not supplied, tau values are automatically generated. Tau values must be divisible by 1/|fs|. Invalid values are ignored. For tau values really used in the calculation see the output.

License: BSD License

Provides GUF: yes

Provides MCM: no

Input Quantities

Required: fs or Ts or t, y

Optional: tau

Descriptions:

`Ts` – Sampling time
`fs` – Sampling frequency
`t` – Time series
`tau` – Observation time
`y` – Sampled values

Output Quantities:

`adev` – Allan deviation
`tau` – Observation time of resulted values

Example

Allan Deviation

Example for algorithm ADEV.

ADEV is an algorithm to compute the Allan deviation for a set of time-domain frequency data.

See also W. J. Riley, "The Calculation of Time Domain Frequency Stability". Implementation: M. A. Hopcroft, `mhopeng@gmail.com`, Matlab Central.’

Contents

- Generate sample data
- Call algorithm
- Display results

Generate sample data

A random numbers with normal probability distribution function will be generated into input data `DI.y.v`. Next a drift will be added.

```
DI = [];
DI.y.v = 1.5 + 3.*randn(1, 1e3);
DI.y.v = DI.y.v + [1:1:1e3]./100;
```

Lets suppose a sampling frequency is 1 Hz. The algorithm will generate all possible tau values automatically.

```
DI.fs.v = 1;
```

Call algorithm

Use QWTB to apply algorithm ADEV to data DI.

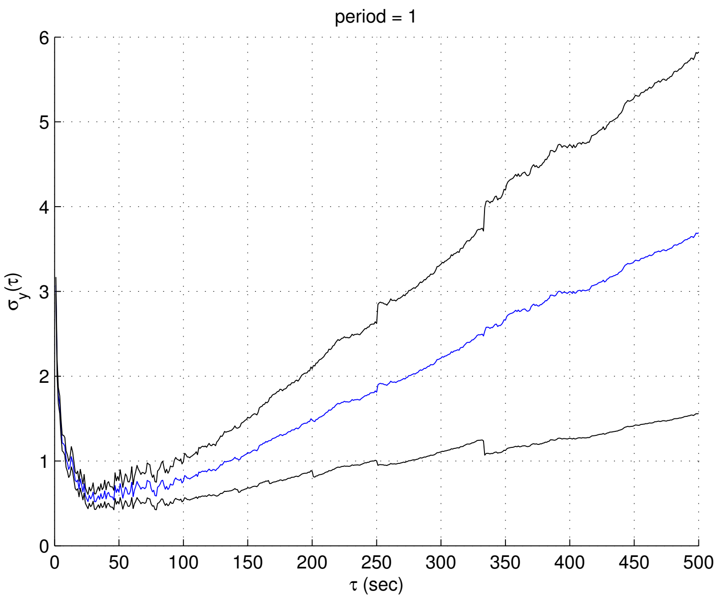
```
DO = qwtb('ADEV', DI);
```

```
QWTB: no uncertainty calculation
```

Display results

Log log figure is the best to see allan deviation results:

```
figure; hold on
loglog(DO.tau.v, DO.adev.v, '-b')
loglog(DO.tau.v, DO.adev.v + DO.adev.u, '-k')
loglog(DO.tau.v, DO.adev.v - DO.adev.u, '-k')
xlabel('\tau (sec)');
ylabel('\sigma_y(\tau)');
title(['period = ' num2str(1/DI.fs.v)]);
grid('on'); hold off
```


period = 1
$\sigma_y(\tau)$
τ (sec)
0
50
100
150
200
250
300
350
400
450
500
0
1
2
3
4
5
6

FourPSF – Standard Four Parameter Sine Wave Fit according IEEE Std 1241-2000

Description

Id: FourPSF

Name: Standard Four Parameter Sine Wave Fit according IEEE Std 1241-2000

Description: Fits a sine wave to the recorded data using 4 parameter (frequency, amplitude, phase and offset) model. The algorithm is according IEEE Standard for Terminology and Test methods for Analog-to-Digital Converters 1241-2000

Citation: IEEE Std 1241-2000, Implementation written by Zoltán Tamás Bilau, modified by Janos Markus. Id: sfit4.m,v 3.0 2004/04/19 11:20:09 markus Exp. Copyright (c) 2001-2004 by Istvan Kollar and Janos Markus. Modified 2016 Rado Lapuh

Remarks: If sampling time |Ts| is not supplied, wrapper will calculate |Ts| from sampling frequency |fs| or if not supplied, time series |t| is used to calculate |Ts|.

License: UNKNOWN

Provides GUF: no

Provides MCM: no

Input Quantities

Required: Ts or fs or t, y

Descriptions:

Ts – Sampling time
fs – Sampling frequency
t – Time series
y – Sampled values

Output Quantities:

A – Amplitude of main signal component
O – Offset of signal
f – Frequency of main signal component
ph – Phase of main signal component

Example

Four parameter sine wave fitting

Example for algorithm FourPSF.

FourPSF is an algorithm for estimating the frequency, amplitude, phase and offset of the sine waveform according standard IEEE Std 1241-2000';

Contents

- Generate sample data
- Call algorithm
- Display results

Generate sample data

Two quantities are prepared: t and y, representing 1 second of sinus waveform of nominal frequency 1 kHz, nominal amplitude 1 V, nominal phase 1 rad and offset 1 V sampled at sampling frequency 10 kHz.

```
DI = [];
Anom = 2; fnom = 100; phnom = 1; Onom = 0.2;
DI.t.v = [0:1/1e4:1-1/1e4];
DI.y.v = Anom*sin(2*pi*fnom*DI.t.v + phnom) + Onom;
```

Call algorithm

Use QWTB to apply algorithm FourPSF to data DI.

```
CS.verbose = 1;
DO = qwtb('FourPSF', DI, CS);
```

```
QWTB: no uncertainty calculation
QWTB: FourPSF wrapper: sampling time was calculated
    from first two elements of time series
```

Display results

Results is the amplitude, frequency, phase and offset of sampled waveform.

```
A = DO.A.v
f = DO.f.v
ph = DO.ph.v
O = DO.O.v
```

```
A =

    2.0000


f =

   100


ph =

    1.0000


O =
    0.2000
```

Errors of estimation in parts per milion:

```
Aerrppm = (DO.A.v - Anom)/Anom .* 1e6
ferrppm = (DO.f.v - fnom)/fnom .* 1e6
pherrppm = (DO.ph.v - phnom)/phnom .* 1e6
Oerrppm = (DO.O.v - Onom)/Onom .* 1e6
```

```

Aerrppm =

   2.2204e-09


ferrppm =

     0


pherrppm =

   8.8818e-09


Oerrppm =

   1.3878e-09
```

iDFT2p – 2-point interpolated DFT frequency estimator

Description

Id: iDFT2p

Name: 2-point interpolated DFT frequency estimator

Description: An algorithm for estimating the frequency, amplitude, phase and offset of the fundamental component using interpolated discrete Fourier transform. Rectangular or Hann window can be used for DFT.

Citation: Krzysztof Duda: Interpolation algorithms of DFT for parameters estimation of sinusoidal and damped sinusoidal signals. In S. M. Salih, editor, Fourier Transform - Signal Processing, chapter 1, pages 3-32, InTech, 2012. `http://www.intechopen.com/books/fourier-transform-signal-processing/interpolated-dft` Implemented by Rado Lapuh, 2016.

Remarks: If sampling time |Ts| is not supplied, wrapper will calculate |Ts| from sampling frequency |fs| or if not supplied, first two elements of time series |t| are used to calculate |Ts|. The optional parameter |window| can be set to values ‘rectangular‘ or ‘Hann‘. If parameter is not supplied, Hann window will be used.

License: Implementation: MIT License

Provides GUF: no

Provides MCM: no

Input Quantities

Required: Ts or fs or t, y

Optional: window

Parameters: window

Descriptions:

Ts – Sampling time
fs – Sampling frequency
t – Time series
window – DFT window: 'Hann' or 'rectangular'
y – Sampled values

Output Quantities:

A – Amplitude of main signal component
O – Offset of main signal component
f – Frequency of main signal component
ph – Phase of main signal component

Example

2-point interpolated DFT frequency estimator

Example for algorithm iDFT3p.

iDFT2p is an algorithm for estimating the frequency, amplitude, and phase of the fundamental component using interpolated discrete Fourier transform. Rectangular or Hann window can be used for DFT.'; See also Krzysztof Duda: Interpolation algorithms of DFT for parameters estimation of sinusoidal and damped sinusoidal signals. In S. M. Salih, editor, Fourier Transform - Signal Processing, chapter 1, pages 3-32, InTech, 2012. `http://www.intechopen.com/books/fourier-transform-signal-processing/interpolated-dft` Implemented by Rado Lapuh, 2016.';

Contents

Generate sample data

Two quantities are prepared: Ts and y, representing 0.5 second of sinus waveform of nominal frequency 100 Hz, nominal amplitude 1 V and nominal phase 1 rad, sampled with sampling time 0.1 ms, with offset 0.1 V. The sampling is not coherent.

```
DI = [];
Anom = 1; fnom = 100; phnom = 1; Onom = 0.1;
DI.Ts.v = 1e-4;
t = [0:DI.Ts.v:0.5];
DI.y.v = Anom*sin(2*pi*fnom*t + phnom) + Onom;
```

Call algorithm

First a rectangular window will be selected to estimate main signal properties. Use QWTB to apply algorithm iDFT3p to data DI and put results into DOr.

```
DI.window.v = 'rectangular';
DOr = qwtb('iDFT2p', DI);
```

```
QWTB: no uncertainty calculation
```

Next a Hann window will be selected to estimate main signal properties Results will be put into DOh.

```
DI.window.v = 'Hann';
DOh = qwtb('iDFT2p', DI);
```

```
QWTB: no uncertainty calculation
```

Display results

Results is the amplitude, frequency and phase of sampled waveform. For rectangular window, the error from nominal in parts per milion is:

```
f_re = (DOr.f.v - fnom)./fnom .* 1e6
A_re = (DOr.A.v - Anom)./Anom .* 1e6
ph_re = (DOr.ph.v - phnom)./phnom .* 1e6
O_re = (DOr.O.v - Onom)./Onom .* 1e6
```

```
f_re =

   -0.8083


A_re =

   40.3148


ph_re =

  217.9412


O_re =

   1.6826e+03
```

For Hann window:

```
f_he = (DOh.f.v - fnom)./fnom .* 1e6
A_he = (DOh.A.v - Anom)./Anom .* 1e6
ph_he = (DOh.ph.v - phnom)./phnom .* 1e6
O_he = (DOh.O.v - Onom)./Onom .* 1e6
```

```
f_he =

   1.8426e-04
```

```
A_he =

   -0.0046


ph_he =

    6.2442


O_he =

   -0.6862
```

iDFT3p – 3-point interpolated DFT frequency estimator

Description

Id: iDFT3p

Name: 3-point interpolated DFT frequency estimator

Description: An algorithm for estimating the frequency, amplitude, phase and offset of the fundamental component using interpolated discrete Fourier transform. Rectangular or Hann window can be used for DFT.

Citation: Krzysztof Duda: Interpolation algorithms of DFT for parameters estimation of sinusoidal and damped sinusoidal signals. In S. M. Salih, editor, Fourier Transform - Signal Processing, chapter 1, pages 3-32, InTech, 2012. `http://www.intechopen.com/books/fourier-transform-signal-processing/interpolated-dft` Implemented by Rado Lapuh, 2016.

Remarks: If sampling time |Ts| is not supplied, wrapper will calculate |Ts| from sampling frequency |fs| or if not supplied, first two elements of time series |t| are used to calculate |Ts|. The optional parameter |window| can be set to values ‘rectangular‘ or ‘Hann‘. If parameter is not supplied, Hann window will be used.

License: Implementation: MIT License

Provides GUF: no

Provides MCM: no

Input Quantities

Required: Ts or fs or t, y

Optional: window

Parameters: window

Descriptions:

Ts – Sampling time
fs – Sampling frequency
t – Time series
window – DFT window: ‘Hann‘ or ‘rectangular‘
y – Sampled values

Output Quantities:

A – Amplitude of main signal component

O – Offset of main signal component

f – Frequency of main signal component

ph – Phase of main signal component

Example

3-point interpolated DFT frequency estimator

Example for algorithm iDFT3p.

iDFT3p is an algorithm for estimating the frequency, amplitude, phase and offset of the fundamental component using interpolated discrete Fourier transform. Rectangular or Hann window can be used for DFT.’; See also Krzysztof Duda: Interpolation algorithms of DFT for parameters estimation of sinusoidal and damped sinusoidal signals. In S. M. Salih, editor, Fourier Transform - Signal Processing, chapter 1, pages 3-32, InTech, 2012. `http://www.intechopen.com/books/fourier-transform-signal-processing/interpolated-dft` Implemented by Rado Lapuh, 2016.’;

Contents

– Generate sample data
– Call algorithm
– Display results

Generate sample data

Two quantities are prepared: Ts and y, representing 0.5 second of sinus waveform of nominal frequency 100 Hz, nominal amplitude 1 V and nominal phase 1 rad, sampled with sampling time 0.1 ms, with offset 0.1 V. The sampling is not coherent.

```
DI = [];
Anom = 1; fnom = 100; phnom = 1; Onom = 0.1;
DI.Ts.v = 1e-4;
t = [0:DI.Ts.v:0.5];
DI.y.v = Anom*sin(2*pi*fnom*t + phnom) + Onom;
```

Call algorithm

First a rectangular window will be selected to estimate main signal properties. Use QWTB to apply algorithm iDFT3p to data DI and put results into DOr.

```
DI.window.v = 'rectangular';
DOr = qwtb('iDFT3p', DI);
```

```
QWTB: no uncertainty calculation
```

Next a Hann window will be selected to estimate main signal properties Results will be put into DOh.

```
DI.window.v = 'Hann';
DOh = qwtb('iDFT3p', DI);
```

```
QWTB: no uncertainty calculation
```

Display results

Results is the amplitude, frequency and phase of sampled waveform. For rectangular window, the error from nominal in parts per milion is:

```
f_re = (DOr.f.v - fnom)./fnom .* 1e6
A_re = (DOr.A.v - Anom)./Anom .* 1e6
ph_re = (DOr.ph.v - phnom)./phnom .* 1e6
O_re = (DOr.O.v - Onom)./Onom .* 1e6
```

```
f_re =

    0.0166


A_re =

   41.2567


ph_re =

   88.3681


O_re =

   1.6826e+03
```

For Hann window:

```
f_he = (DOh.f.v - fnom)./fnom .* 1e6
A_he = (DOh.A.v - Anom)./Anom .* 1e6
ph_he = (DOh.ph.v - phnom)./phnom .* 1e6
O_he = (DOh.O.v - Onom)./Onom .* 1e6
```

```
f_he =

  -3.7790e-06
```

```
A_he =

     3.9679e-07


ph_he =

       6.2737


O_he =

     -0.6862
```

INL-DNL – Integral and Differential Non-Linearity of ADC

Description

Id: INL-DNL

Name: Integral and Differential Non-Linearity of ADC

Description: Calculates Integral and Differential Non-Linearity of an ADC. The histogram of measured data is used to calculate INL and DNL estimators. ADC has to sample a pure sine wave. To estimate all transition levels the amplitude of the sine wave should overdrive the full range of the ADC by at least 120

Citation: Estimators are based on Tamás Virosztek, MATLAB-based ADC testing with sinusoidal excitation signal (in Hungar- ian), B.Sc. Thesis, 2011. Implementation: Virosztek, T., Pálfi V., Renczes B., Kollár I., Balogh L., Sárhegyi A., Márkus J., Bilau Z. T., ADCTest project site: `http://www.mit.bme.hu/projects/adctest` 2000-2014

Remarks: Based on the ADCTest Toolbox v4.3, November 25, 2014.

License: UNKNOWN

Provides GUF: no

Provides MCM: no

Input Quantities

Required: bitres, codes

Descriptions:

bitres – Bit resolution of an ADC

codes – Sampled values represented as ADC codes (not converted to voltage)

Output Quantities:

DNL – Differential Non-Linearity
INL – Integral Non-Linearity

Example

Integral and Differential Non Linearity of ADC

Example for algorithm INL-DNL

INL-DNL is an algorithm for estimating Integral and Differential Non-Linearity of an ADC. ADC has to sample a pure sine wave. To estimate all transition levels the amplitude of the sine wave should overdrive the full range of the ADC by at least 120%. If not so, non estimated transition levels will be assumed to be 0 and the results may be less accurate. As an input ADC codes are required.';

See also 'Virosztek, T., Pálfi V., Renczes B., Kollár I., Balogh L., Sárhegyi A., Márkus J., Bilau Z. T., ADCTest project site: `http://www.mit.bme.hu/projects/adctest` 2000-2014';

Contents

- Generate sample data
- Call algorithm

Generate sample data

Suppose a sine wave of nominal frequency 10 Hz and nominal amplitude 1.5 V is sampled by ADC with bit resolution of 4 and full range of 1 V. First quantity bitres with number of bits of resolution of the ADC is prepared and put into input data structure DI.

```
DI = [];
DI.bitres.v = 4;
```

Waveform is constructed. Amplitude is selected to overload the ADC.

```
t=[0:1/1e4:1-1/1e4];
Anom = 3.5; fnom = 2; phnom = 0;
wvfrm = Anom*sin(2*pi*fnom*t + phnom);
```

Next ADC code values are calculated. It is simulated by quantization and scaling of the sampled waveform. In real measurement code values can be obtained directly from the ADC. Suppose ADC range is -2..2.

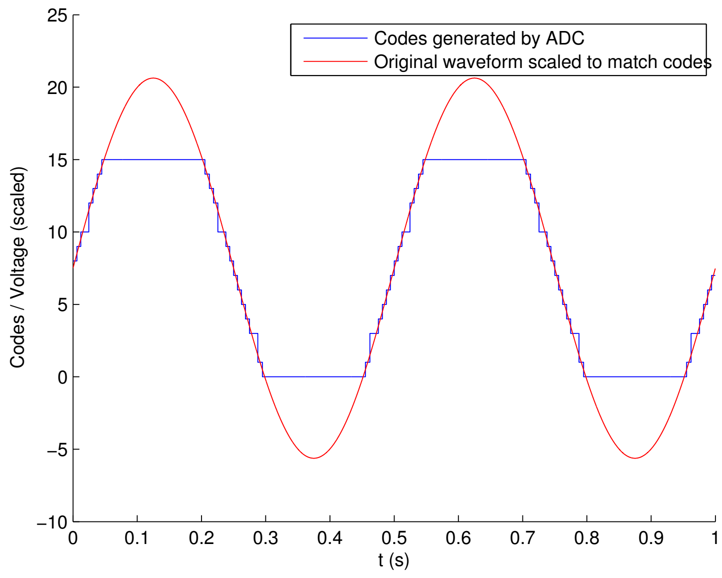
```
codes = wvfrm;
rmin = -2; rmax = 2;
levels = 2.^DI.bitres.v - 1;
codes(codes<rmin) = rmin;
codes(codes>rmax) = rmax;
codes = round((codes-rmin)./(rmax-rmin).*levels);
```

Now lets introduce ADC error. Instead of generating code 2 ADC erroneously generates code 3 and instead of 11 it generates 10.

```
codes(codes==2) = 3;
codes(codes==11) = 10;
codes = codes + min(codes);
```

Create quantity codes and plot a figure with sampled sine wave and codes.

```
DI.codes.v = codes;
figure
hold on
stairs(t, codes);
wvfrm = (wvfrm - rmin)./(rmax-rmin).*levels;
plot(t, wvfrm, '-r');
xlabel('t (s)')
ylabel('Codes / Voltage (scaled)');
legend('Codes generated by ADC','Original waveform
   scaled to match codes');
hold off
```



Call algorithm

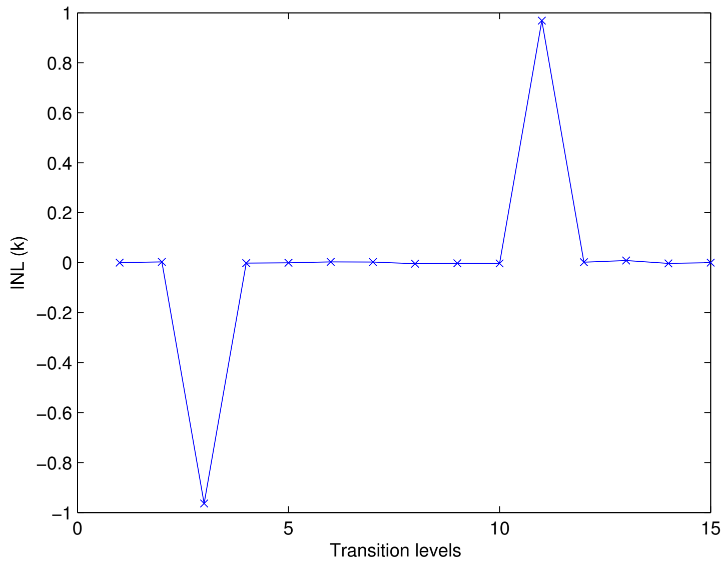
Apply INL algorithm to the input data DI.

```
DO = qwtb('INL-DNL', DI);
```

```
QWTB: no uncertainty calculation
```

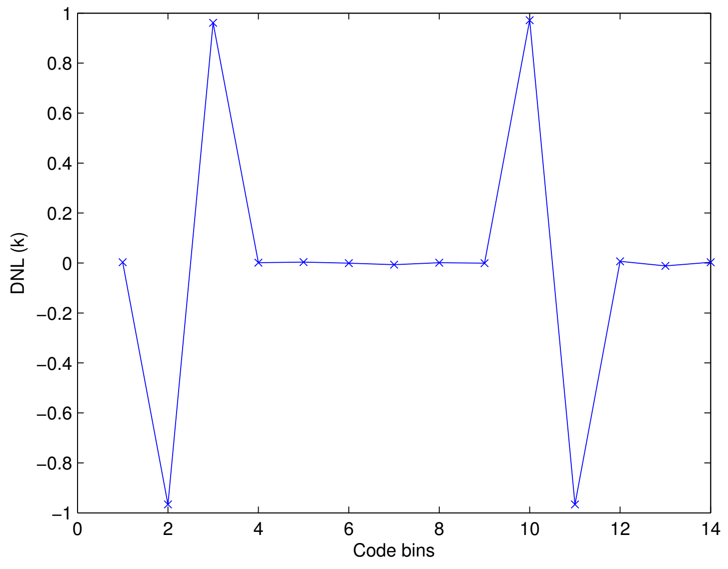
Plot results of integral non-linearity. One can clearly observe defects on codes 3 and 11.

```
figure
plot(DO.INL.v, '-x');
xlabel('Transition levels')
ylabel('INL (k)')
```



Plot results of differential non-linearity. One can clearly observe defects on transitions 2-3 and 10-11.

```
figure
plot(DO.DNL.v, '-x');
xlabel('Code bins')
ylabel('DNL (k)')
```



MADEV – Modified Allan Deviation

Description

Id: MADEV

Name: Modified Allan Deviation

Description: Compute the modified Allan deviation for a set of time-domain frequency data.

Citation: D.W. Allan and J.A. Barnes, "A Modified Allan Variance with Increased Oscillator Characterization Ability", Proc. 35th Annu. Symp. on Freq. Contrl., pp. 470-474, May 1981. Implementation: Implementation by M. A. Hopcroft, mhopeng@gmail.com, Matlab Central, online: `http://www.mathworks.com/matlabcentral/fileexchange/26637-allan-modified` Test data by W. J. Riley, "The Calculation of Time Domain Frequency Stability", online: `http://www.wriley.com/paper1ht.htm`

Remarks: If sampling frequency |fs| is not supplied, wrapper will calculate |fs| from sampling time |Ts| or if not supplied, first two elements of time series |t| are used to calculate |Ts|. If observation time(s) |tau| is not supplied, tau values are automatically generated. Tau values must be divisible by 1/|fs|. Invalid values are ignored. For tau values really used in the calculation see the output.

License: BSD License

Provides GUF: yes

Provides MCM: no

Input Quantities

Required: fs or Ts or t, y

Optional: `tau`

Descriptions:

`Ts`	– Sampling time
`fs`	– Sampling frequency
`t`	– Time series
`tau`	– Observation time
`y`	– Sampled values

Output Quantities:

`madev`	– Modified Allan deviation
`tau`	– Observation time of resulted values

Example

Modified Allan Deviation

Example for algorithm MADEV.

MADEV is an algorithm to compute the modified Allan deviation for a set of time-domain frequency data.

See also W. J. Riley, "The Calculation of Time Domain Frequency Stability". Implementation: M. A. Hopcroft, `mhopeng@gmail.com`, Matlab Central.'

Contents

- Generate sample data
- Call algorithm
- Display results

Generate sample data

A random numbers with normal probability distribution function will be generated into input data `DI.y.v`. Next a drift will be added.

```
DI = [];
DI.y.v = 1.5 + 3.*randn(1, 1e3);
DI.y.v = DI.y.v + [1:1:1e3]./100;
```

Lets suppose a sampling frequency is 1 Hz. The algorithm will generate all possible tau values automatically.

```
DI.fs.v = 1;
```

Call algorithm

Use QWTB to apply algorithm MADEV to data DI.

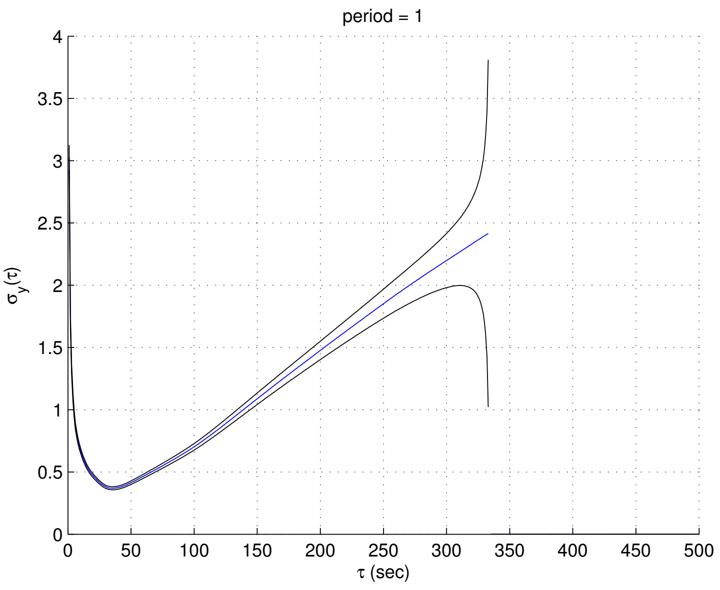
```
DO = qwtb('MADEV', DI);
```

```
QWTB: no uncertainty calculation
```

Display results

Log log figure is the best to see modified allan deviation results:

```
figure; hold on
loglog(DO.tau.v, DO.madev.v, '-b')
loglog(DO.tau.v, DO.madev.v + DO.madev.u, '-k')
loglog(DO.tau.v, DO.madev.v - DO.madev.u, '-k')
xlabel('\tau (sec)');
ylabel('\sigma_y(\tau)');
title(['period = ' num2str(DI.fs.v)]);
grid('on'); hold off
```


period = 1
σ_y(τ)
τ (sec)
0
0.5
1
1.5
2
2.5
3
3.5
4
0
50
100
150
200
250
300
350
400
450
500

OADEV – Overlapping Allan Deviation

Description

Id: OADEV

Name: Overlapping Allan Deviation

Description: Compute the overlapping Allan deviation for a set of time-domain frequency data.

Citation: D.A. Howe, D.W. Allan and J.A. Barnes, "Properties of Signal Sources and Measurement Methods', Proc. 35th Annu. Symp. on Freq. Contrl., pp. 1-47, May 1981. Implementation: Implementation by M. A. Hopcroft, mhopeng@gmail.com, Matlab Central, online: `http://www.mathworks.com/matlabcentral/fileexchange/26441-allan-overlap` Test data by W. J. Riley, "The Calculation of Time Domain Frequency Stability", online: `http://www.wriley.com/paper1ht.htm`

Remarks: If sampling frequency |fs| is not supplied, wrapper will calculate |fs| from sampling time |Ts| or if not supplied, first two elements of time series |t| are used to calculate |Ts|. If observation time(s) |tau| is not supplied, tau values are automatically generated. Tau values must be divisible by 1/|fs|. Invalid values are ignored. For tau values really used in the calculation see the output.

License: BSD License

Provides GUF: yes

Provides MCM: no

Input Quantities

Required: `fs` or `Ts` or `t`, `y`
Optional: `tau`
Descriptions:

`Ts` – Sampling time
`fs` – Sampling frequency
`t` – Time series
`tau` – Observation time
`y` – Sampled values

Output Quantities:

`oadev` – Overlapping Allan deviation
`tau` – Observation time of resulted values

Example

Allan Overlapping Deviation

Example for algorithm OADEV.

OADEV is an algorithm to compute the overlapping Allan deviation for a set of time-domain frequency data.

See also W. J. Riley, "The Calculation of Time Domain Frequency Stability". Implementation: M. A. Hopcroft, `mhopeng@gmail.com`, Matlab Central.’

Contents

- Generate sample data
- Call algorithm
- Display results

Generate sample data

A random numbers with normal probability distribution function will be generated into input data `DI.y.v`. Next a drift will be added.

```
DI = [];
DI.y.v = 1.5 + 3.*randn(1, 1e3);
DI.y.v = DI.y.v + [1:1:1e3]./100;
```

Lets suppose a sampling frequency is 1 Hz. The algorithm will generate all possible tau values automatically.

```
DI.fs.v = 1;
```

Call algorithm

Use QWTB to apply algorithm OADEV to data DI.

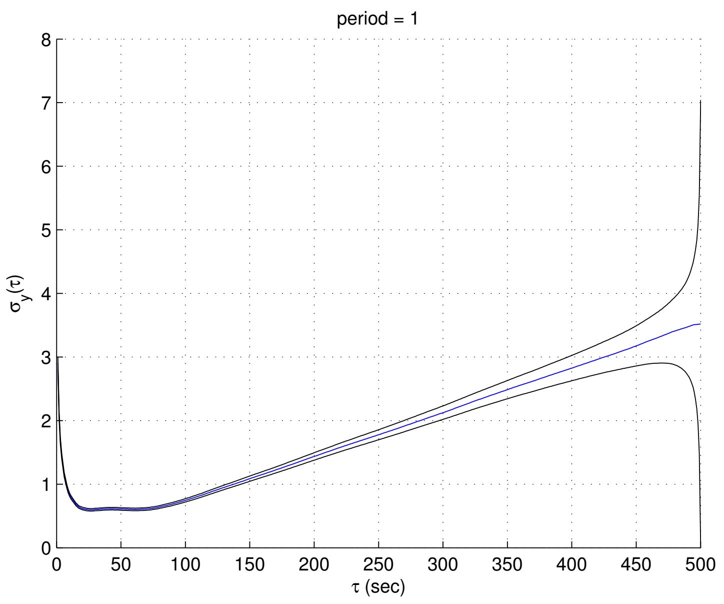
```
DO = qwtb('OADEV', DI);
```

```
QWTB: no uncertainty calculation
```

Display results

Log log figure is the best to see allan deviation results:

```
figure; hold on
loglog(DO.tau.v, DO.oadev.v, '-b')
loglog(DO.tau.v, DO.oadev.v + DO.oadev.u, '-k')
loglog(DO.tau.v, DO.oadev.v - DO.oadev.u, '-k')
xlabel('\tau (sec)');
ylabel('\sigma_y(\tau)');
title(['period = ' num2str(DI.fs.v)]);
grid('on'); hold off
```


period = 1
$\sigma_y(\tau)$
τ (sec)
0
1
2
3
4
5
6
7
8
0
50
100
150
200
250
300
350
400
450
500

PSFE – Phase Sensitive Frequency Estimator

Description

Id: PSFE

Name: Phase Sensitive Frequency Estimator

Description: An algorithm for estimating the frequency, amplitude, and phase of the fundamental component in harmonically distorted waveforms. The algorithm minimizes the phase difference between the sine model and the sampled waveform by effectively minimizing the influence of the harmonic components. It uses a three-parameter sine-fitting algorithm for all phase calculations. The resulting estimates show up to two orders of magnitude smaller sensitivity to harmonic distortions than the results of the four-parameter sine fitting algorithm.

Citation: Lapuh, R., "Estimating the Fundamental Component of Harmonically Distorted Signals From Noncoherently Sampled Data," Instrumentation and Measurement, IEEE Transactions on , vol.64, no.6, pp.1419,1424, June 2015, doi: 10.1109/TIM.2015.2401211, URL: `http://ieeexplore.ieee.org/stamp/stamp.jsp?tp=&arnumber=7061456&isnumber=7104190`

Remarks: If sampling time |Ts| is not supplied, wrapper will calculate |Ts| from sampling frequency |fs| or if not supplied, first two elements of time series |t| are used to calculate |Ts|.

License: MIT License

Provides GUF: no

Provides MCM: no

Input Quantities

Required: Ts or fs or t, y

Descriptions:

Ts – Sampling time
fs – Sampling frequency
t – Time series
y – Sampled values

Output Quantities:

A – Amplitude of main signal component

f – Frequency of main signal component

ph – Phase of main signal component

Example

Phase Sensitive Frequency Estimator

Example for algorithm PSFE.

PSFE is an algorithm for estimating the frequency, amplitude, and phase of the fundamental component in harmonically distorted waveforms. The algorithm minimizes the phase difference between the sine model and the sampled waveform by effectively minimizing the influence of the harmonic components. It uses a three-parameter sine-fitting algorithm for all phase calculations. The resulting estimates show up to two orders of magnitude smaller sensitivity to harmonic distortions than the results of the four-parameter sine fitting algorithm.

See also Lapuh, R., "Estimating the Fundamental Component of Harmonically Distorted Signals From Noncoherently Sampled Data," Instrumentation and Measurement, IEEE Transactions on , vol.64, no.6, pp.1419,1424, June 2015, doi: 10.1109/TIM.2015.2401211, URL: `http://ieeexplore.ieee.org/stamp/stamp.jsp?tp=&arnumber=7061456&isnumber=7104190'`

Contents

- Display results

Generate sample data

Two quantities are prepared: Ts and y, representing 1 second of sinus waveform of nominal frequency 100 Hz, nominal amplitude 1 V and nominal phase 1 rad, sampled with sampling time 0.1 ms.

```
DI = [];
Anom = 1; fnom = 100; phnom = 1;
DI.Ts.v = 1e-4;
t = [0:DI.Ts.v:1-DI.Ts.v];
DI.y.v = Anom*sin(2*pi*fnom*t + phnom);
```

Add noise:

```
DI.y.v = DI.y.v + 1e-3.*randn(size(DI.y.v));
```

Call algorithm

Use QWTB to apply algorithm PSFE to data DI.

```
DO = qwtb('PSFE', DI);
```

```
QWTB: no uncertainty calculation
```

Display results

Results is the amplitude, frequency and phase of sampled waveform.

```
f = DO.f.v
A = DO.A.v
ph = DO.ph.v
```

```
f =

   100.0000


A =

      1.0000


ph =

      1.0000
```

Errors of estimation in parts per milion:

```
ferrppm = (DO.f.v - fnom)/fnom .* 1e6
Aerrppm = (DO.A.v - Anom)/Anom .* 1e6
pherrppm = (DO.ph.v - phnom)/phnom .* 1e6
```

```
ferrppm =

      0.1228


Aerrppm =

      9.5634


pherrppm =

   -45.3739
```

SFDR – Spurious Free Dynamic Range

Description

Id: SFDR

Name: Spurious Free Dynamic Range

Description: Calculates Spurious Free Dynamic Range of an ADC. A FFT method with Blackman windowing is used to calculate a spectrum and SFDR is estimated in decibels relative to carrier amplitude. ADC has to sample a pure sine wave. SFDR is calculated according IEEE Std 1057-2007

Citation: Implementation: Virosztek, T., Pálfi V., Renczes B., Kollár I., Balogh L., Sárhegyi A., Márkus J., Bilau Z. T., ADCTest project site: `http://www.mit.bme.hu/projects/adctest` 2000-2014

Remarks: Based on the ADCTest Toolbox v4.3, November 25, 2014.

License: UNKNOWN

Provides GUF: no

Provides MCM: no

Input Quantities

Required: y

Descriptions:

y – Sampled values

Output Quantities:

SFDRdBc – Spurious Free Dynamic Range in decibels relative to carrier (dBc)

Example

Spurious Free Dynamic Range by means of Fast fourier transform

Example for algorithm SFDR.

Calculates SFDR by calculating FFT spectrum.

Contents

- Generate sample data
- Call algorithm
- Display results

Generate sample data

First quantity y representing 1 second of signal containing spurious component is prepared. Main signal component has nominal frequency 1 kHz, nominal amplitude 2 V, nominal phase 1 rad and offset 1 V sampled at sampling frequency 10 kHz.

```
DI = [];
fsnom = 1e4; Anom = 4; fnom = 100; phnom = 1; Onom =
    0.2;
t = [0:1/fsnom:1-1/fsnom];
DI.y.v = Anom*sin(2*pi*fnom*t + phnom);
```

A spurious component with amplitude at 1/100 of main carrier frequency is added. Thus by definition the SFDR in dBc has to be 40.

```
DI.y.v = DI.y.v + Anom./100*sin(2*pi*fnom*3.5*t + phnom
    );
```

Call algorithm

Use QWTB to apply algorithm SFDR to data DI.

```
DO = qwtb('SFDR', DI);
```

```
QWTB: no uncertainty calculation
```

Display results

Result is the SFDR (dBc).

```
SFDR = DO.SFDRdBc.v
```

```
SFDR =

    40.0000
```

SINAD-ENOB – Ratio of signal to noise and distortion and Effective number of bits (in time space)

Description

Id: SINAD-ENOB

Name: Ratio of signal to noise and distortion and Effective number of bits (in time space)

Description: Algorithm calculates Ratio of signal to noise and distortion and Effective number of bits in time space, therefore it is suitable for noncoherent measurements. Requires estimates of the main signal component parameters: frequency, amplitude, phase and offset. If these values are estimated by four parameter sine wave fit, the SINAD and ENOB will be calculated according IEEE Std 1241-2000. A large sine wave should be applied to the ADC input. Almost any error source in the sine wave input other than gain accuracy and dc offset can affect the test result.

Citation: IEEE Std 1241-2000, pages 52 - 54

Remarks: If Time series |t| is not supplied, wrapper will calculate |t| from sampling frequency |fs| or if not supplied, sampling time |Ts| is used to calculate |t|.

License: MIT License

Provides GUF: no

Provides MCM: no

Input Quantities

Required: t or fs or Ts, y, f, A, ph, O, bitres, FSR

Descriptions:

A – Amplitude of main signal component
FSR – Full scale range of an ADC
O – Offset of signal
Ts – Sampling time
bitres – Bit resolution of an ADC
f – Frequency of main signal component
fs – Sampling frequency
ph – Phase of main signal component
t – Time series
y – Sampled values

Output Quantities:

ENOB – Effective number of bits

SINADdB – Ratio of signal to noise and distortion in decibels relative to the amplitude of the main signal component

---

Example

Ratio of signal to noise and distortion and Effective number of bits (time space)

Example for algorithm SINAD-ENOB.

Algorithm calculates Ratio of signal to noise and distortion and Effective number of bits in time space. First signal is generated, then estimates of signal parameters are calculated by Four parameter sine wave fitting and next SINAD and ENOB are calculated. This example do not take into account a quantisation noise.

Contents

Generate sample data

Two quantities are prepared: t and y, representing 1 second of sinus waveform of nominal frequency 1 kHz, nominal amplitude 1 V, nominal phase 1 rad and offset 1 V sampled at sampling frequency 10 kHz.

```
DI = [];
Anom = 2; fnom = 100; phnom = 1; Onom = 0.2;
DI.t.v = [0:1/1e4:1-1/1e4];
DI.y.v = Anom*sin(2*pi*fnom*DI.t.v + phnom) + Onom;
```

Add a noise with normal distribution probability:

```
noisestd = 1e-4;
DI.y.v = DI.y.v + noisestd.*randn(size(DI.y.v));
```

Lets make an estimate of frequency 0.2 percent higher than nominal value:

```
DI.fest.v = 100.2;
```

Calculate estimates of signal parameters

Use QWTB to apply algorithm 4PSWF to data DI.

```
CS.verbose = 1;
DO = qwtb('4PSWF', DI, CS);
```

```
QWTB: no uncertainty calculation
Fitting started

Local minimum found.

Optimization completed because the size of the
   gradient is less than
the default value of the function tolerance.
```

```
Fitting finished
```

Copy results to inputs

Take results of FPSWF and put them as inputs DI.

```
DI.f = DO.f;
DI.A = DO.A;
DI.ph = DO.ph;
DI.O = DO.O;
```

Suppose the signal was sampled by a 20 bit digitizer with full scale range FSR of 6 V (+- 3V). (The signal is not quantised, so the quantization noise is not present. Thus the simulation and results are not fully correct.):

```
DI.bitres.v = 20;
DI.FSR.v = 3;
```

Calculate SINAD and ENOB

```
DO = qwtb('SINAD-ENOB', DI, CS);
```

```
QWTB: no uncertainty calculation
```

Display results:

Results are:

```
SINADdB = DO.SINADdB.v
ENOB = DO.ENOB.v
```

```
SINADdB =

    83.0767


ENOB =

    13.0912
```

Theoretical value of SINADdB is 20*log10(Anom./(noisestd.*sqrt(2))). Theoretical value of ENOB is log2(DI.range.v./(noisestd.*sqrt(12))). Absolute error of results are:

```
SINADdBtheor = 20*log10(Anom./(noisestd.*sqrt(2)));
ENOBtheor = log2(DI.FSR.v./(noisestd.*sqrt(12)));
SINADerror = SINADdB - SINADdBtheor
ENOBerror = ENOB - ENOBtheor
```

```
SINADerror =

     0.0664


ENOBerror =

     0.0110
```

SP-FFT – Spectrum by means of Fast Fourier Transform

Description

Id: SP-FFT

Name: Spectrum by means of Fast Fourier Transform

Description: Calculates frequency and phase spectrum by means of Fast Fourier Transform algorithm. Result is normalized.

Citation:

Remarks: If sampling frequency |fs| is not supplied, wrapper will calculate |fs| from sampling time |Ts| or if not supplied, first two elements of time series |t| are used to calculate |fs|.

License: MIT License

Provides GUF: no

Provides MCM: no

Input Quantities

Required: `fs` or `Ts` or `t`, `y`

Descriptions:

`Ts` – Sampling time
`fs` – Sampling frequency
`t` – Time series
`y` – Sampled values

Output Quantities:

A – Amplitude spectrum
f – Frequency series
ph – Phase spectrum

Example

Signal Spectrum by means of Fast fourier transform

Example for algorithm SP-FFT.

Calculates frequency and phase spectrum by means of Fast Fourier Transform algorithm. Result is normalized.

Contents

Generate sample data

Two quantities are prepared: y and fs, representing 1 second of signal containing 5 harmonic components and one inter-harmonic component. Main signal component has nominal frequency 1 kHz, nominal amplitude 2 V, nominal phase 1 rad and offset 1 V sampled at sampling frequency 10 kHz.

```
DI = [];
fsnom = 1e4; Anom = 2; fnom = 100; phnom = 1; Onom =
    0.2;
t = [0:1/fsnom:1-1/fsnom];
DI.y.v = Anom*sin(2*pi*fnom*t + phnom);
for i = 2:45
        DI.y.v = DI.y.v + Anom./i*sin(2*pi*fnom*i*t +
    phnom + i - 1);
end
DI.y.v = DI.y.v + 1*sin(2*pi*fnom*1.456*t + phnom);
DI.fs.v = fsnom;
```

Call algorithm

Use QWTB to apply algorithm SP−FFT to data DI.

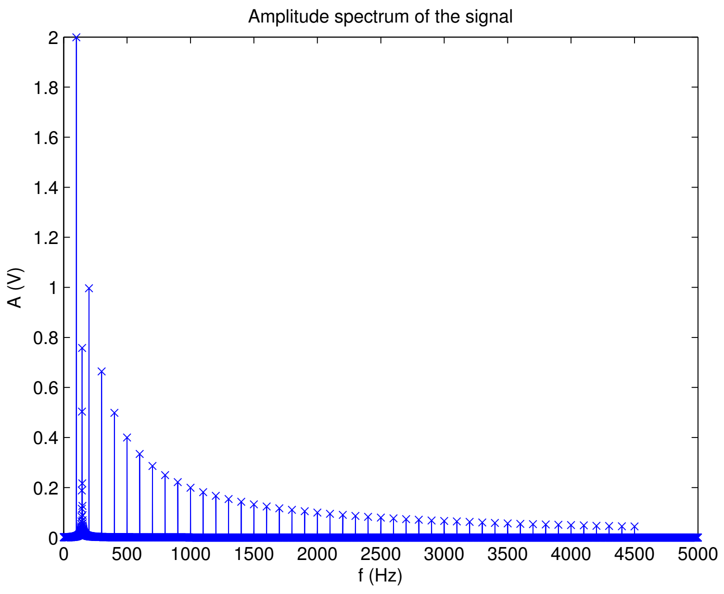
```
DO = qwtb('SP−FFT', DI);
```

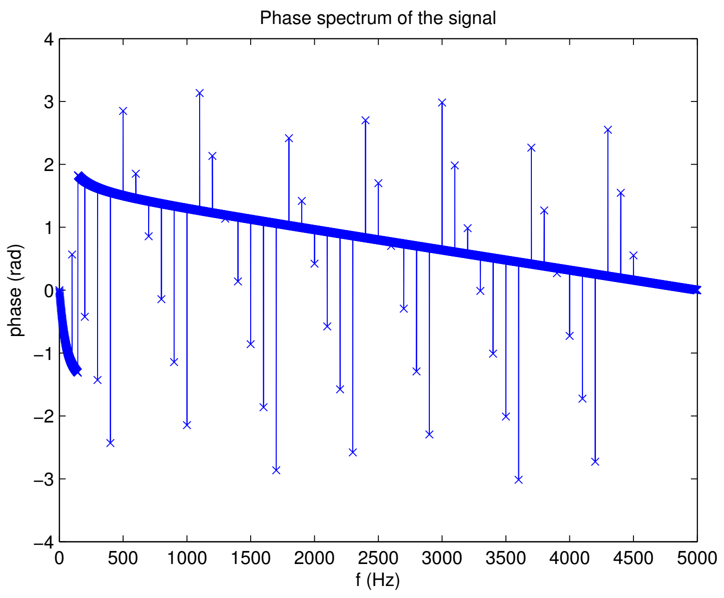
```
QWTB: no uncertainty calculation
```

Display results

Results is the amplitude and phase spectrum.

```
figure
plot(DO.f.v, DO.A.v, '−x')
xlabel('f (Hz)'); ylabel('A (V)'); title('Amplitude
   spectrum of the signal');
figure
plot(DO.f.v, DO.ph.v, '−x')
xlabel('f (Hz)'); ylabel('phase (rad)'); title('Phase
   spectrum of the signal');
```




Phase spectrum of the signal
phase (rad)
f (Hz)
4
3
2
1
0
−1
−2
−3
−4
0
500
1000
1500
2000
2500
3000
3500
4000
4500
5000