

# *Smart Sound/Vibration Analyzer*

*(Real-Time Based Measurement System)*

## Manual

First Edition(Ver. 1.7)



SV Corp.

This document is on revising. We will upload the revised document to FTP Server(Refer next page) at every revising. Of course, you can download a upgraded program(Smart Sound/Vibration Analyzer.)

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## I. Composition of Smart Sound/Vibration Analyzer

- AC Adapter: 1set
- USB Interface Cable: 1 ea
- Smart Sound/Vibration Analyzer: 1 ea
- S/W CD(RealWave S/W, Firmware Update S/W, RealWave Desk S/W) : 1ea
- Manual: 1 ea
- Portable Aluminum Hardcase : 1 ea



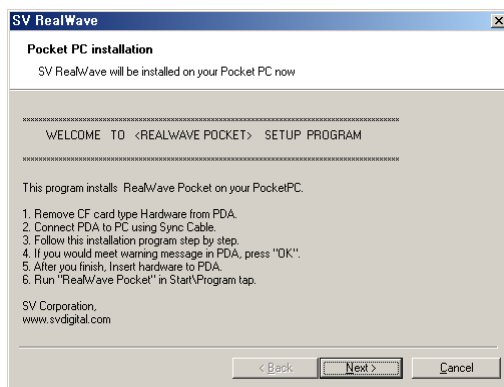
[Fig. I-1] System Diagram

## II. Installation & Operation Procedure

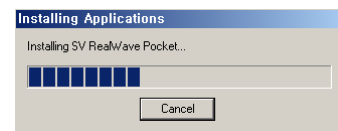
### 1. S/W Installation

Before installing of the RealWave program on PDA, you must install ActiveSync on your PC. Please refer to the chapter VIII to install ActiveSync. After installation of ActiveSync, connect PC with PDA via USB cable and Insert the CD. Run the RealWave.exe file from the RealWave folder of the CD-Rom drive. When the window appears like [Fig. II-1-1] and click 'Next', it can be proceed to [Fig. II-1-2] and [Fig. II-1-3]. And then click 'OK' to finish installation [Fig. II-1-3].

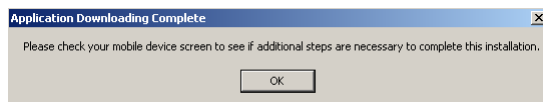
Realwave.ARM.CAB files use when installing a program from PDA directly.



[Fig. II-1-1] Installation window 1



[Fig. II-1-2] Installation window 2



[Fig. II-1-3] Installation window 3



[Fig. II-1-4] Run of RealWave program 1



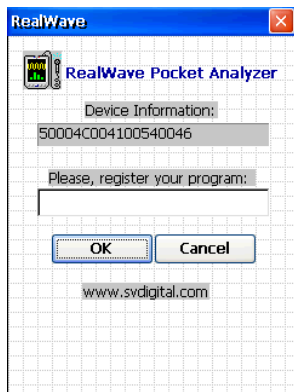
[Fig. II-1-5] Execute RealWave program 2

To run Launch RealWave program, click the tap [Start >> Programs], and then click 'RealWave' icon on the window of [Fig. II-1-5].

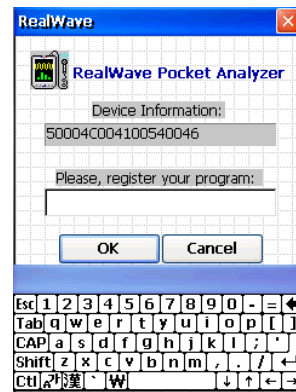
## 2. Run and Registration

### 2.1 Unregistered PDA

If running RealWave program at unregistered PDA, please enter given license number on the register window when appears a window like [Fig. II-2-1,2]. The license number is limited at only one PDA. Once registered, RealWave program doesn't work properly if the license number is different. Please contact us when you have a problem with the license number.



[Fig. II-2-1] Registration window



[Fig. II-2-2] Entering license number

### 2.2 Pre-registered PDA

If you have PDA included package, program will start without register procedure like [Fig. II-2-1]. In case of no H/W or failure to recognize, icons in the red box are not activated like [Fig. II-2-3]. But in case of the opposite, icons are activated and ready to measure like [Fig. II-2-4].



[Fig. II-2-3] Failure to recognize H/W



[Fig. II-2-4] Properly recognized H/W

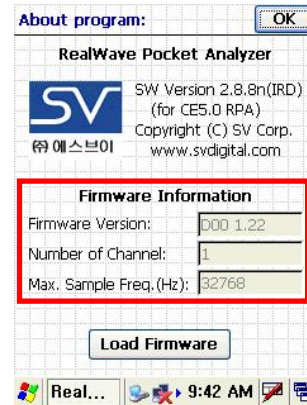
### 3. Firmware Upgrade

#### 3.1 Check firmware version

If you select [About RealWave] menu in [File], dialog window appears in [Fig. II-3-2]. You can check the information of hardware including firmware version.



[Fig. II-3-1] Select [About RealWave...] menu



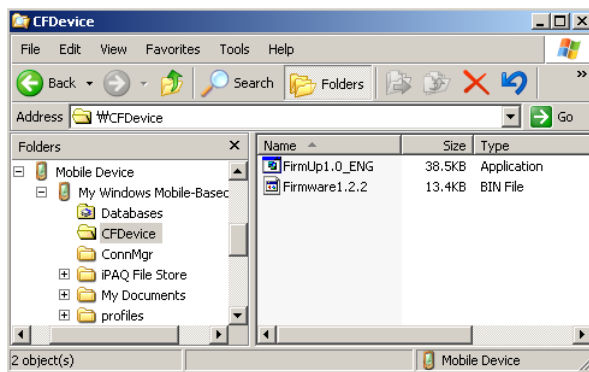
[Fig. II-3-2] Check firmware version

#### 3.2 Firmware Upgrade

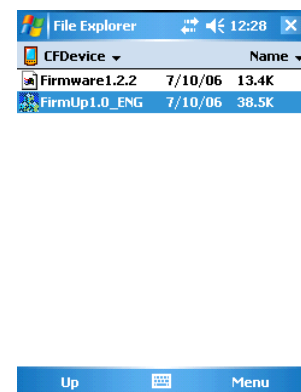
Smart Sound/Vibration Analyzer H/W has firmware inside, so the firmware has to be upgraded when the new version is released. First, 'FirmUp1.0\_ENG.exe' file in the RealWave software CD has to be copied at a proper folder in PDA. Here is the way to copy it:

- 1) Copy the 'FirmUp1.0\_ENG.exe' in the RealWave software CD.
- 2) Connect PDA with PC using a USB cable.
- 3) After ActiveSync connected, double click the 'Mobile Devices' folder on the windows explorer.  
Several folder in PDA will be found.
- 4) Paste 'FirmUp1.0\_ENG.exe' to the proper PDA folder.

[Fig. II-3-3] ~[Fig. II-3-4] shows a copy of 'FirmUp1.0\_ENG.exe' file.



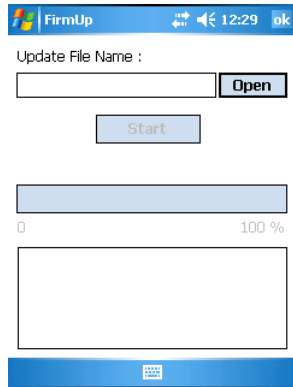
[Fig. II-3-3] Copy and Paste in the Window Explore



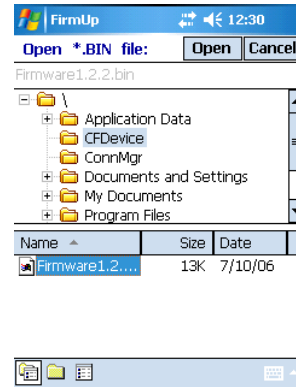
[Fig. II-3-4] Run 'FirmUp.exe' file on PDA



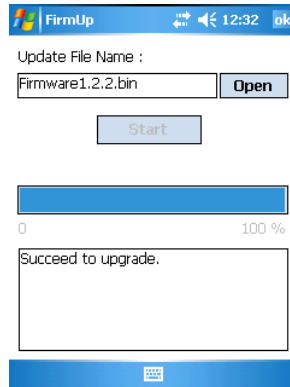
When running 'FirmUp1.0\_ENG.exe' file on PDA, the program runs like [Fig. II-3-5]. Press [Open] button on the window of [Fig. II-3-6]. Select folder and file (\*.bin) to update and press [Open] button on the window of [Fig. II-3-7].



[Fig. II-3-5] FirmUp window




[Fig. II-3-6] Open a file to update



[Fig. II-3-7] Completion to open a file to update

At [Fig. II-3-7], press [Start] button to start update. You must insert front-end H/W before update. It is highly recommended to use AC adaptor or USB cable, but not to use PDA while updating. It is very harmful that H/W is removed from CF card slot or shut down compulsorily while updating. If it causes, you can't to update it and the hardware has to be repaired.

You must turn off PDA after finishing upgrade(Or remove hardware from PDA and then insert hardware again).

 **Attention before updating firmware!**

1. Use of AC adaptor or USB cable is highly recommended.
  - ※ It is very harmful to use PDA while updating.
  - ※ If the firmware upgrade is failed, the hardware can't be used anymore and must be repaired.




**Attention after updating firmware!**


1. Turn off the power of PDA, and then Turn on.


### III. Basic Functions of Program


#### 1. Icons


##### 1. 1 Vibration Level Meter

 Icon for vibration level meter mode [Fig. III-1-1].  and  icons are activated.



 Icon to set measurement option of vibration level meter mode[Fig. III-1-2]

 Icon to control the result display[Fig. III-1-3]

 Icon to hear sounds in real-time

 Icon to display data only (without saving) in real-time and be called 'ARM button'

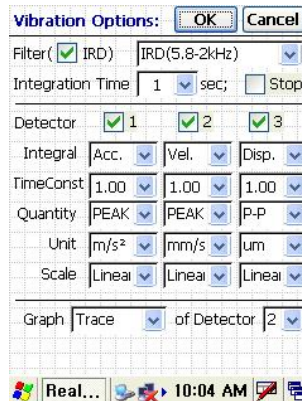
 Icon to measure and record wave file in real-time.

To accomplish sound hearing function, press  and  icons so that you can hear sounds in real-time



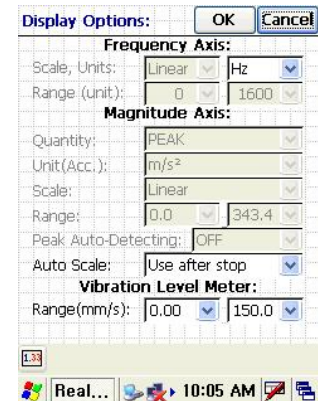
[Fig. III-1-1]

Vibration Level Meter



[Fig. III-1-2]

Vibration Options




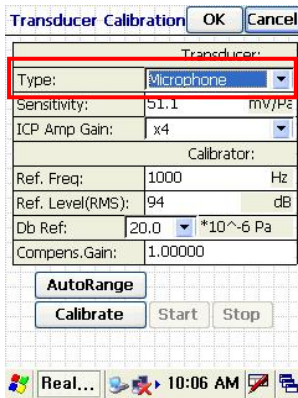
[Fig. III-1-3]

Display Options

##### 1. 2 Sound Level Meter

Upon clicking [File>> Transducer Calibration...] tap, [Fig. III-1-4] will appear, and if you choose Microphone in Type option then you can use sound level meter.

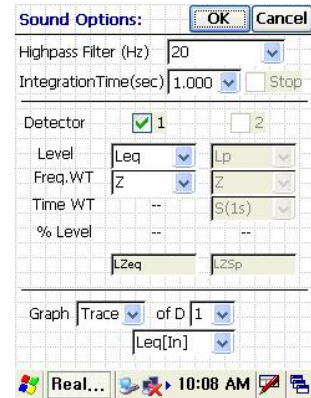
Function of buttons are similar to Vibration Level Meter, icon  will appears measurement option of sound level meter ([Fig. III-1-6]).



[Fig. III-1-4] Setting the Sound Level Meter Mode









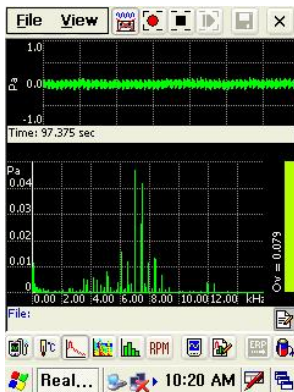
[Fig. III-1-5] Sound Level Meter



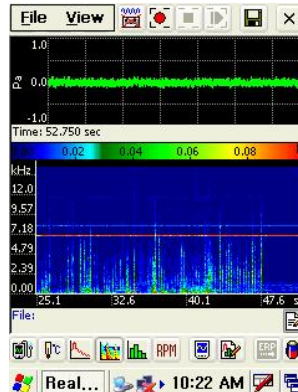
[Fig. III-1-6] Sound Options

### 1. 3 FFT Analyzer

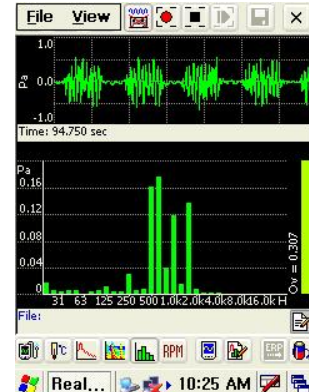
 Icon for FFT Analyzer. If pressing the icon, graph control icon , option setting icon  and sound hearing mode icon  are activated. Likewise, you can also hear the sounds of accelerometer and microphone with  and  icons.



[Fig. III-1-7] FFT Analyzer




[Fig. III-1-8] Spectrogram




[Fig. III-1-9] Octave Analyzer

### 1. 4 Spectrogram

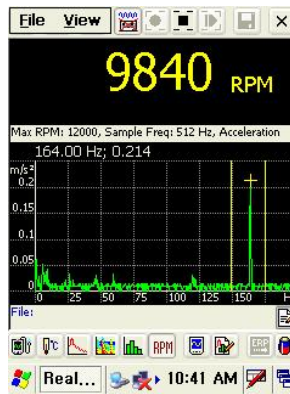
 Icon for spectrogram. The activated icons are identical with FFT Analyzer.

### 1. 5 Octave Analyzer

 Icon for FFT Based Octave Analysis. The activated icons are identical with FFT Analyzer.


### 1. 6 FFT Based RPM Meter

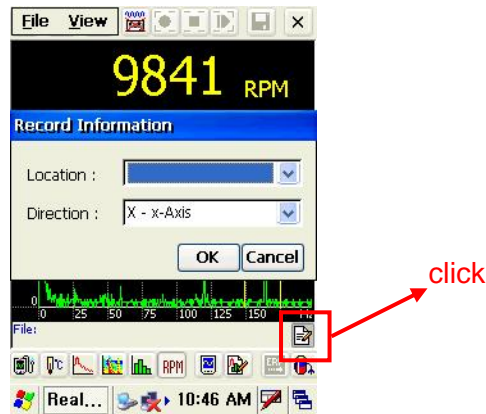
 Icon to measure RPM. Graph control icon  and measurement option control icon  are activated only. And Arm button  is only activated because it doesn't support data store function.



[Fig. III-1-10] RPM Meter

### 1. 7 Entering record information

 Icon to enter record information. Record information window appears like [Fig. III-1-13].




[Fig. III-1-11] Record information window

### 1. 8 Measurement and Playback

 : ARM mode – Icon to display data only in real-time without data store.

 : Record mode – Icon to store wave file and data in real-time.

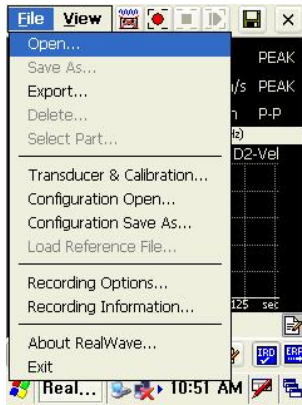
 : Stop mode– Icon to stop arm, record and playback.

 : Playback mode – Icon to analyze data playing wave file. The measured data with this function is the same as the one of record mode.

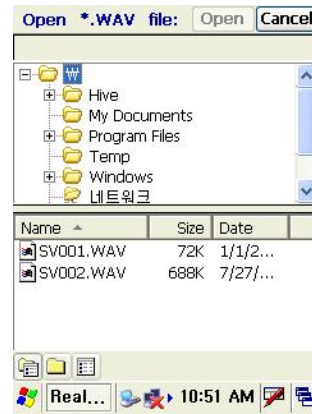
## 2. Control of Wave File

### 2. 1 Open a file

When clicking [File] menu on lower left side of the screen, wave file can be controlled with [Open], [Save As...] and [Delete] tap at [Fig. III-2-1]. To open file, click [Open] tap and choose a folder and a file on the window as you wish at [Fig. III-2-2].




[Fig. III-2-1] Open a wave file



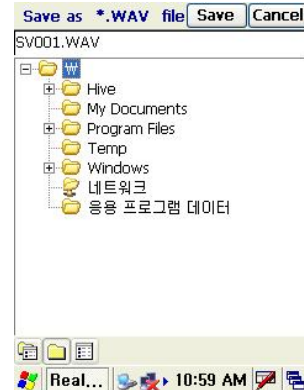
[Fig. III-2-2] Opening window of wave file

## 2. 2 Save As...

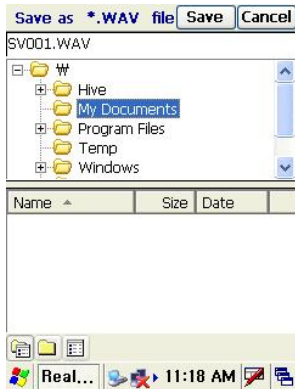
After opening a file, file name can be checked by clicking  icon to convert window like [Fig. III-2-3]. Upon clicking [File>>Save As...] tap, one of the window from [Fig. III-2-4] to [Fig. III-2-6] will appear, so you can define a folder and file name at the window. [Fig. III-2-7] shows how to enter a file name.



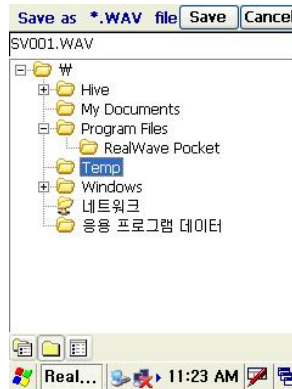
[Fig. III-2-3] Confirmation of a file name



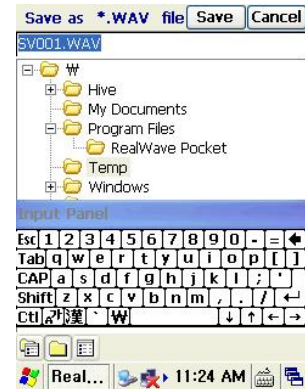
[Fig. III-2-4] Selection of a folder to save as another name



[Fig. III-2-5] Selection of a folder and file to save as another name



[Fig. III-2-6] Selection of a file to save as another name



[Fig. III-2-7] Entering a file name

## 2. 3 Deleting currently opened file

[File>>Delete] is showed [Fig. III-2-8] and you can erase .wav file which are selected currently or erase all files which include export files. It will not be activated if there's no opened file. (When selecting [File>>Delete] tap, currently opened files are deleted.)

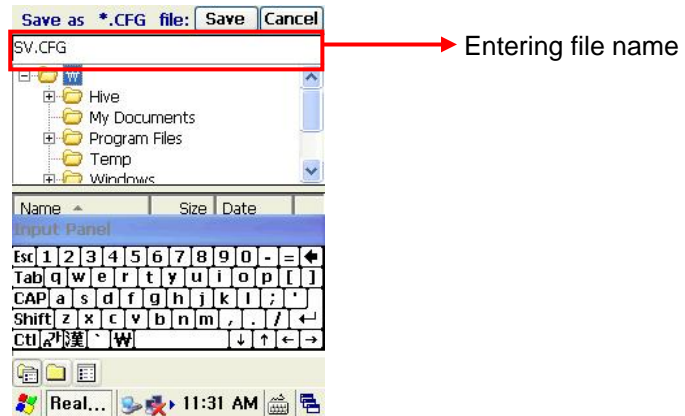


[Fig. III-2-8] Deleting currently opened file

## 3. Open & save configuration.

[File>>Configuration Save As...] is to save all of currently configured options (measurement and graphic display option) of the program. [File>>Configuration Open...] is to open and change to the saved configuration. Extension of the saved file is \*.CFG.

The functions of 'Open' and 'Save As...' is similar to that of chapter 2 "Control of Wave File".

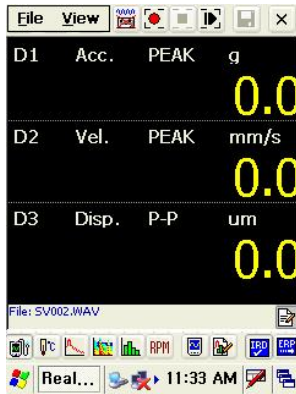


[Fig. III-3-1] Entering a configuration file name.

## 4. Data Export

### 4. 1 Vibration Level Meter Mode

Each different data is exported according to the display option. When displaying only vibration value like [Fig. III-4-1], it doesn't support to export data. On the other hand, when displaying vibration value and trace like [Fig. III-4-2], it supports to save the changed values and export as an image file.



[Fig. III-4-1] Vibration value display



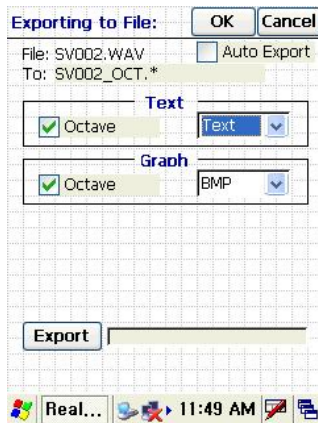
[Fig. III-4-2] Vibration value and Trace display

To export data, click [File>>Export] tap, and then the window [Fig. III-4-3] is opened. Check the data or image file which wants to export, select file type and click 'Export' button. At last, just close window by clicking [Ok] button.

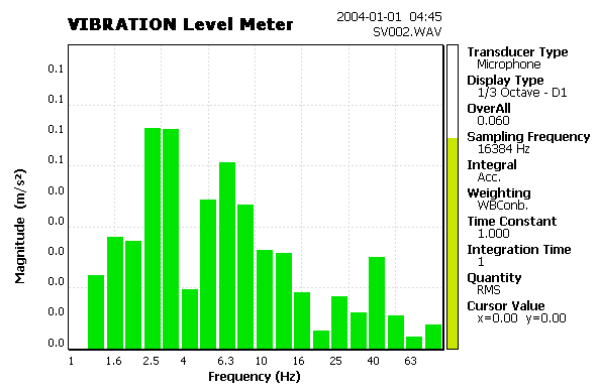
- The supported data file type : Text, Excel, MATLAB
- The supported image file type : BMP, JPEG

[Fig. III-4-4] shows exported file and filename is similar to wave file's name. For example, exported octave image file is named as [wave file name\_OCT.bmp].

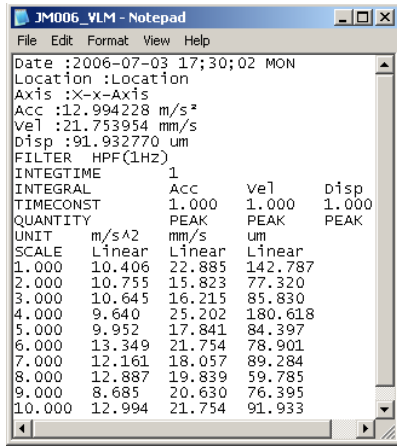
If you should select [Automatic Export], dialog box shown in [Fig. III-4-6] would appear after every measurement. Pressing [Export] button have data be exported automatically as option.



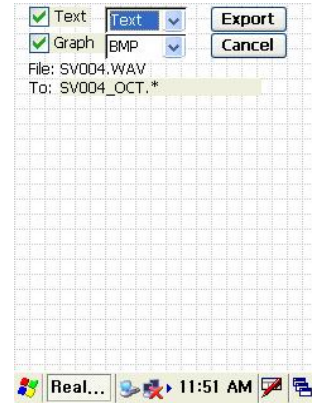
[Fig. III-4-3] Export window



[Fig. III-4-4] Exported octave image file



[Fig. III-4-5] Exported octave data file (TEXT)



[Fig. III-4-6] Automatic Export After measurement

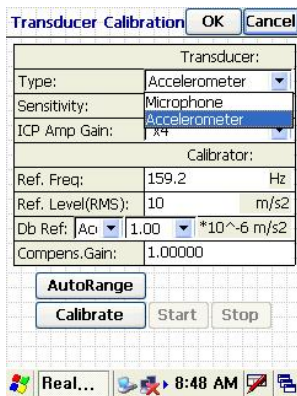
## 4. 2 Sound Level Meter, FFT, Spectrogram, Octave Mode

Sound Level Meter, FFT and Octave analysis mode support both data and graph image export. But spectrogram mode supports graph image export only. In the exported data, the frequency and corresponding values are arranged vertically.

## 5. Transducer type and Calibration

### 5. 1 Selection of transducer type and entering sensitivity

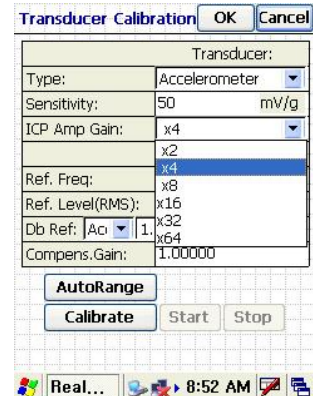
By clicking [File>>Transducer Calibration...] tap, [Fig. III-5-1] window is opened.



[Fig. III-5-1] Selecting transducer



[Fig. III-5-2] Entering sensitivity



[Fig. III-5-3] Gain adjustment

ICP(IEPE) type microphone and accelerometer is supported and input sensitivity data as shown on [Fig. III-5-2].


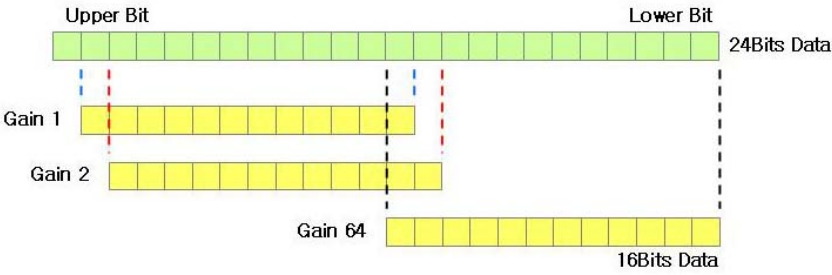
### 5. 2 AutoRange and Gain adjustment

This system supports digital gain adjustment by 1, 2, 4, 8, 16, 32 and 64 times. It is preferable to



adjust gain if the signal is low. To use AutoRange function is more desirable than hand-operated adjustment.

[AutoRange] calculates input signal and automatically adjusts gain. So it would be better to accomplish AutoRange before calibration. [Fig. III-5-3] shows AutoRange process.

<p>HighTech</p> 	<p>RealWave H/W has an analog gain with 10 times, but digital gain is used at gain adjustment. Analog amplifier accomplishes the role of Pre-Amplifier. Work of the analog amplifier depends on the cable connected to the sensor.</p>
<p>1</p>	<p>Digital gain usually uses the method that multiplies an original signal by any gain, but it also amplifies noise with signal. So this system doesn't use the method.</p> <p>This system uses the 24bits for ADC(Analog to Digital Convert) and 16bits data is extracted from 24bits data. Therefore, the Digital Gain means to extract data shifting bit position as figure below when extracting 16bits data from 24bits data.</p>
	
<p>Unlike conventional method using fixed magnification, It is very effective to amplify signal without the amplification of noise.</p>	

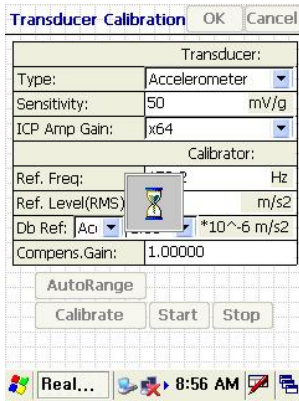
### 5. 3 Calibration

After selecting type of transducer and entering sensitivity and adjust gain, click [Calibrate] button. A graph window will appear like [Fig. III-5-5]. It is only the status to display data in real-time without measurement. By clicking [Start] button, it is calibrated during 10 seconds. After calibration, [Fig. III-5-6] will appear and the result is calculated by compensated gain.

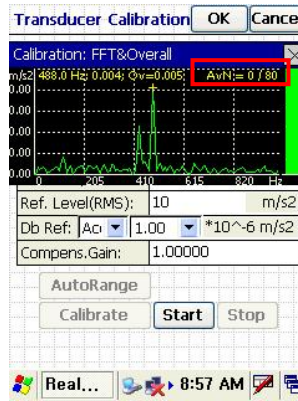
Generally, calibrated sensitivity can be calculated as follows:

$$\text{Compensated Sensitivity} = \text{Compensated Gain} \times \text{Inputted Sensitivity} \quad \text{<Eq. III-5-1>}$$

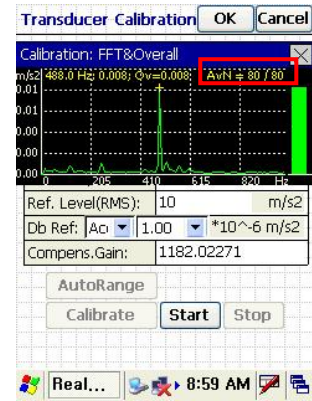
If it is calibrated inadequately, you can return to the pre-calibrate status by clicking [1.0] button. In case calibrated gain value is smaller than 0.5 or exceeds 1.5, an alarm message will be pop up and the transducer cable or the power of calibrator need to be checked.



[Fig. III-5-4] AutoRange



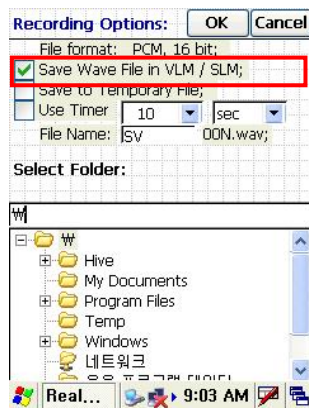
[Fig. III-5-5] Calibration window



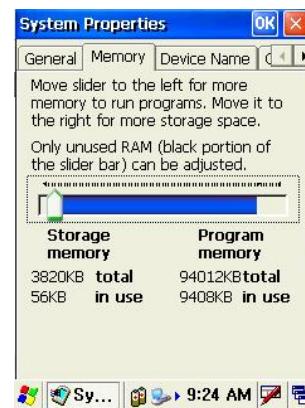
[Fig. III-5-6] Completion of calibration

## 6. Recording Options

When selecting [File>>Recording Options...] tap, a recording option window will appear like [Fig. III-6-1]. [Save Wave File In VLM / SLM] option is for if saving wave file in vibration level meter / sound level meter mode or not. By checking the option, vibration values / sound values and wave files are stored simultaneously. If it is not checked, actual wave data will not exist although file name will remain.

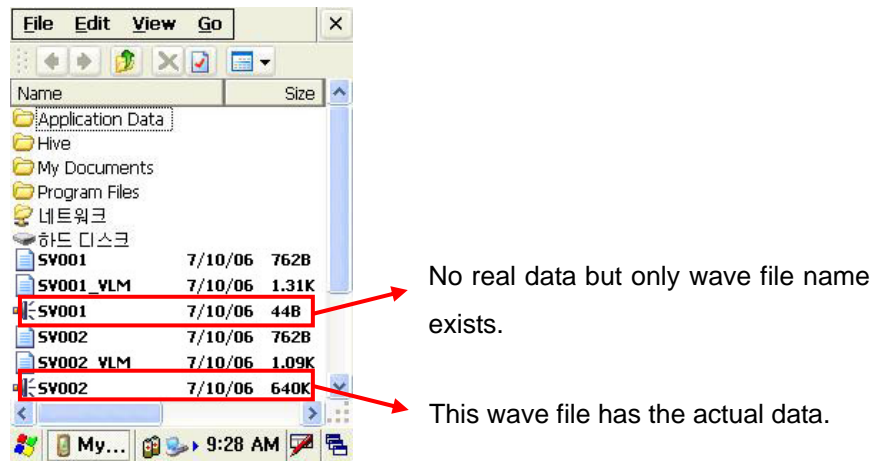


[Fig. III-6-1] Recording options window




[Fig. III-6-2] Memory information window

[Fig. III-6-3] shows the difference between checking [Save Wave File In VLM] option and no checking the option on the vibration level meter mode. The size of 'SV001.wav' is only 44Bytes at [Fig. III-6-3]. It means there's no actual recorded data. On the contrary, 'SV002.wav' has 640kBytes. It means that there's actual data.



[Fig. III-6-3] Saving wave file on vibration level meter mode


- [Use Timer] option is to define time duration. Measurement is accomplished during the defined duration and stop automatically. For example, if the timer is set 10 sec., it will measure only for 10 seconds.
- [File Name] option is to define file name. Once measurement is accomplished, each recording session automatically generates numeric name after the given file name. For example, let us consider a series of measurements with the name “SV”, numeric name automatically generated as “SV001.wav”, “SV002.wav”, etc.
- [Select Folder] option is to select folder to save files.
- [Memory Info...] button shows memory information like [Fig. III-6-2].

 To prevent losing data in case of battery discharge, it is recommended to set saving folder to the 'SD Card'.

## 7. Recording Information

Recording Information window is for entering the recording information (Name, object or axis of measurement etc.).

- [Name] : Operator
- [Object] : Object of measurement
- [Location] : Location of measurement
- [Direction] : Measurement directions of accelerometer as x, y, z-axis.

In [Location] option, you should enter the location information where measurement is accomplished in advance, and then choose pre-defined location by clicking  icon. To add new location information, it is easily added by clicking [add] button after entering new location information. To delete it, select a list and click [Del] button. Do not use special character \ / : \* ? “ < > for [Location], after the measuring it will use for folder name in desk program,

Recording Info: [OK] [Cancel]

Name: SV Corp

Date: 2011-07-28 09:33:09 THU

Object: Object

Location: [ ] [Add] [Del.]

Direction

X-Axis: x-Axis

Y-Axis: y-Axis

Z-Axis: z-Axis

Real... 9:34 AM

[Fig. III-7-1] Recording Info.

Recording Info: [OK] [Cancel]

Name: SV Corp

Date: 2011-07-28 09:33:09 THU

Object: Car

Location: Seat [Add] [Del.]

Direction

X-Axis: Processing Course

Y-Axis: Processing-LEFT

Z-Axis: Perpendicular

Real... 9:37 AM

[Fig. III-7-2] Completion to enter Rec. Info.

Recording Info: [OK] [Cancel]

Name: SV Corp

Date: 2011-07-28 09:33:09 THU

Object: Car

Location: Engine [Add] [Del.]

Direction


X-Axis: Mission

Y-Axis: Processing-LEFT

Z-Axis: Perpendicular

Real... 9:40 AM

[Fig. III-7-3] Example of pre-defined location

 Do not use special character \ / : \* ? " < > for [Location]

## IV. Real Time Measurement in Vibration Level Meter/Sound Level Meter

### 1. Basic Theory of Vibration Level Meter / Sound Level Meter

#### 1. 1 Understanding of Integration Time and Time Constant

The RMS(Root-Mean-Square) calculation routine is based on “IEC60804:2000-Integrated Averaging Sound Level Meter” and “ISO8041:1990-Human response to vibration-Measuring instrumentation”.

Generally, RMS is calculated by the equation below :

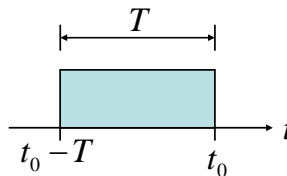
$$a_{rms} = \sqrt{\frac{1}{T} \int_0^T a^2(t) dt} \quad \text{<Eq. IV-1-1>}$$

where  $a(t)$  is instantaneous acceleration and  $T$  is measuring time.  $a_{rms}$  is calculated for  $T$  second. The calculation method in real-time is named Running RMS Acceleration which is described at ISO(International Standard Organization) standard.

The RMS calculation equation in real-time can be arranged as below :

$$a_{rms}(t_0) = \sqrt{\frac{1}{T} \int_{t_0-T}^{t_0} a^2(t) dt} \quad \text{<Eq. IV-1-2>}$$

where  $t_0$  is current time,  $T$  is measurement period(time) for calculation. Sometimes, this period is called by integral time. This calculation for RMS is linear Averaging RMS.



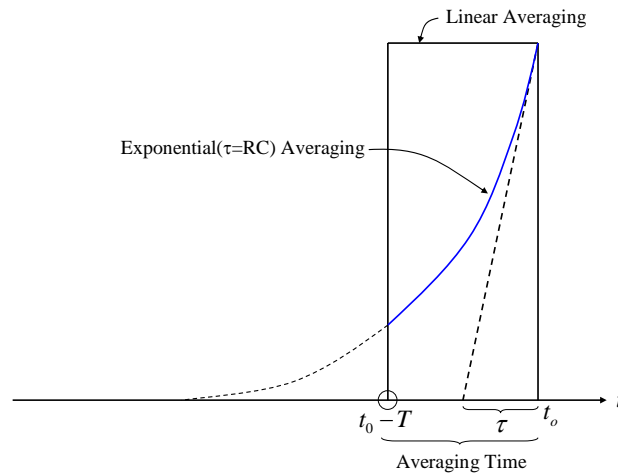
[Fig. IV-1-1] The Calculation Method with Linear Averaging RMS

The linear averaging RMS Method is used for RMS, but is inferior to the exponential averaging RMS in real-time calculation. The equation below represents the exponential averaging RMS.

$$a_{rms,\tau}(t_0) = \sqrt{\frac{1}{\tau} \int_{-\infty}^{t_0} a^2(t) e^{\frac{t-t_0}{\tau}} dt} \quad \text{<Eq. IV-1-3>}$$

where  $\tau$  is the time constant and we can read it with ‘tau’ pronunciation. As [Fig. IV-1-2] is shown, the exponential averaging RMS equation is different to the linear averaging RMS, but the results are similar and the former emphasizes recent data of signal as [Fig. IV-1-2].

Generally, the time constant is set with 0.035sec(Impulse), 0.125sec(Fast), 1sec(Slow). Each term of times writes in the parenthesis. If you want to measure the impulse signal, you have to use 0.035 sec time constant. If you use more time constant (more than 10sec), the RMS has crushed values and the response time is also slow.



[Fig. IV-1-2] Running RMS Calculation with Exponential Averaging

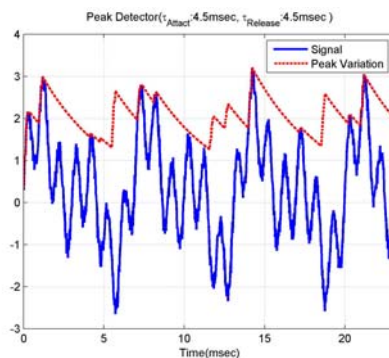
### 1. 2 Detector with Exponential Averaging Function

The full procedure of RMS(Root-Mean-Square) Detector is described in [1. 1] paragraph. When peak values are calculated, we also use Peak Detector with exponential averaging. [Fig. IV-1-3] shows how to detect peak values in real-time.

Peak Detector has two time constants. One is the attack time constant and another is the release (or decay) time constant. The attack time constant means how peak values increase quickly. And the release time constant means how the peak values decrease slowly after a peak.

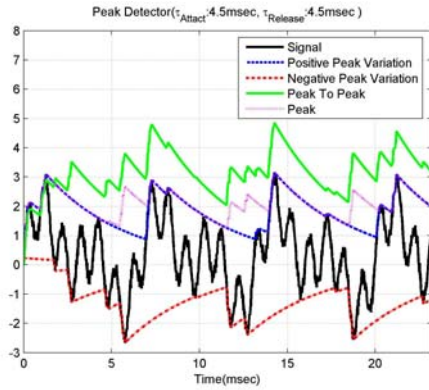
Both time constants are defined from the user definable time constant in the program. If the release time constant is smaller than 1.5sec, this is set by 1.5sec automatically which is defined in “BS60651(IEC60651:1979) – Specification for Sound Level Meters”.

[Fig. VI-1-3] shows the variation of peak as the time signal in our system. This peak is calculated from the absolute of time signal.

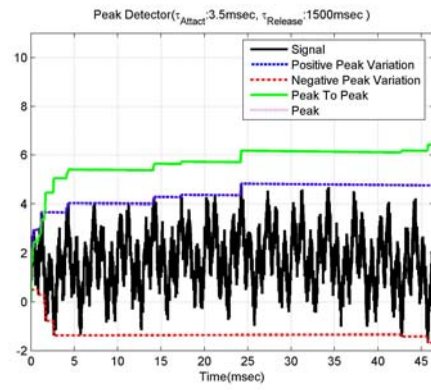


[Fig. IV-1-3] The Peak Variation of Time Signal

For getting the peak-to-peak(P2P) value, our system also use detector. At first, the positive and negative peak values are calculated, and then we can get P2P from the difference of positive P2P and negative P2P. The positive peak values are calculated from the positive signal and the negative peak values are calculated from the negative signal. [Fig. IV-1-4] and [IV-1-5] shows how to calculate P2P from signal.



[Fig. IV-1-4] A Case of Small Release Time  
Constant



[Fig. IV-1-5] A Case of Big Release Time  
Constant

### 1. 3 Calculation of the Maximum of Signal (Max. Detector)

Peak Values are decreased with the release time constant as time goes. If you want to observe the peak value(or maximum) during measuring, you have to use the maximum detector. You can't observe the peak value with peak detector. The value is found from the signal during the integration time. For example, suppose that the integration time is 5sec. This system finds the maximum value from time signal with 5sec and then saves and displays it to file at every 5sec. the integration time is shown as  $T$  at [Fig. IV-1-1]

### 1. 4 Integrator

The theory of discrete integrator is classified into 3 types. There are Trapezoidal, Forward Euler and Backward Euler. Integrator is written like below in the  $s$ -domain :

$$H(s) = \frac{1}{s} \quad \text{<Eq. IV-1-4>}$$

After this equation is rearranged with the bilinear Z Transform method, the integrator is written as below in the  $z$ -domain :

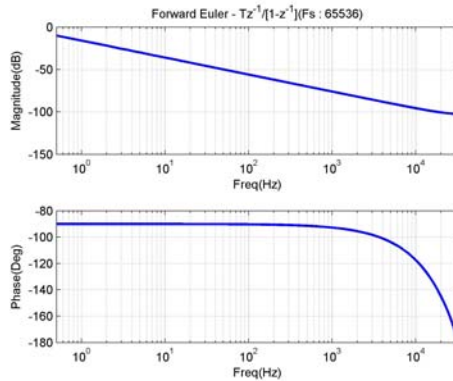
$$s = \frac{2}{T} \frac{1-z^{-1}}{1+z^{-1}} \quad \text{<Eq. IV-1-5>}$$

$$H(z) = \frac{T}{2} \cdot \frac{1+z^{-1}}{1-z^{-1}} \quad \text{<Eq. IV-1-6>}$$

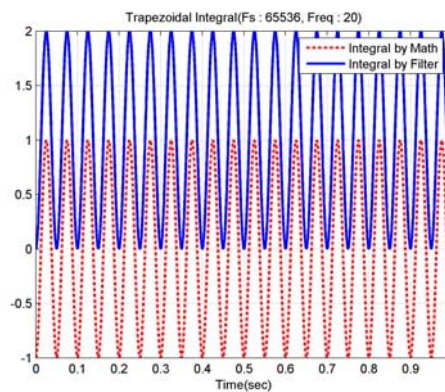
where  $T = \frac{1}{F_s}$ ,  $F_s$  is Sampling Frequency. RealWave uses the “Forward Euler” method, which is simplified from Trapezoidal Method. We can write it as below :

$$H(z) = \frac{Tz^{-1}}{1 - z^{-1}} \quad \text{<Eq. IV-1-7>}$$

The frequency response function of integrator is shown in the [Fig. IV-1-6].

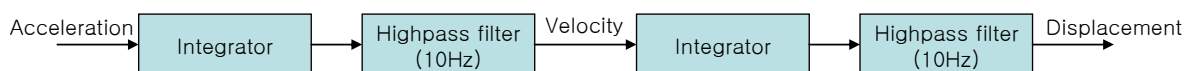


[Fig. IV-1-6] Frequency Response of Forward Euler Integrator



[Fig. IV-1-7] Integrated Result

Discrete integrator generates offset of signal (see [Fig. IV-1-7]). We must use highpass filter for removing the offset. The highpass filter with 10Hz cutoff frequency (2-pole Butterworth IIR Filter) is used in RealWave. The procedure for velocity and displacement is shown in the [Fig. IV-1-8]



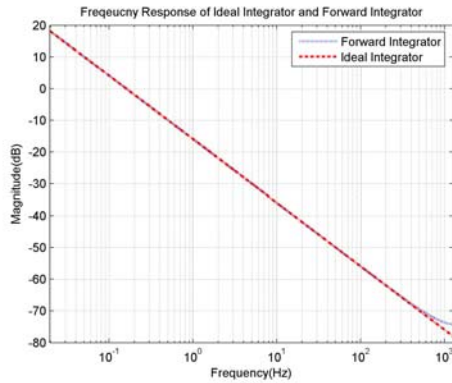
[Fig. IV-1-8] Flow-chart for integration

Discrete integrator has original error as shown in the [Fig. IV-1-9]. Specially, that error increases

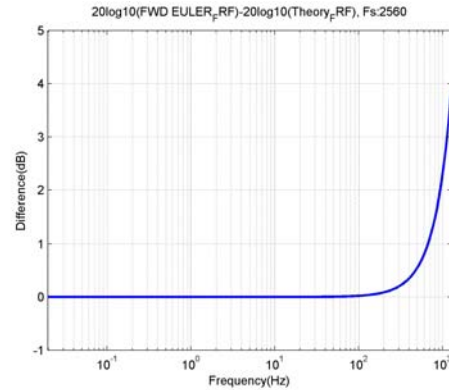


gradually in the high frequency region. But the integrated values(velocity or displacement) have a very small values less than -50dB. Finally, the errors rarely affect the integrated value.

The integrated values have linearity in the low frequency which is in range of 23% of sampling frequency. If a sampling frequency is set 10 times of the wanted frequency, we can get a good linearity in the wanted frequency. Therefore, the sampling frequency is set 1024Hz because the useful frequency range in the human vibration level meter is 80Hz.



[Fig. IV-1-9] Frequency Response



[Fig. IV-1-10] Phase Response

### 1. 5 Sound Pressure Level Calculation

Differently with Vibration Level Meter, you can calculate dB value of measured sound signal which uses microphone using such below basic theory through the chapter 1.1 to 1.4.

Sound Pressure Level (SPL) is composed as below.

$$SPL(L_p) = 20 \cdot \log_{10} \left( \frac{p'_{rms}}{p_{ref}} \right)$$

$p'_{rms}$  : Root Mean Square (Pa)

$p_{ref}$  : Reference Sound Level (20  $\mu Pa$ )

Equivalent continuous Sound Pressure Level ( $L_{eq}$ ) is expressed as below.

$$L_{eq} = 10 \cdot \log_{10} \left( \frac{1}{T} \sum \left( \frac{p'}{p_{ref}} \right)^2 \right)$$

$p'$  : Real Sound Pressure (Pa)

$p_{ref}$  : Reference Sound Level (20  $\mu Pa$ )

$T$  : Measurement Time

You can get Equivalent continuous A-weighted sound pressure level when A-weighted  $p'_A$  replaces  $p'$  from this equation.

## 2. Setting of measuring option

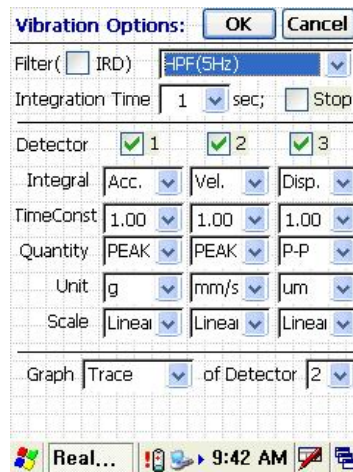
### 2. 1 Selection of Filter

#### 2. 1. 1 Filter selection of vibration level meter

A window is opened to press  button like [Fig. IV-2-1]. You can see 9 type filters.

Here are the list below :

- 1Hz Highpass Filter
- 5Hz Highpass Filter
- 10Hz Highpass Filter
- 1KHz Highpass Filter
- 2Hz ~ 300Hz Bandpass Filter
- 2Hz ~ 1kHz Bandpass Filter
- 10Hz ~ 500Hz Bandpass Filter
- 10Hz ~ 1kHz Bandpass Filter
- WBCombined Filter (ISO6954)



[Fig. IV-2-1] Parameter Selection Window in Vibration Level Meter

And another option is supplied for low frequency analysis as below :

- BW 1.6kHz : it means 1.6kHz bandwidth for analysis, Sampling Frequency is 4096Hz.
- BW 200Hz : it means 200Hz bandwidth for analysis, Sampling Frequency is 512Hz.
- BW 100Hz : it means 100Hz bandwidth for analysis, Sampling Frequency is 256Hz.

Highpass Filter uses 2-pole Butterworth Filter, Bandpass filter uses 4-pole Butterworth Filter. And Whole-Body Combined filter is made with being based on the ISO6954 and ISO8041.

The frequency response of the each filter is shown in the [Appendix 2 : The frequency response of Vibration Level meter] and [Appendix 3 : The frequency response of whole-body vibration filter]. And if you select WBCombined filter, Quantity value become initialize to RMS value.

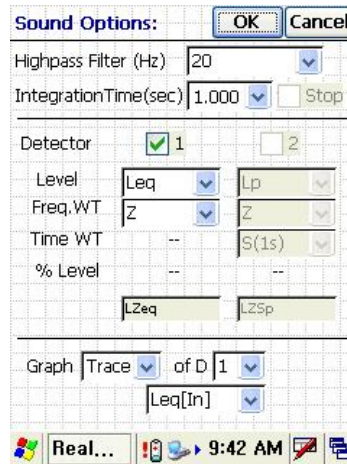


If you select WBCombined filter, Quantity value become initialize to RMS value.

## 2. 1. 2 Filter selection of sound level meter

A window is opened to press  button like [Fig. IV-2-2]. You can see 2 type filters.

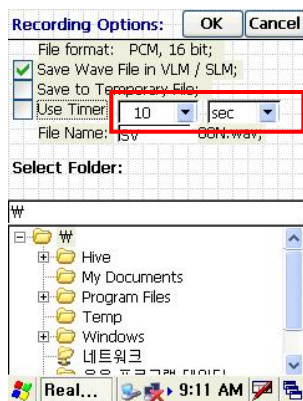
- 1Hz Highpass Filter
- 20Hz Highpass Filter



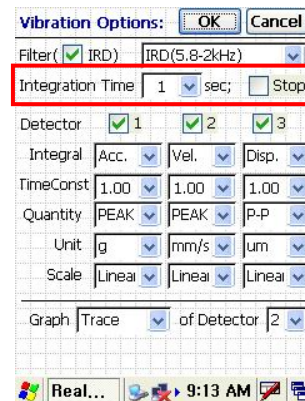
[Fig. IV-2-2] Parameter Selection Window in Sound Level Meter

## 2. 2 Integration Time

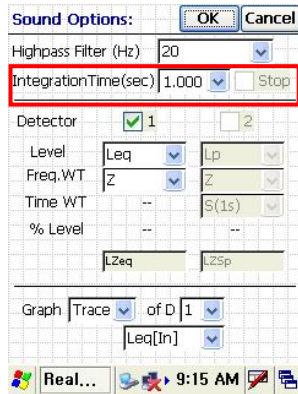
This parameter is the  $T$  of <Eq. IV-1-1> and <Eq. IV-1-2> and means the measuring time or the observation time. The calculated values are saved to file at every integration time. If you select stop check box in [Fig. IV-2-4] and [Fig. IV-2-5], RealWave will stop measuring in the integration time. Also if you select [Timer] in the [Recording Options] and [Stop] in the integration time simultaneously, the system will stop at a short time. For example, if the timer is set by 10sec and the integration time is set 2sec without stopping function, the system will save the 5 data at every 2sec and stop after 10sec.



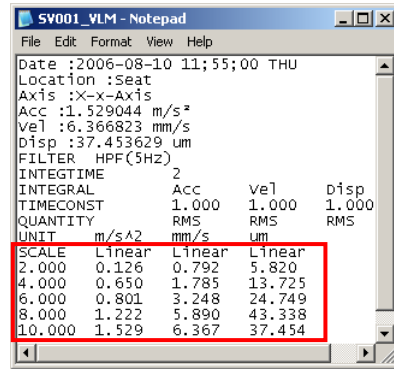
[Fig. IV-2-3] Timer Setting



[Fig. IV-2-4] VLM Integration Time Setting



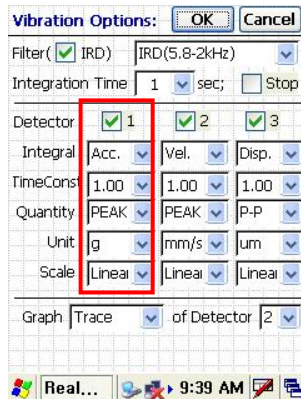
[Fig. IV-2-4] SLM Integration Time Setting



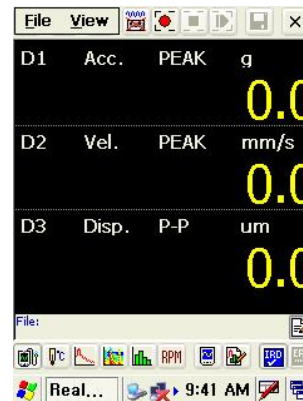
[Fig. IV-2-6] Measured Results

### 2. 3 Detector

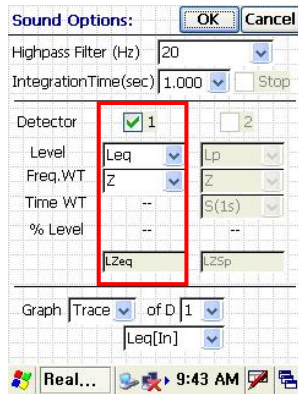
The results of 3 detectors are displayed in vibration level meter and 2 detectors are displayed in sound level meter simultaneously. Only the data of the selected detector are displayed and stored. For example, if you want to measure acceleration in vibration level meter, you can select only a detector like [Fig. IV-2-7]. And if you want to measure Leq in sound level meter, you can select only a detector like [Fig. IV-2-9]. And [Fig. VI-2-8], [Fig. VI-2-10] is shown the activated display window. The options of each detector are listed vertically as shown in [Fig. VI-2-7] and [Fig. VI-2-9]. RealWave supports various units with gravity acceleration(g) and decibel(dB) (see [Fig. IV-2-11] and [Fig. IV-2-12]).



[Fig. IV-2-7] One Detector Selection in VLM



[Fig. IV-2-8] Activated Window for One Detector in VLM

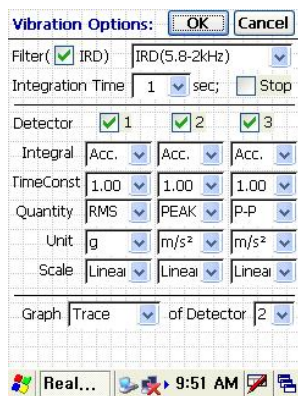


[Fig. IV-2-9] One Detector Selection in SLM

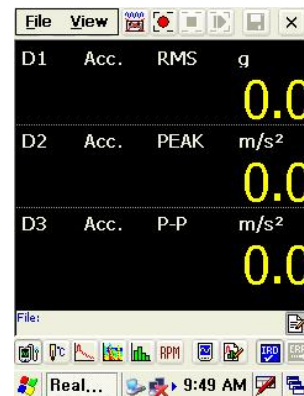


[Fig. IV-2-10] Activated Window for One Detector in SLM

You can measure acceleration, velocity, displacement with various physical quantities at the same time in the vibration level meter. The various quantities are RMS(Root-Mean-Square), Peak, Peak To Peak, Max and Min.

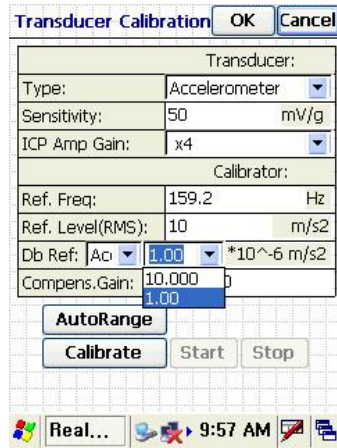


[Fig. IV-2-11] Setting with various units in VLM



[Fig. IV-2-12] Display with various units in VLM

You can change unit with  $m$ ,  $mm$ ,  $\mu m$ ,  $g$ . For example, you can set a unit of acceleration to  $m/s^2$ ,  $cm/s^2$ ,  $mm/s^2$ ,  $\mu m/s^2$ ,  $g$  and display vibration levels with linear scale or logarithms scale(dB:Decibel). The reference acceleration of decibel unit is  $10^{-6} m/s^2$ . Of course, you can choose  $10^{-5} m/s^2$  in the [Acc. Decibel Ref:] of [File >>Transducer Calibration...] tap. This function works when you select accelerator in the transducer. The references of the decibel calculation are listed in the <Table IV-2-1>.

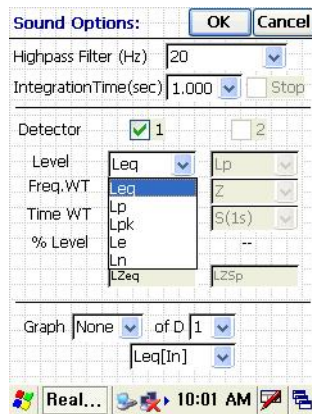


[Fig. IV-2-13] Setting the reference of decibel in VLM

<Table IV-2-1> The Reference of Decibel

	Korea, etc.	ISO
Acceleration	$10^{-5} \text{ m/s}^2$	$10^{-6} \text{ m/s}^2$
Velocity	$10^{-9} \text{ m/s}$	$10^{-9} \text{ m/s}$
Displacement	$-10^{-12} \text{ m}$	$10^{-12} \text{ m}$

You can measure the Leq, Lp, Lmax, Lmin, Lpk in sound level meter.

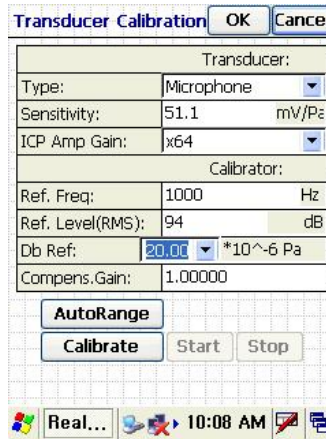


[Fig. IV-2-14] Setting with various units in SLM.



[Fig. IV-2-15] Display with various units in SLM

Reference level value of sound level meter is  $20 \mu Pa$ .



[Fig. IV-2-16] Setting the reference of decibel in SLM

## 2. 4 Display

You can choose one detector among 3 detectors in Vibration Level Meter mode and among 2 detectors in Sound Level Meter mode in the [Detector] of [Display] Option. You can see 4 types- [None], [Trace], [Octave], [FFT]-graph as display option from [Fig. IV-2-17] to [Fig. IV-2-21]



[Fig. IV-2-17] Display-None in VLM



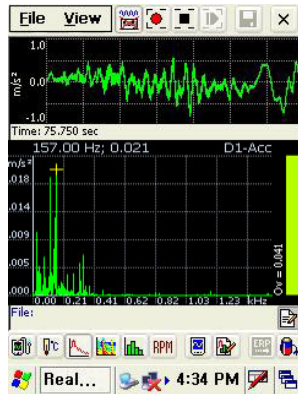
[Fig. IV-2-18] Display-None in SLM



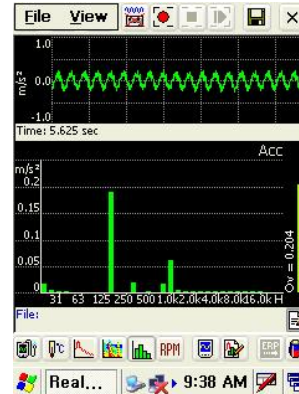
[Fig. IV-2-19] Display-Trace in VLM




[Fig. IV-2-20] Display-Trace in SLM




[Fig. IV-2-21] Display-FFT in VLM




[Fig. IV-2-22] Display-Octave in VLM

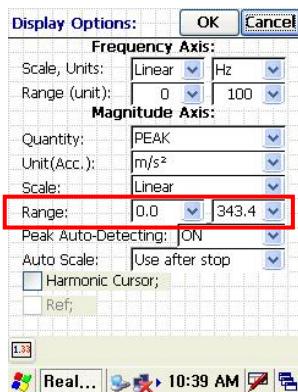
But octave and FFT graph is shown only in the WBCombined Filter. The 1/1, 1/3 and 1/6 Octave analysis which is selected in FFT option window is supported. So after you changed to FFT mode, you have to press the FFT option button  to set octave. If you want to see the detailed contents, see the chapter V-5.

 The FFT and Octave Graph of the Vibration Level Meter are supported in the Human Vibration Filter, BW 1.6kHz, BW 200Hz and BW 100Hz. They are not supported in another Filter. Also, the option of octave is set in the FFT option window.

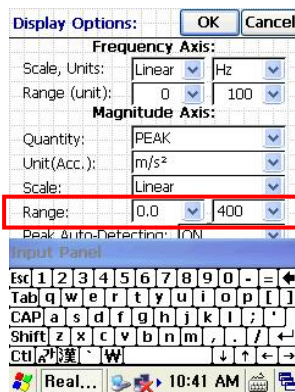
### 3. Display Option

#### 3.1 Trace Graph

If you press the  button, the display option window appears as [Fig. IV-3-1]. The option in the red square is only for vibration level meter. The range of y-axis in the trace graph is defined at [Range] as [Fig. IV-3-1] or you can input this range with keyboard. If user clicks the Y-Axis by stylus pen, the range of Y-Axis is automatically adjusted.



[Fig. IV-3-1] Adjustment of the range of trace



[Fig. IV-3-2] Entering the range of trace with keyboard

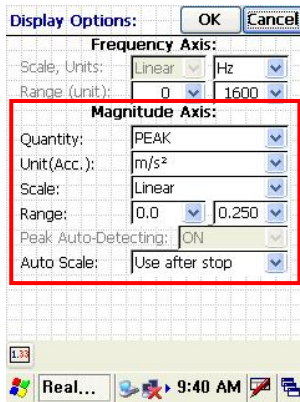


[Fig. IV-3-3] Auto-scaling for y-axis

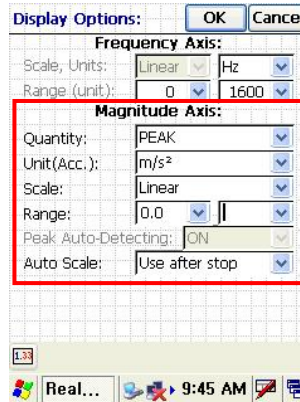


### 3. 2 Octave graph

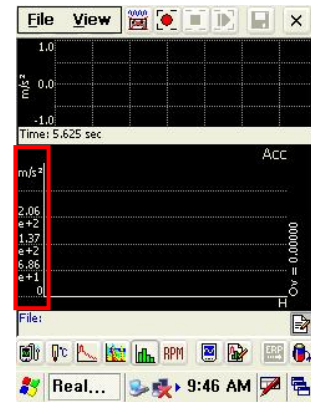
Control of octave graph is similar to controlling trace graph. For more detailed information about the control, see the pictures from [Fig. IV-3-4] to [Fig. IV-3-6].



[Fig. IV-3-4] Adjustment of the scale of octave



[Fig. IV-3-5] Entering the range

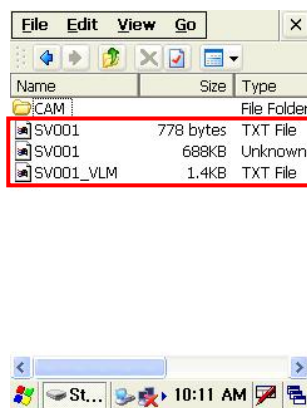


[Fig. IV-3-6] Auto-scaling for y-axis

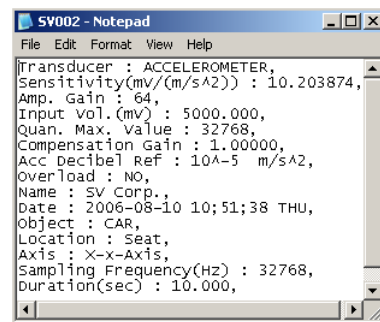
### 4. Measurement and Storage

Measurement starts with pressing button. If you don't use [Timer] or [Integration Time] option, you have to press stop button for stopping. Stored files are three which are a wave file, text file with measurement option and text file with vibration level. Saving the wave file depends on [Save Wave File In VLM / SLM] option in [File>>Recording Options...] tap. If you check the saving option, a wave file will be saved. Otherwise, a wave file exists without data.

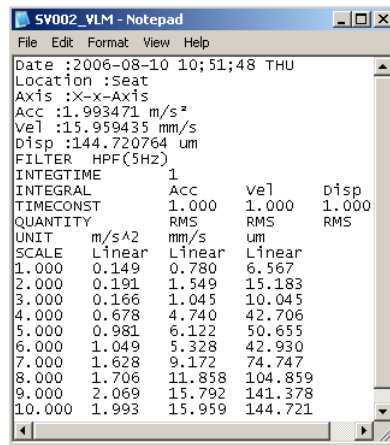
If you want to get more information, see the chapter III-6.



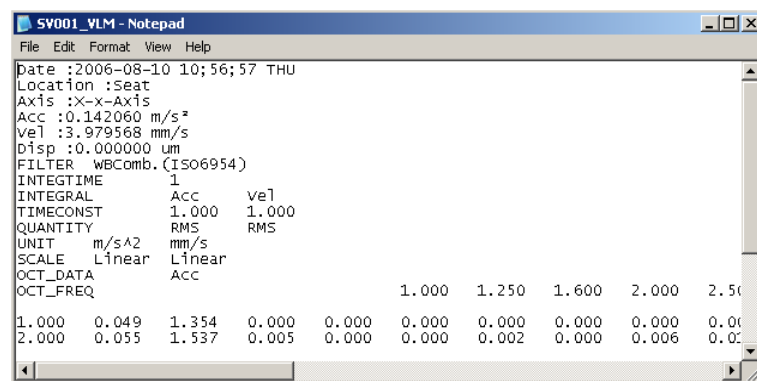
[Fig. IV-4-1] The Three Saved Files



[Fig. IV-4-2] File with Measurement Information





[Fig. IV-4-3] File with Vibration Level



[Fig. IV-4-4] File with Vibration Level and Octave Results


If you want to get results with vibration level and octave data together as [Fig. IV-4-4], you should set the octave graph at [Display] in vibration level meter.

## 5. Playback and Analysis



You can play the wave file using playback button  after opening (or loading) wave file. The measured wave file is so loading basically that you can play that without loading (see [Fig. IV-5-1]). Playback button  is activated always when wave file is loading. The analysis is carried out playing wave file as the measured options. If you play wave file again, the results of analysis are overwritten at every playing.



[Fig. IV-5-1] Playback Button is Activated after Loading Wave File

 Measured and Analyzed Data File is overwritten at every playback. If you want to analyze with another option and playback, you have to change the results file name absolutely.

## 6. Input of Measurement Information

Press toolbar switch button  in red circle of [Fig. IV-6-1] after measurement. You can see  button which is measurement information input button. If you click it, dialog box appears for selecting the information of measurement as [Fig. IV-6-2]. Texts (or Information) of dialog box are inputted previously in [File>>Recording Informations...] option. Also the information is written in the result file of vibration level.



[Fig. IV-6-1] Measurement Information Button



[Fig. IV-6-2] Selection of Measurement Information

## 7. Octave Analysis in Vibration Level Meter

The option of octave analysis in the human vibration filter of vibration level meter is listed below :

- Sampling Frequency : 512Hz

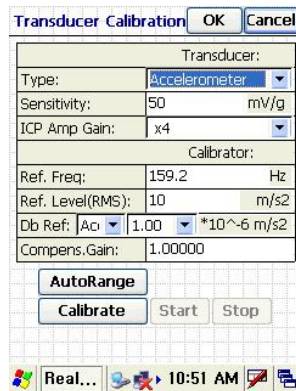
- Buffer Size : 2048
- Hanning Window
- Exponential Averaging : Time constant is same to that of detector.
- Frequency Range : 1~100Hz
- Nth Octave : 1/1, 1/3, 1/6 Octave is useful. This parameter is changeable in [Nth Octave] of FFT Analysis(or Spectrogram, Octave Analysis) Option.

If you want to get the more information of octave analysis, refer to the chapter V-3.

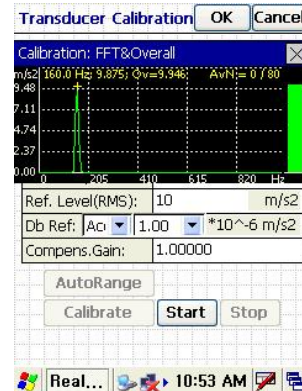
## 8. Setting options for Human Vibration Measurement

### 8.1. Selecting type of sensor, Entering sensitivity and Calibration

- ① Select [Accelerometer] in [File>>Transducer Calibration...] tap.
- ② Entering Sensitivity at the option [Sensitivity].
- ③ With connecting sensor to calibrator, click [Calibrate] and [Start] button to accomplish calibration.
  - ⇒ [Compens. Gain] value is changed from 1 to another value.
- ④ Adjustment of [ICP Amp Gain] just before measuring : With connecting sensor and adhering the sensor at measurement position, select [ICP Amp Gain] or [AutoRange] button to set the gain automatically.



[Fig. IV-8-1] Selecting Sensor and Entering Sensitivity

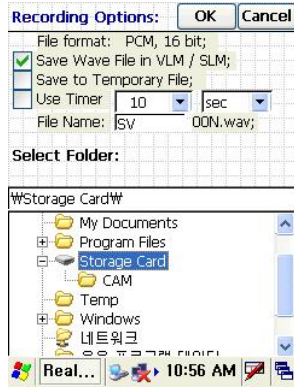


[Fig. IV-8-2] Calibration

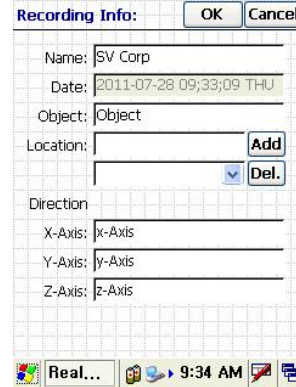
### 8.2. Setting the measurement time and file name

When selecting [File>>Recording Option...] tap, a window appears like [Fig. IV-8-3].

- ① [Save Wave File in VLM / SLM] option should be checked to store wave files.
- ② [Use Timer] option should be checked to use timer. Measurement for 1 minute is recommended at ISO6954:2000(E). If main component is smaller than 2Hz, it is recommended to set the measurement for 2 minutes.
- ③ Enter the file name as you wish at [File Name].
- ④ Select the folder at [Select Folder:]



[Fig. IV-8-3] Recording Option



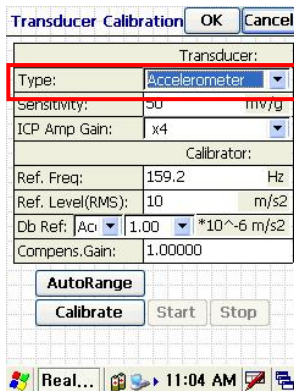
[Fig. IV-8-4] Entering Recording information

### 8.3. Entering Recording information

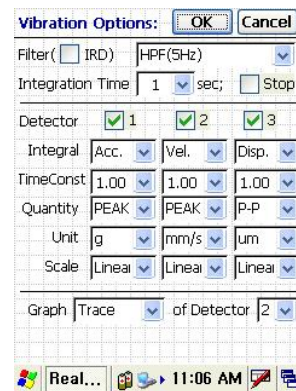
- ① Select [File>>Recording Information...]
- ② Enter name, object, location and direction of measurement in advance.

### 8.4. Selecting Vibration Level Meter and Setting the measurement options

- ① Change the analysis mode to vibration level meter.
- ② Select the vibration options as [Fig. IV-8-5]. Set [Integration Time] for 1 second and [Time Constant] for 1 second. Therefore, the data is stored per 1 second for 1 minute.



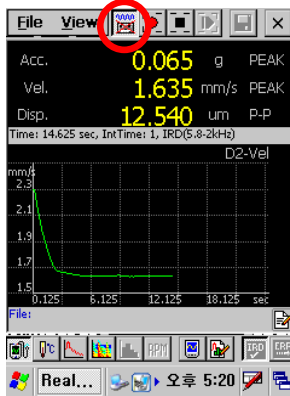
[Fig. IV-8-5] Conversion of Vibration Level Meter Mode



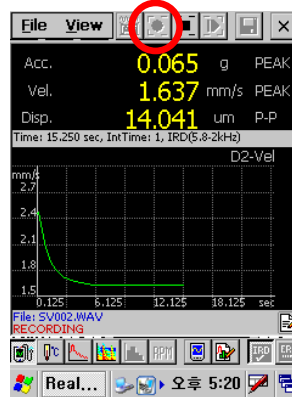
[Fig. IV-8-6] Window for Vibration Options

## 8.5. Measurement

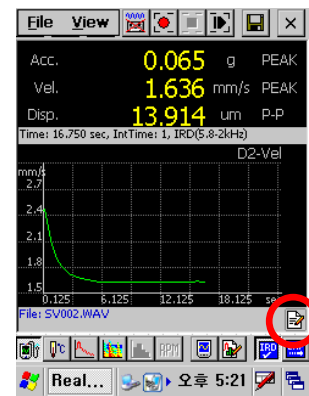
- ① Confirm the status of measurement to press [ARM] button.
- ② Start measurement to press [Record] button.
- ③ Enter recording information to press [Entering recording information] button.



[Fig. IV-8-7] ARM Button



[Fig. IV-8-8] Recording Button

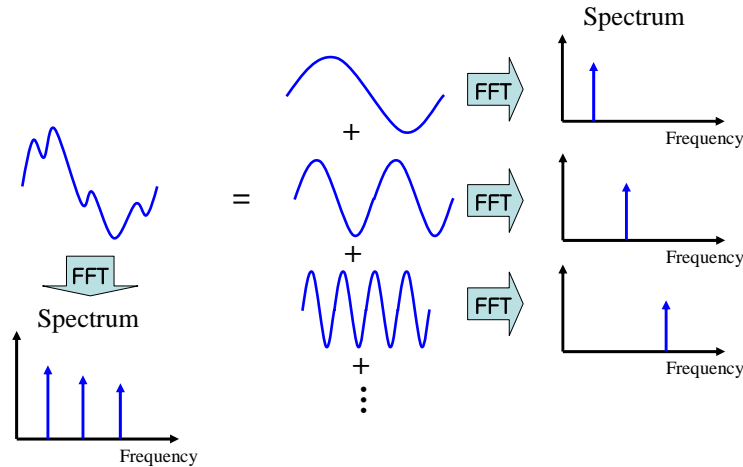


[Fig. IV-8-9] Entering recording information Button

## V. Real Time FFT Spectrum·Spectrogram and Octave Analysis

### 1. Understanding of Frequency Analysis

The shape of the time signal is so complex that we can't see the information of frequency in the measured time signal. Therefore, we have to calculate FFT (Fast Fourier Transform) analysis to get the information of frequency-the periodicity of time signal. Frequency analysis is classifying the magnitude of time signal as frequency like [Fig. V-1-1]



[Fig. V-1-1] Concept of Frequency Analysis

### 2. Fast Fourier Transform

Frequency analysis usually uses Fast Fourier Transform which is abbreviated to FFT. The basic assumption to calculate FFT is that time signal has infinite periodicity. Here is the calculating equation below :

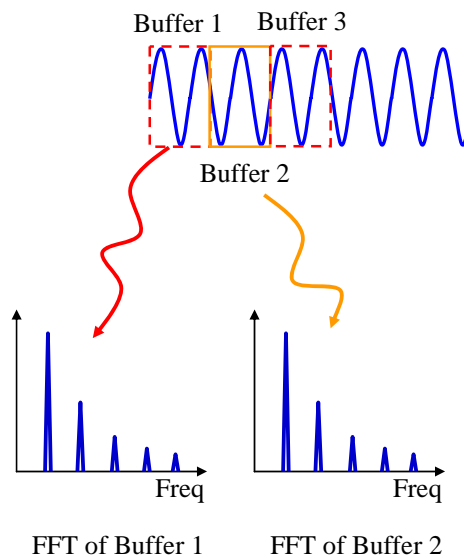
$$X[k] = \sum_{n=0}^{N-1} \left[ x(n) e^{-j2\pi \frac{k}{N} n} \right], \quad 0 \leq k \leq N-1 \quad \text{<Eq. V-2-1>}$$

where  $x(n)$  is discrete time signal,  $N$  is the buffer size of FFT. And  $X[k]$  is the calculated result of FFT. In general,  $N$  has the number of  $2^n$  -512, 1024, 2048, 4096. The parameters of FFT are listed in <Table V-2-1>.

The calculation of FFT accomplishes after an allocated buffer is filled with data. This procedure is shown in [Fig. V-2-1].  $FFT_1$  is calculated after the first buffer is filled with data, and then  $FFT_2$  is calculated after the second buffer is filled with data. This procedure is continuous, so we can see the FFT results as real-time data.

<Table V-2-1> Defined Parameters for FFT

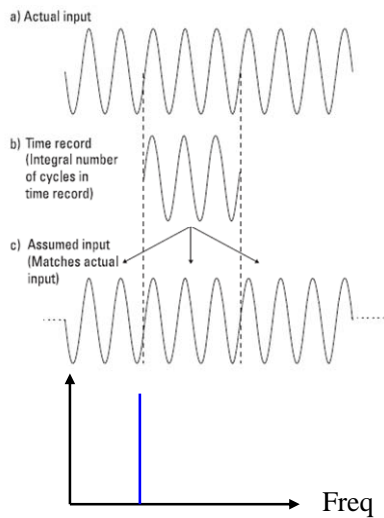
Parameters	Definition	Equation
$F_s$	Sampling Frequency	User Definition
$F_{span}$	Useful Frequency Range	$F_{span} = \frac{F_s}{2.56}$
$N$	Buffer Size	User Definition
$L$	Display Line	$L = \frac{N}{2.56} + 1$ or $L = \frac{N}{2.56}$
$\Delta f$	Frequency Resolution	$\Delta f = \frac{F_s}{N}$
$\Delta t$	Sampling Interval(sec)	$\Delta t = \frac{1}{F_s}$
$T$	Recording Time or Total Time of Saved Data in Buffer	$T = N \cdot \Delta t$



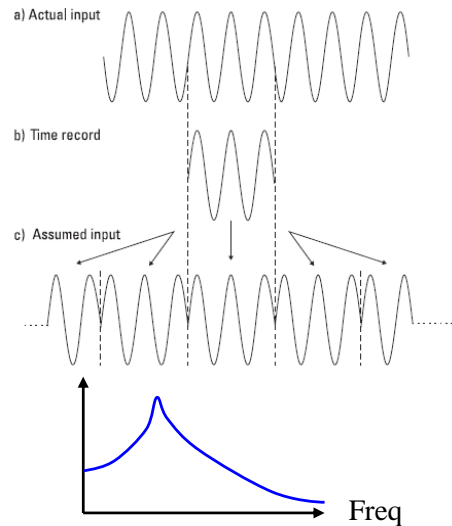
[Fig. V-2-1] Instantaneous FFT results from each FFT calculation

As previously mentioned, the basic assumption of FFT analysis is that time signal has infinite periodicity. But buffer size is so finite that energy leakage causes. So we call it “leakage error”. As shown in [Fig. V-2-2], time signal with infinite periodicity generates just one peak in FFT results. However, as shown in [Fig. V-2-3], time signal with finite buffer causes finite periodicity. If we calculated FFT using this buffer, we got the broaden peak of FFT.





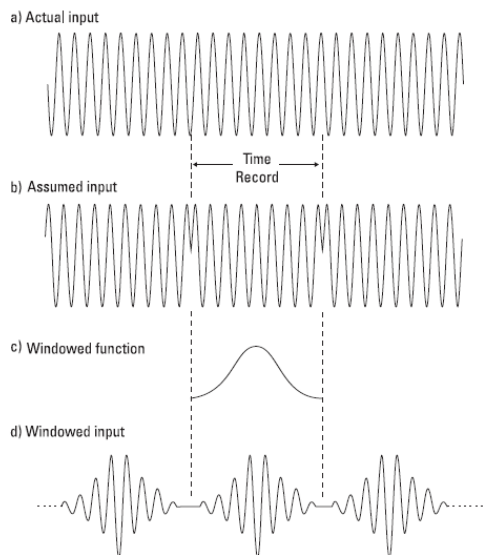
[Fig. V-2-2] Ideal FFT



[Fig. V-2-3] Actual FFT

## 2. 1 Windowing

We must use windowing function to solve leakage error. The procedure shown in [Fig. V-2-4] is the windowed time signal which is original time multiplied by windowing function. The figure (b) is shown the finite periodicity of time signal that is recognized in FFT algorithms, but figure (d) represents the infinite periodicity of time signal by multiplying original time signal and windowing function. Therefore, a windowed time signal has not leakage error.



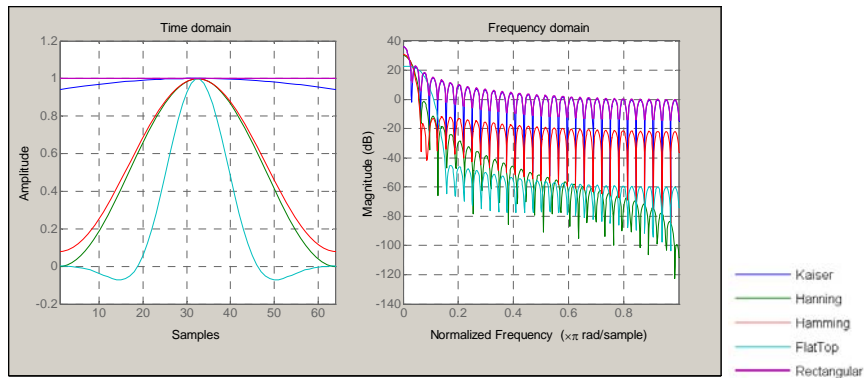
[Fig. V-2-4] Windowing Procedure

Windowing is multiplied time signal and the coefficients of window which has number  $N$ . Here is the equation below :

$$x_w(i) = w(i) \times x(i), \quad 0 \leq i \leq N-1$$

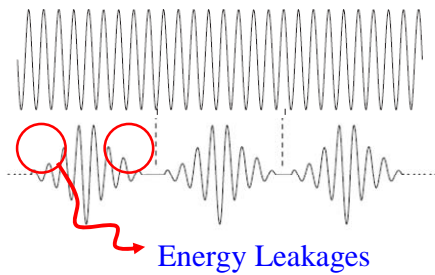
<Eq. V-2-2>

where  $N$  is buffer size. There are all kinds of window function, but generally hanning, flat-top and rectangular(no window) window is used.



[Fig. V-2-5] Various Windowing Functions

However windowing causes the leakage of time signal's energy as [Fig. V-2-6].

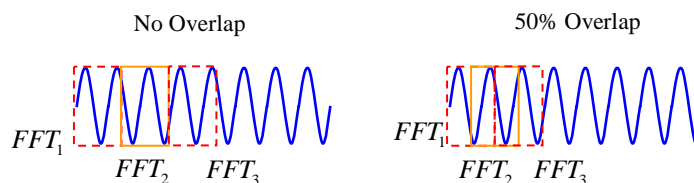


[Fig. V-2-6] Windowing causes the leakage of energy of time signal

## 2. 2 Overlap

Overlapping is done to protect energy leakage. If you hope to understand the overlapping, you have to know no overlapped time signal. For example, let's suppose that the first time data  $x_1$  has  $0 \sim N-1$ , the second time data  $x_2$  has  $N \sim 2N-1$  and so on.  $FFT_1$  is the FFT spectrum of  $x_1$  and  $FFT_2$  is the FFT spectrum of  $x_2$ . See [Fig. V-2-7]. The left figure is no overlap that means the time signal buffer for FFT is not repeated. 50% Overlapping uses the second half data of  $x_1$  and the first half data of  $x_2$  together.

Over 66% overlapping doesn't usually have the loss of energy in the time signal.



[Fig. V-2-7] Overlapping

## 2. 3 Averaging of FFT Spectrum

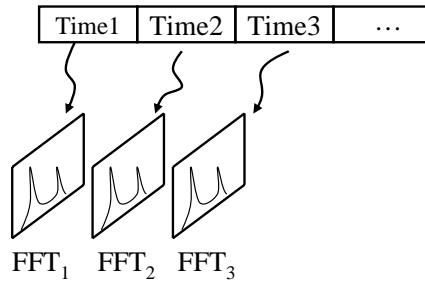
The reason of averaging spectrum is to remove noises in the spectrum. There are linear averaging, exponential averaging and peak hold averaging.

### 2. 3. 1 Linear Averaging

Let's suppose that  $FFT_1$  is FFT spectrum of the first time signal buffer,  $FFT_2$  is FFT spectrum of the second time signal buffer and so on as [Fig. V-2-8]. Linear averaging is the arithmetic average of spectrums at each frequency. On other words, you have to sum all of spectrums at special frequency, and then divide summed values with averaging number like the equation below.

$$\overline{FFT}(k) = \frac{1}{n_a} \sum_{i=1}^{n_a} FFT_i(k), \quad 0 \leq k \leq \frac{N}{2} \quad \text{<Eq. V-2-3>}$$

where  $\overline{FFT}$  is the averaged spectrum of FFT.  $n_a$  is averaged number.  $N$  is buffer size.



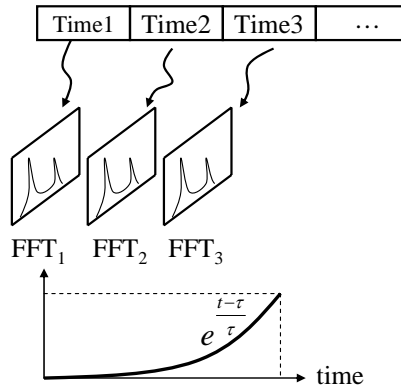
[Fig. V-2-8] Instantaneous FFT Spectrum

### 2. 3. 2 Exponential Averaging

Exponential averaging is calculated from the exponential averaged spectrum at each frequency using <Eq. V-2-4>. Exponential averaging emphasizes recent spectrum in all of spectrums.

$$\overline{FFT}_i(k) = \left(1 - e^{-\frac{T}{\tau}}\right) \cdot \sum_{i=0}^{\infty} e^{-\frac{iT}{\tau}} \overline{FFT}_{i-1}(k), \quad 0 \leq k \leq \frac{N}{2} \quad \text{<Eq. V-2-4>}$$

where  $\overline{FFT}_i$  is the averaged spectrum of FFT at present.  $\overline{FFT}_{i-1}$  is the proceeding averaged spectrum.  $T$  is the total time of one buffer per second. And  $\tau$  is time constant per second.



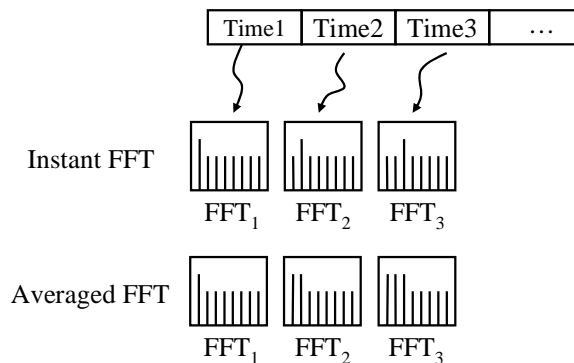
[Fig. V-2-10] Exponential Averaging

### 2. 3. 3 Peak Hold Averaging

This averaging keeps the biggest peak value of spectrum at the same frequency to repeat with averaging number. This theory is shown in [Fig. V-2-10] and <Eq. V-2-5>.

$$\overline{FFT(k)} = \text{MAX} [FFT_i(k)], \quad 0 \leq i \leq n_a - 1, \quad 0 \leq k \leq \frac{N}{2} \quad \text{<Eq. V-2-5>}$$

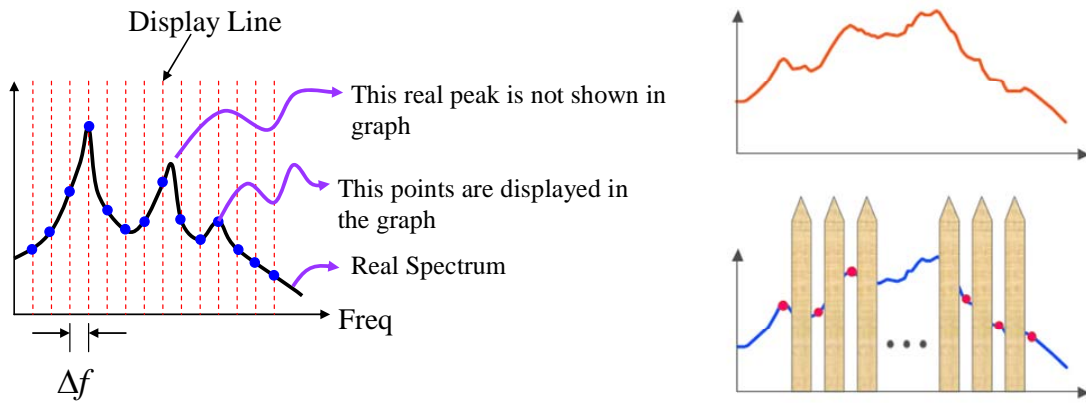
where  $n_a$  is averaging number(or iteration number for averaging).



[Fig. V-2-10] Peak Hold Averaging

### 2. 4 Picket Fence Error

The resolution of frequency is defined as dividing the sampling frequency by buffer size. Therefore, if the peak of spectrum exists in the smaller region than resolution, the peak was not shown with real value. The energy of peak divides into both side spectrums. In this situation, we can't get a real frequency and magnitude of peak. This is called 'Picket Fence Error'. This error is shown in [Fig. V-2-11]. If you want to get the exact frequency and magnitude of spectrum, you have to compensate both parameters with some theory.

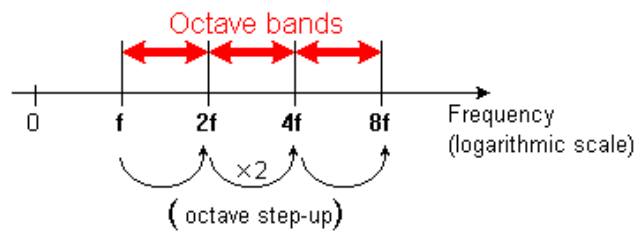


[Fig. V-2-11] Picket Fence Error

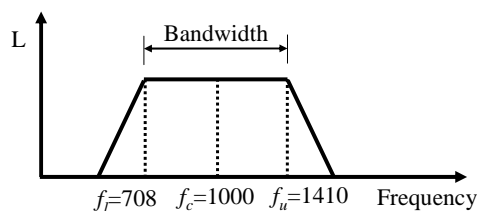
### 3. FFT Based Octave Analysis

FFT analysis is named narrow band frequency analysis, but octave analysis which is describing in this paragraph is called broad band analysis. Octave means the twice frequency of reference's frequency. Reference frequency is defined as 1kHz in the International Standards-ANSI S1.11. Therefore, we can make bandpass filters with center frequency which is based on octave of 1kHz.

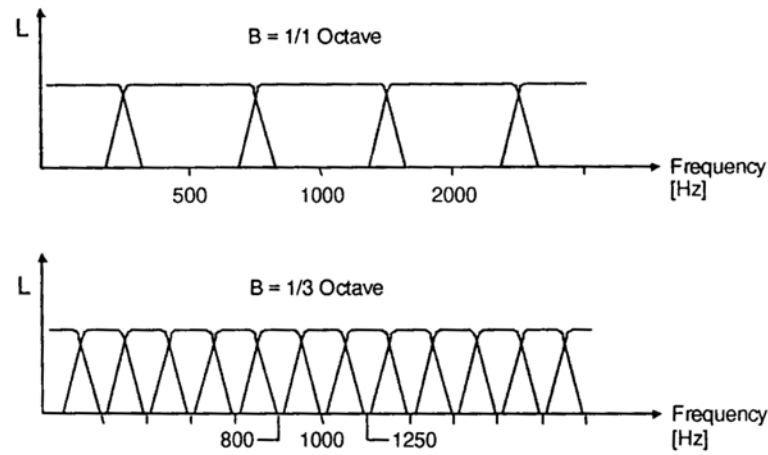
Octave analysis uses a lot of defined bandpass filter. The 1/1 octave bandpass filter with 1kHz center frequency is shown in [Fig. V-3-2]. This 1/1 octave filter means bandpass filter with a full octave band. 1/3 octave filters are realized by bandpass filter with bandwidth that is divided an octave by 3. Therefore, you can get 1/1 octave data from 1/3 octave data.



[Fig. V-3-1] Concept of Octave

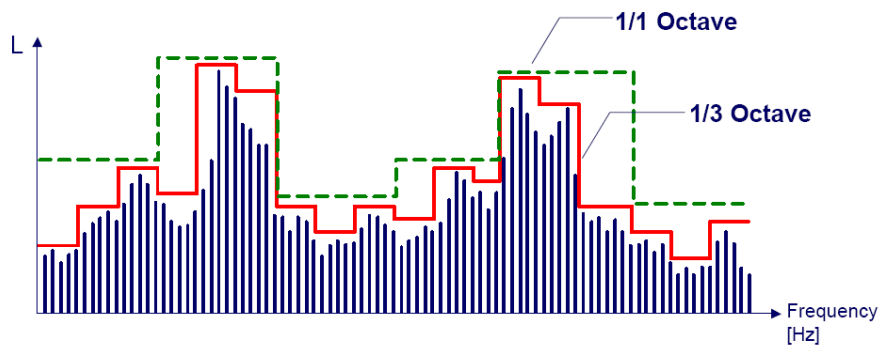


[Fig. V-3-2] 1/1 Octave Based Bandpass Filter with 1kHz Center Frequency and an octave bandwidth



[Fig. V-3-3] 1/1 and 1/3 Octave Bandpass Filters

1/1 and 1/3 octave bandpass filters are shown in [Fig. V-3-3]. As previously mentioned, we can calculate the octave data from FFT spectrum. The relationship is shown in [Fig. V-3-4].



[Fig. V-3-4] Relationship between FFT Spectrum and Octave Analysis

Equation for calculating is <Eq. V-3-1>.

$$y_{1/3Octave} = \sqrt{\sum_{i=0}^{N/2} y_i^2 \cdot \text{Window Correction}} \quad \text{<Eq. V-3-1>}$$

where  $y_i$  is the spectrum of FFT.  $y_{1/3Octave}$  is the octave value of each band.  $i$  is frequency index in the spectrum.

#### 4. Weighting

Microphone is reacted uniformly as the frequency of sound. So when you measure the sound using a microphone, you can't get the sound level as you hear the sound because the ear of human being hears sound differently according to the frequency.

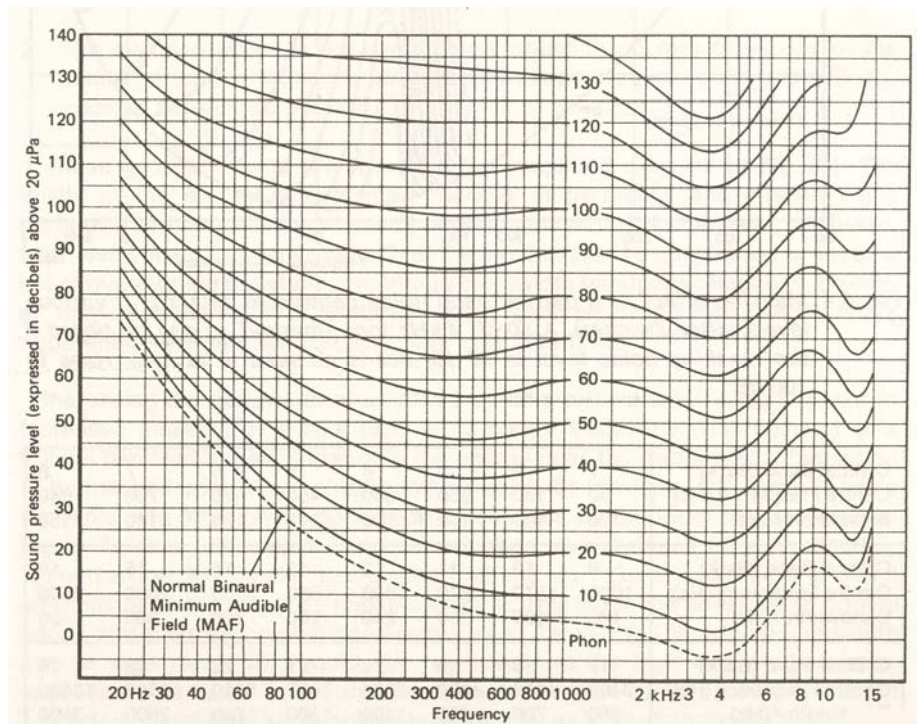
Equal Loudness Contour represents the curve that the human being feels equal loudness of sound

as frequency at specific sound pressure level(SPL). Sometimes, Phon unit is used instead of SPL. Phon is same to the SPL at 1kHz. For example, the curve passing on 80dB at 1kHz is called '80phon curve'. The definition of weighting functions is listed in <Table V-4-1>.

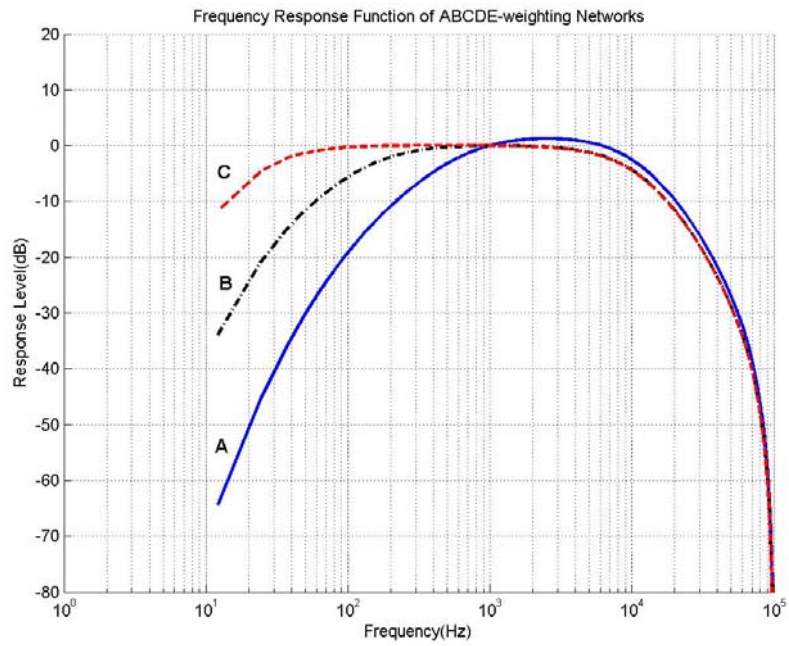
<Table V-4-1> Various Weighting Filter

Weighting Term	Curve Definition	Remarks
A-weight	Reciprocal of 40phon Curve	Useful in Noise Analysis
B-weight	Reciprocal of 70phon Curve	-
C-weight	Reciprocal of 100phon Curve	-
D-weight	-	Useful in Airplane Noise Analysis
E-weight	-	-

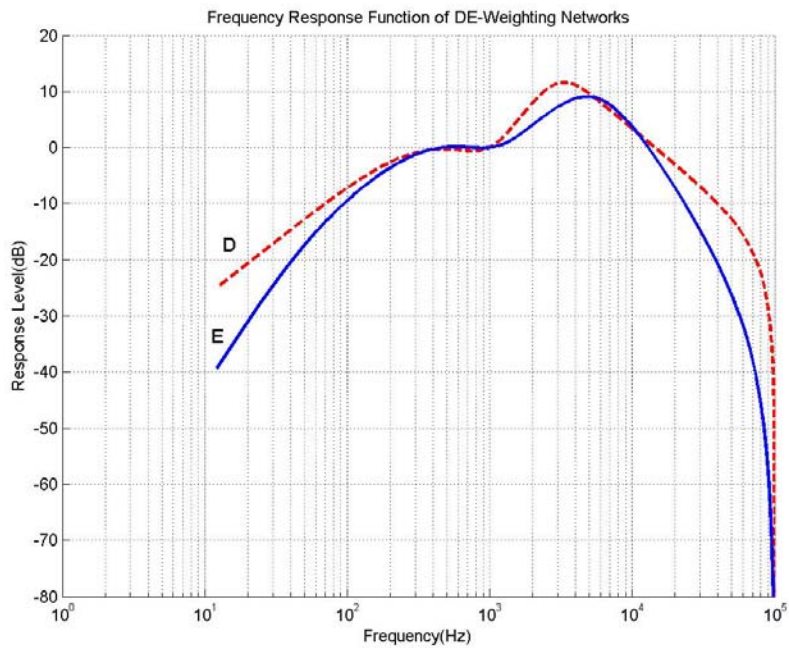
Normal Equal Loudness Contour is shown in [Fig. V-4-1], the frequency responses of each weighting function are shown in [Fig. V-4-3].



[Fig. V-4-1] Normal Equal Loudness Contours



[Fig. V-4-2] A, B, C-Weighting Frequency Response

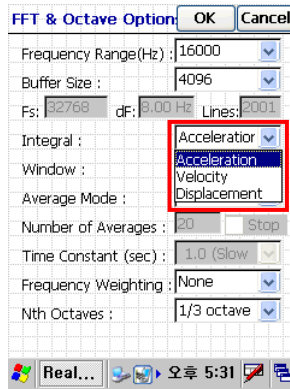


[Fig. V-4-3] D, E-Weighting Frequency Response

## 5. Integration in Frequency Domain


Velocity and displacement spectrum can be analyzed in FFT and Octave, Spectrogram analysis when accelerator is used. For this, you select [Integral] option in [Fig. V-5-1]. The detailed theory of integral in frequency domain sees [VI. 3. 2] verse.

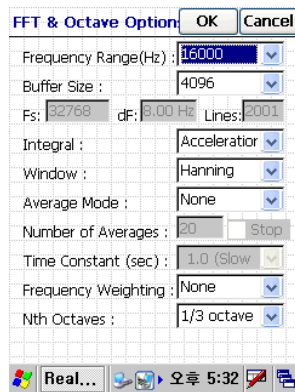




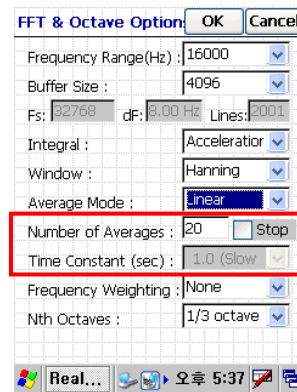
[Fig. V-5-1] Integration in Frequency Domain

## 6. Setting the Options of Measurement

Setting the options of measurement of frequency analysis, spectrogram and octave analysis is defined in the same option's window. If you press 'FFT & Octave option' button , the window of [Fig. V-6-1] appears.



[Fig. V-6-1] Setting the Option for FFT Analysis



[Fig. V-6-2] Inputting Averaging Number or Time Constant

Frequency range means the useful frequency and is same to frequency span as mentioned. The maximum of frequency range is 16 kHz. Sampling frequency  $F_s$  is dependent on the frequency span and maximum value is 32768Hz. You have to also choose the buffer size which has the max 4096. A frequency resolution( $dF$ ) is dependent on the relationship between sampling frequency and buffer size. The number of lines is dependent on the buffer size.

The relationships of parameters are listed in <Table V-2-1>.


Window function is supported with rectangular, hanning and flat-top window. Average mode is offered with NONE, linear and exponential averaging, peak hold averaging. If you select the linear or peak hold averaging, the box for number of average is activated. The system with [ Stop] will stop measuring automatically when the numbers of average reaches the input number. If you set [Use Timer] in [File>>Recording Options...] and [ Stop] simultaneously, the system will stop at short time.

Selecting the exponential averaging have been activated the combo box of time constant. You can choose the proper time constant and this function doesn't offer the automatic stopping with [☑ Stop].

Frequency weighting is used to analyze the noise with A, B, C, D, W weighting filter. A-weighting is an ordinary weighting in noise analysis. The combo box of [Nth Octaves] has 1/1 octave, 1/3 octave and 1/6 octave.

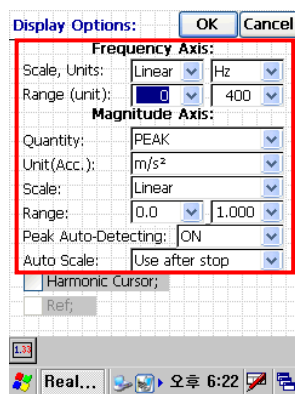
FFT based octave analysis is dependent on the options of FFT analysis. So you have to set the option of FFT properly. For example, if the linear averaging with 20 iterations is set, octave analysis is calculated from linear averaged spectrum with 20.

## 7. Setting the option of display

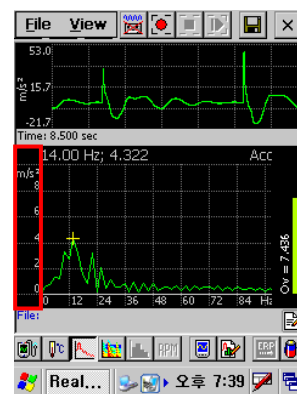
Pressing the display option button  generates the window of [Fig. V-7-1]. The adjusting frequency and magnitude axis is used in frequency analysis, spectrogram and octave analysis. The range of frequency axis defines the start frequency and end frequency in the FFT graph. And the scale of frequency axis is defined linear axis or logarithms axis. Frequency range is editable in linear scale, but it is impossible in logarithms scale. The units in magnitude axis is displayed with RMS( $m/s^2$  or  $Pa$ ), Peak( $m/s^2$  or  $Pa$ ), Power( $m^2/s^4$  or  $Pa^2$ ) and scale is displayed with linear / logarithms / decibel scale.

For example, if you want to see the decibel of RMS value, you have to set that the unit is RMS and the scale is dB.

There is another way for adjusting magnitude axis. If user clicks the Y-Axis by stylus pen, the range of Y-Axis is automatically adjusted.



[Fig. V-7-1] Option Window Display

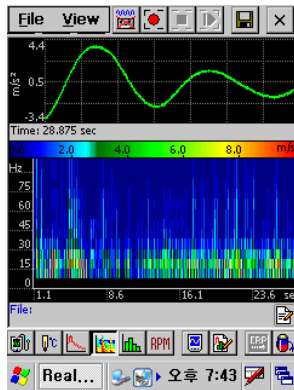


[Fig. V-7-2] Auto-scaling of Magnitude Axis

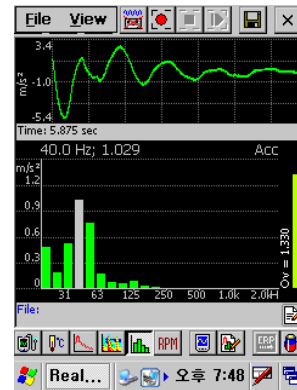
Cross marker represents the maximum peak of spectrum with frequency and magnitude after [Peak Auto-Detecting] is turned on. And if you press any peak of FFT graph for 1second, big cross marker appears with current frequency and magnitude instead of the value of peak auto-detecting. Pressing the screen for a short time turned off that function. The rightmost bar graph is overall level.

[Auto Scale] and [Peak Auto-Detecting] function is not offered and only the adjustment of frequency is available in the spectrogram graph.

In octave analysis, [Auto Scale] function is offered but [Peak Auto-Detecting] function is not offered. However, the bandwidth being pressed for 1 second is changed from green to white bar, and then you can read frequency and magnitude in the leftmost top of octave graph. Refer to [Fig. V-7-4].









[Fig. V-7-3] Spectrogram Graph



[Fig. V-7-4] Octave Graph




## 8. Record and Playback

Press the ARM button  to see an analyzed data in real-time without recording. And press the stop button  to stop arming. If you want to hear the sound in real-time, you have to press hearing button  and arming button  simultaneously.

Press the record button  to record and analyze the data. After recording is finished, wave file is loaded and playback button  is activated. If you press playback button, playing the wave file and analyzing the data executes at the same time.

To export a measured data, [Export] menu in [File>>Export...] tap should be selected. Refer to the chapter III-4 to get the details.

## 9. Mode Conversion of FFT Analysis, Spectrogram and Octave Analysis

Press FFT button  for frequency analysis, and press the spectrogram button  for spectrogram and press octave button  for octave analysis. Recording, playback and hearing function is same to vibration level meter mode.

## VI. FFT Based RPM Meter

### 1. Understanding about the RPM Measurement

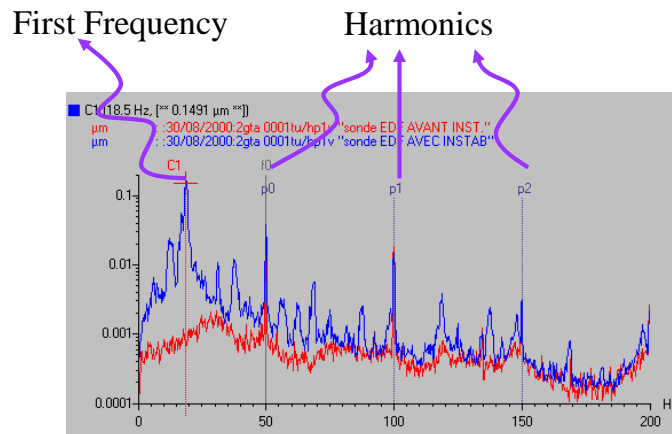
The RPM is usually measured using a tachometer. Tachometer counts the revolution of a rotator using optical sensor and reflector or using the photo coupler and the hole of rotator or magnetic sensor and the teeth of rotator. Anyway, all of this method contacts a rotator directly. This is good way to get an exact revolution. On the other hand, it is a defeat. To solve the defeat, we have devised the FFT based RPM meter.

Spectrum in a rotational vibration has a peak which corresponds to RPM and a lot of peaks against to rotor/teeth/bearing. But a first peak generates by an input rotation. Therefore, this peak is called '1st order'. The peaks which is an integer times of the frequency of a first peak are called harmonic components.

The relationship of frequency and RPM represents in <Eq. VI-1-1>.

$$Freq = \frac{RPM}{60} \quad \text{<Eq. VI-1-1>}$$

where RPM is the abbreviation of a revolution per minute. Therefore, we can calculate RPM from the first frequency of FFT spectrum.



[Fig. VI-1-1] Fundamental Frequency and Harmonic Frequency

### 2. Frequency Analysis

Frequency analysis is done with the listed option below and user can't change the option.



- Hanning Window
- Linear Averaging : this parameter is defined at [Average Number] in option setting window.
- Buffer size for FFT is variable as the maximum RPM. Refer to <Table VI-2-1>

<Table VI-2-1> Relationship of Maximum RPM, Sampling Frequency and Buffer Size

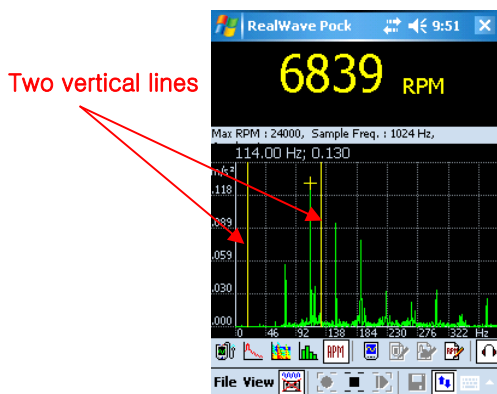
Max. RPM	Sampling Frequency(Hz)	Buffer Size
12000	512	1024
24000	1024	2048
48000	2048	4096
96000	4096	4096

As mentioned in V-2-4 paragraph, a frequency analysis generates a picket fence error. We can't get an exact frequency which corresponds to rotational RPM. As this problem, the system has a compensation algorithm.

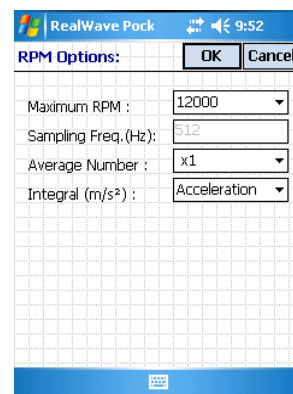
### 3. Measuring with RPM Meter

Button  changes another mode into RPM mode and, at the same time, RPM option setting button  is activated. This mode doesn't offer the recording except arming.

RPM Meter has the RPM window and FFT graph window as [Fig. VI-3-1]. FFT graph window has two vertical lines for setting range of a first frequency. To calculate RPM from the maximum peak in a defined region by two vertical lines, you have to drag lines with stylus pen.



[Fig. VI-3-1] RPM Meter



[Fig. VI-3-2] Option Setting Window

The option setting window is shown in [Fig. VI-3-2] and maximum rpm and averaging number shall be selected. Sampling frequency is calculated from maximum RPM as <Eq. VI-3-1> equation.

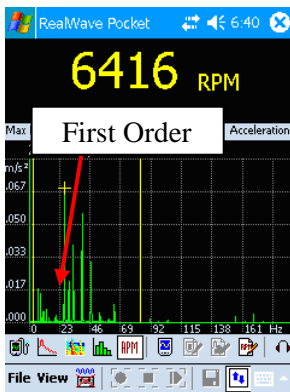
$$F_s = \frac{Max. RPM}{60} \times 2.56 \quad \text{<Eq. VI-3-1>}$$

Cross mark '+' represents the maximum value in all frequency with no correlation of vertical lines. If you set an unfit range, you would get a fault RPM from a fault frequency.

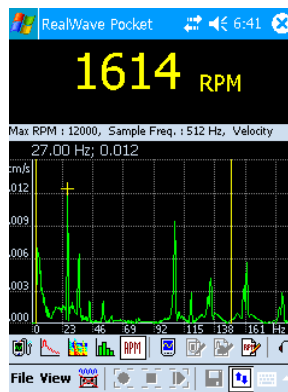
### 3.1 Using the Velocity Graph or Displacement Graph to get a fit RPM

It is difficult that you find the fundamental frequency in the harmonic frequency of spectrum of acceleration. The spectrum of rotational vibration with the fluctuation from 1610PRM to 1660RPM is shown from [Fig. VI-3-3] to [Fig. VI-3-5]. As shown in [Fig. VI-3-3], you can't find the first order(or fundamental frequency) easily and harmonic frequencies are so big that the calculated RPM is not correct. If you find in the spectrum of a velocity or displacement, you can find it easily. But in the integrating process, the noise of a low frequency is amplified. Consequently, though you see a first order, you can be mixed between noise and first frequency. To solve this problem, the highpass filter with 4Hz cutoff frequency is used. In other words, the frequency less than 4Hz has 0 magnitudes.

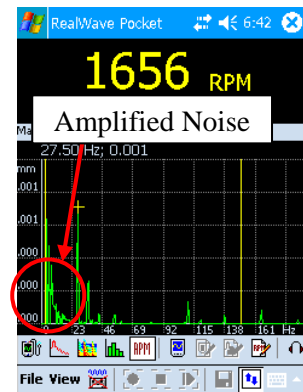
To get the graph of acceleration/velocity/displacement, you can set it in [Integral] of [Fig. VI-3-2]. A magnitude has RMS, units is fixed with  $m/s^2$ ,  $cm/s$  and  $mm$ .



[Fig. VI-3-3] Acceleration



[Fig. VI-3-4] Velocity



[Fig. VI-3-5] Displacement

### 3.2 Integral in the spectrum of FFT

The relationships of acceleration, velocity and displacement are listed below equations:

$$velocity = \frac{Acceleration}{2\pi f} \quad \text{<Eq. VI-3-2>}$$

$$Displacement = \frac{Acceleration}{(2\pi f)^2} \quad \text{<Eq. VI-3-3>}$$

where  $f$  is frequency. These equations represent the relationship of magnitude except phase. We can rewrite these equations with frequency in the spectrum as below.

$$FFT_{vel}(i) = \frac{FFT_{Acc}(i)}{2\pi f_i} \quad \text{<Eq. VI-3-4>}$$

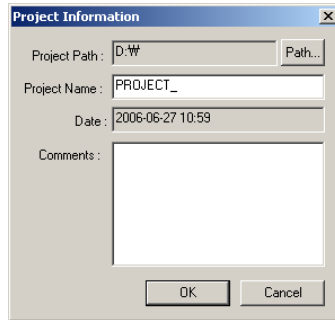
$$FFT_{Disp}(i) = \frac{FFT_{Acc}(i)}{(2\pi f_i)^2}, \quad 0 \leq i \leq \frac{N}{2} \quad \text{<Eq. VI-3-5>}$$

where  $FFT(i)$  is a magnitude at each frequency.  $f_i$  is frequency.  $i$  is the index of frequency and  $N$  is a buffer size.

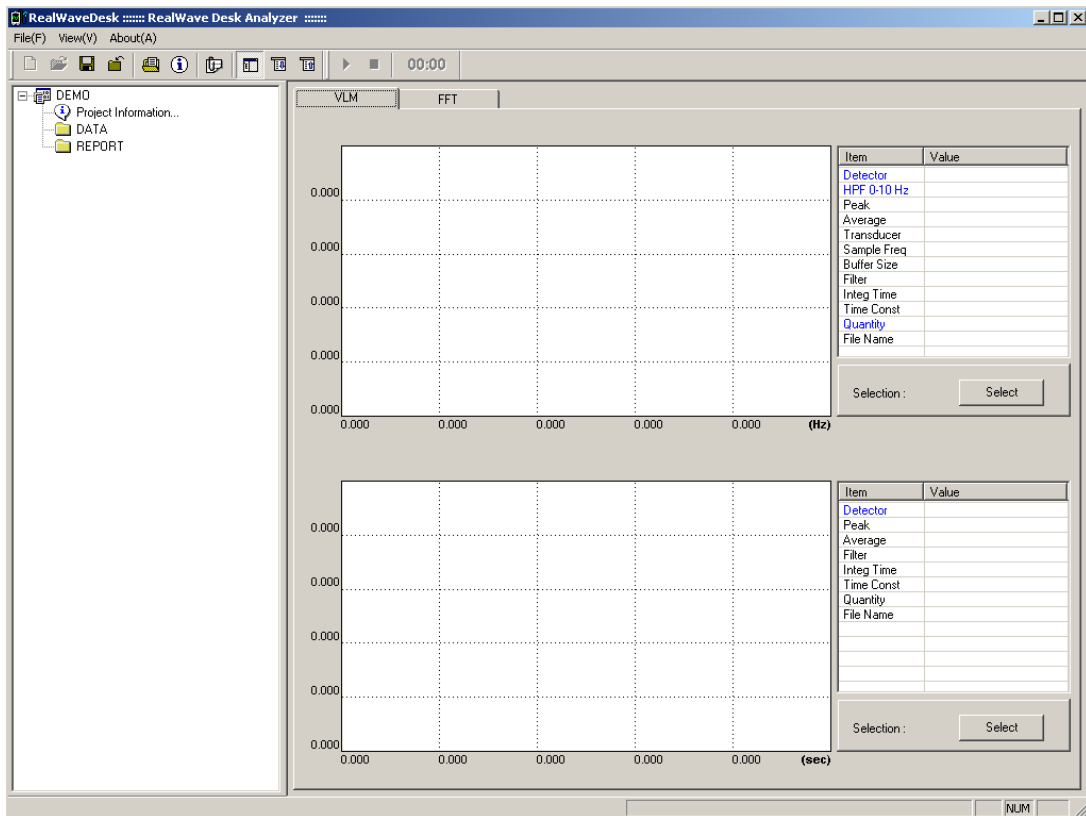
## VII. The way to use RealWave Desk for PC

### 1. Create a Project / Open

When user creates a new project of file menu, project information dialog is opened. Choose the route of the project which has to be installed, enter the project name and just put the OK button [Fig. VII-1-1]. After then a new project like [Fig. VII-1-2] will be created.

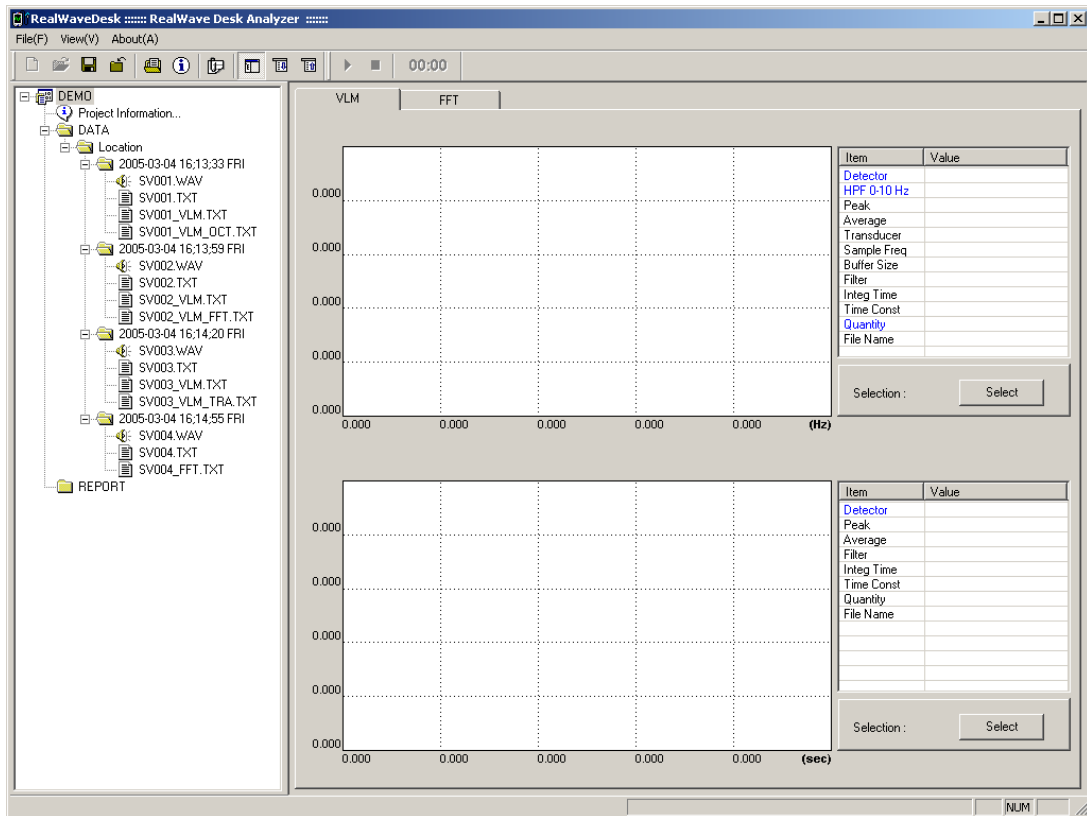


[Fig. VII-1-1] Enter the information of Project




[Fig. VII-1-2] Program

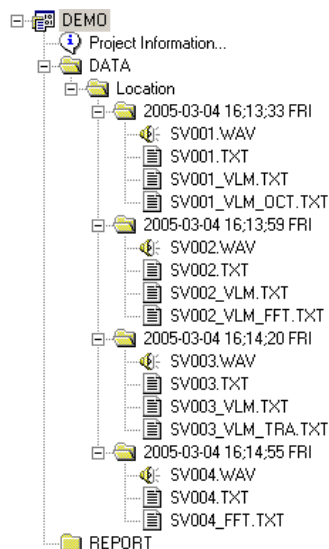
When user opens a project of file menu and choose a project file (\*.PRJ), the chosen project is loading like [Fig. VII-1-3].



[Fig. VII-1-3] Project loading

## 2. Adding a file / Delete

Click a file additional icon  on the toolbar or after choose data folder of the tree control, then click the mouse on the right button when user add a wave file of file menu and choose the wave file which has to be added at the Project, the wave file and the result file of analysis are added like [Fig. VII-2-1]



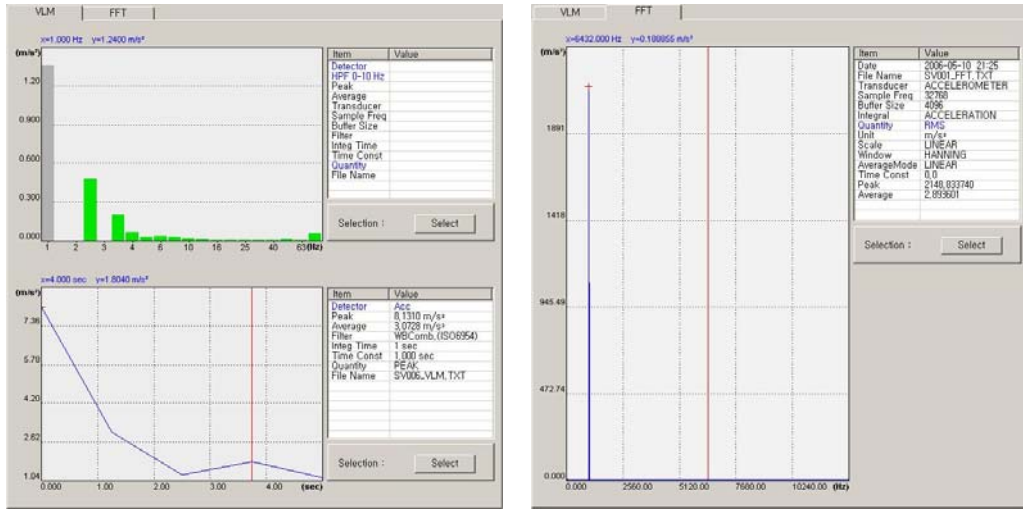
[Fig. VII-2-1] Project tree control

If user wants to the added file, user has to select the “Delete” menu in the menu window, which is displayed by clicking of mouse right button. All file of folder and sub-folder will be deleted.



### 3. Control of Graph

When double clicking the folder that a data from tree control, a graph is appeared on the right of the window like [Fig. VII-3-1].



[Fig. VII-3-1] VLM / FFT Control of Graph

- VLM Trace : The Trace graph will be displayed on the bottom side in VLM. User can see the cursor value by clicking of mouse left button. Also user can select the specific range by clicking of mouse right button and dragging
- VLM Octave : The Octave graph will be displayed on the upper side in VLM .User can see the Octave graph corresponding to the cursor position in automatically trace graph.
- VLM FFT : The FFT graph will be displayed on the upper side in VLM peak position is detected in the FFT graph. Also user can select the specific range by clicking of mouse right button and dragging.
- FFT : User can display the cursor value by clicking of mouse left button, also user can select the specific range as same way in VLM FFT.
- Selection Button : After user select the specific range in the graph control. The peak & average values will be calculated and displayed by locking the “Select” button.

### 4. The Control of Property list

The property of measured data which is important from saved file is displayed in list control like [Fig. VII-4-1]. Combo box will be displayed by double click of blue color character in the list, and then user can change the parameters, displayed graph will be changed as defined parameters.

Item	Value	Item	Value
Detector	Acc	Detector	Acc
Peak	14,3640 m/s <sup>2</sup>	HPF 0-10 Hz	None
Average	4,9372 m/s <sup>2</sup>	Peak	0,680210 m/s <sup>2</sup>
Filter	WBComb.(ISO6954)	Average	0,023411 m/s <sup>2</sup>
Integ Time	1 sec	Transducer	ACCELEROMETER
Time Const	1,000 sec	Sample Freq	2048 Hz
Quantity	PEAK	Buffer Size	2048
File Name	SV005_VLM.TXT	Filter	WBComb.(ISO6954)
		Integ Time	1 sec
		Time Const	1,000 sec
		Quantity	PEAK
		File Name	SV005_VLM_FFT.TXT

[Fig. VII-4-1] The control of property list

## 5. Control of Sound

If user roading the data from project tree, sound control menu of toolbar is activated like [Fig. VII-5-1]. Sound play, stop, play time are displayed.



[Fig. VII-5-1] Control of Sound

## 6. List of Result Report

The measuring result data is displayed as a report form of ISO6954 like [Fig. VII-6-1] from current project file.

No.	Location	Direction	Acceleration	Acc. Unit	Velocity	Vel. Unit
1	Location	X-x-Axis	0.0910	m/s <sup>2</sup>	2.5062	mm/s
2	Location	X-x-Axis	0.0976	m/s <sup>2</sup>	2.5376	mm/s
3	Location	X-x-Axis	0.0796	m/s <sup>2</sup>	2.2887	mm/s

[Fig. VII-6-1] Reporting of Result

- New : Tap to load a result date from current project.

- Open : Tap to load a stored result list file.
- Save As : Tap to save a current result list file as another name.
- Delete : Tap to delete a date of selected list.
- Excel : Tap to change details of list as MS Office Excel file like [Fig. VII-6-2].

The screenshot shows a Microsoft Excel window titled 'Book1'. The spreadsheet contains a table with the following data:

Measurement Results				
No.	Transducer Location	Direction	Overall frequency-weighted r.m.s value	
			Acceleration	Velocity
1	Location	X-x-Axis	0.091 m/s <sup>2</sup>	2.5062 mm/s
2	Location	X-x-Axis	0.0976 m/s <sup>2</sup>	2.5376 mm/s
3	Location	X-x-Axis	0.0796 m/s <sup>2</sup>	2.2887 mm/s

[Fig. VII-6-2] Result Report with MS Office Excel

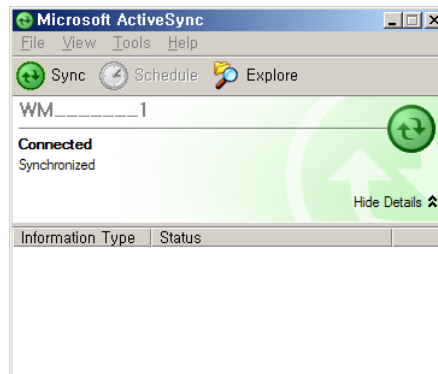
Measurement Results				
No.	Transducer Location	Direction	Overall frequency-weighted r.m.s value	
			Acceleration	Velocity
1	Location	X-x-Axis	0.091 m/s <sup>2</sup>	2.5062 mm/s
2	Location	X-x-Axis	0.0976 m/s <sup>2</sup>	2.5376 mm/s
3	Location	X-x-Axis	0.0796 m/s <sup>2</sup>	2.2887 mm/s

[Fig. VII-6-3] Printed Result at MS Office Excel

## VIII. Installation of ActiveSync and Communication with PDA

### 1. What is Microsoft ActiveSync?

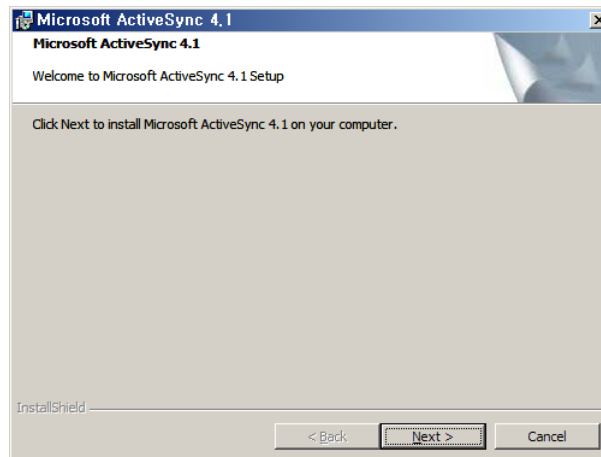
Microsoft ActiveSync is a program which synchronizes with PC and PDA. If PDA is connected with PC, many data is transported each other. If selecting the folder named Mobile Device, user can see the internal data of PDA. ActiveSync program must be installed at PC which will be used before connecting PDA at a USB port of PC.



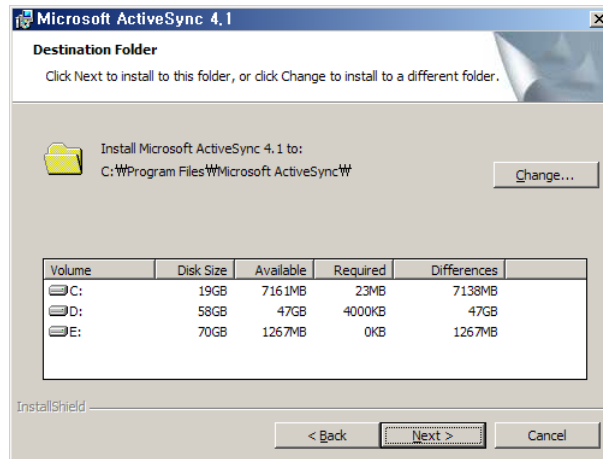
[Fig. VIII-1-1] ActiveSync Executed Window

### 2. Installation of ActiveSync

Run "MSASYNC.EXE" at the folder named "ActiveSync" in CD and follow from [Fig. VIII-2-1] to [Fig. VIII-2-9].



[Fig. VIII-2-1]



[Fig. VIII-2-2]

Up to here, installation of the program is completed. First, PC and PDA has to be connected each other using USB cable or cradle because connection of PDA will be carried out from now on.

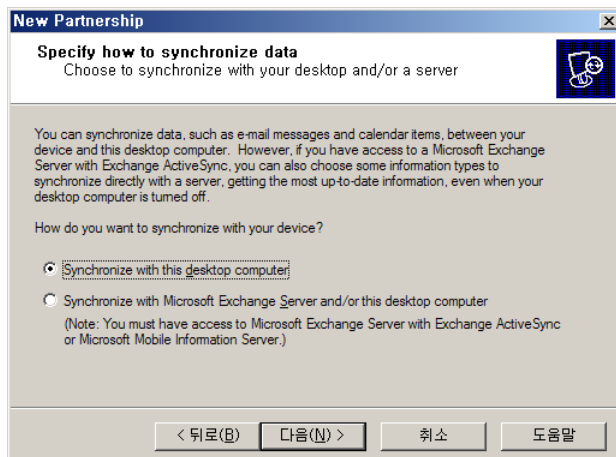


[Fig. VIII-2-3]

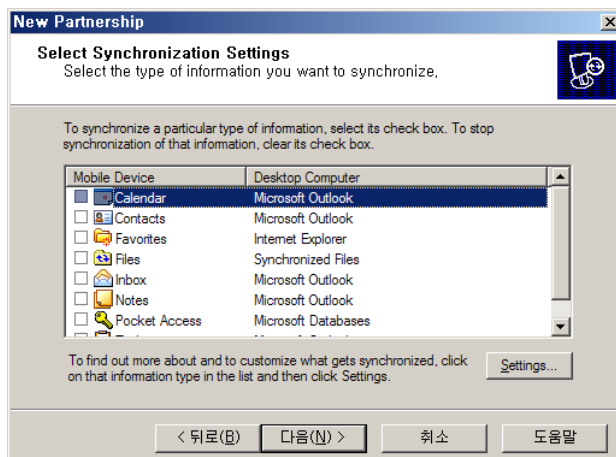
Press the NEXT button after connection as [Fig. VIII-2-3]. In the following, relationship of partner has to be established to use PDA at the office. When standard relationship of partner is set up, Sync item of data is selected and PDA is connected with PC, the data of PC is copied at PDA and the one of PDA is copied at PC, so the data of each device is identical. Generally, this is called to be Sync (Synchronization). The synchronized data is Microsoft Outlook data, E-mail and the files to be set up synchronization. If user just set up standard relationship of partner, it is useful because the connection is completed directly without any questions for connection.



[Fig. VIII-2-4]

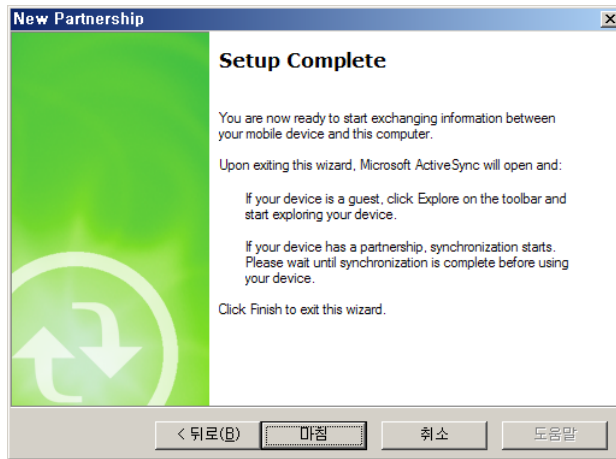


[Fig. VIII-2-5]

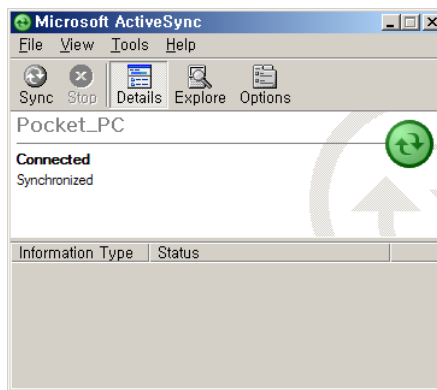


[Fig. VIII-2-6]

All check boxes should be unchecked like [Fig. VIII-2-6] and push the NEXT button because this step is for what the data of PC and PDA is transferred each other. And then, the connection comes to an end. Generally, it is not used for office but for measurement, so it is useful to delete the item of data synchronization.

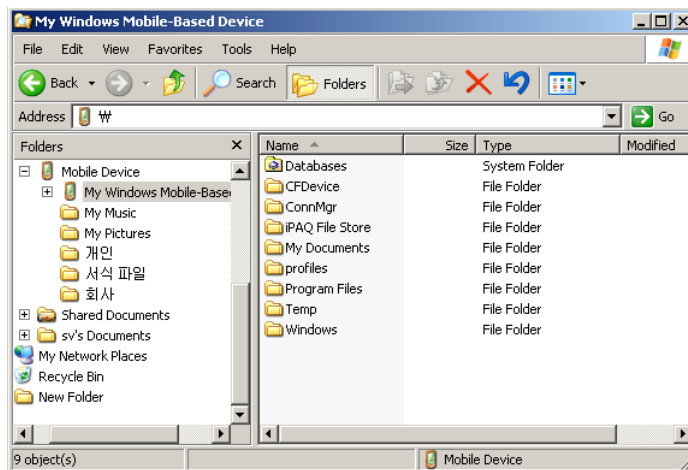


[Fig. VIII-2-7]



[Fig. VIII-2-8] The State of Connection

If the connection is completed, user can see and copy the entire data of PDA with the folder, [Mobile Device] like [Fig. VIII-2-9] at Window Explorer.



[Fig. VIII-2-9] PDA Folders at Window Explorer

[Appendix 1]

# The Theory for calculating RMS by ISO2631 and ISO8041



Annex D

D. 1 Nomenclature

$a$  : Instantaneous magnitude of signal

$a_0$  : Reference magnitude of signal( =  $1\mu m/s^2$  or  $1pW$  ) if power is measured

$\tau$  : Time constant for running averaging

$t$  : Time

$t_o$  : Time of observation

$t_1$  : Start time of integration

$t_2$  : Stop time of integration

$e$  or  $\exp$  : Exponential Function

$T(=t_2 - t_1)$  : Measurement period for linear averaging

D.2 Integration

The equivalent continuous vibration value or the equivalent continuous vibration level are based on normal start-stop integration (linear integration) of the squared signal. The time weighting characterizes a running integration of the squared signal. The result is the running r.m.s value.

$$a_{rms,\tau}(t_o) = \left[ \frac{1}{\tau} \int_0^{t_o} a^2(t) e^{-\frac{t-t_o}{\tau}} dt \right]^{\frac{1}{2}}$$

It is a function of the instant time  $t_o$ . All recent parts of the signal  $a(t)$  contribute to the value.

They are weighted by a factor decreasing exponentially with the time distance from  $t_o$ .

The corresponding time weighted vibration level, in decibels, is given by

$$L_{rms,\tau}(t_o) = 10 \log \left[ \frac{1}{\tau} \int_0^{t_o} \frac{a^2(t)}{a_o^2} e^{-\frac{t-t_o}{\tau}} dt \right] dB$$

NOTE – These equations describe the behavior of a hot-wire ampere meter and can be realized by relatively simple means or digital algorithms. The running r.m.s value responds to fluctuations of the signal as slowly as the constant allows. A considerable reduction of data is achieved without losing essential information about changes of amplitude or level.

The equivalent continuous vibration value or level can also be derived from the running level by true integration or by summing over discrete samples,  $L_i$  being constant or nearly constant for a time interval  $t_i$  as follows:

$$a_{eq,T} = a_0 \left[ \sum_{i=1}^n \frac{t_i}{T} 10^{0.1L_i} \right]^{\frac{1}{2}}$$

where

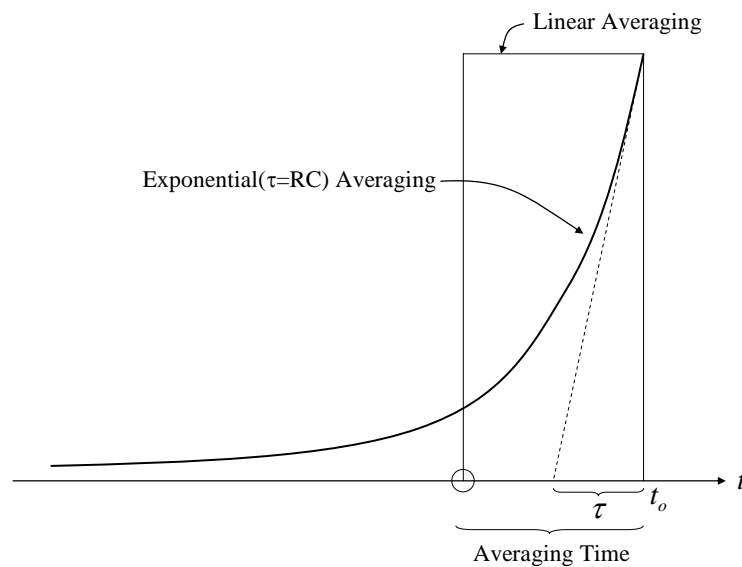
$$T = \sum_{i=1}^n t_i$$

From a statistical distribution giving the percentages  $p_i$  of the total measuring time  $T$ , during which the level is within an interval (e.g. 1dB) around a central value  $L_i$ , the equivalent value is given by

$$a_{eq,T} = a_0 \left[ \sum_{i=1}^n \frac{p_i}{100\%} 10^{0.1L_i} \right]^{\frac{1}{2}}$$

where

$$\sum_{i=1}^n p_i = 100\%$$



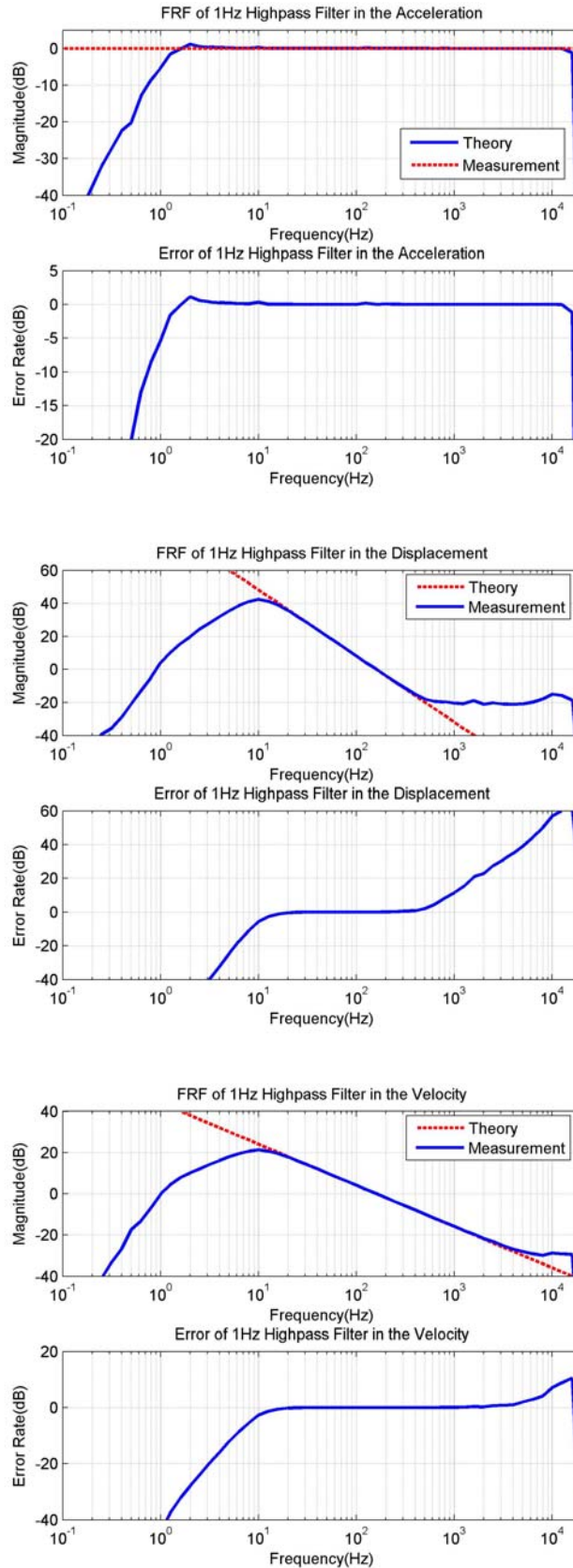
[Figure] Weighting curves for linear and exponential integration

[Appendix 2]

# Frequency Response of Filter in the Vibration Level Meter

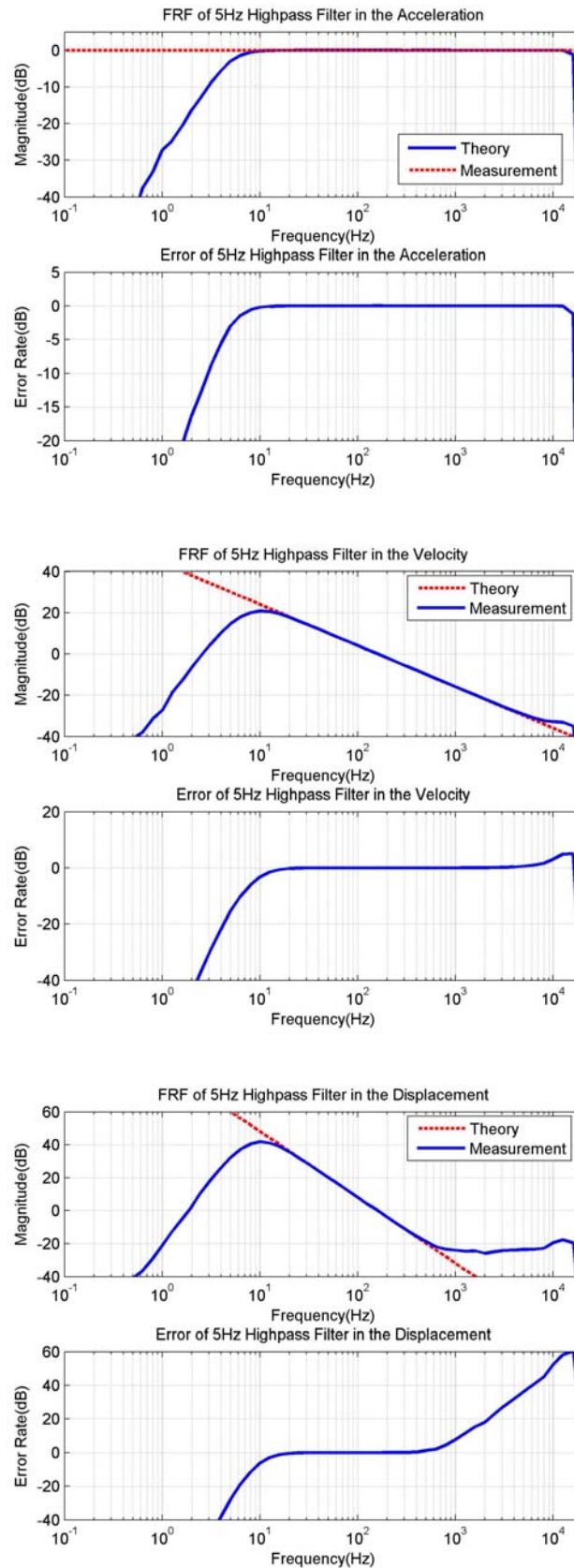
# 1. 1Hz Highpass Filter

A 2-pole Butterworth IIR Filter was used for this filter and integrator has highpass filter with 10Hz Cutoff frequency.



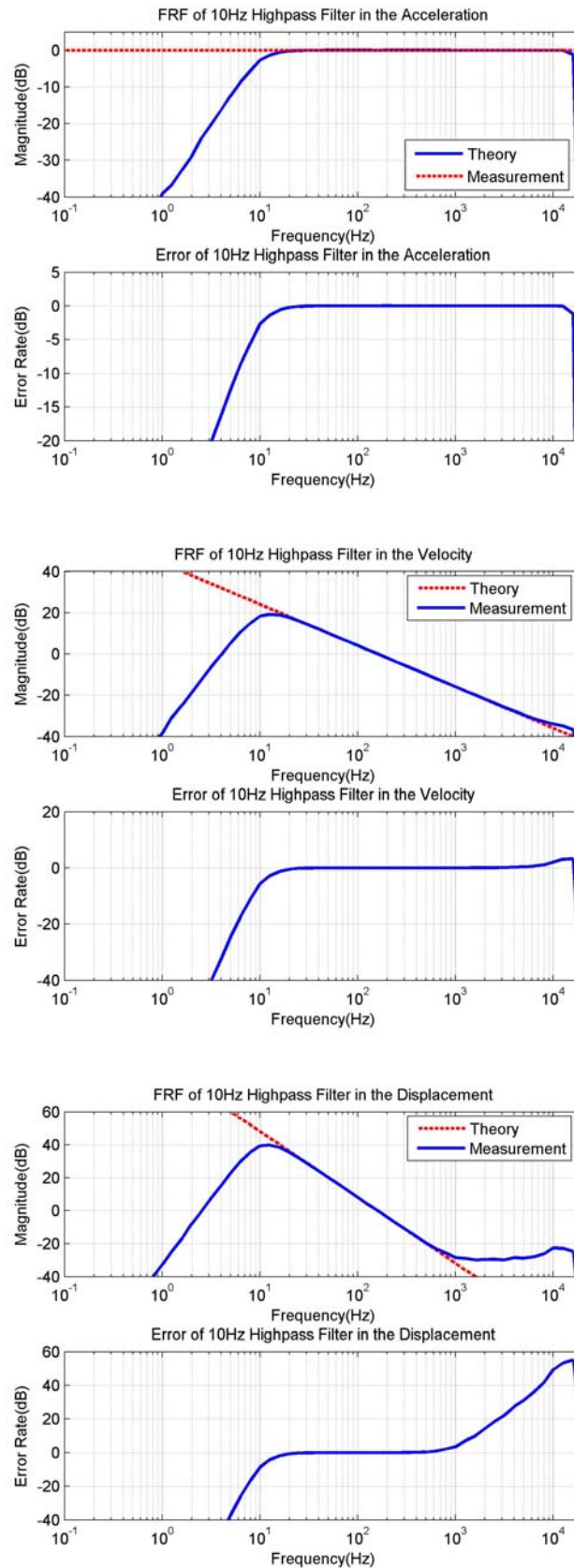
## 2. 5Hz Highpass Filter

A 2-pole Butterworth IIR Filter was used for this filter and integrator has highpass filter with 10Hz Cutoff frequency.



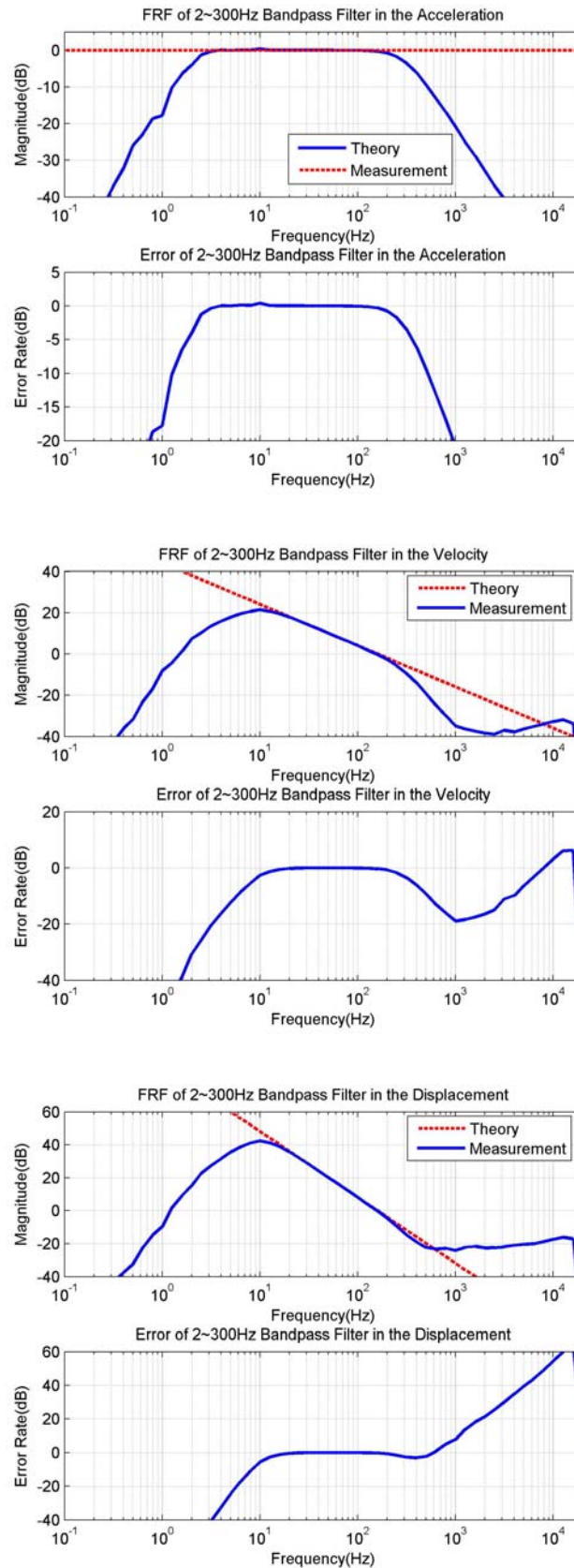
### 3. 10Hz Highpass Filter

A 2-pole Butterworth IIR Filter was used for this filter and integrator has highpass filter with 10Hz Cutoff frequency.



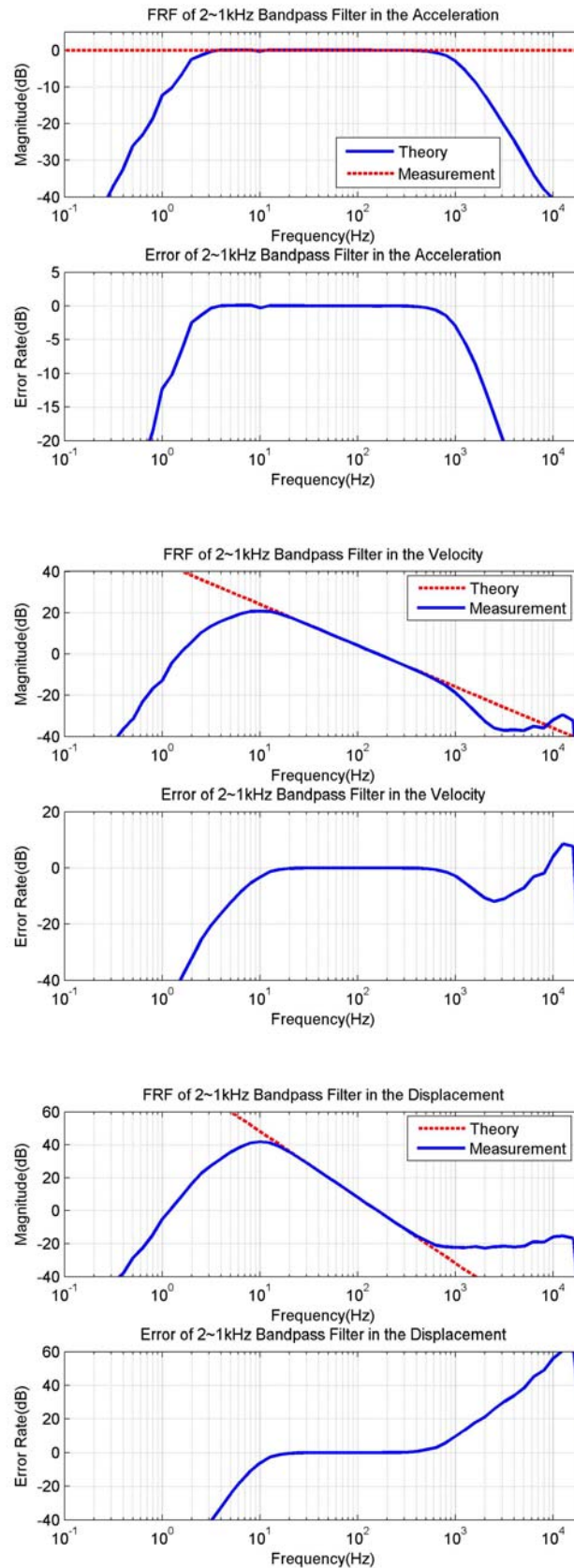
## 4. 2Hz~300Hz Bandpass Filter

A 4-pole Butterworth IIR Filter was used for this filter and integrator has highpass filter with 10Hz Cutoff frequency.



## 5. 2Hz~1kHz Bandpass Filter

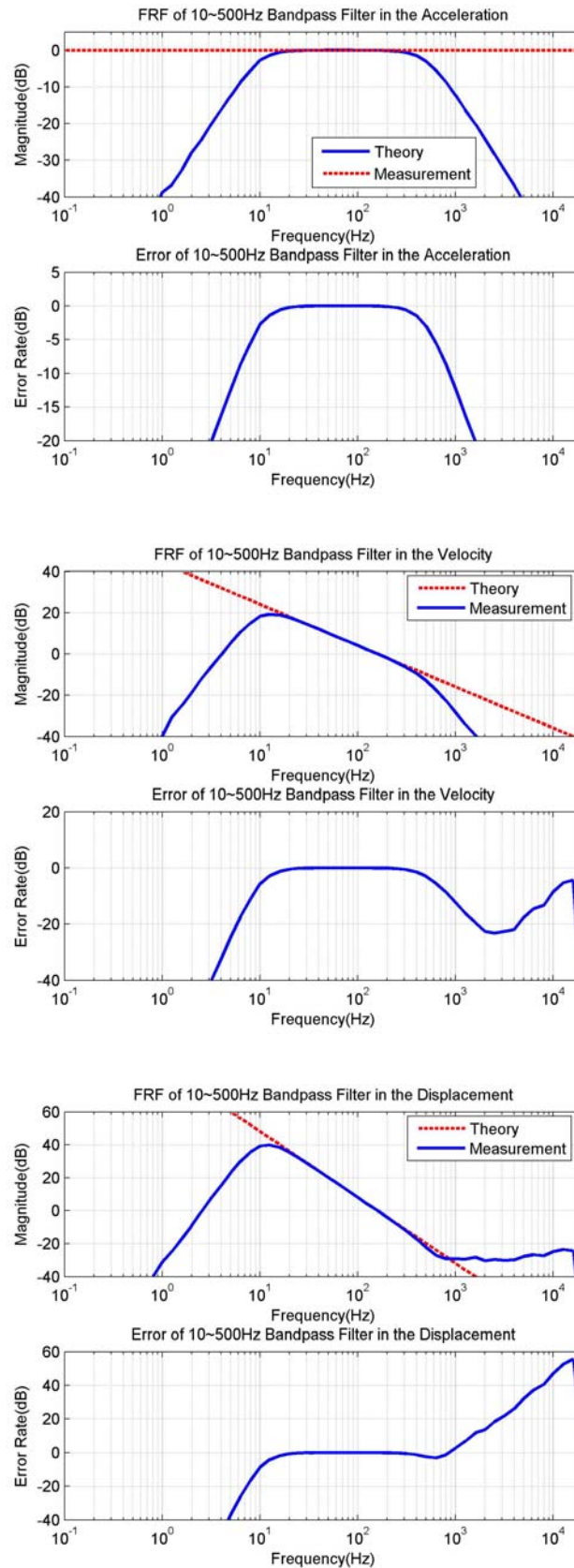
A 4-pole Butterworth IIR Filter was used for this filter and integrator has highpass filter with 10Hz Cutoff frequency.





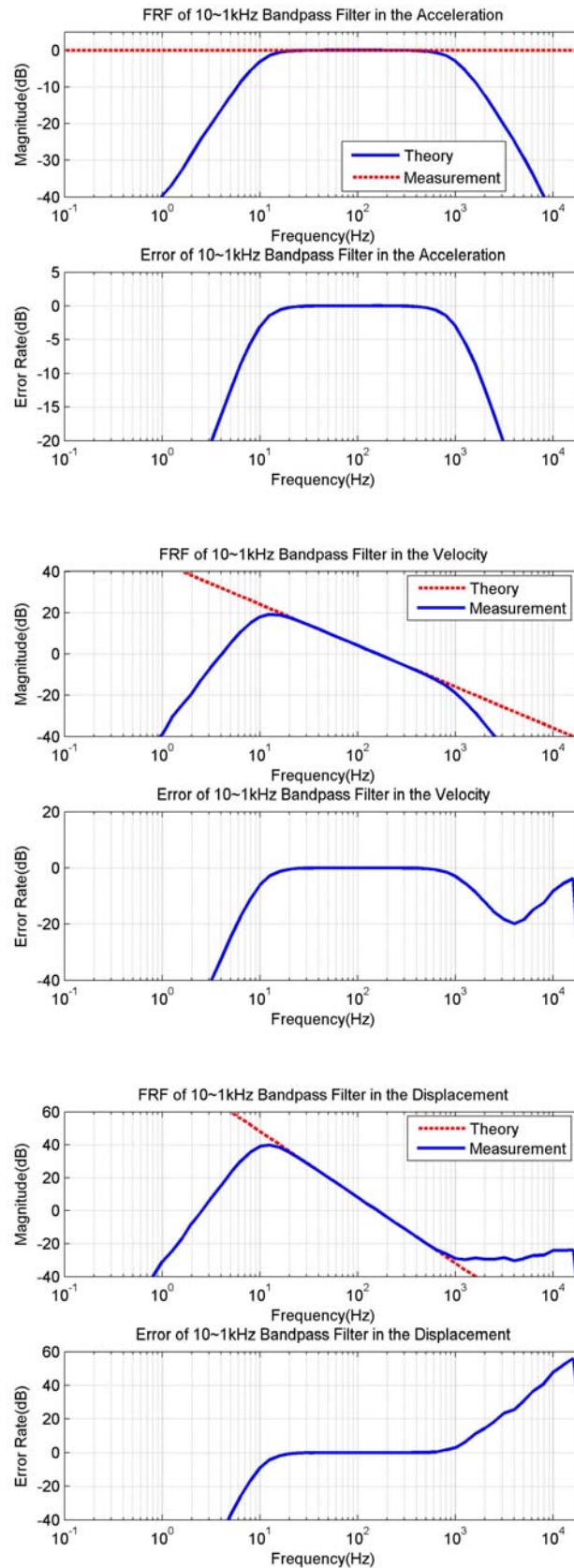
## 6. 10Hz~500Hz Bandpass Filter

A 4-pole Butterworth IIR Filter was used for this filter and integrator has highpass filter with 10Hz Cutoff frequency.



## 7. 10Hz~1kHz Bandpass Filter

A 4-pole Butterworth IIR Filter was used for this filter and integrator has highpass filter with 10Hz Cutoff frequency.



# Theory of WBCombined Filter and Frequency Response (ISO6954 Filter)

## 1. Theory

<Table 1. Parameters of the transfer functions of frequency weightings >

Weighting	Band-limiting		Acceleration-velocity transition			Upward step			
	$f_1$ Hz	$f_2$ Hz	$f_3$ Hz	$f_4$ Hz	$Q_4$	$f_5$ Hz	$Q_5$	$f_6$ Hz	$Q_6$
$W_k$	0.4	100	12.5	12.5	0.63	2.37	0.91	3.35	0.91
$W_d$	0.4	100	2.0	2.0	0.63	$\infty$	-	$\infty$	-
$W_f$	0.08	0.63	$\infty$	0.25	0.86	0.0625	0.80	0.1	0.80
WBCombined	0.7943	100	-	5.684	-	$\infty$	-	$\infty$	-
$W_c$	0.4	100	8.0	8.0	0.63	$\infty$	-	$\infty$	-
$W_e$	0.4	100	1.0	1.0	0.63	$\infty$	-	$\infty$	-
$W_j$	0.4	100	$\infty$	$\infty$	-	3.75	0.91	5.32	0.91
$W_h$	6.310	1258.9	15.915	15.915	0.64	$\infty$	-	$\infty$	-
$W_b$	0.4	100	16	16	0.55	2.5	0.9	4.0	0.95

The frequencies  $f_1$  to  $f_6$  and the resonant quality factors  $Q_4$  to  $Q_6$  are parameters of the transfer functions determining the overall frequency weighting, referred to acceleration as input quantity. The transfer function is expressed as a product of several factors.

a) Band-limiting(Two-pole filter with Butterworth characteristics,  $Q_1 = Q_2 = 1/\sqrt{2}$  )

High-pass :

$$H_h(s) = \frac{1}{1 + \frac{\sqrt{2}\omega_1}{s} + \frac{\omega_1^2}{s^2}} = \frac{s^2}{\omega_1^2 + \sqrt{2}\omega_1 s + s^2}$$

Low-pass :

$$H_l(s) = \frac{1}{1 + \frac{\sqrt{2}}{\omega_2} s + \frac{1}{\omega_2^2} s^2} = \frac{\omega_2^2}{\omega_2^2 + \sqrt{2}\omega_2 s + s^2}$$

b) a-v transition(proportionality to acceleration at lower frequencies, proportionality to velocity at higher frequencies)

$$H_t(s) = \frac{1 + \frac{1}{\omega_3} s}{1 + \frac{1}{Q_4 \omega_4} s + \frac{1}{\omega_4^2} s^2}$$

For W.B.Combined this expression is replaced by

$$H_t(s) = \frac{1}{1 + \frac{1}{\omega_4} s} = \frac{\omega_4}{\omega_4 + s}$$

c) Upward step (steepness approximately 6dB per octave, proportionality to jerk)

$$H_s(s) = \left( \frac{\omega_5}{\omega_6} \right)^2 \cdot \frac{1 + \frac{1}{Q_5 \omega_5} s + \frac{1}{\omega_5^2} s^2}{1 + \frac{1}{Q_6 \omega_6} s + \frac{1}{\omega_6^2} s^2} = \frac{\omega_5^2 + \frac{\omega_5}{Q_5} s + s^2}{\omega_6^2 + \frac{\omega_6}{Q_6} s + s^2}$$

Where

$$\omega_n = 2\pi f_n \quad (n = 1 \text{ to } 6)$$

$$s = j\omega (\text{imaginary angular frequency}) = j2\pi f$$

The product  $H_h(s) \cdot H_l(s)$  represents the band-limiting transfer function. Its corner frequencies (intersection of asymptotes) are given by  $f_1$  and  $f_2$ .

The product  $H_t(s) \cdot H_s(s)$  represents the actual frequency weighting transfer function for a certain application.  $H_t(s) = 1$  for frequency weighting  $W_j$ ;  $H_s(s) = 1$  for frequency weighting  $W_c$ ,  $W_d$ ,  $W_e$ ,  $W_h$  and W.B.Combined. This is indicated by infinity frequencies and absence of quality factors in Table 1.

The total weighting function is

$$H(s) = H_h(s) \cdot H_l(s) \cdot H_t(s) \cdot H_s(s)$$

The weighting curves in annex C show the modulus (magnitude)  $|H|$  of  $H$  versus the frequency  $f$  in a double-logarithmic scale.

NOTE 1. If the equations are interpreted in the time domain, the imaginary angular frequency  $p$

leads directly to the differential operator  $dt$ , which may be approximated by  $\Delta t$  in a digital realization of the weighting if the sampling interval  $\Delta t$  is small enough.

NOTE 2. Frequency weighting  $W_b$  is considered to be the appropriate weighting curve for  $z$  direction comfort in rail vehicles.  $W_b$  is defined in BS6841:1987, and is equivalent to :

$$H(s) = 1.024 \cdot H_h(s) \cdot H_l(s) \cdot H_t(s) \cdot H_s(s)$$

Where the parameter values  $f_1$  to  $f_6$  and  $Q_4$  to  $Q_6$  for  $H_h(p)$ ,  $H_l(p)$ ,  $H_t(p)$  and  $H_s(p)$  are :

$$f_1 = 0.4\text{Hz}; f_2 = 100\text{Hz}; f_3 = f_4 = 16\text{Hz}; Q_4 = 0.55;$$

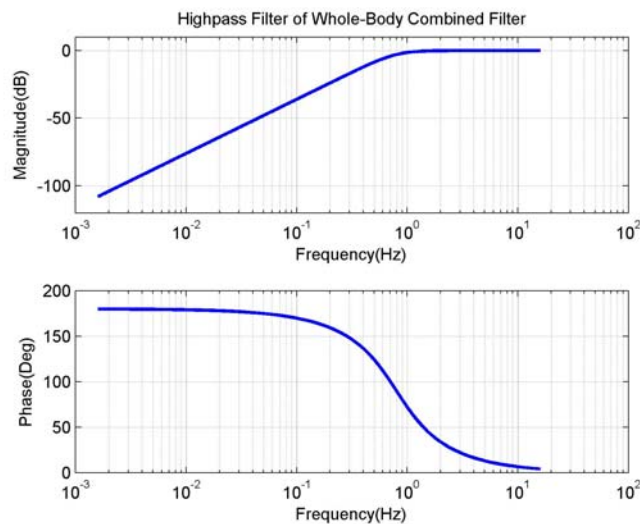
$$f_5 = 2.5\text{Hz}; Q_5 = 0.9; f_6 = 4.0\text{Hz}; Q_6 = 0.95$$

NOTE 3. Parameter values in [Table 1] define exactly the frequency weighting  $W_k$ ,  $W_d$ ,  $W_t$ ,  $W_c$ ,  $W_e$  and  $W_i$  according to ISO2631-1:1997.

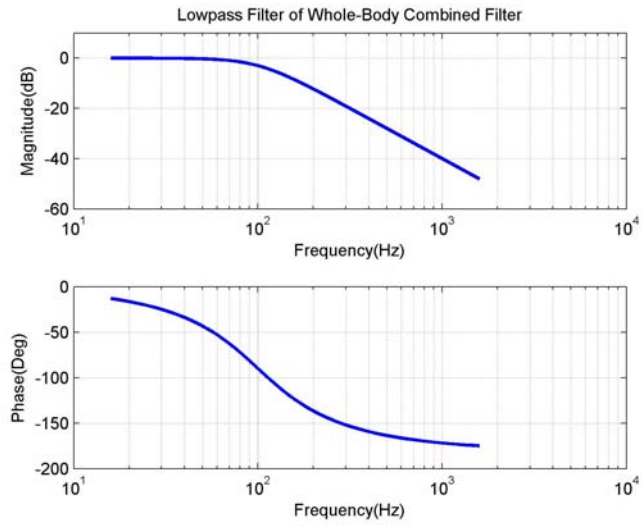
NOTE 4. Frequency weighting W.B.Combined and  $W_h$  ,(formerly H.A) are unchanged by this amendment. The parameter values in Table 1 are chosen to reproduce the weighting curves of ISO 8041:1990 using formulae of ISO2631:1997.

Above analog frequency response function is transferred to digital frequency response function by bilinear transformation method which is below :

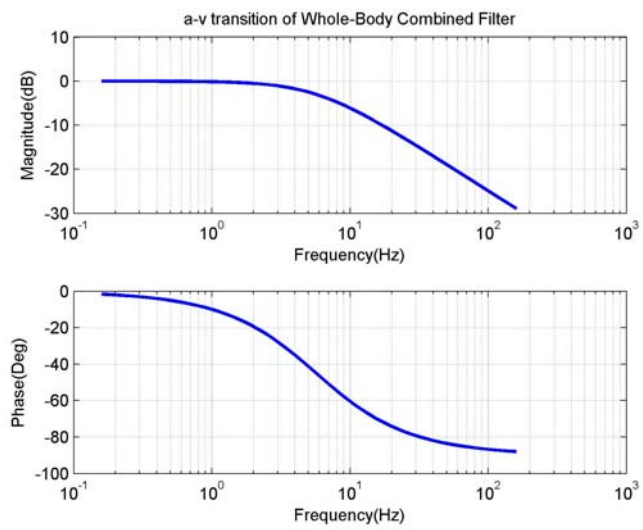
$$s = 2F_s \left( \frac{1 - z^{-1}}{1 + z^{-1}} \right)$$



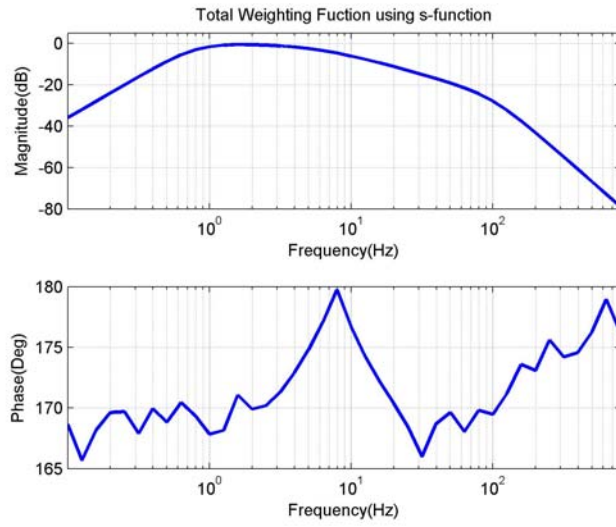
<Fig. 1 Highpass Filter with 0.7943Hz Cutoff>



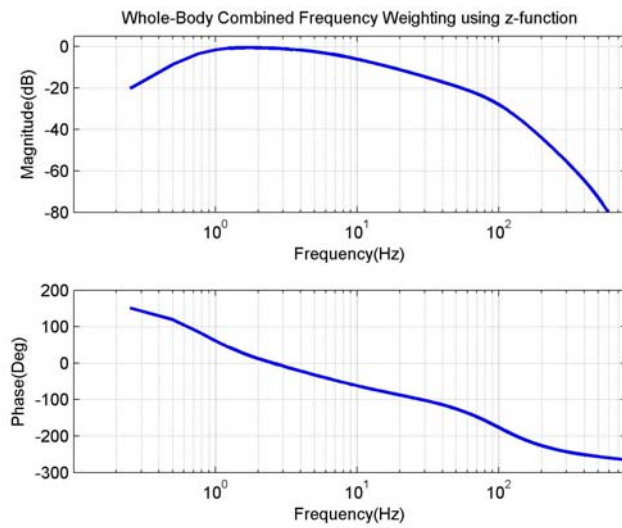
<Fig. 2 Lowpass Filter with 100Hz Cutoff>



<Fig. 3 a-v transition Frequency Response>

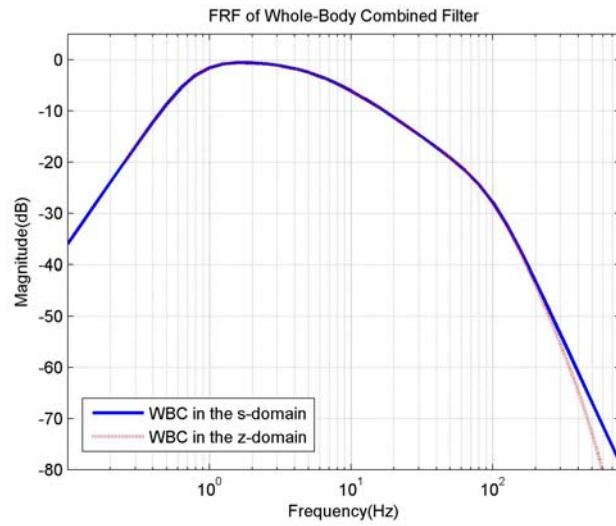


<Fig. 4 Frequency Response of Whole-body Combined Filter using s-function>

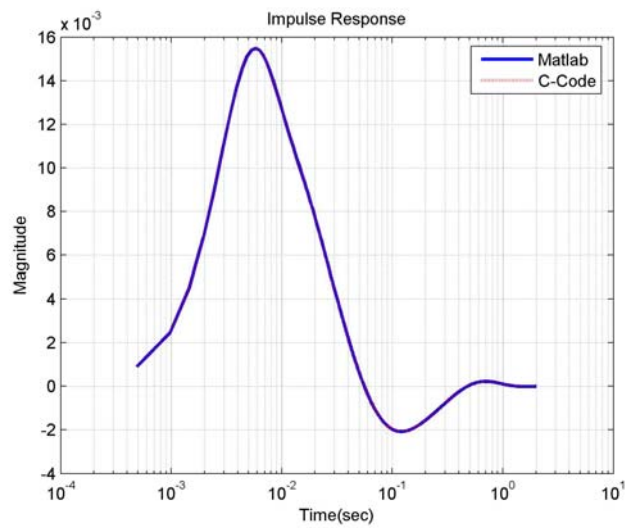


<Fig. 5 Frequency Response of Whole-body Combined Filter using z-function>





<Fig. 6 Comparison of Whole-body Combined Filter>



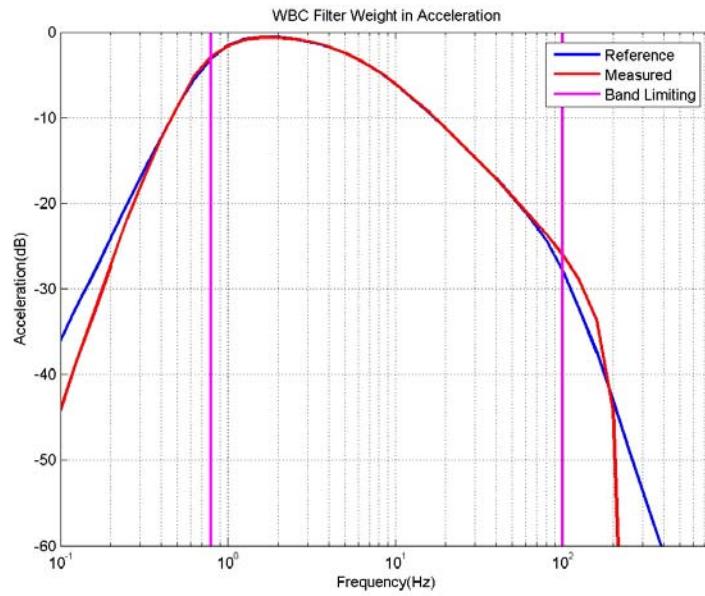
<Fig. 7 Impulse Response Function in the Matlab and C-code>

A simulated impulse response function is same to a theoretical impulse response function in C-code's algorithms as shown in [Fig. 7]. Where the sampling frequency for simulation is 2048Sample/sec, a buffer size is 4096.

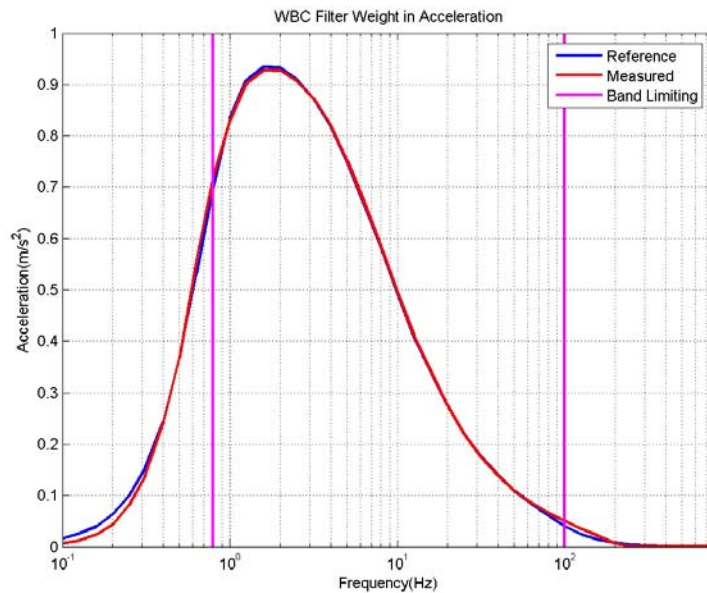
<Table 2. Error between Standard and Calculated Filter>

Frequency		WBCombined(dB)		Error(%)
Nominal	TRUE	Calculated	Standard	
0.100	0.1	-36.002	-36	0.005555
0.125	0.12589	-32.004	-32	0.012498
0.160	0.15849	-28.01	-28.01	0
0.200	0.19953	-24.022	-24.02	0.008326
0.250	0.25119	-20.051	-20.05	0.004987
0.315	0.31623	-16.121	-16.12	0.006203
0.400	0.39811	-12.286	-12.29	-0.03256
0.500	0.50119	-8.672	-8.67	0.023063
0.630	0.63096	-5.5082	-5.5	0.148869
0.800	0.79433	-3.094	-3.09	0.129282
1.000	1	-1.5876	-1.59	-0.15117
1.250	1.2589	-0.84682	-0.85	-0.37552
1.600	1.5849	-0.59086	-0.59	0.145551
2.000	1.9953	-0.61239	-0.61	0.390274
2.500	2.5119	-0.81796	-0.82	-0.2494
3.150	3.1623	-1.1884	-1.19	-0.13463
4.000	3.9811	-1.7404	-1.74	0.022983
5.000	5.0119	-2.5008	-2.5	0.03199
6.300	6.3096	-3.4885	-3.49	-0.043
8.000	7.9433	-4.7032	-4.7	0.068039
10.000	10	-6.1234	-6.12	0.055525
12.500	12.589	-7.7138	-7.71	0.049262
16.000	15.849	-9.4352	-9.44	-0.05087
20.000	19.953	-11.253	-11.25	0.02666
25.000	25.119	-13.141	-13.14	0.00761
31.500	31.623	-15.088	-15.09	-0.01326
40.000	39.811	-17.102	-17.1	0.011695
50.000	50.119	-19.228	-19.23	-0.0104
63.000	63.096	-21.581	-21.58	0.004634
80.000	79.433	-24.385	-24.38	0.020504
100.000	100	-27.931	-27.93	0.00358
125.000	125.89	-32.371	-32.37	0.003089
160.000	158.49	-37.551	-37.55	0.002663
200.000	199.53	-43.176	-43.18	-0.00926
250.000	251.19	-49.017	-49.02	-0.00612
315.000	316.23	-54.952	-54.95	0.00364
400.000	398.11	-60.925	-60.92	0.008207
500.000	501.19	-66.914	-66.91	0.005978
630.000	630.96	-72.91	-72.91	0
800.000	794.33	-78.908	-78.91	-0.00253

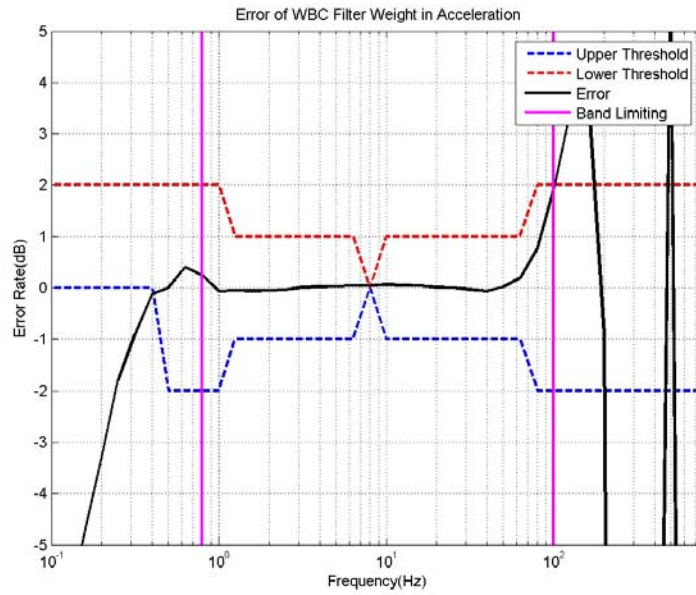
## 2. Frequency Response and weight factor of Weighting Filter as acceleration input



<Fig. 8 Comparison of a theoretical acceleration and a measured acceleration (y-axis : Decibel)>



<Fig. 9 Comparison of a theoretical acceleration and a measured acceleration (y-axis : Linear)>



<Fig. 10 Error Range and Acceleration's Error >

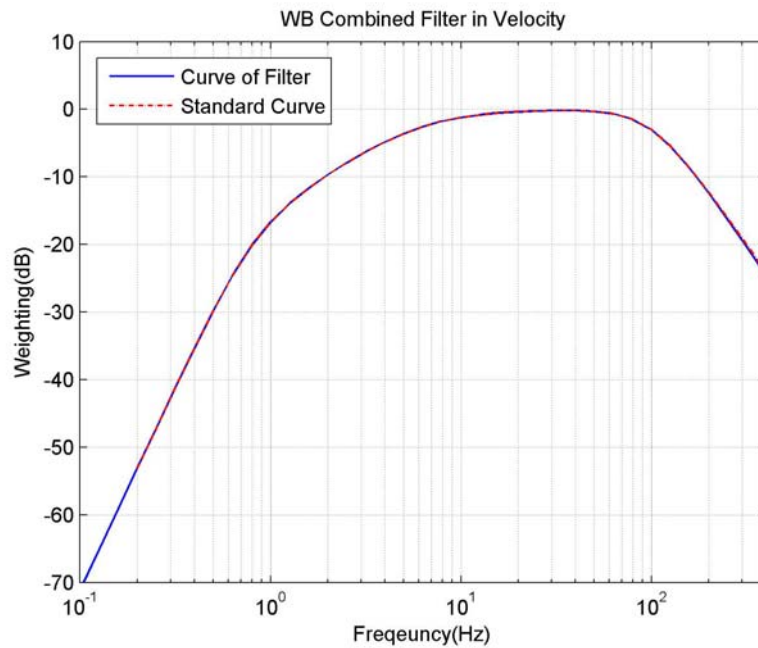
Error curve is in a upper and lower threshold line within band limiting frequencies.

<Table 3. Combined frequency weighting, defined from 1Hz to 80Hz, in one-third-octave bands>

Frequency		WBCombined Weight		Tolerance
TRUE	Nominal	Factor $W_m$	dB	dB
0.1000	0.100	0.015800	-36.03	+ 2
0.1259	0.125	0.025100	-32.01	+ 2
0.1585	0.160	0.040000	-27.96	+ 2
0.1995	0.200	0.062900	-24.03	+ 2
0.2512	0.250	0.099400	-20.05	+ 2
0.3162	0.315	0.156000	-16.14	+ 2
0.3981	0.400	0.243000	-12.29	+ 2
0.5012	0.500	0.368000	-8.68	± 2
0.6310	0.630	0.530000	-5.51	± 2
0.7943	0.800	0.700000	-3.10	± 2
1.0000	1.000	0.833000	-1.59	± 2
1.2590	1.250	0.907000	-0.85	± 1
1.5850	1.600	0.934000	-0.59	± 1
1.9950	2.000	0.932000	-0.61	± 1
2.5120	2.500	0.910000	-0.82	± 1
3.1620	3.150	0.872000	-1.19	± 1
3.9810	4.000	0.818000	-1.74	± 1
5.0120	5.000	0.750000	-2.50	± 1
6.3100	6.300	0.669000	-3.49	± 1

7.9430	8.000	0.582000	-4.70	0 0
10.0000	10.000	0.494000	-6.13	± 1
12.5900	12.500	0.411000	-7.72	± 1
15.8500	16.000	0.337000	-9.45	± 1
19.9500	20.000	0.274000	-11.24	± 1
25.1200	25.000	0.220000	-13.15	± 1
31.6200	31.500	0.176000	-15.09	± 1
39.8100	40.000	0.140000	-17.08	± 1
50.1200	50.000	0.109000	-19.25	± 1
63.1000	63.000	0.083400	-21.58	± 1
79.4300	80.000	0.060400	-24.38	± 2
100.0000	100.000	0.040100	-27.94	± 2
125.9000	125.000	0.024100	-32.36	± 2
158.5000	160.000	0.013300	-37.52	± 2
199.5000	200.000	0.006940	-43.17	± 2
251.2000	250.000	0.003540	-49.02	+ 2
316.2000	315.000	0.001790	-54.94	+ 2
398.1000	400.000	0.000899	-60.92	+ 2
501.2000	500.000	0.000451	-66.92	+ 2
631.0000	630.000	0.000226	-72.92	+ 2
794.3000	800.000	0.000113	-78.94	+ 2

### 3. Frequency Response and weight factor of Weighting Filter as velocity input

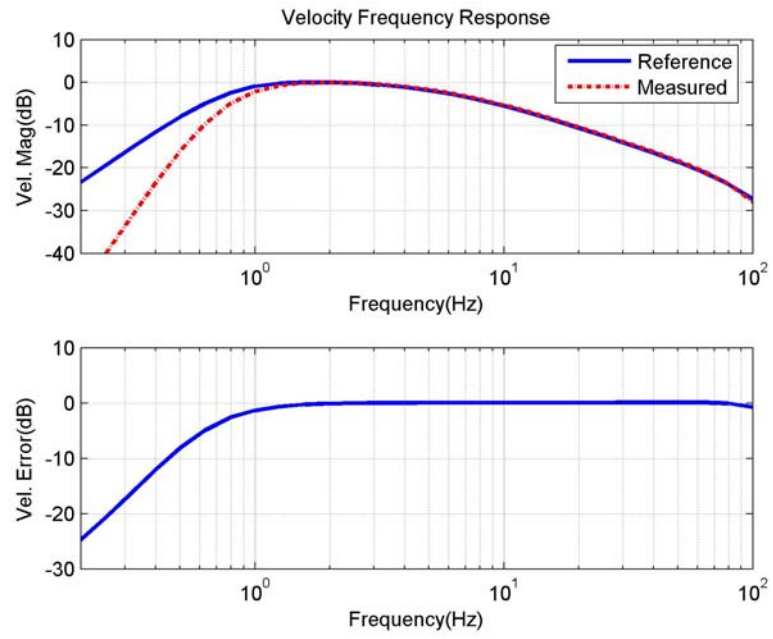


<Fig. 11> Velocity Weighting Factor

<Table 4> Velocity Weight Factor in Human Vibration

True Frequency(Hz)	Velocity Weighting		
	Reference	Calculated	Error(%)
0.1995	0.00221	0.00223	-0.679
0.2512	0.00439	0.00438	0.150
0.3162	0.00870	0.00868	0.260
0.3981	0.01700	0.01701	-0.053
0.5012	0.03250	0.03246	0.135
0.6310	0.05890	0.05889	0.025
0.7943	0.09790	0.09779	0.115
1.000	0.147	0.147	0.313
1.259	0.201	0.201	0.040

1.585	0.260	0.261	-0.196
1.995	0.327	0.327	-0.028
2.512	0.402	0.402	-0.055
3.162	0.485	0.485	-0.035
3.981	0.573	0.573	-0.037
5.012	0.661	0.661	-0.024
6.310	0.743	0.743	0.013
7.943	0.813	0.813	-0.021
10.000	0.869	0.869	-0.036
12.590	0.911	0.911	-0.033
15.850	0.941	0.941	-0.001
19.950	0.961	0.961	0.003
25.120	0.973	0.973	-0.042
31.620	0.979	0.979	-0.035
39.810	0.978	0.978	0.026
50.120	0.964	0.964	0.037
63.100	0.925	0.925	-0.014
79.430	0.844	0.843	0.118
100.000	0.706	0.705	0.201
125.900	0.533	0.531	0.439
158.500	0.370	0.366	0.959
199.500	0.244	0.240	1.635
251.200	0.156	0.153	2.103
316.200	0.100	0.096	3.856
398.100	0.063	0.059	6.184



<Fig. 12> Error between theoretical value and measured value