中華民國音響學會
第五屆學術研討會論文集

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行 政 院 勞 工 委 員 會
行　政　院　衛　生　署
行 政 院 環 境 保 護 署
合辦

中華民國八十一年十一月二十日

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DIGITAL FILTERING AND WAVELET TRANSFORM TECHNIQUES IN THE ANALYSIS OF RAPIDLY VARYING TRANSIENT SIGNALS

Roger Upton, (Bruel & Kjaer)

Abstract: the analysis of acoustic phenomena often requires the ability to follow signals whose spectral content can vary rapidly with time. One means of achieving this is to use the relatively recently introduced technique of fractional octave digital filtering combined with multispectrum. However, as the spectral variation becomes more and more rapid, the conventional digital filtering/multispectrum technique begins to fail, and an alternative technique has to be used. One such technique is the wavelet transform. This paper compares the two techniques in the analysis of rapidly varying transient signals and shows an example of the application of wavelet transform in the analysis of blasting noise.

1. Comparison of the Techniques

1.1 Use of Multispectrum with Fractional Octave Digital Filtering

Digital filtering has been used for many years as the basis for real-time fractional octave analyzers. These analyzers operate on a sample-to-sample basis to achieve their results. An advantage of this process is that the filter bandwidth and the averaging time used are independent of each other. This is in contrast to the FFT-based analyzer where blockwise analysis means that the filter bandwidth and the averaging time are always linked.

Because of the above, the real-time digital filtering process lends itself to rapid data acquisition. Further, it is possible to apply a gating function after the filtering process to study the time domain variation of the RMS filter output without corrupting the frequency domain data. This leads to the multispectrum technique, where multiple spectra are collected to describe the time domain variation of an event.

(Note that where such techniques are used with FFT analyzers, the gate is applied ahead of the analysis process. This means that the gate-width will have a fundamental influence on the amount of information which can be obtained from the ensuing analysis process.)

1.2 Collection of Multispectra

Fig. 1 > Selectable analysis parameters in the collection of multispectra

A multispectrum is no more than an array of spectra collected as a function of some other parameter. Where the signal being analyzed is an acoustic transient, that other parameter will usually be time. Fig.1 then shows which analysis parameters can be selected in acquisition of the multispectrum.

Multispectra can be collected with either exponential or linear averaging. A typical application of multispectrum with exponential averaging is measurement of reverberation time, where spectra at the filter/detector outputs are collected at constant time intervals to create a three-dimensional map tracing the reverberation decay at each frequency as a function of time. In such multispectrum collection the time interval between spectra and the averaging time will usually be different.

Fig. 2 > Multispectrum showing sound intensity as a function of position in the cycle of a ships diesel engine.

A typical application of multispectrum with linear averaging is collection of data to examine the changes in a measurement parameter, for instance, as a function of position in a machine cycle. Here the time interval between spectra and the averaging time will usually be the same. Fig. 2 gives an example of such a multispectrum for 1/3 octave sound intensity data as a function of position in the cycle of a ships diesel engine.

1.3 Limitations of Multispectrum Collection

Fig. 3 > Multispectrum map of a 1/3 octave analysis of an impulse.

Multispectrum is a very powerful method of data collection in the examination of transient data. However, as such transients become shorter and shorter, limitations due to the filtering/detection process begin to appear. These limitations can be summarised as being:

a) the impulse response of the filters increases with reducing frequency thus requiring a longer averaging time.

b) the filtering process introduces a frequency-dependent time delay in the analysis process.

Both of these limitations are illustrated in Fig. 3, which shows a map of a 1/3 octave multipectrum collected in the analysis of a 0,7ms impulse. This quite clearly shows the frequency dependent time delay and the spreading of the data at low frequency due the increasing duration of the filter impulse response.

1.4 The Wavelet Transform

The wavelet transform, (or WT), is of relatively recent origin and shows much promise in acoustic applications. It can be described as a time-frequency decomposition method using a frequency dependent Gaussian window. A frequency dependent window is used because time domain events change more rapidly at high frequencies where wavelengths are shorter than at low frequencies. As a consequence of using a frequency dependent window the WT gives a constant percentage bandwidth frequency analysis as is extensively used in acoustics. Further, the WT can be used to produce fractional octave analysis.

The WT is defined as follows:

$$S(b,a)=|a|^{-1/2}\int s(t)\overline{\psi}((t-b)/a)\,dt$$

where ψ is a special function called a basic wavelet which is translated in time and expanded or contracted using the parameter a, ($a \neq 0$). Strictly speaking, the WT is a time-scale decomposition, since the function S(b,a) gives the contributions to the signal around time b at scale a, but choice of an appropriate wavelet causes the WT to give a time-frequency representation of the signal.

1.5 Examples of Wavelet Transform Analysis

Fig 4 > 1/12 octave wavelet transform analysis of a delta function.

Fig. 4 shows a 1/12 octave wavelet transform analysis of a delta function. The first point to notice is that the analysis is centred on the same time scale as the original time signal. This is because the wavelet transform is a zero-phase method and is not subject to the frequency dependent time delay seen earlier. This in turn means that events in the analysis can be directly related to events in the time signal.

The second point to notice is that the time window of the analysis gets steadily narrower with increasing frequency without ever reaching a limiting value. This is because the wavelet transform uses envelope detection, (as opposed to RMS detection and averaging used in traditional methods), to analyze with optimum efficiency at all frequencies using the minimum window allowed by the well-known time-frequency limitation. This in turn means that wavelet transform methods can be up to a factor of 10 better than filter methods at resolving closely spaced transient time domain events.

Fig 5> Analysis of the transient response of a loudspeaker using wavelet transform.

Finally, as an example of application of wavelet transform to analysis of an acoustic signal, Fig. 5 shows an analysis of the transient response of a loudspeaker. The differences in the transient decay as a function of frequency can clearly be seen.

2. Comparison in the Analysis of Blasting Noise

2.1 The Time Signals Involved

Fig. 6> A typical example of the time signals being analyzed.

The signals used in this comparison were taken from a study of blasting noise made at a quarry in Japan. In particular, signals recorded at 100m, 150m, and 200m were used because these all displayed a double peak after the main peak, in which the two components of the double peak were separated by about 20ms. The signal recorded at a distance of 100m is shown in Fig. 6. It was felt that these signals would give a true test of digital filtering/multispectrum and wavelet transform techniques and their relative abilities to resolve the double peak in the subsequent analyses.

2.2 Results of Analyses using 1/3 Octave Digital Filtering and Multispectrum Technique

Fig. 7> Multispectrum map of the signal at 100m analyzed using multispectrum collection at 50ms intervals.

Fig. 8> As Fig. 2, but using multispectrum collection at 5ms intervals.

An analysis of this type of signal using digital filtering/multispectrum techniques presents the user with a conflict in selecting the analysis parameters. On the one hand a long averaging time is necessary to correctly average the low frequency energy in the signal, while on the other hand a short averaging time is necessary to resolve the double peak. Fig.s 7 and 8 show analyses of the signal recorded at 100m collected into multispectra using linear averaging, the averaging time and the time interval between the spectra collected into the multispectra being equal. Fig. 7, where the time interval between spectra is 50ms, shows good data from about 100Hz and upwards, but fails to resolve the double peak, while Fig. 8 where the time interval between spectra is 5ms resolves the double peak, but gives bad data below about 1kHz, (these cut-off frequencies between good and bad data are based on having a BT equal to 1). Note also the frequency dependent time delay, described in Section 1.3 of this paper, in both sets of results.

A solution to the above problem might be to use a frequency dependent averaging time. However, even this might not be sufficient due to the impulsive nature of the signals and the resulting need to average over 2 to 3 filter response times in order to correctly integrate the entire filter response. At 1kHz, for instance, this would require an averaging time of about 12ms, which would be too long to resolve the double peak. Further, this would introduce an additional frequency dependent time delay to the analysis.

2.3 Results of the Analyses using 1/3 Octave Wavelet Transform.

Fig. 9> Wavelet transform analysis of the signal at 100m.

Fig. 10> Wavelet transform analysis of the signal at 150m.

Fig. 11> Wavelet transform analysis of the signal at 200m.

Fig.s 9 through 11 show the results of wavelet transform analysis of the blasting noise signals recorded at distances of 100m, 150m, and 200m, respectively. They show none of the problems of the analyses made using traditional digital filtering/multispectrum technique, in that each is able to resolve the double peak while at the same time giving a correct analysis of the low frequency information. At the same time because of the zero phase nature of the wavelet transform, it is easy to correlate events in the time signal with corresponding events in the resulting analysis, thereby easing interpretation of the results. In particular, these results were used to confirm a new waveform prediction method being developed for blasting noise.

3. Conclusion

Multispectrum analysis forms a powerful means of examining the three-dimensional behaviour of transient signals. Where, however, this method starts to reach its limits, WT analysis starts to take over.

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Collection of Multispectra

Selectable parameters

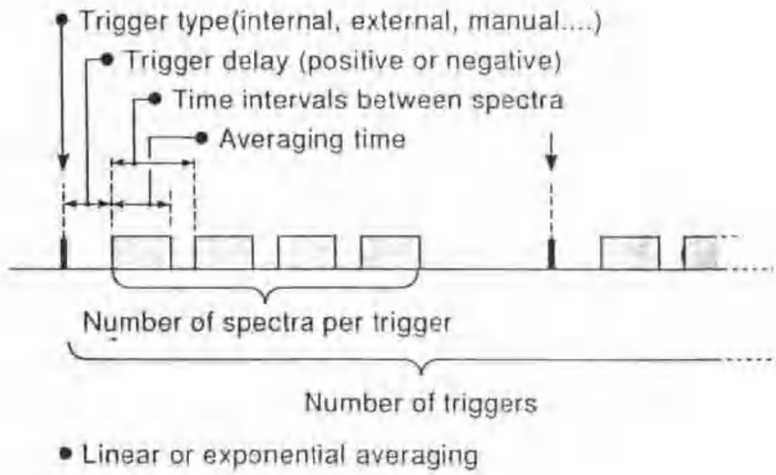
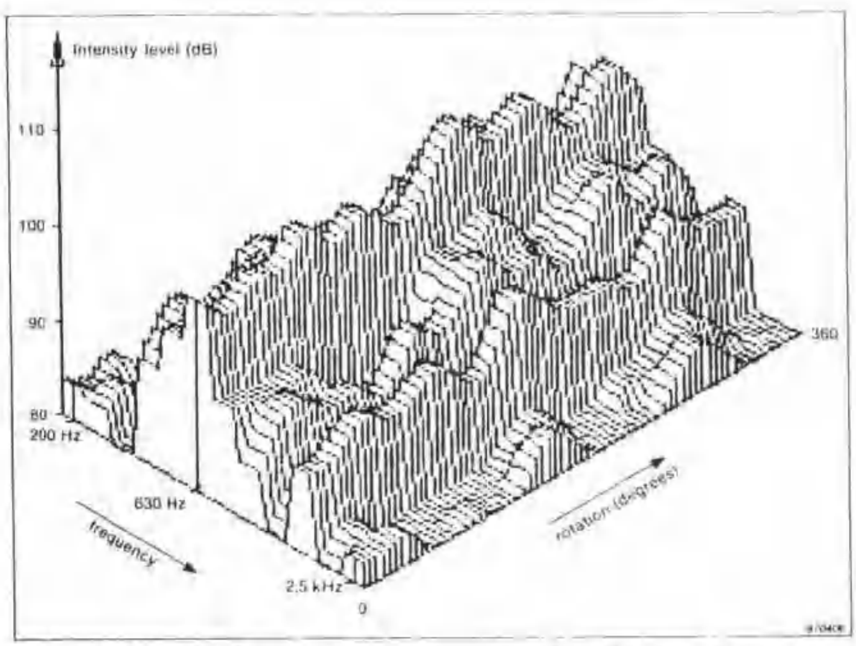
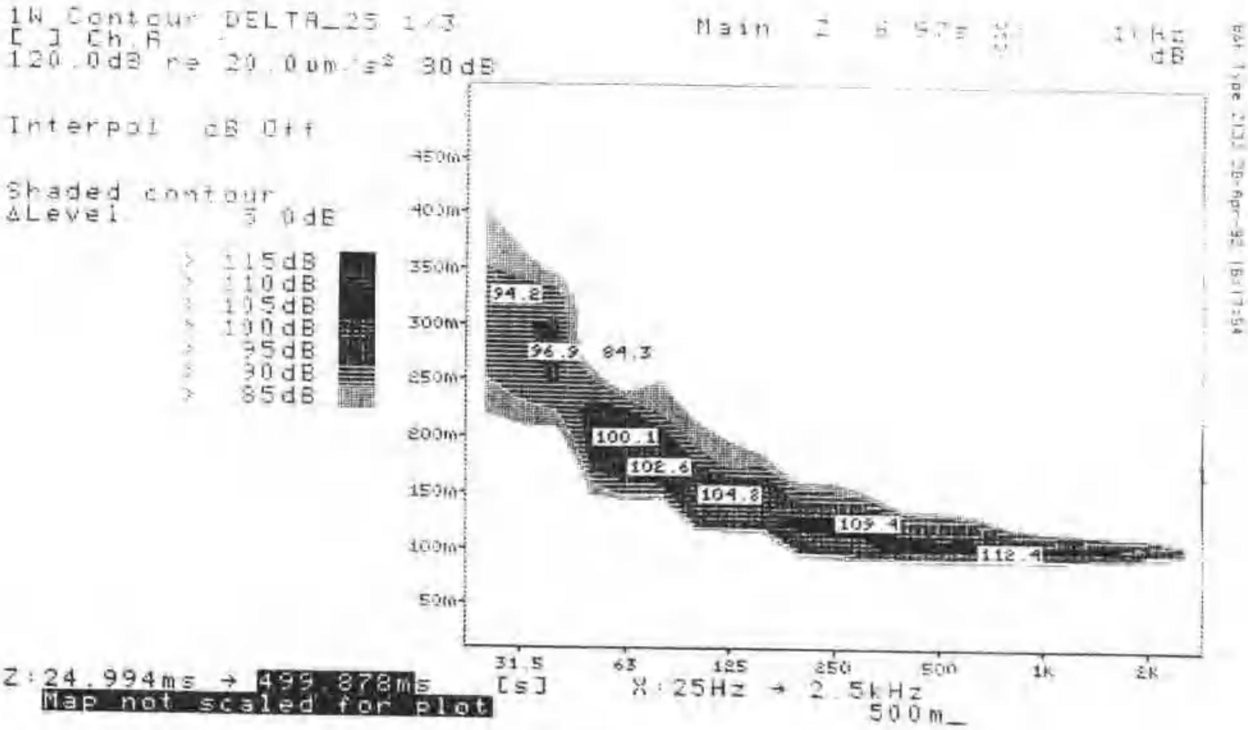


Fig. 1



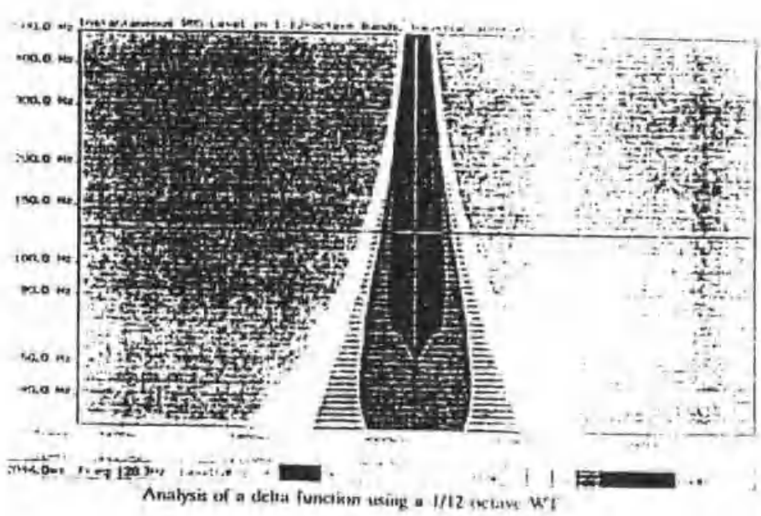
3D plot produced from gated-intensity measurements on the cam-shaft cover

Fig. 2



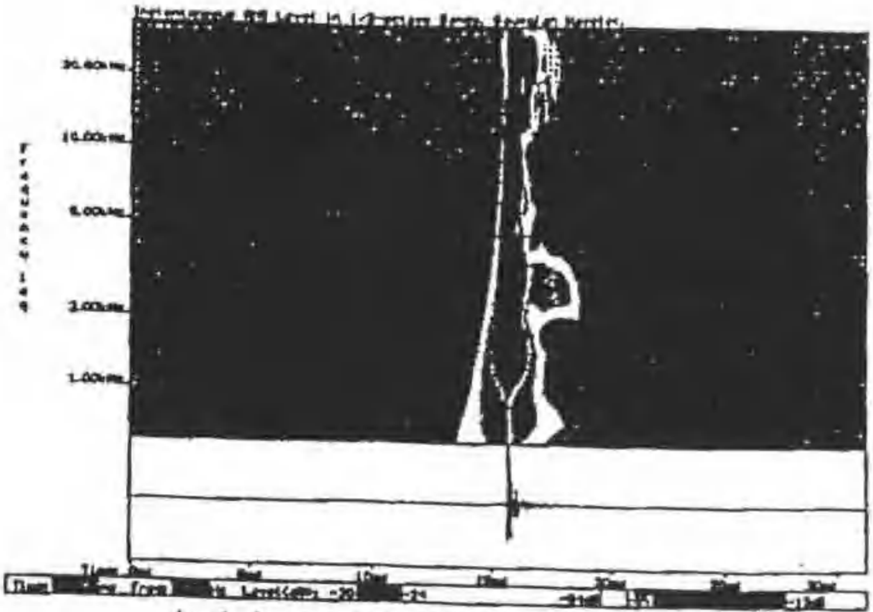
1W Contour DELTA_25 1/3
[ ] Ch.A
120.0dB re 20.0µm/s² 80dB
Interpol dB Off
Shaded contour
ΔLevel 5.0dB
> 115dB
> 110dB
> 105dB
> 100dB
> 95dB
> 90dB
> 85dB
Main
dB
450m
400m
350m
300m
250m
200m
150m
100m
50m
94.8
96.9
84.3
100.1
102.6
104.8
109.4
112.4
31.5
63
125
250
500
1K
2K
Z:24.994ms → 499.878ms
Map not scaled for plot
[s]
X:25Hz → 2.5kHz
500m

Fig. 3



Analysis of a delta function using a 1/12 octave WT

Fig. 4



Analysis of the transient response of
a loudspeaker using wavelet transform.

Fig. 5

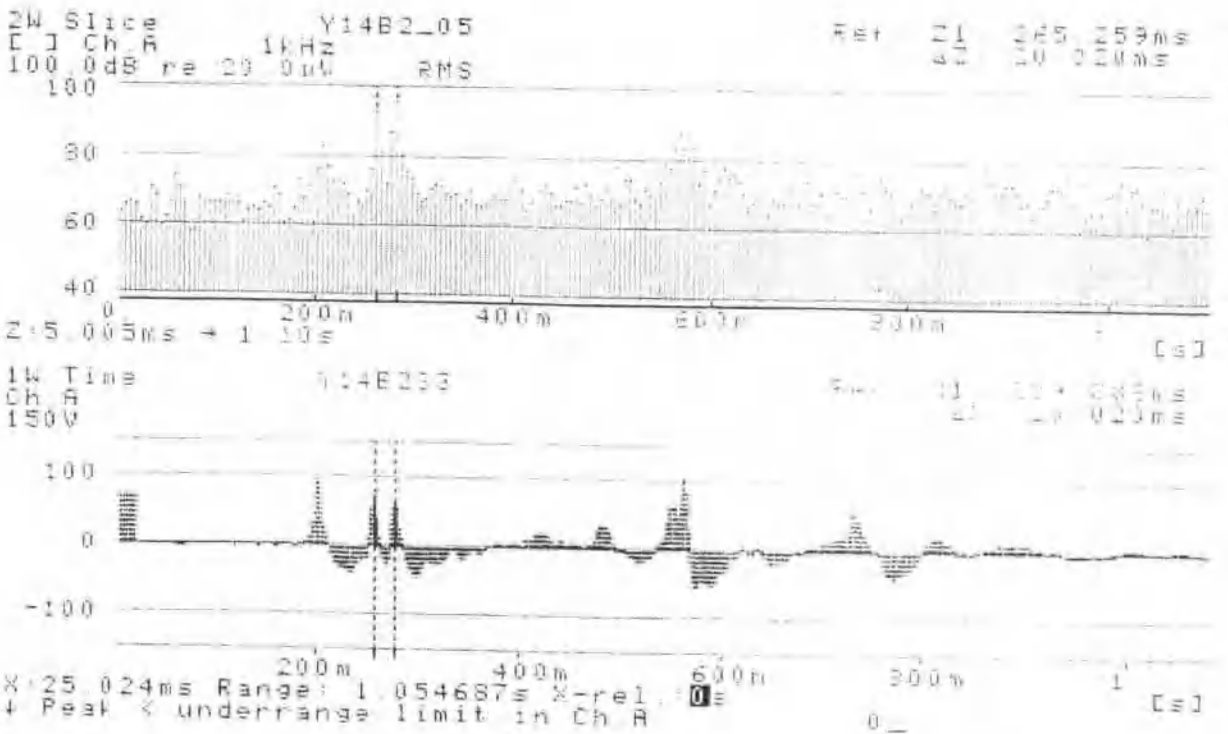
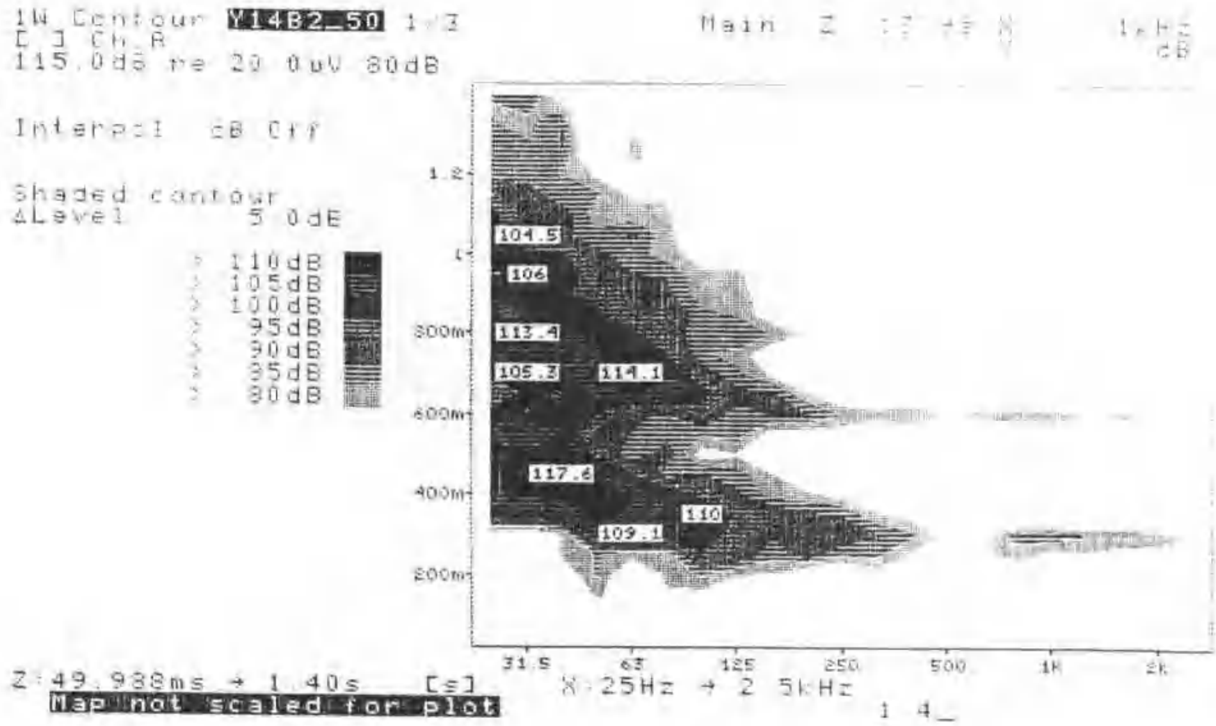
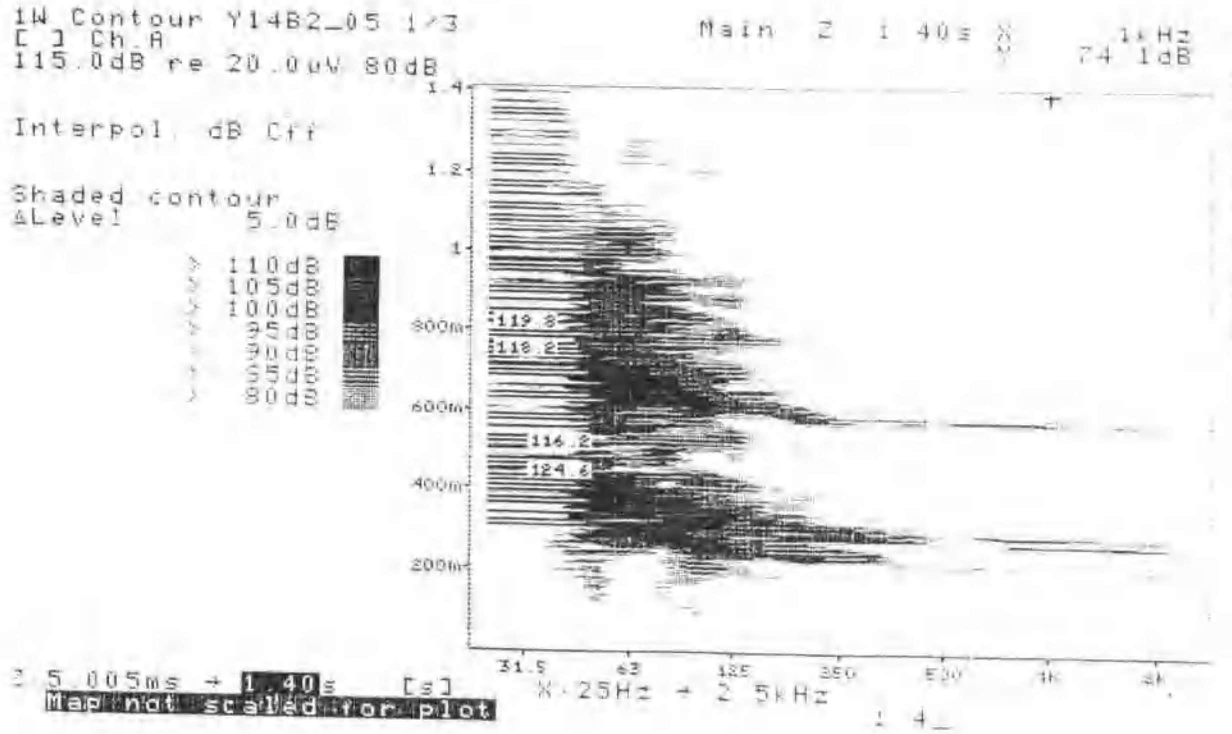


Fig. 6



1W Contour Y14B2_50 1/3
[ ] Ch.A
115.0dB re 20.0uV 80dB
Interpol dB Off
Shaded contour
ΔLevel 5.0dB
> 110dB
> 105dB
> 100dB
> 95dB
> 90dB
> 85dB
> 80dB
Main
104.5
106
113.4
105.3
114.1
117.6
110
109.1
Z:49.988ms → 1.40s [s]
Map not scaled for plot
X:25Hz → 2.5kHz

Fig. 7



1W Contour Y14B2_05 1/3
[ ] Ch.A
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Interpol dB Off
Shaded contour
ΔLevel 5.0dB
> 110dB
> 105dB
> 100dB
> 95dB
> 90dB
> 85dB
> 80dB
Main Z 1.40s
1kHz
74.1dB
119.8
118.2
116.2
124.6
Z 5.005ms → 1.40s [s]
Map not scaled for plot
X:25Hz → 2.5kHz

Fig. 8

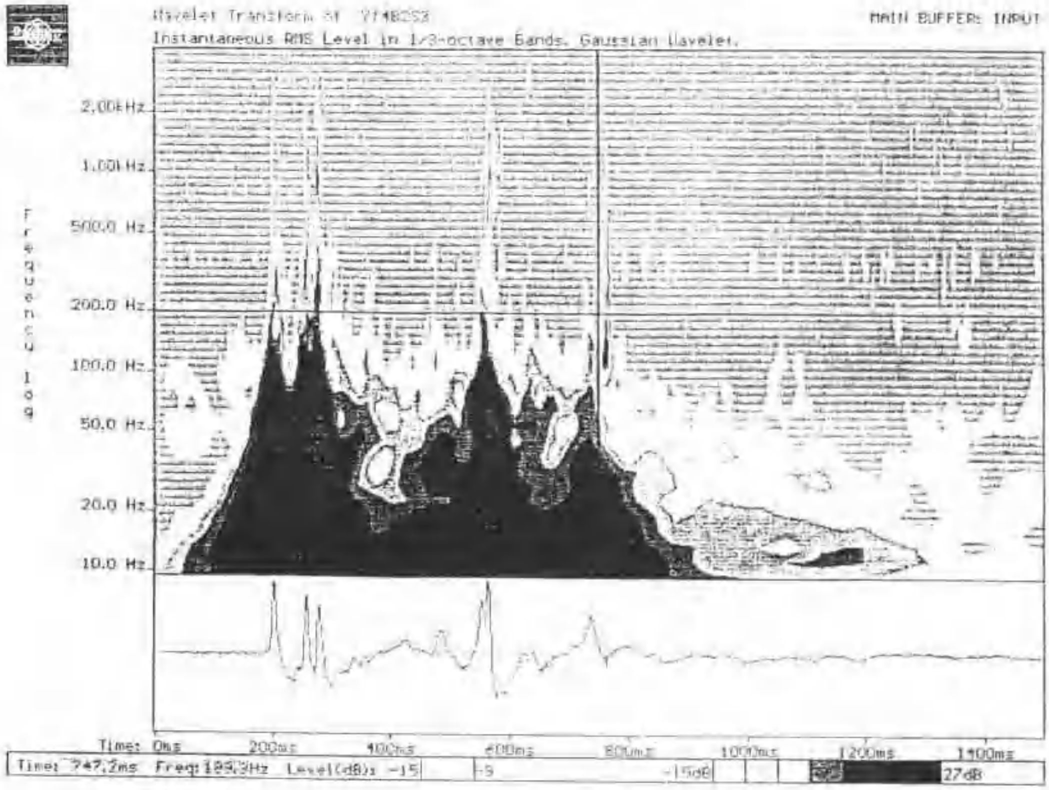
Wavelet Transform of 7748253
MAIN BUFFER: INPUT
Instantaneous RMS Level in 1/3-octave Bands. Gaussian Wavelet.
2.00kHz
1.00kHz
500.0 Hz
200.0 Hz
100.0 Hz
50.0 Hz
20.0 Hz
10.0 Hz
Frequency log
Time: 0ms
200ms
400ms
600ms
800ms
1000ms
1200ms
1400ms
Time: 747.2ms Freq:189.3Hz Level(dB): -15
-9
-15dB
27dB

Fig. 9

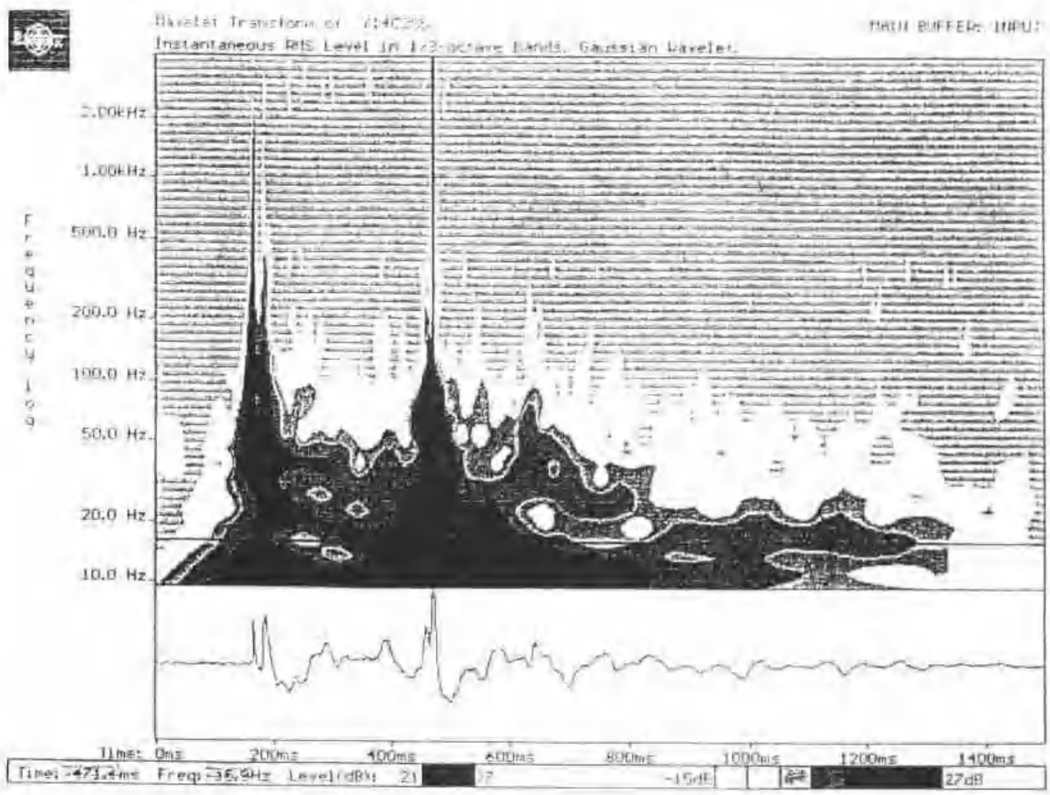
MAIN BUFFER: INPUT
Instantaneous RMS Level in 1/3-octave Bands. Gaussian Wavelet.
2.00kHz
1.00kHz
500.0 Hz
200.0 Hz
100.0 Hz
50.0 Hz
20.0 Hz
10.0 Hz
Frequency log
Time: 0ms
200ms
400ms
600ms
800ms
1000ms
1200ms
1400ms
-15dB
27dB

Fig. 10

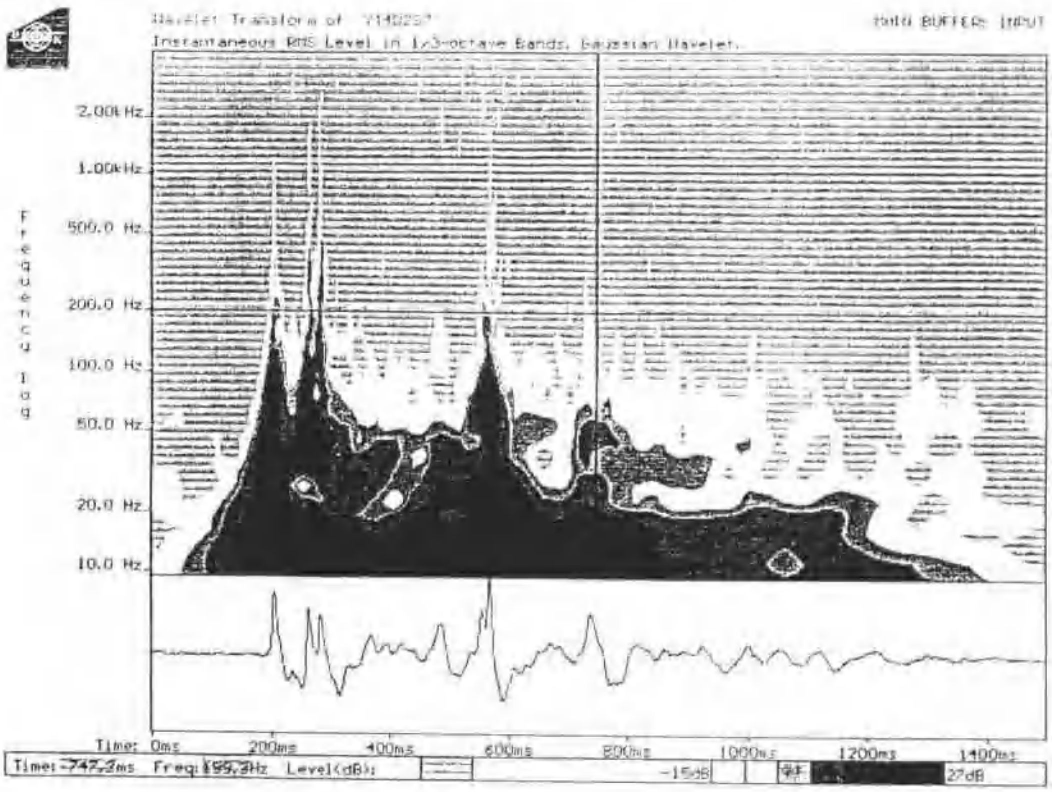
Wavelet Transform of 'Y14D257'
DUAL BUFFER: INPUT
Instantaneous RMS Level in 1/3-octave Bands, Gaussian Wavelet.
2.00kHz
1.00kHz
500.0 Hz
200.0 Hz
100.0 Hz
50.0 Hz
20.0 Hz
10.0 Hz
Frequency log
Time: 0ms
200ms
400ms
600ms
800ms
1000ms
1200ms
1400ms
Level(dB):
-15dB
27dB

Fig. 11

NEW PORTABLE SOUND LEVEL MEASURING SYSTEM BY UTILIZING DIGITAL TECHNOLOGY

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1. INTRODUCTION

Recent developments and spreads of digital technology are amazing. Now a days, it is well known that many instruments even for domestic use, a micro-computer system is built in. The acoustic measurement is neither exception. The active sound control system for noise reduction or sound intensity measurement for visualization of sound field, adopting digital technology are attractive.

Here, we are going to introduce the advanced instruments for sound level measurement and its recording. These instruments are new type sound level meter and level recorder. One of the biggest point of the advance is the realization of down sized and light weight measuring instruments for on-site measurement. At the same time, these instruments have the auto measuring/recording function, including measuring condition.

The advance was attained by adopting the IC memory card and the down sizing technology. Both size and weight of instruments are unexpectedly important, particularly in the outdoor measurement. Using these instruments, measuring engineer will be free from burden like as instruments transportation and setting up or data processing by hand-work. After all, it will be possible for them to pay attention to measurement itself more than before.

2. BACKGROUND IN SOUND LEVEL MEASURING FIELD

Sound environment is constructed by many sound sources like automobiles, aircrafts, trains, and machines. In many cases, it is very difficult to read the indication of sound level meter directly, because the sound levels generally change very rapidly.

Under the circumstances mentioned above, it is commonly used the method of reading sound level from recorded level on paper by using a set of level recorder and sound level meter. But this method still includes some problems.

In some cases of multi-point and simultaneous measurement, for example, the measuring engineers should pay attention to many directions, from prepairing and transporting instruments to data processing after the measurement. Those jobs must be very heavy load of theirs. And also the measuring cost should not be ignored.

2.1 ON-SITE PROBLEM

Generally speaking, in-situ measuement is very hard and troublesome, in the case of outdoor measurement, particularly. Some unexpected troubles may occur like rainfall while measuring. It might be useless to say, the bigger size measuring system means the harder to carry and operate.

Fig.1 is a simple schematic diagram for usual sound level recording.
Fig.1-A shows the typical measuring system for sound level measurement. This system is useful for recording sound levels only on paper for long time. On the other hand, the weak point of this system is very hard to calculate the recorded data. For, data reading or calculation must be done by men.

| Sound Level Meter AC OUT | → | Level Recorder | [on site] |

A : *Typical System* ; SOUND LEVEL RECORD ONLY

| Sound Level Meter DC OUT | → | Data Recorder (record) | [on site] |

note:AC OUT when wave form

| Data Recorder (play back) | → | A/D Converter | → | Computer (with software) | [at work station] |

B : *Bigger System* ; SOUND LEVEL or WAVE FORM RECORD & DATA CALCULATION

Fig.1 FORMER CONSTRUCTION for SOUND LEVEL RECORD

Fig.1-B shows the improved measuring system for sound level recording and data processing. Constructing the system like this figure, measuring engineers are free from recording and calculation. Still more, a data recorder is also possible to connect to a level recorder. This system, however, still includes other weak points as follows. Actually, these points have very big meaning when in-situ measurement.

1: BIG SIZE & HEAVY WEIGHT
2: TAKE LONG TIME TO PLAY BACK
3: UNEASY TO OPERATE
4: BIG POWER CONSUMPTION

2.2 PROBLEM at WORKSTATION

Every sound level data should be processed to estimate the noise. And suitable estimative value should be chosen according to the object of measurement. A-weighted sound pressure level L_{pA} is one of the estimative value, but almost of all estimative value is calculted from L_{pA}, like as L_{Aeq}, L_{AE} or L_x. Using former type sound level meter that measures L_{pA} only, measureing engineer has to calculate by himself.
Measuring condition data is also important as well as sound level data. Many significant informations should be recorded while the measurement. Frequency weighting and level range of sound level meter or dynamic characteristics and paper speed of level recorder, none of them is dispensable. Not only the setting condition of instruments but the time data is important. At the same time, such a condition is likely to be changed, depending on actual condition on site.

Every field engineer who uses the former type system is forced to write down the measuring condition change by their own hands every time. Such a change procedure job is very troublesome, and all the results of measurement will lose the meaning, if there may be some mistakes in it.

3. PROPOSED IMPROVEMENTS

"How much does the technology contribute to sound measuring engineer?" It was our main theme of the discussion. And then, our conclusion was down sizing of the system and building up the automatic measuring and data calculation system by introducing an IC memory card.

3.1 DOWN SIZED SYSTEM

Face to down sizing of the measuring system, we have reviewed every recording media like magnetic tape/disk, every kind of IC memory and optical memory disk. At the same time, we have discussed about new type level recorder, maintaining the merits of recording on paper. And then, we'd reached to adopt the IC memory card. IC memory card is one of the latest media, so commercial price still expensive a little, this problem will be improved by mass production soon.

The two biggest merits of IC memory card are needless mechanism and very low power consumption.

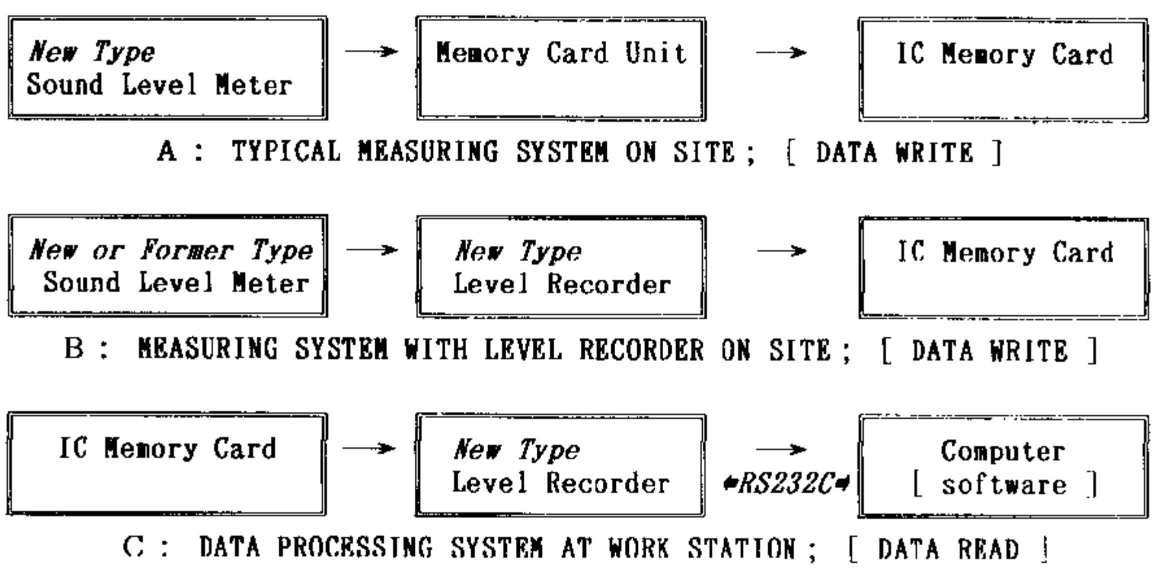

Fig2. PROPOSED CONSTRUCTION for SOUND LEVEL RECORD

Fig.2 is a simple schematic diagram of proposed construction for sound level recording or reading. The new type sound level meter is model NL-04;IEC type 2 / -14;IEC type 1 integrating sound level meter in this figure. And also the new type level recorder is model LR-06. The out-looking of NL-04 and LR-06 are shown