Requirements and Limitations of Digital Impulse Analysing System

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Introduction

Requirements for digital recorders in high-voltage testing are given by the standards, but what is the actual advantage of higher resolution (number of bits) or faster sampling (Mega samples per second) for the user? If just a visual judgement and comparison of impulses on a piece of paper is performed the difference between average and high-end digital recorders can hardly be recognised. However, transformations from time to frequency domain, mostly performed by FFT (Fast Fourier Transformation) and used in Transfer and Coherence analysis, are more sensitive to the recorder's performance data [1]. Actual amplitudes of the excitation impulse are very small at high frequencies which requires a good amplitude resolution for an accurate measurement.

As important as technical performance are frequent calibrations, which are also influenced by the quality standard ISO 9000. However frequent calibrations are a very expensive procedure. To minimise cost and work, a unique channel insert system has been developed. This channel insert system allows for an easy calibration procedure and a continuous device availability at 60% of the cost of conventional continuously operational systems.

Frequency Spectra in HV Testing

Most common impulses in high voltage laboratories are switching impulses (SI), full, chopped and front chopped lightning impulses (LI, CW and FCW). The relatively slow Switching Impulse (250 / 2500µs) contains mainly a spectral density at frequencies below 100 kHz whereas the front chopped lightning impulse (chopped in the front at <1µs) shows a significant spectral density at frequencies up to several MHz. In general the statement is true that the smaller the time parameters the larger the spectral density at higher frequencies. Chopped impulses contain, due to high *dU/dt*, which occurs during the chopping process, the largest spectral density at high frequencies. As an example the frequency spectra of a SI, LI and FCW impulse are shown in Figure 1.

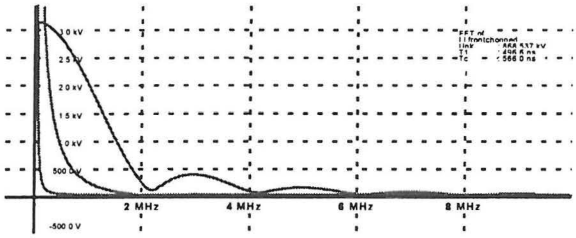


Figure 1

From bottom left to top right: spectral density of SI LI and FCW. The later one showing largest spectral density at high frequencies.

Performance Data of Digital Recorders

When using a sophisticated analysis tool such as the Transfer Function it is important to obtain the best achievable S/N (signal to noise) ratio [2]. A closer look to amplitude resolution, sampling rate and their influence on the final result is given below.

Amplitude resolution describes the smallest increment a digital recorder can resolve. For an 8 bit recorder this corresponds to 1/256 or 0.40%. Every additional bit improves the amplitude resolution by a factor of 2. Therefore 10 bits correspond to 0.10% and 12 bit to 0.02%. This improvement is shown in Figure 2 and 3.

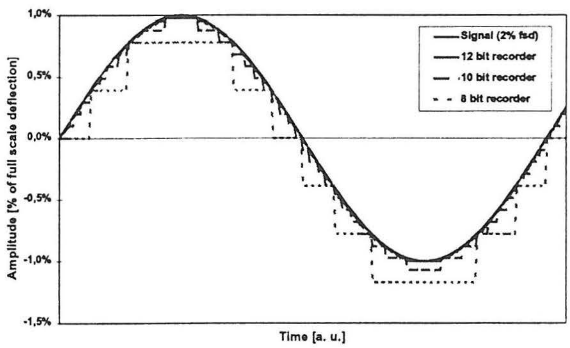


Figure 2

Comparison between 8, 10 and 12 bit amplitude resolution. A 2 % modulation is approached with 5 steps (8 bit), 20 steps (10 bit) and 80 steps (12 bit). The latter one can hardly be distinguished from the original.

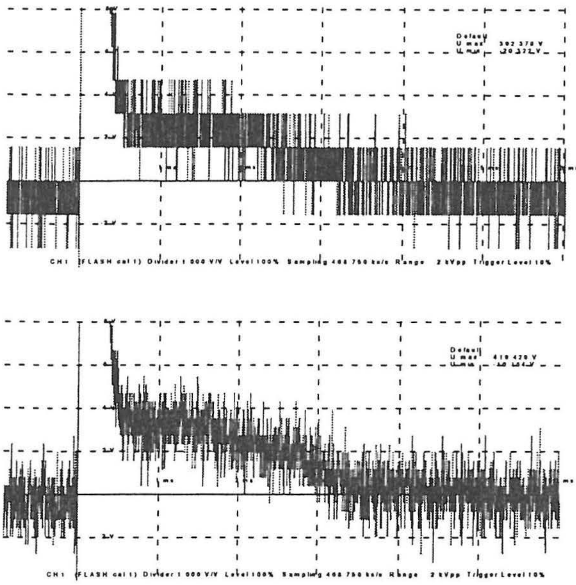


Figure 3

Enlargement of real recordings performed with 10 bit (top) and a 12 bit (bottom) digitisers. The 10 bit digitiser clearly shows the steps of individual bits.

Sampling rate describes how often the analogue signal is measured per second. The faster this is done the higher frequencies can be measured by the digital recorder. For a given sampling rate F_{sample} the maximum frequency which can be recorded is given by the Nyquist theorem:

$$F_{Nyquist} = \frac{F_{sample}}{2}$$

Frequencies larger than the $F_{nyquist}$ are reproduced with strong aliasing components. In Figure 4 an example of aliasing is shown.

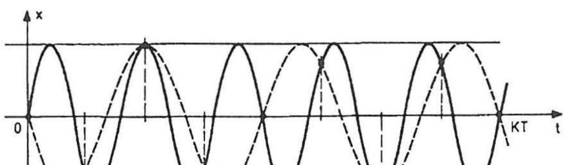


Figure 4

Aliasing effect: A signal of frequency 5/8 f_{sample}(solid line) sampled with f_{sample}. Connecting the resulting samples with a line (dashed) shows an aliasing signal of 3/8 f_{sample}.

Influence on Transfer / Coherence Analysis

Transfer Functions (TF) are often used to compare e.g. power transformers of the same electrical and mechanical design [3]. TF are in first order insensitive to the wave shape of the exciting impulse. Physically TF reflects the electrical characteristics of the winding and reveals its natural oscillations. Each resonant pole on the TF plotted against frequency corresponds to a natural resonance of a winding section [4]. An example for a TF (and CF) of a power transformer is given in Figure 5.

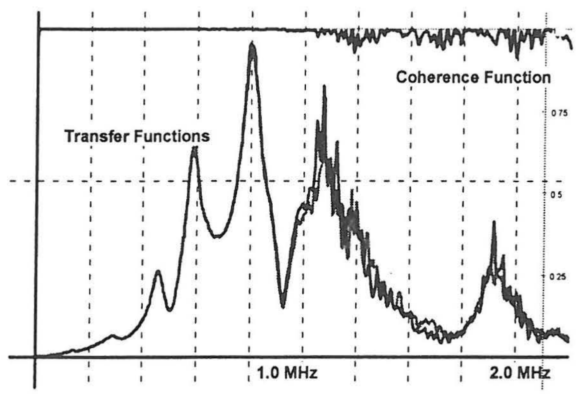


Figure 5

Coherence Function (top) and Transfer Function (bottom). This example shows that this TF is reliable up to approximately 1 MHz.

To understand the performance data of digital recorders most important to obtain best sensitivity for Transfer Functions, the mathematics are described briefly.

To receive the transfer function the frequency spectras of the recorded voltage *U(t)* and current *I(t)* signals are calculated by FFT (Fast Fourier Transformation) resulting in *U(f)* and *I(f)*. The quotient of *I(f)* and *U(f)* is called Transfer Function *TF*

$$TF(f) = \left|\frac{I(f)}{U(f)}\right| = \frac{1}{|Z(f)|}$$

where *Z(f)* reflects the impedance spectrum of the transformer under test.

Now what is the influence of technical performance data on the calculation of a TF? As we have seen at paragraph *"Frequency Spectra of High-Voltage Impulses"* amplitude of excitation *U(f)* decreases strongly towards higher frequencies dropping already at 1MHz far below the 1% level. Therefore the error in the denominator *U(f)* at small amplitudes is the most important quantity. This is generally the digitising error of the A/D (analogue to digital) converter, e.g. for a 12 bit recorder 1/4096 respectively 0.02%.

The Coherence Function (CF) is an additional tool, which, mainly in connection with the TF, provides an indication of the areas where the ingress of noise made the signal processing unreliable. The coherence function is derived from all the time domain records used in the transfer function calculation. Assuming a linear behaviour of the examined winding, and an ideal, noise-free measuring system, the coherence shall be equal to unity over all the analysed frequency band [4, 5].

A coherence function value lower than one indicates noise from interference or quantisation or a winding fault.

The coherence function $\gamma(f)$ is calculated as follows:

$$\gamma(f) = \frac{\left|G_{xy}(f)\right|}{\sqrt{G_{xx}(f) \cdot G_{yy}(f)}}, \quad G_{xx}(f) = \sum_{i=1}^{n} \frac{\left|U_i(f)\right|^2}{n}$$

$$G_{yy}(f) = \sum_{i=1}^{n} \frac{\left|I_i(f)\right|^2}{n}, \quad G_{xy}(f) = \sum_{i=1}^{n} \frac{U_i^*(f) \cdot I_i(f)}{n}$$

Gxx(f), Gyy(f):	*auto power spectrum (auto spectral density function)*
Gxy(f):	*cross power spectrum (cross spectral density function)*
n:	*number of sets*
U(f):	*frequency spectrum (complex value) of the recorded voltage waveform*
I(f):	*frequency spectrum (complex value) of the recorded current waveform*
U:*	*complex conjugate value*

Gxx(f), Gyy(f), Gxy(f) are approximated that means averaged functions.

The resulting figure γ is reel and has a value between 0 and 1. A zero means that there is no correlation between the recorded signal, e.g. when there is only noise. On the contrary a value of one indicates that the signals are correlated, e.g. for ideal non distorted and noise free measurements.

The number of records taken for the evaluation influence the uncertainty of the coherence value. For practical purposes a number of records greater or equal three should be taken for reliable analysis results.

One important requirement for the use of the coherence analysis is that all the records which are used for the Coherence Function shall be taken on the same test object and with an identical test circuit configuration. No modifications in the test arrangement (e.g. moving of dividers, changing of cables and connections) shall be performed during the tests.

Mathematically the CF is derived from all the time domain records used in the TF calculation and therefore has the same requirements from the digital recorder - mainly a good amplitude resolution.

As we have already seen during the calculation of TF, the digitising error becomes the most critical quantity. This is not very surprising as the same basis, namely *U(f)* and *I(f)*, are involved as with the TF calculation.

Oversampling and Resolution Enhancement

With DSO (Digital Scanning Oscilloscopes) the use of oversampling, resolution enhancement (or sometimes just referred to as signal processing) has become a very common tool. Some manufacturers are, by means of such processes, pushing up the amplitude resolution of their DSO's by 2, 3 or even 4 bits.

As those procedures rely on multiple measurements respectively averaging of the same measurement quantity. The final result can, assuming noise with a gaussian distribution, be improved according the following formula valid for statistical processes:

$$\Delta F_N = \frac{\Delta F_1}{\sqrt{N}}$$

ΔF_N: *uncertainty for N measurements*

ΔF_1: *uncertainty of individual measurement*

Results for some typical factors of oversampling and influence on amplitude resolution and bandwidth are shown in the following table:

Oversampling factor (N)	**gain of resol. in amplitude (bit)**	**reduction of bandwidth to**
2	0.5	0.50 $F_{Nyquist}$
4	1	0.25 $F_{Nyquist}$
8	1.5	0.12 $F_{Nyquist}$
16	2	0.06 $F_{Nyquist}$
32	2.5	0.03 $F_{Nyquist}$
64	3	0.016 $F_{Nyquist}$
general	½ $\log_2(N)$	$F_{Nyquist}/N$

Table1

Oversampling can enhance the resolution of signals in the time domain at the cost of a reduction in overall bandwidth.
However a 1 GS/s, 8 bit recorder with 500 MHz bandwidth can, supposing the right amount of noise is present, be "processed" to a 10 bit recorder with 30 MHz bandwidth.

The procedure we have described above is also called running average filtering and the simplest approach to resolution enhancement. A more sophisticated approach using weighted functions instead of a simple average allows optimisation of e.g. the impulse response respectively the behaviour at the cut off frequency. There are infinite possibilities and approaches to design such filters but two main categories: Finite Impulse Response (FIR) filters and Infinite Impulse Response (IIR) filters.

FIR filter are phase conserving. Impulse response and bandwidth can be chosen and they have no resonance or oscillation phenomena to be taken into account. They are more difficult to calculate and require more CPU power.

IIR filter are not phase conserving (apart from some exceptions) can get into resonance and show hysteresis effects. They are much easier to calculate and require little CPU power. [6].

Characteristic for an FIR filter with +2bit

Let us look a bit closer at such a +2bit resolution enhancement filter. A full lightning impulse gets smoothed significantly and high frequency noise is reduced. The peak is reduced by a fraction of a percent (see Figure 6).

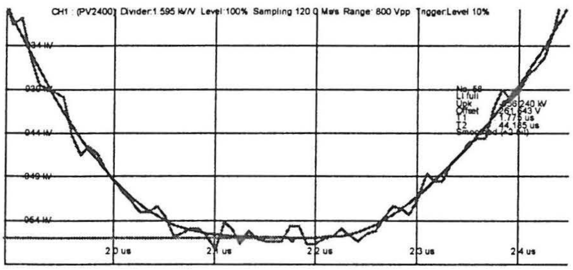


Figure 6

A full lightning impulse: raw data and signal processed +2 bit result. The +2 bit result is clearly smoother but will also suppress any high frequent oscillations e.g. in the front.

The filter characteristic of such a +2 bit resolution enhancement filter is shown below:

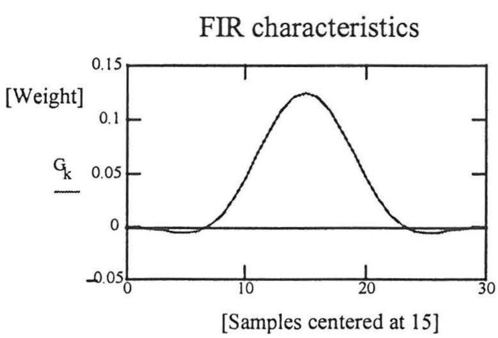


Figure 7

FIR filter for +2 bit resolution enhancement. The total filter length is 31 samples.

The corresponding frequency response graph shows that this specific filter reduces the bandwidth to approximately 6% of $F_{nyquist}$, which for a 100 MS/s Recorder ($F_{nyquist}$= 50MHz) corresponds to 3 MHz bandwidth.

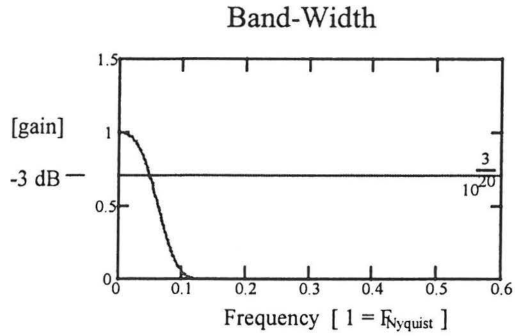


Figure 8

Bandwidth for a +2 bit resolution enhancement filter. The -3db cut-off frequency for this filter is at 6% of $F_{Nyquist}$

If we consider IEC 61083-1 and check for the required bandwidth respectively rise time (relation: $F_{bandwidth}$ = $0.35/T_{rise}$) we find the following requirement: " T_{rise} shall be not more than 0.03 T_x, where T_x is the time interval to be measured", considering the shortest LI full wave (0.84/50 µs) as time interval, a rise time of 25 ns or bandwidth of 14 MHz is required. Assuming 2 bit resolution enhancement an initial sampling rate of 0.5 GHz is required to fulfil the IEC standard - only covering LI full wave impulses.

Now what is the influence of the above described procedure if we perform Transfer Function analysis? If measurements are transformed from time to frequency domain resolution enhancement is, as we have seen above, equivalent to low pass filtering with cut-off frequencies given in Table 1. High frequency signals and high frequency noise are both suppressed. Low frequency signals and low frequency noise are not affected (see Figure 8).

Therefore oversampling or resolution enhancement is, in connection with analysis in the frequency domain, generally of no help as both, signal and noise are eliminated towards higher frequencies.

Calibration Procedure required by Standards

In many fields manufacturers of calibrated measuring equipment e.g. multi-meters or digital oscilloscopes quote a certain period during which the calibration is guaranteed, e.g. one year.

Following the standard for high-voltage test techniques, IEC 60-2, recommends that the Performance Test is repeated annually and in any case it shall be repeated at least once every five years. The Performance Check shall be made at intervals based on the recorded stability of the Measuring System as shown in the Record of Performance. IEC 1083-1 (1991) on the other hand requires a Performance Check daily before and after measurements. A good description for performing a System calibration is also given at [7].

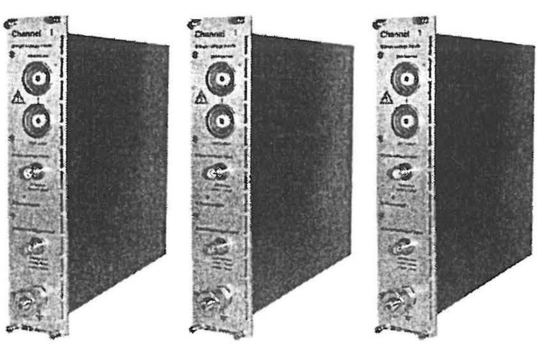
However the mentioned standards are in practise adopted differently by users - mainly because calibration procedures are cost and work intensive. With increasing accuracy of High-End Impulse Analysing Systems, calibration on a regular basis should be equally important to the user as technical performance data. What is the advantage of a highly precise measurement device if it is not calibrated?

Most users would like to perform calibrations frequently but the involved cost and time should be reduced. Therefore a new system which allows for an easier calibration procedure has been developed.

Device Architecture for easy Calibration

Many users of digital recorders do not posses a calibrator to perform the above tests. If no calibrator is available locally, the measuring device must be return the manufacturer or a laboratory capable of performing the described tests. However many test labs use their measuring equipment continuously, and interruption of testing is not possible. To reach a continuous availability for a measuring device either a second redundant system or a device with a new structure is required.

To address this problem a new so called **Channel Insert System** has been developed. The system is based on individual measuring channels which include for each channel all accuracy sensitive elements in **fully closed** aluminium casings (see Figure 6). Inserting / exchanging a channel is performed in less than 30 seconds and absolutely no work other than putting the Channel in the relevant slot is required.



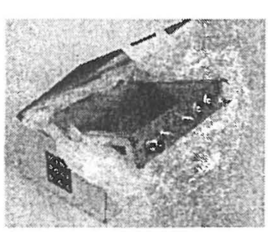


Figure 6

Individual Channel Inserts, small compact and easy to send back for calibration.

It has been proven with a series of comparisons that accuracy of a system which strictly follows the Channel Insert design philosophy is only dependant on the accuracy of the Channel Inserts. Measuring an impulse is entirely performed by the Channel Inserts which pass a completely digitised impulse measurement via bus system to the CPU.

Caution:

If individual channels are not covered by a complete casing, mechanical work, sensitive cable connections or other adjustments have to be performed, it is most likely that the reliability of an exchange will suffer. E.g. if individual measuring cards (no complete casing) and/or mechanical work is required, an exchange by non-experienced users can influence the accuracy or damage such a channel. We therefore strongly recommend that non of the above work will be accepted if a channel calibration, as described later, is performed.

A schematic block diagram of a system featuring Channel Inserts is shown on the next page in Figure 7.

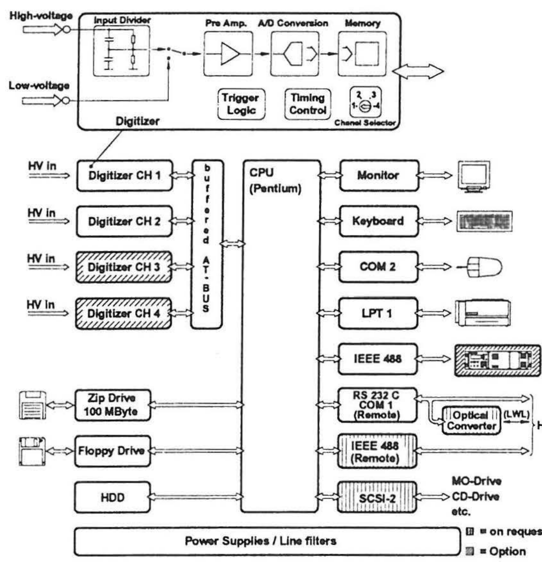


Figure 7

System Architecture of new digital recorders featuring the Channel Insert System. All accuracy sensitive elements are built into individual Channel Inserts. Important: the system is operational with any number of Channel Inserts.

Additional software is only responsible for analysis, display, saving and reporting of the measured impulse. The software can be checked separately for its conformity with the so called software validation IEC 61083-2. A software validation passes the same kind of measurement data to CPU for evaluation as the Channel Inserts do for real measurements. Therefore calibrating Channel Inserts and performing software validation ensures overall system performance.

Proposed Calibration Procedure

With such a **Channel Insert System** a so called "circular calibration" is proposed which will be explained using a system with 3 channels.

- A new 3 channel system is installed
- Channel 1 is sent back for calibration
- Channel 2 is sent back for calibration
- Channel 3 is sent back for calibration
- Channel 1 is sent back for calibration

The calibration interval could e.g. be scheduled once every year as foreseen in the new draft of IEC 61083-1 (1998). Additional comparison should be performed by parallel measurements using the recently calibrated channel (reference) and one of previously calibrated channels (channel under test). The channels have to agree in time and amplitude parameters within the limits required by the standard IEC 61083.

Using the Channel Insert System with the above described "circular calibration procedure" has the advantage of a fully operational system at the cost of one additional channel. This is approximately 60% of the cost compared to a fully operational conventional system which involves a second redundant device.

Conclusions

Technical performance data of digital recorders for measurements in high-voltage impulse tests have been continuously improved during the past few years. Especially in connection with sophisticated analysis such as Transfer and Coherence Function the amplitude resolution of the analogue to digital converter seems to be the most important quantity.

To ensure initial accuracy and technical performance data on a long term, frequent calibrations of digital recorders are required. The aim to hierarchically trace uncertainties from the user to national standards as well as the influence of the quality standard ISO 9000 are pointing in this direction. The Channel Insert System described in this paper will allow for a simple calibration procedure and a continuous device availability at a very economical price.

References

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