# Time-domain EMI Measurements at Open Test Sites using Ambient Noise Cancelation Methods

***Arnd Frech[1], Stephan Braun[1], Amer Zakaria[1], Peter Russer[1]***

[1]Institute for High-Frequency Engineering, Technische Universität München,
Arcisstraße 21, 80333 Munich, Germany
frech@tum.de, stephan.braun@tum.de, azakaria@ee.umanitoba.ca, russer@tum.de

## Abstract

Time-domain as well as frequency domain adaptive filter algorithms have been investigated and compared in respect of their capabilities of suppressing ambient noise during time-domain measurements of electromagnetic interference at open test sites. The fundamentals and the application of the most advantageous algorithm for time-domain EMI measurements is described in this paper. An advanced digital signal processing technique for fast measurements of electromagnetic interference from electronic devices is presented. Measurements of electromagnetic interference of a device under test can be performed in a test site polluted by electromagnetic ambient noise by using digital signal processing. Frequency domain adaptive filtering using the overlap-save method is applied and measurement results of a real device under test are carried out showing the successful cancelation of ambient noise at an urban test site in a broad frequency range up to 1000 MHz.

## 1. Introduction

Traditionally measurements of electromagnetic interference (EMI) are performed in shielded anechoic chambers or at open area test sites. While anechoic chambers are very large and expensive to build, open area test sites typically have to be located in remote areas far from cities where no electromagnetic ambient noise like television, radio broadcasting, and mobile phone stations distorts the EMI measurements. A system suppressing ambient noise during the measurement in frequency domain has been presented in [1]. It is a two channel system based on conventional coherent EMI receivers carrying out the measurements in frequency domain. The drawback of frequency domain EMI measurement systems is the long measurement time. The time-domain electromagnetic interference (TDEMI) measurement system provides a novel approach for considerable reduction of the measurement time [2]. This system measures radiated emissions in time-domain and calculates the spectrum in a frequency band up to 1 GHz. In this paper algorithms for ambient noise cancelation in TDEMI measurement systems are presented. These algorithms allow to combine the advantage of measurement time reduction of a TDEMI measurement system with the ability to perform EMI measurements in the presence of ambient noise. Fig. 1 depicts the block diagram of a two antenna TDEMI measurement system with ambient noise cancellation. This system is able to perform EMI measurements in time-domain in presence of ambient noise is shown without any prior knowledge about the ambient noise signal and without the necessity of turning the device under test (DUT) off. The system consists of two ultra broadband biconical logarithmic periodic antennas. While one antenna receives the emission of the DUT and the ambient noise signal the other antenna is used as a reference antenna receiving only the ambient noise. The signals are low-pass filtered and digitized by an analog-to-digital converter (ADC) unit. The acquired signals are processed digitally on a personal computer to perform the cancelation of the ambient noise. The ambient noise signal received with the first antenna is correlated to the ambient noise signal received with the second antenna, whereas the emissions of the DUT are uncorrelated to these ambient noise signal. Digital adaptive filters are used to filter the signal of the antennas and to perform a cancelation of the ambient noise. Adaptive filters can adjust their characteristics to exploit these relationships when the ambient noise is changing.

## 2. The Time-domain EMI Measurement System

Spectral estimation is achieved by recording the complete signal at once and performing the fast Fourier transformation (FFT). This method allows to reduce the measurement time by several orders of magnitude [2]. As shown in Fig. 1, the input EMI signal is received via an ultra broadband antenna and digitized by an AD converter unit. The acquired data is processed via digital signal processing (DSP) unit, the amplitude spectrum is calculated and visualized.

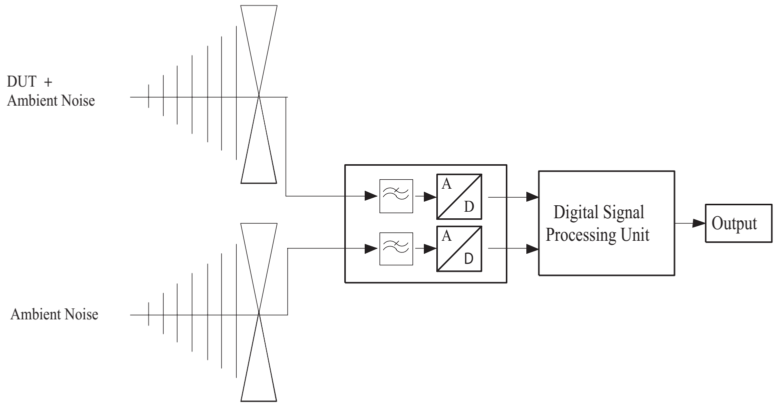


**Fig. 1** Block diagram of ambient noise cancelation system.

The digital signal processing is either performed by field programmable gate arrays (FPGA) or by a personal computer. Thereby in both cases the discrete spectral estimation is achieved via the discrete Fourier transformation (DFT), which formula is given as:

$$X[k] = \sum_{n=0}^{N-1} x[n] e^{\frac{-j2\pi kn}{N}}. \tag{1}$$

According to the CISPR standards the calculated spectrum must have a 6 dB bandwidth of 9 kHz, 120 kHz or 1MHz. A short time FFT (STFFT) is applied to each data block one by one. The collected data sequence of length $N$, depending on the 6 dB bandwidth, is multiplied with a Gaussian window function $w[n]$ which models the IF filter of an EMI receiver. The STFFT calculates as follows:

$$Z[k,\tau] = \sum_{n=0}^{N-1} x[n-\tau] w[n] e^{\frac{-j2\pi kn}{N}}, \tag{2}$$

where $\tau$ describes the dependency on time and with the STFFT we obtain a spectrogram showing a discretization in frequency and time. The signal at each discrete frequency is processed by the detector mode**.**

## 3. Ambient Noise Cancelation by Adaptive Filtering

The ambient noise cancelation system, shown in Fig. 2, exploits the correlationships between the signals detected by the two antennas. Besides the electromagnetic noise present in the environment antenna 1 receives also the emissions from the DUT $d_1[n]$, whereas antenna 2 is isolated from the DUT and only receives the ambient noise signal $a_2[n]$. Assuming that the detected emissions from the DUT are uncorrelated to the ambient electromagnetic noise, and the ambient noise detected by the two separated antennas are correlated to each other adaptive filter methods can be applied to suppress the unwanted noise. The ambient noise signal $a_2[n]$ is digitally processed by an adaptive filter and subtracted from the signal $s_1[n]$ received by antenna 1. Depending on the result e[n] of this superposition the discrete coefficients of the filter are adjusted once more by least mean square (LMS) algorithm and $a_2[n]$ is weighted such that the output e[n] of the system is ideally representing the emissions of the DUT. The complex LMS algorithm can be found in detail in [3]. Providing a better stability for higher order filters filter with finite impulse response (FIR) are used.

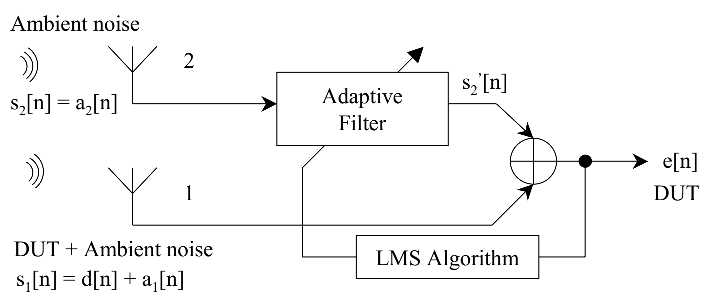


**Fig. 2** Block diagram of the ambient noise cancelation.

## 3.2 Frequency Domain Adaptive Filtering

To reduce the computational effort frequency domain adaptive filters (FDAF) can be applied [4]. Instead of a sample by sample approach FDAF algorithms are based on a block by block processing by collecting blocks of the sampled input data and then performing the adaptation process with fixed filter coefficients for each collected block of samples. The FFT is applied on the collected blocks for fast convolution which allows an efficient implementation of the filter algorithm performing the adaptation process by computing the new filter coefficients in the frequency domain. The filter output e[n] can be either computed in the time domain or in the frequency domain.

### 3.2.1 Overlap-save Method

The algorithm which is used for ambient noise cancelation is the overlap-save FDAF algorithm. The block diagram of this algorithm is shown in Fig.3. The sampled input signals are $s_1[n]$ which contains the DUT signal in addition to the ambient noise and ambient noise signal $a_2[n]$ measured at the reference antenna where n describes the sampled time index. Two blocks of size M of the two input signals are collected, where M is the order of the adaptive filter. A new collected block of $a_2[n]$ is concatenated with a previously collected block. The 2M block of $a_2$ is transformed via FFT to the frequency domain block $\mathbf{A}_2[k]$ and afterwards sample-by-sample multiplied with $\mathbf{H}[k]$, where $\mathbf{H}[k]$ is the frequency domain transformation of the impulse response of filter $\mathbf{h}[k]$ of size M where k is the block number index. The output signal of the multiplication $\mathbf{A}'_2[k]$ is transformed to the time-domain via inverse FFT (IFFT) and the first M samples are discarded while the last M samples are saved in the vector $\mathbf{a}'_2[k]$. The first M samples are discarded because they are the unwanted results of circular convolution, while the last M samples are a subset of the resulting output of the linear convolution of $\mathbf{a}_2[k]$ and $\mathbf{h}[k]$. Afterwards $\mathbf{a}'_2[k]$ is subtracted from the collected block $\mathbf{s}_1[k]$ resulting in the signal block $\mathbf{e}[k]$ which is the desired signal of the DUT. Besides being the output of the algorithm and therefore represents the emissions of the DUT, this signal will be used to update the filter coefficients as follows. M zeros are padded to the beginning of the error signal block. The new 2M block is transformed to the frequency domain via FFT. The frequency domain signal $\mathbf{E}[k]$ is multiplied by the complex conjugate of $\mathbf{A}_2[k]$. Performing the IFFT of this multiplication output, the first M terms are attained while the last M terms are discarded. The first M terms describe a subset of the linear correlation process between $\mathbf{a}_2[k]$ and $\mathbf{e}[k]$, while the last M terms are unwanted results of circular correlation. The attained M samples are saved in vector $\nabla[k]$, where this vector is called the block gradient estimate or the gradient constraint. The gradient constraint $\nabla$ [k] is padded with M zeros and transformed to the frequency domain via FFT resulting in $\mathbf{\Psi}$. Finally, the adaptive filter coefficients are updated using the following criterion:

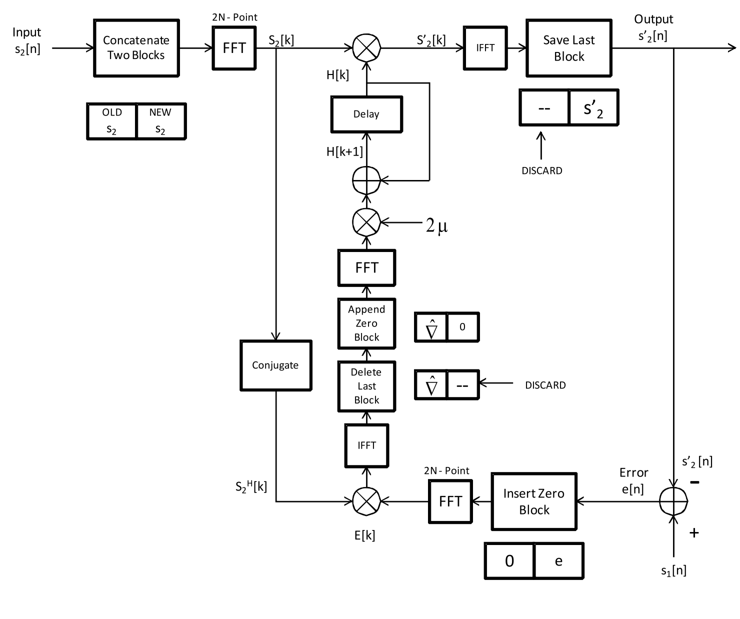


**Fig. 3** Block diagram of the overlap-save FDAF method

$$\mathbf{H}[k+1] = \mathbf{H}[k] + 2\mu\mathbf{\Psi}\ ,\ dim\{\mathbf{H}[k+1]\} = 2M - by - 1 \tag{3}$$

Where μ is the step-size and determines the rate of convergence and stability of the adaptive filter algorithm.

## 4. Measurement Results

Using the overlap-save method, in this section the results of the ambient noise canceling algorithm are presented which has been investigated during several measurements. In Fig. 4, the spectrum of the final output of the algorithm is shown for DUT turned off. The filter order was selected to 8192. In Fig. 4 it can be observed that the algorithm succeeded in canceling the ambient noise in the FM band, GSM band, some line spectra between 100 and 200 MHz, and between 400 and 500 MHz. The suppression is about 20 dB to 30 dB. Stationary signals are suppressed up to 40 dB. Moreover, the algorithm was able to attenuate the ambient noise signal in the 210 to 230 MHz band, the DVB-T band (575 to 590 MHz), and 680 to 760 MHz band with an attenuation of about 10 dB. For a portable computer as DUT which now was turned on, the resulting frequency spectrum is shown in Fig.5. The measurement detector was set to the maximum peak mode, and the spectrum was calculated with an 120 kHz IF bandwidth as required by CISPR 16-1-1 for band C, and D. Analyzing Fig. 5, it can be noticed that the algorithm was successful in canceling the ambient noise in the FM band, GSM band, and other frequency components along the spectrum and the DUT signal was preserved. Due to multi-path propagation, e.g. if a minimum occurs at one receiving antenna, the correlation between the ambient noise received at antenna 1 and antenna 2 can be reduced. A solution to improve the performance of the cancelation would be to determine a separate step size for every frequency bin in the spectrum of the ambient noise.

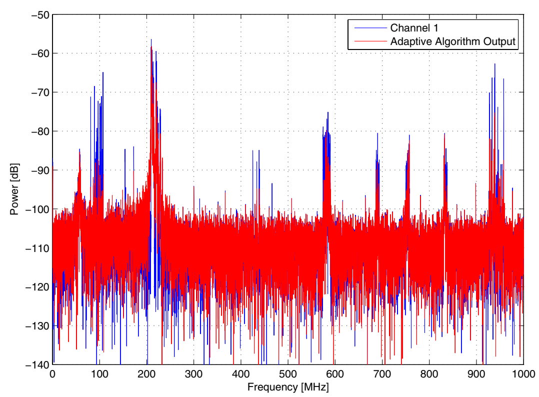


**Fig. 4** DUT off, single shot mode.

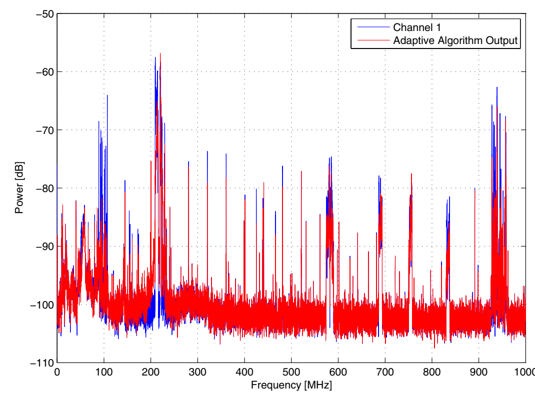


**Fig. 5** DUT (Laptop) on, maximum peak detector mode.

## 5. Conclusion

Ambient noise cancelation applied on a TDEMI measurement system has been presented. The frequency domain adaptive filter algorithm using the overlap-save method has been successful in suppressing and canceling ambient noise signals over a broad frequency range. The FM radio broadcasting band and the GSM band were suppressed with an attenuation about 30 dB. Limitations of the algorithm have been observed due to the non-continuous processed data stream of the data acquisition unit, especially for non-stationary ambient noise signals.

## 6. Acknowledgments

The authors would like to thank the Deutsche Forschungsgemeinschaft (DFG) for supporting this project.

## 7. References

1. T. Shinozuka, and A. Sugiura, "Reduction of Ambient Noise in EMI Measurement," *IEEE International Symposium On Electromagnetic Compatibility Digest*, Nagoya, Japan, September 8-10, 1989, pp. 24-28.
2. S. Braun, M. Al-Qedra, and P. Russer, "A novel Realtime Time-domain Measurement System based on Field Programmable Gate Arrays," *17th International Zurich Symposium On Electromagnetic Compatibility*, Singapore, Feb. 27 - March 3, 2006, pp. 501-504. Best Student Paper
3. B. Widrow, J. McCool, and M. Ball, "The Complex LMS Algorithm," *Proceedings of the IEEE*, vol.63, no.4, April 1975, pp. 719-720.
4. J.J. Shynk, "Frequency-domain and Multirate Adaptive Filtering," *Signal Processing Magazine, IEEE*, vol.9, no.1, Jan 1992, pp. 14-37.