ULTRAWIDEBAND TRANSMISSION IN PHYSICAL CHANNELS: A BROADBAND INTERFERENCE VIEW

HENRYK LASOTA, IWONA KOCHAŃSKA, ROMUALD MAZUREK

Gdańsk University of Technology
ul. Narutowicza 11/12, 80-233 Gdańsk, Poland
hlas@pg.gda.pl, iwona.kochanska@eti.pg.gda.pl, rmaz@wp.pl

The superposition of multipath components (MPC) of an emitted wave, formed by reflections from limiting surfaces and obstacles in the propagation area, strongly affects communication signals. In the case of modern wideband systems, the effect should be seen as a broadband counterpart of classical interference which is the cause of fading in narrowband systems. This paper shows that in wideband communications, the time- and frequency-domain approach of the linear time-invariant (LTI) systems theory does not fully reveal the significance of channel-induced distortions. A survey of MPC interference phenomena is presented, with time-evolving impulse responses and space-dependent notransfer functions. Linear prediction technique based on the autoregressive model is successfully applied as a complementary tool for speech signal distortion analysis in public address systems.

INTRODUCTION

In wireless acoustic and electromagnetic communication systems the signal crosses space in the form of a wave that propagates through a physical medium that fills the communication channel. Phenomena occurring in underwater acoustic communication (UAC), wideband and ultrawideband radiocommunication (WBC/UWBC), and public speech communication (PSC) channels are qualitatively similar but quantitatively different.

Wireless communication takes place, in general, in a multipath propagation environment, limited by the bottom and the water surface (UAC), the earth's surface, troposphere layers, and buildings (WBC) or premises' walls (UWBC and PSC). Scattering on objects and reflections on borders result in the effect of constructive and destructive interference leading to both the signal waveform's and the spectrum's deformations. The specific nature of interference depends on the relative position of the transmitter and the receiver, as well as the reflecting or scattering objects. In these conditions every space location of the receiver represents a specific realization of the communication channel, and UAC, WBC, and UWBC systems are designed so as to avoid as much as possible the adverse effects of interference.

In both kinds of acoustic channels (air and water) there is a significant wave attenuation growing with frequency squared. For the purpose of this survey, however, we will neglect the influence of attenuation on communication signals, adopting an all-pass model of a physical medium, as is usually made in the case of electromagnetic waves in air with negligible attenuation.

1. UNDERWATER ACOUSTIC COMMUNICATIONS (UAC)

In underwater acoustic channels (UAC) multipath propagation is determined by channel geometry. It depends on the UAC system range and frequency. Moreover, strong refraction effects affect the multipath phenomenon. The UAC channels are naturally sparse, meaning that most channel energy is concentrated in a few multipath components (MPCs). Large delay spread is observed and significant Doppler influence, hence they fall into the category of doubly (time- and frequency-) spread channels. In the case of a shallow water channel, the coherence bandwidth does not exceed a range of tens of Hertz [1]. In the presence of significant surface waves, the MPCs associated with surface scattering fluctuate rapidly over time, in the sense that the magnitude, arrival time, and Doppler shift of each MPC, all change dynamically. This restricts communication systems' design options. When taking into account the limitations due to the physical properties of underwater propagation medium and the technology of ultrasonic transducers, it becomes clear that underwater applications should be rather limited to WB systems only.

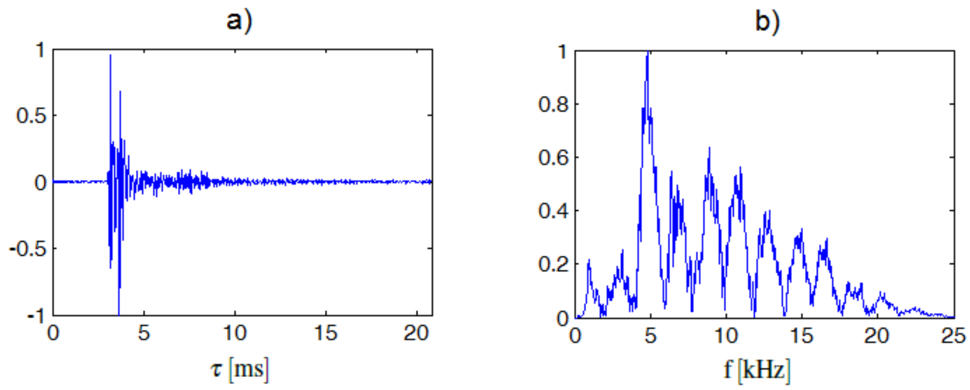

Fig.1. Impulse response (a) and transfer function (b) estimates of a shallow UAC channel.

Modern UAC implements the techniques developed originally for wideband radiocommunication techniques (WBC). In communication systems with the orthogonal frequency-division multiplexing (OFDM) technique, multiple narrowband signals of mutually orthogonal carriers are transmitted simultaneously. In systems implementing the frequency-hopping spread spectrum (FHSS) method, a narrowband waveform carrier is being switched in a pseudorandom manner. Finally, in the case of direct-sequence spread spectrum (DSSS) systems, the waveform is spread in both time and frequency domains by pseudo-random (PN) sequences. All these WB transmission techniques allow the communication system bandwidth to be filled with narrowband signals carrying the information. In OFDM and FHSS systems the signal spectrum is designed to meet the coherence bandwidth requirements, making the communication signal free from interference-induced distortions [2].

2. MODERN RADIOCOMMUNICATION SYSTEMS

A free-space wireless channel is inherently broadband. Historically, however, it has mostly carried quasi-monochromatic waves of broadcast communication systems. Such signals have been used in narrowband (NB) communication systems with classical (analog) modulation schemes. More complex signalling and modulation schemes, i.e. OFDM, FHSS, and DSSS techniqes, are used in modern communication systems, such as terrestrial digital video broadcasting systems (DVB-T), long term evolution (LTE) cellular systems, or wireless local area networks (WLAN), occupying a wide frequency band (WB). In the physical layer of OFDM and FHSS based systems, however, a superposition of modulated carrier sinusoid waveforms, i.e. narrowband signals is transmitted. Thus, the multipath propagation induced interference has a narrowband character only. A specific receiver architecture called RAKE, takes advantage of the narrowband interference phenomenon, increasing the energy of the received signal before detection [3].

According to the U.S. Federal Communications Commission (FCC), UWB refers to radio technology with a bandwidth exceeding the lesser of 500 MHz or 20% of the arithmetic center frequency. The term: Ultra Wide Band (UWB) signal signifies a number of synonymous terms such as: impulse, carrier-free, baseband, time-domain, nonsinusoidal, orthogonal-function and large-relative-bandwidth radio/radar signals [4]. Spread-spectrum communication systems using ultra-short impulses have seen a renewed interest because of their fine resolution in delay to the order of a tenth of nanosecond though at the cost of an ultra wide frequency band. Low transmission power and large bandwidth together render the power spectral density of the transmitted signal extremely low, which allows the frequency-overlay of a UWB system with other existing radio systems [5].

Several possible monocycles can be used in UWB systems: Gaussian pulse, Gaussian monocycle and Scholtz's monocycle. The multipath propagation induced interference has a broadband character. In Fig.2 an impulse response of the Saleh-Valenzua radiocommunication channel model is shown [6]. Numerous MPCs determine a significant distortion of the transfer function. The transfer function, however, has been calculated on the assumption of channel excitation with continuous waveforms, lasting long enough to calculate (or measure) the influence of the consecutive MPCs interference on the measurement signal. It is equivalent to calculating the Fourier transform of the respective IR with the assumption of "exciting" the channel with a superposition of harmonic signals of time duration not less than the channel delay spread.

Fig.2. The Saleh-Valenzua impulse response model in an indoor channel (a) and a corresponding steady-state transfer function of UWB radiocommunication system (b) [6].

In UWB systems the shortness of communication pulses is the key to the mitigation of possible influence of broadband interference. If the monocycle time duration is shorter than the delay between consecutive MPCs, the UWB system does not suffer from the multipath propagation problem [6].

3. PUBLIC SPEECH COMMUNICATIONS

Speech communication, including the natural one with no use of technical means, should be seen as the transmission of signals carrying a specifically coded message, reaching a listener via a less or more difficult channel. The signal can be emitted directly by the speaker, a single loudspeaker or via a multi-speaker PSC system such as a public address system (PAS), a sound system for emergency purposes (SSEP) or other sound reinforcement system (SRS). The listener analyses the signal and decodes the message based on a specific protocol that is, in this case, the language known to both sides of communication. The clarity of the voice message is the basis of its intelligibility. Clear speech means one that is composed of well recognizable elements, not distant from pattern elements well known to both parts. These speech elements are syllables and phonemes.

The speech messages transmission in PSC system means, in fact, an UWB (or even ExtraWB) communication in tough acoustic channels. The reasons of sound modifications in the hearing space recognized in acoustic standards [7] are: nonlinear distortions introduced by electroacoustic equipment; linear distortions due to irregular frequency characteristics of electroacoustic transducers; and reverberation phenomena due to sound reflection from auditorium walls, other limiting surfaces, and scattering from furniture elements. No attention seems, however, to be paid by sound engineers to the phenomenon of interference in the field of multiple loudspeaker systems, though this mechanism is another source of significant signal deformation in auditory rooms, strongly altering overall PSC quality [8], [9].

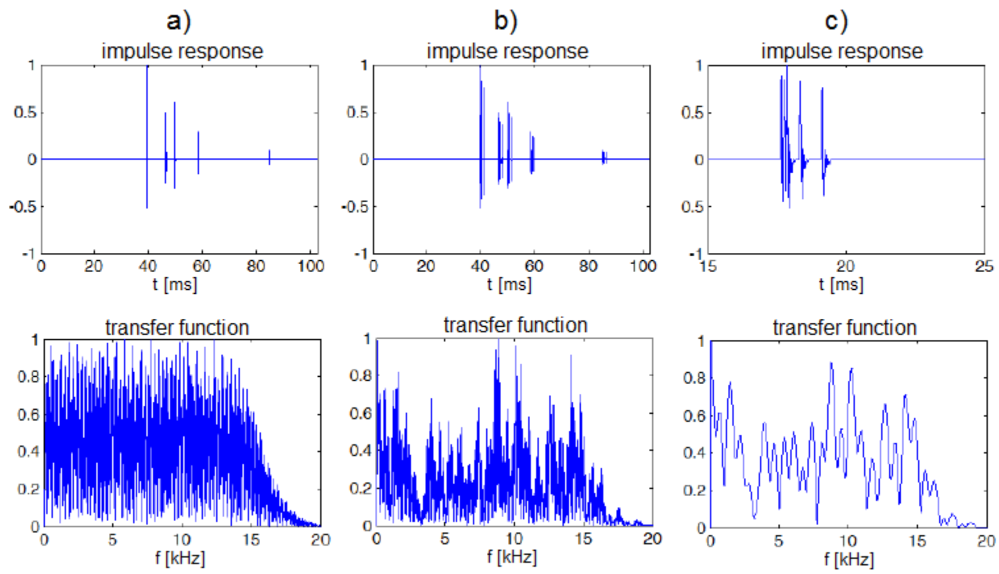

Fig.3. Impulse responses and transfer functions of: a) single loudspeaker in reverberation room, b) 5-loudspeaker array in reverberation room, c) 5-loudspeaker array with reverberation tails suppressed.

Fig. 3a) presents a transmission channel's IR between a single loudspeaker and a listener in a reverberating environment. Fig. 3b) shows the IR of the transmission channel between a 5-loudspeaker linear array and a listener situated at the same listening point. In the first case the TR has an uniformly random character in the frequency band of the system, while in the second one there is an additional variability indicating the occurrence of an additional factor associated, as it appears, with source multiplication. Fig. 3c) shows the IR and TF corresponding to a simulation experiment with the 5-loudspeaker system in free space, with mere direct signals from the sources and the reverberation tails digitally suppressed (notice the extension of the time scale). It can be remarked that the IR of Fig. 3b) is the time convolution of the IRs of Figs 3a) and 3c), and the TF of Fig. 3b) is a product of the respective TFs.

The latter result indicates the presence of an effect that is additional to common reverberations, due to distance differences separating the listening point from the consecutive sources emitting wide frequency spectrum signals. It is clear that the linear superposition of time-shifted broadband waves of the same form and slightly different magnitudes is accompanied by constructive and destructive interference leading to a deep modification in the received signal waveform and its frequency spectrum. Subsequent frequency components of the signal are alternately attenuated or amplified, occurring in the irregular shape of the PSC system transfer function.

The distribution of sources in a PSC systems is, from a listener's point of view, mostly irregular, say random. In the general case for multiple source systems, analytical functions describing the distribution of the acoustic field cannot be determined and it is not possible to use the concept of the directivity pattern [10]. A listening point is usually placed in the area between the radiating elements and there is no zone in which far-field requirements could be fulfilled. In the whole frequency range the situation corresponds to a near zone in which respective transfer functions are highly dependent on the listener's position. Hence it is worth noting that, in fact, every space location of the listening area represents a specific realization of the source-listener communication channel. This results in a quasi-infinite, contiguous set of individual channels [8], [11].

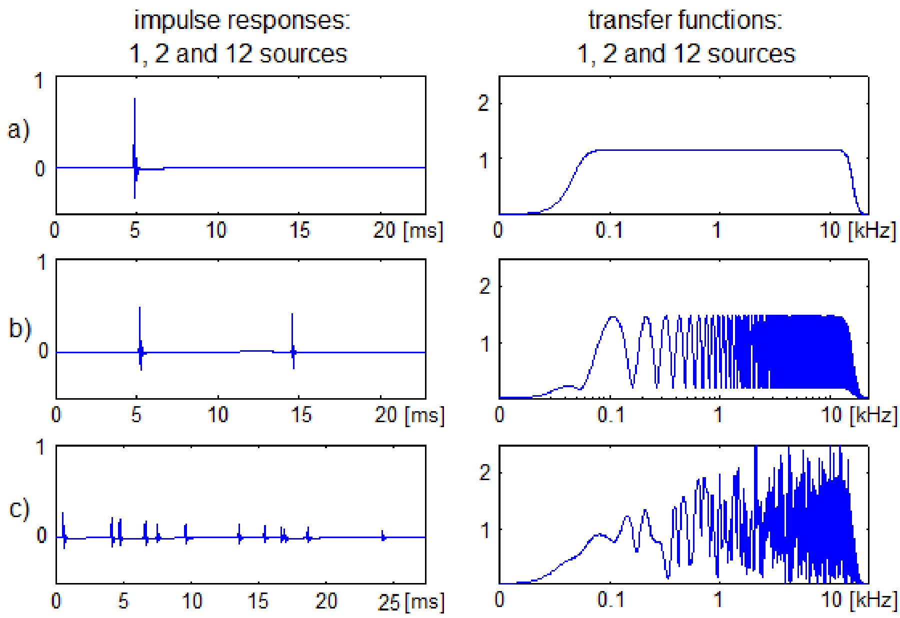

Fig.4. Impulse responses and transfer functions of simulated sound reinforcement system: a) one loudspeaker, b) two loudspeaker, c) twelve loudspeaker configurations [8].

Fig.4. presents the results of a simulation experiment. The IRs of Fig. 4a) are related to the listening area of a hypothetical auditory room of dimensions 10 m x 7.5 m x 4 m (no reflections, flat transfer functions in the frequency band from 50 Hz to 15 kHz). In the case of multiple sound sources, the corresponding TFs have been calculated as the Fourier transforms of the IRs composed of the individual loudspeakers' IRs, properly time-shifted and amplitude-scaled. Strong constructive and destructive interference effects appear alternately in contiguous frequency bands. When two sources are active, broadband interference can be observed in a

pure form (Fig. 4b), while twelve loudspeakers produce highly irregular interference effects in the whole frequency band (Fig. 4c).

5. SPEECH PHONEME “CODE” – LINEAR PREDICTION

For the measurement of distortion of speech signals in the listening area the linear prediction technique has been applied, assuming that the speech waveform can be modeled as an autoregressive (AR) process of the order of $L = 10$ [12]. Under this assumption the speech signal can be described as a response of all-pole filter of transmittance $H(z)$ for laryngeal tone excitation:

$$H(z) = \frac{G}{1 - \sum_{l=1}^{L} a_l z^{-l}}, \tag{1}$$

where a_l, $l = 1, \ldots, L$, are the linear prediction coefficients and G is the gain of the system. A set of coefficients $\{a_l\}$ is computed from Durbin's recursion, on the basis of the autocorrelation function $R_x(k)$ of the speech signal $x(t)$ [3], [13].

A so-called LPC-10 spectrum being the frequency response of the filter Eq. (1) of the order of 10, estimates the resonant characteristics of the vocal tract. Fig. 5 shows the LPC-10 and FFT spectra of the “o” phoneme original waveform and the one measured in a given listening point in space insonified by the 5 loudspeaker SC system. It can be clearly seen that the LPC-based spectrum estimate reflects better the perception and differentiation of a phoneme by a listener than the FFT-based high resolution spectrum representation. At the same time, strong narrowband deformations easily seen in the FFT spectrum, are less accentuated in the LPC estimate, exactly as is observed in human perception [9].

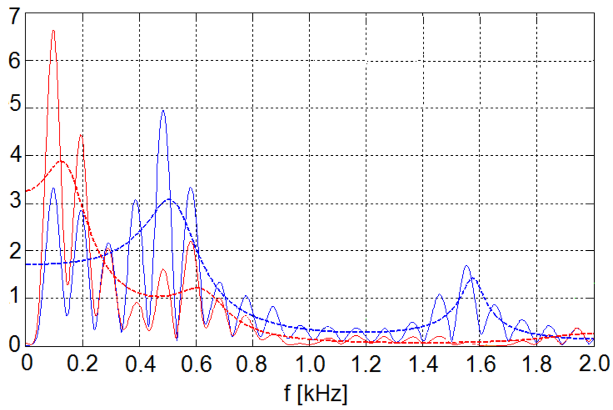

Fig.5. LPC-10 spectra (dot lines) and FFT spectra (solid lines) of Polish phoneme “o” waveforms: undistorted (red lines) and measured (blue lines) in listening point of space insonified by a 5 loudspeaker PSC system [12].

Polish phonemes recorded in noiseless conditions have been convolved with impulse responses measured at various hearing points situated in an auditorium of dimensions of 10 m x 7.5 m x 4 m, insonified by 12 loudspeakers regularly distributed beneath the ceiling (3 rows x 4 loudspeakers). Phonemes have been analyzed in the 50-4000 Hz band that covers 4-5 basic vowel formants of the decisive role in speech recognition processes. In Fig. 6 row a) shows the IRs of each configuration, row b) presents its TFs, and row c) presents the vowel spectrum estimate calculated with the LPC-10 algorithm.

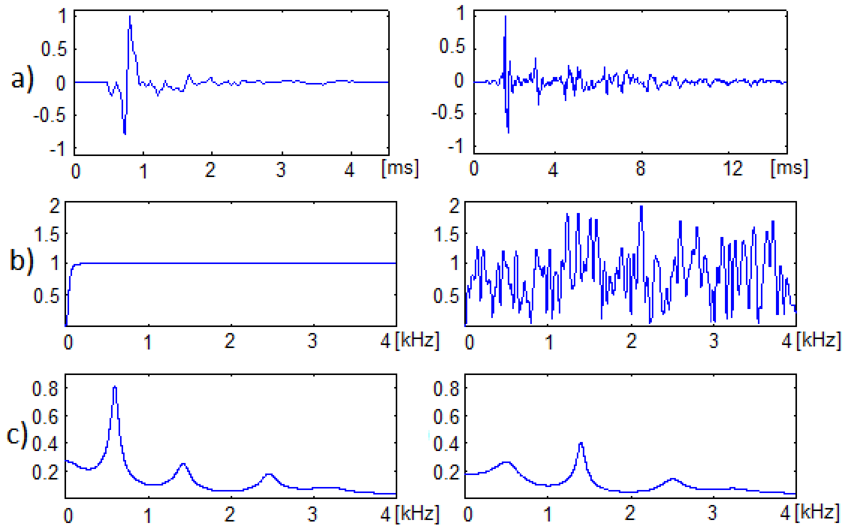

Fig.6. Impulse responses (a), transfer functions (b), and LPC-10 spectrum estimates of a Polish phoneme "e" (c) measured in an auditory room insonified with a single-source (left) and a twelve-source PSC system (right) [9].

The measurements confirm the influence of broadband interference on overall system performance. It is clearly seen in the deep modification of the TF of 12 loudspeaker PCS system compared with the TF of single loudspeaker insonification. The LPC estimate analysis shows strong changes in the shape, level and even frequency of formants in comparison with those of original phonemes. Additionally, relative levels of subsequent formants change. As a result, the formant dominating in the original phoneme at some hearing points, can become weaker than the neighbouring one. At some points, the TF irregularity can result in a strong attenuation of the dominating formant. Further studies have shown that in some cases, new "artificial" formants appear, e.g. on the slope of the eliminated original one. In consequence, the resulting formant adopts a new frequency that can disturb the phoneme intelligibility, making it similar to some other one. Phoneme distortion is thus interpreted as a change in the voice generation mechanism, similar to a possible change of the speaker.

6. CONCLUSIONS

The review of the paper is based on the authors' experience in such different applications as speech communication (SC) systems (PAS, SSEP), underwater WB acoustic communications, and outdoor and indoor UWB radiocommunications. The presence of multipath components in received signals and the width of their spectra are common points of these systems that can be seen as a kind of generalized interference reducing the transmission performances of acoustic and wireless communication systems. Echo and reverberation arising in all these systems as a result of a multipath propagation, manifest themselves in the transfer functions of the channel as irregularities of an accidental nature. In SC systems MPC superposition occurs during all the phoneme phases, often changing its character.

So-called wideband (WB) communication systems, such as modern cellular and wireless network systems, as well as modern UAC systems, transmit in the physical layer parallel narrowband signals, generated as modulated sinusoid waveforms. The UWB technique, known as impulse radio, develops new transmission methods for indoor applications in buildings and small premises. It employs truly wideband signals designed to be so short as to mitigate the broadband interference phenomenon. The speech signal, however, is long enough to be unconditionally subject to the interference phenomenon, especially in the case of PAS using a plurality of synchronous loudspeakers. The identification of the channel influence on wideband communication signals requires a fresh approach based on the parametric identification of dynamic processes. New algorithms are worth developing and testing, combining classical time and frequency domain signal processing with autoregression methods. The LTI-based view with its transfer functions, although not directly applicable to modern communication problems, may be helpful in forming system-design intuition in particular situations, revealing possibilities and constraints of concerned physical channels with dominating reverberation, reflections, or multi-source interference.

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