Implementation of a Jakes Channel Simulator for Multiple-Input-Multiple-Output Wireless Systems

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Background

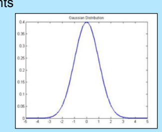
The wireless link between two communication terminals – what is called a channel – is relatively unprotected and vulnerable to factors including noise, attenuation, delay, Doppler effect, reflection, and interference. Thus, the strength of the received signal can be a function of many, or practically infinite factors. Due to the complexity of the environment of a channel, it is practically impossible to predict the signal strength at any given location and time. Therefore, statistical methods are widely adopted to model the fading channels, one of which is the Jakes' fading model. The goal of this project is to develop a tool that simulates the Jake's fading model in an efficient and easy-to-use manner. Such a tool will be useful in the evaluation of systems and algorithms used in wireless communications.

What is a Fading Channel?

- A channel resulting from a moving receiver on a random path
- Varying signal levels as a consequence:
  - Gaussian distribution for real and imaginary components

Each of the real and components of the channel are distributed in a Gaussian manner with zero mean , described by:

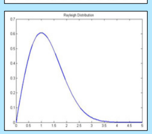
$$y=\frac{1}{\sigma\sqrt{2\pi}}e^{-\frac{x^2}{2\sigma^2}}$$



  - Rayleigh distribution for amplitudes

The resulting amplitudes of the channel are distributed in a Rayleigh manner, described by:

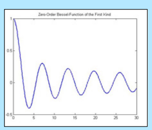
$$y=\frac{x}{b^2}e^{-\frac{x^2}{2b^2}}$$



  - Time-Domain Correlation

According to Jakes' model [1], The channel coefficients will have an auto-correlation defined by the following formula:

$$R(\tau)=E\{h(t)h^*(t-\tau)\}=J_0(2\pi f_D\tau)=J_0\left(\frac{2\pi f v\tau}{c}\right)$$



, where $R(\tau)$ is the autocorrelation at the lag of τ, J_0 is the zero-order Bessel function of the first kind, f is the carrier frequency, v is the speed of the receiver, and c is the speed of light.

Available Methods

1. Sum of Sinusoids

Proposed by Jakes [1], this method involves summing multiple sinusoidal signals in time, which will result in a deterministic output with the expected distributions and autocorrelation function. The following formula describes the summation using exponential form:

$$T(t)=\frac{E_0}{\sqrt{N}}\left\{\sqrt{2}\sum_{n=1}^{N_0}\left[e^{i(\omega_m t\cos(\alpha_n)+\phi_n)}+e^{-i(\omega_m t\cos(\alpha_n)+\phi_{-n})}\right]+e^{i(\omega_m t+\phi_N)}+e^{-i(\omega_m t+\phi_{-N})}\right\}$$

$$N_0=\frac{1}{2}\left(\frac{N}{2}-1\right)$$

, where T(t) is the expected channel, φ is the phase (uniformly distributed), α is the incident angle (uniformly distributed from –90 ° to 90°), ω_m is the Doppler shift, and N is the number of sources chosen for the simulation. There are proposed methods for simulating multiple channels with the ideally zero cross correlations between channels[2].

Evaluation:

This method produces acceptable auto correlations within a time window, but it has the disadvantage of having significant cross-channel coefficients when time-efficient algorithms are used to simulate multiple channels.

2. Filtering of White Gaussian Noise

There are known methods for producing pseudo-random white Gaussian noise. Such a signal has an ideally flat spectrum over the entire range of frequency, and the autocorrelation of an impulse function (at all lags except zero the autocorrelation is equal to zero). When the white Gaussian noise is filtered, its spectrum is modified, and so is its autocorrelation. In this project, an IIR (Infinite Impulse Response) filter design is used [3], which has the following transfer function in general format:

$$H(s)=\frac{a_N s^N + a_{N-1}s^{N-1}+\ldots+a_1 s + a_0}{b_N s^N + b_{N-1}s^{N-1}+\ldots+b_1 s + b_0}$$

, where H(s) is the ratio between input and output in s-domain (frequency domain), and a_i and b_i are the coefficients of the transfer function. In the next step this transfer function is converted into the discrete-time domain by a procedure called bilinear transform. This will result in a similar transfer function in format, but this time using the z-domain coefficients instead of s-domain ones.

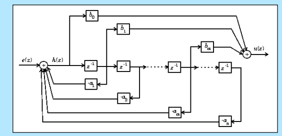
The problem of finding the right set of coefficients is categorized as an "optimization problem", where the goal is to find the best set of coefficients that will result in the closest match with the ideal autocorrelation function within the given window size. In a first attempt, optimization of the filter coefficients for a better match in the frequency domain was attempted which did not give good results due to the difficulty of designing filters close-enough to the ideal spectrum. As a result, this optimization has to be done for time-domain.

Filtering Algorithm:

Methods for filtering a signal can be divided into time-domain and frequency-domain approaches. In this project the time domain approach was chosen due to lower MIPS requirements (computation time) since there is no need for conversion from time domain to frequency domain and vice versa. Direct Form II method was chosen for an IIR filter, in order to have higher precision, lower MIPS requirements, and lower memory usage for filtering. For a z-domain transfer function of the form

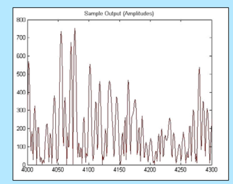
$$H(s)=\frac{a_N s^N + a_{N-1}s^{N-1}+\ldots+a_1 s + a_0}{b_N s^N + b_{N-1}s^{N-1}+\ldots+b_1 s + b_0},$$

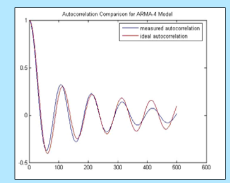
the structure that represents the direct form II algorithm is as follows (also known as ARMA structure):



Results

Using a 4th order ARMA filter the following results were obtained:





In comparison with the first approach of summing of sinusoids, the ARMA4 method resulted is slightly larger variations from the ideal autocorrelation, but a longer window of matching. For MIMO channels, this algorithm is simply repeated multiple times, which guarantees that there will be no correlation between channels, as opposed to the noticed correlation in the first method. Although slightly more complex in structure and design, the ARMA algorithm is actually more efficient in computation, and has better flexibility in comparison with the first proposed algorithm.

Reference

[1] W. C. Jakes, Jr. *Microwave Mobile Communications.* John Wiley & Sons, NY, 1974.
[2] M. Yan, "Doppler Frequency and Rayleigh Fading Process," http://dsp.ucsd.edu/%7Eyanming/research/ResearchOverView/ChannelEstimation.htm.
[3] B. C. Banister, J. Zeidler, "Feedback Assisted Stochastic Gradient Adaptation of MultiAntenna Transmission" , IEEE Transactions on Wireless Communications. Vol 4, No. 3, May 2005