Chapter 1 : The BJT as a Signal Amplifier - MP Study

Time-Varying Signals. Objective In this experiment we'll introduce the tools used to generate and measure signals that vary with time. You'll use a function generator and oscilloscope, view the output of an AC voltage divider circuit, and simulate the circuit in PSpice.

This is an open access article distributed under the Creative Commons Attribution License, which permits unrestricted use, distribution, and reproduction in any medium, provided the original work is properly cited. Received June 2nd, ; revised July 5th, ; accepted July 13th, Keywords: First, the Hilbert transform based decomposition is analyzed for the analysis of nonstationary signals. Based on above analysis, a hypothesis under certain condition that AM-FM components can be separated successfully based on Hilbert transform and assisted signal is developed, which is supported by representative experiments and theoretical performance analysis on error bound that is shown to be proportional to the product of frequency width and noise variance. The assisted signals are derived from the refined time-frequency distributions via image fusion and least squares optimization. Experiments on man-made and real-life data verify the efficiency of the proposed method and demonstrate the advantages over the other main methods. Introduction The decomposition of signals blended in noises is a reallife problem in measurement and other signal processing fields, which includes two tasks: The filtering of signals from observed noisy data, while preserving their original features respectively, remains a challenging problem in both signal processing and statistics. A number of filtering methods have been proposed, particularly for the case of additive white Gaussian noise []. Frequently, linear methods such as the Wiener filtering [1,4,10,11] are used because linear filters are easy to design and implement. However, linear filtering methods are not very effective when signals are nonstationary. To overcome the shortcomings, nonlinear methods have been proposed such as wavelet thresholding [5,7]. The idea of wavelet thresholding relies on the assumption that signal magnitudes dominate the magnitudes of the noise in a wavelet representation so that wavelet coefficients can be set to zero if their magnitudes are less than a predetermined threshold [1,7]. A limit of the wavelet approach is that the basis functions are fixed and, thus, do not necessarily match all real signals. On the other hand, the second taskâ€"the decomposition of multi-components is also an important problem in signal processing. The adaptive methods for signal separation should be explored [2,] especially for the separation of multiple components blended in noise data. In most of the reported papers [], the two tasks are separate and they cannot do the both tasks at the same time. In this paper, we will combine these two tasks to single one. Among many types of signals, the AM-FM amplitude modulation and frequency modulation signals are one of the most important ones and play an important role in various signal processing fields [1]. In telecommunications, amplitude modulation and frequency modulation convey information over a carrier wave by varying its instantaneous amplitude and frequency. AM-FM signals are also widely used in telemetry, radar, seismic prospecting and newborn EEG seizure monitoring, broadcasting music and speech, two-way radio systems, magnetic tape-recording systems and some video-trans-mission systems. Furthermore, many real-life signals can also be taken as the AM-FM signals approximately. In some cases, once the multiple AM-FM components are superposed and noised, the information carried by every AM-FM component is blended and hidden and cannot be recognized clearly. In this work, an elementary fundamental problem how to separate the AM-FM components blended in noises is addressed. The frequency domain separation method is the most traditional method to separate the superposed AMFM components in frequency domain or fractional frequency domain [17] through finding the separation points or peak points. For the case that the superposed AM-FM components have distinguishable frequency regions, the separation is done well. Unfortunately, once the superposed AM-FM components have overlapped and undistinguishable frequency regions, the separation method in frequency domain will fail to work. After the introduction of the HHT by Huang et al. If these conditions are not satisfied, EMD fails to separate them. Although IEMD can improve the performance, there are still some problems. Second, the employment of this IEMD method is cumbersome for more complicated cases such as the components with crossed instantaneous frequencies. The masking method [13,14] uses the mask signals to help decompose the superposed AM-FM components. In fact this method makes full use of the characters of filter bank for EMD [3] to extract the component whose frequency is close to the mask signal, and then eliminates the mask signal in IMFs to obtain the components. This method has two problems. The mask signal is based on Fourier spectral, therefore this method fails to handle the case of time-varying instantaneous frequencies that have overlapped frequency regions in frequency domain. In addition, this method still fails to separate two components with crossed frequencies. Differently, the Hilbert transform based method can separate the superposed AM-FM components whose frequencies are very close in principle. Unfortunately, the frequency of the assisted signal used in the Hilbert transform based method needs to be found in the Fourier transform domain via the valley values that can distinguish the different frequency modes. In other words, in the case of more overlapping of frequencies, this method will fail. In this paper, we will first analyze the performance of the Hilbert transform based method in great details and find its limitation, and then find its potential value for analysis of the more complicated cases such as timevarying instantaneous frequencies with overlapped and crossed points. The main aim in Section 2 to analyze the Hilbert transform based method is twofold: After that, in Section 3, the time-varying bandpass filter based on Hilbert transform and assisted signals is proposed, in addition the error bounds are analyzed theoretically and verified via simulations. The estimation of time-varying assisted signals is addressed in Section 4 via image fusion and least square optimization. In Section 5 experiments on manmade and real-life data are shown. Section 6 concludes this paper. Component Separation The decomposition theorem [16] by G. Wang is defined as follows. Let denote a real time series of n significant frequency components of the real time variable t. It can be decomposed into n signals whose Fourier spectra are equal to over n mutually exclusive frequency ranges:

Chapter 2 : Signal - Wikipedia

signals sampling time and frequency domains systems filters convolution ma, ar, arma filters system identification graph theory fft dsp processors speech signal processing data communications. Fourier Transform Analysis of Signals and Systems -.

Definitions[edit] Definitions specific to sub-fields are common. For example, in information theory , a signal is a codified message, that is, the sequence of states in a communication channel that encodes a message. In the context of signal processing , arbitrary binary data streams are not considered as signals, but only analog and digital signals that are representations of analog physical quantities. For example, the words " Mary had a little lamb " might be the message spoken into a telephone. The telephone transmitter converts the sounds into an electrical voltage signal. The signal is transmitted to the receiving telephone by wires; at the receiver it is reconverted into sounds. In telephone networks, signaling , for example common-channel signaling , refers to phone number and other digital control information rather than the actual voice signal. Signals can be categorized in various ways. The most common distinction is between discrete and continuous spaces that the functions are defined over, for example discrete and continuous time domains. Discrete-time signals are often referred to as time series in other fields. Continuous-time signals are often referred to as continuous signals even when the signal functions are not continuous ; an example is a square-wave signal. A second important distinction is between discrete-valued and continuous-valued. Particularly in digital signal processing a digital signal is sometimes defined as a sequence of discrete values, that may or may not be derived from an underlying continuous-valued physical process. In other contexts, digital signals are defined as the continuous-time waveform signals in a digital system, representing a bit-stream. In the first case, a signal that is generated by means of a digital modulation method is considered as converted to an analog signal, while it is considered as a digital signal in the second case. Another important property of a signal actually, of a statistically defined class of signals is its entropy or information content. Analog and digital signals[edit] A digital signal has two or more distinguishable waveforms, in this example, high voltage and low voltages, each of which can be mapped onto a digit. Characteristically, noise can be removed from digital signals provided it is not too large. Two main types of signals encountered in practice are analog and digital. The figure shows a digital signal that results from approximating an analog signal by its values at particular time instants. Digital signals are quantized , while analog signals are continuous. Analog signal An analog signal is any continuous signal for which the time varying feature variable of the signal is a representation of some other time varying quantity, i. For example, in an analog audio signal , the instantaneous voltage of the signal varies continuously with the pressure of the sound waves. It differs from a digital signal , in which the continuous quantity is a representation of a sequence of discrete values which can only take on one of a finite number of values. For example, an aneroid barometer uses rotary position as the signal to convey pressure information. In an electrical signal, the voltage , current , or frequency of the signal may be varied to represent the information. Any information may be conveyed by an analog signal; often such a signal is a measured response to changes in physical phenomena, such as sound , light , temperature , position, or pressure. The physical variable is converted to an analog signal by a transducer. For example, in sound recording, fluctuations in air pressure that is to say, sound strike the diaphragm of a microphone which induces corresponding fluctuations in the current produced by a coil in an electromagnetic microphone, or the voltage produced by a condenser microphone. The voltage or the current is said to be an "analog" of the sound. Digital signal A binary signal, also known as a logic signal, is a digital signal with two distinguishable levels A digital signal is a signal that is constructed from a discrete set of waveforms of a physical quantity so as to represent a sequence of discrete values. Other types of digital signals can represent three-valued logic or higher valued logics. Alternatively, a digital signal may be considered to be the sequence of codes represented by such a physical quantity. Digital signals are present in all digital electronics , notably computing equipment and data transmission. A received digital signal may be impaired by noise and distortions without necessarily affecting the digits With digital signals, system noise, provided it is not too great, will not affect system operation whereas noise always degrades the operation of analog signals to some degree. The resulting stream of numbers is stored as digital data on a discrete-time and quantized-amplitude signal. Computers and other digital devices are restricted to discrete time. Time discretization[edit] Discrete-time signal created from a continuous signal by sampling One of the fundamental distinctions between different types of signals is between continuous and discrete time. In the mathematical abstraction, the domain of a continuous-time CT signal is the set of real numbers or some interval thereof , whereas the domain of a discrete-time DT signal is the set of integers or some interval. What these integers represent depends on the nature of the signal; most often it is time. If for a signal, the quantities are defined only on a discrete set of times, we call it a discrete-time signal. A simple source for a discrete time signal is the sampling of a continuous signal, approximating the signal by a sequence of its values at particular time instants. A discrete-time real or complex signal can be seen as a function from a subset of the set of integers the index labeling time instants to the set of real or complex numbers the function values at those instants. A continuous-time real or complex signal is any real-valued or complex-valued function which is defined at every time t in an interval, most commonly an infinite interval. Amplitude quantization[edit] Digital signal resulting from approximation to an analog signal, which is a continuous function of time If a signal is to be represented as a sequence of numbers, it is impossible to maintain exact precision - each number in the sequence must have a finite number of digits. As a result, the values of such a signal belong to a finite set ; in other words, it is quantized. Quantization is the process of converting a continuous analog audio signal to a digital signal with discrete numerical values. Examples of signals[edit] Signals in nature can be converted to electronic signals by various sensors. The motion of an object can be considered to be a signal, and can be monitored by various sensors to provide electrical signals. A motion signal is one-dimensional time , and the range is generally three-dimensional. Position is thus a 3-vector signal; position and orientation of a rigid body is a 6-vector signal. Orientation signals can be generated using a gyroscope. Since a sound is a vibration of a medium such as air , a sound signal associates a pressure value to every value of time and three space coordinates. A sound signal is converted to an electrical signal by a microphone , generating a voltage signal as an analog of the sound signal, making the sound signal available for further signal processing. Sound signals can be sampled at a discrete set of time points; for example, compact discs CDs contain discrete signals representing sound, recorded at 44, samples per second ; each sample contains data for a left and right channel, which may be considered to be a 2-vector signal since CDs are recorded in stereo. The CD encoding is converted to an electrical signal by reading the information with a laser , converting the sound signal to an optical signal. A picture or image consists of a brightness or color signal, a function of a two-dimensional location. It can be converted to voltage or current waveforms using devices such as the charge-coupled device. A 2D image can have a continuous spatial domain, as in a traditional photograph or painting; or the image can be discretized in space, as in a raster scanned digital image. Color images are typically represented as a combination of images in three primary colors , so that the signal is vector-valued with dimension three. A video signal is a sequence of images. A point in a video is identified by its two-dimensional position and by the time at which it occurs, so a video signal has a three-dimensional domain. Analog video has one continuous domain dimension across a scan line and two discrete dimensions frame and line. The value of the signal is an electric potential "voltage". The domain is more difficult to establish. Some cells or organelles have the same membrane potential throughout; neurons generally have different potentials at different points. These signals have very low energies, but are enough to make nervous systems work; they can be measured in aggregate by the techniques of electrophysiology. Other examples of signals are the output of a thermocouple , which conveys temperature information, and the output of a pH meter which conveys acidity information. Signal processing Signal transmission using electronic signals A typical role for signals is in signal processing. A common example is signal transmission between different locations. The embodiment of a signal in electrical form is made by a transducer that converts the signal from its original form to a waveform expressed as a current I or a voltage V , or an electromagnetic waveform , for example, an optical signal or radio transmission. Once expressed as an electronic signal, the signal is available for further processing by electrical devices such as electronic amplifiers and electronic filters , and can be transmitted to a remote location by electronic transmitters and received using electronic receivers. Signals and systems[edit] In Electrical engineering programs, a class and field of study known as "signals and systems" S and S is often seen as the "cut class" for EE careers, and is dreaded by some students as such. Depending on the school, undergraduate EE students generally take the class as juniors or seniors, normally depending on the number and level of previous linear algebra and differential equation classes they have taken. Time, Frequency, s and z. Since signals and systems are both studied in these four domains, there are 8 major divisions of study. As an example, when working with continuous time signals t , one might transform from the time domain to a frequency or s domain; or from discrete time n to frequency or z domains. Systems also can be transformed between these domains like signals, with continuous to s and discrete to z. Although S and S falls under and includes all the topics covered in this article, as well as Analog signal processing and Digital signal processing , it actually is a subset of the field of Mathematical modeling. The field goes back to RF over a century ago, when it was all analog, and generally continuous. Today, software has taken the place of much of the analog circuitry design and analysis, and even continuous signals are now generally processed digitally. In past EE curricula S and S, as it is often called, involved circuit analysis and design via mathematical modeling and some numerical methods, and was updated several decades ago with Dynamical systems tools including differential equations, and recently, Lagrangians. The difficulty of the field at that time included the fact that not only mathematical modeling, circuits, signals and complex systems were being modeled, but physics as well, and a deep knowledge of electrical and now electronic topics also was involved and required. Students are expected to understand the tools as well as the mathematics, physics, circuit analysis, and transformations between the 8 domains. Because mechanical engineering topics like friction, dampening etc. Dynamical systems that involve noise, filtering and other random or chaotic attractors and repellors have now placed stochastic sciences and statistics between the more deterministic discrete and continuous functions in the field. Deterministic as used here means signals that are completely determined as functions of time. Signals and Systems Current loop - a signaling system in widespread use for process control.

Chapter 3 : Practical Introduction to Time-Frequency Analysis - MATLAB & Simulink Example

An analog signal is any continuous signal for which the time varying feature (variable) of the signal is a representation of some other time varying quantity, i.e., analogous to another time varying signal.

This is machine translation Translated by Mouseover text to see original. Click the button below to return to the English version of the page. This page has been translated by MathWorks. Click here to see To view all translated materials including this page, select Country from the country navigator on the bottom of this page. MathWorks does not warrant, and disclaims all liability for, the accuracy, suitability, or fitness for purpose of the translation. Translate Open Live Script This example shows how to perform and interpret basic time-frequency signal analysis. In practical applications, many signals are nonstationary. This means that their frequency-domain representation their spectrum changes over time. The example discusses the advantages of using time-frequency techniques over frequency-domain or time-domain representations of a signal. It answers basic questions, such as: When is a particular frequency component present in my signal? How do I increase time or frequency resolution? How can I sharpen the spectrum of a component or extract a particular mode? How do I measure power in a time-frequency representation? How do I visualize the time-frequency information of my signal? How do I find intermittent interference within the frequency content of the signal of interest? Using Time-Frequency Analysis to Identify Numbers in a DTMF Signal You can divide almost any time-varying signal into time intervals short enough that the signal is essentially stationary in each section. Time-frequency analysis is most commonly performed by segmenting a signal into those short periods and estimating the spectrum over sliding windows. Consider the signaling system of a digital phone dial. The signals produced by such a system are known as dual-tone multi-frequency DTMF signals. The sound generated by each dialed number consists of the sum of two sinusoids or tones with frequencies taken from two mutually exclusive groups. Each pair of tones contains one frequency of the low group Hz, Hz, Hz, or Hz and one frequency of the high group Hz, Hz, or Hz and represents a unique symbol. The following are the frequencies allocated to the buttons of a telephone pad: Generate a DTMF signal and listen to it. However, you cannot tell which number it was. Next, visualize the signal in time and in frequency domain over the to Hz band. To measure the length of the burst, you can take the pulse width of the RMS envelope. However, you cannot tell which numbers were dialed. A frequency-domain plot helps you figure this out because it shows the frequencies present in the signal. Locate the frequency peaks by estimating the mean frequency in four different frequency bands. However, the frequency-domain plot does not provide any type of time information that would allow you to figure out the order in which they were dialed. To solve this puzzle, use the pspectrum function to compute the spectrogram and observe how the frequency content of the signal varies with time. Compute the spectrogram over the to Hz band and remove content below the 10 dB power level to visualize only the main frequency components. Yellow colors indicate frequency content with higher power; blue colors indicate frequency content with very low power. A strong yellow horizontal line indicates the existence of a tone at a particular frequency. The plot clearly shows the presence of a Hz tone in all three dialed digits, telling you that they are all on the second column of the keypad. From the plot you can see that the lowest frequency, Hz, was dialed first. The highest frequency, Hz, was next. The middle frequency, Hz, came last. Hence, the dialed number was Longer segments provide better frequency resolution; shorter segments provide better time resolution. When no frequency resolution or time resolution values are specified, pspectrum attempts to find a good balance between time and frequency resolutions based on the input signal length. Consider the following signal, sampled at 4 kHz, that consists of the trill portion of a Pacific blue whale song: Look at the time signal and the spectrogram obtained by pspectrum when no resolution is specified and when time resolution is set to 10 milliseconds. Since we want to localize the time position of the pulses, set overlap percent to 0. On the other hand, a time resolution of 10 milliseconds is enough to localize each trill pulse in time. This becomes even clearer if we zoom into a few pulses: The signal is measured with a sampling interval of 7 microseconds. Analyze the spectrogram of the signal. Reduce the frequency resolution value to 3 kHz to get more details on the frequency variation of each ridge. However, since frequency and time resolution are inversely proportional, the time resolution of the spectrogram is considerably smaller. Time-Frequency Reassignment Even though we have been able to identify four frequency ridges, we can still see that each ridge is spread over several adjacent frequency bins. This is due to the leakage of the windowing method used in both time and frequency. The pspectrum function is capable of estimating the center of energy for each spectral estimate in both time and frequency. If you reassign the energy of each estimate to the bin closest to the new time and frequency centers, you can correct for some of the leakage of the window. Setting this parameter to true computes the reassigned spectrogram of the signal. You can also localize the signal energy using the function fsst, which is discussed in the next section. Reconstructing a Time-Frequency Ridge Consider the following recording, consisting of a chirp signal whose frequency decreases over time and a final splat sound. We use fsst to sharpen the spectrum of a noisy version the splat signal, tfridge to identify the ridge of the chirp sound, and ifsst to reconstruct the chirp. The process denoises the reconstructed signal. Add Gaussian noise to the chirp portion of the "splat" sound. The added noise simulates an audio recording taken with an inexpensive microphone. Examine the time-frequency spectral content. Compute and plot the synchrosqueezed transform of the noisy chirp. Identify the ridge using tfridge. Plot the ridge along with the transform. Include one bin on each side of the ridge. Plot the spectrogram of the reconstructed signal. Play the noisy and denoised signals consecutively to hear the difference. Compute the spectrogram of the signal using a time resolution of 1. The color bar shows that the power level of the signal is around 4 dB. Logarithmic Frequency Scale Visualization In certain applications, it may be preferable to visualize the spectrogram of a signal on a logarithmic frequency scale. You can achieve this by changing the YScale property of the y-axis. For example, consider a logarithmic chirp sampled at 1 kHz. The frequency of the chirp increases from 10 Hz to Hz in 10 seconds. You can also change the display colors with the colormap function. The persistence spectrum is a histogram in power-frequency space. The longer a particular frequency persists in a signal as the signal evolves, the higher its time percentage and thus the brighter or "hotter" its color in the display. Use the persistence spectrum to identify signals hidden in other signals. Consider an interference narrowband signal embedded within a broadband signal. Generate a chirp sampled at 1 kHz for seconds. The frequency of the chirp increases from Hz to Hz during the measurement. The weak sinusoid is obscured by the chirp. Now both signal components are clearly visible. You learned how to change time and frequency resolution to improve your understanding of signal and how to sharpen spectra and extract time-frequency ridges using fsst, ifsst, and tfridge. You learned how to configure the spectrogram plot to get a logarithmic frequency scale and three-dimensional visualization. Finally, you learned how to find interference signals by computing a persistence spectrum.

Chapter 4 : Creating time varying signal in C++ - Stack Overflow

create time-varying voltages with controlled waveshapes, amplitudes, and frequencies. The oscilloscope is essentially a voltmeter that allows us to measure and graph rapidly varying voltages.

Chapter 5 : PPT - Time Varying Signals PowerPoint Presentation - ID

Hello I am using Hilbert spectrum to study my nonstationary earthquakes recorded strong-motion. I was able to analyze the signal into AMPLITUDE ENVELOPE and INSTANTANEOUS FREQUENCY, later I was able to synthesis my signal back with the same amplitude envelope, instantaneous frequency and the phase.

Chapter 6 : discrete signals - kalman filter with time-varying noise? - Signal Processing Stack Exchange

The function generator creates accurate, time-varying voltage signals with user-controlled frequency and amplitude. The oscilloscope is capable of measuring and displaying rapidly changing voltages. You will also study capacitors and inductors in this lab.

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Chapter 7 : How to generate a signal with time varying frequency? - MATLAB Answers - MATLAB Central

To the Graduate Council: I am submitting herewith a thesis written by Xiaolin Luo entitled "Time-varying Autoregressive Modeling of Nonstationary Signals".

Chapter 8 : Time-Varying Bandpass Filter Based on Assisted Signals for AM-FM Signal Separation: A Rev

Review of Parameter Estimation Techniques for Time-Varying Autoregressive Models of Biomedical Signals. A. R. Najeeb, M. J. E. Salami, T. Gunawan, and A. M. Aibinu.

Chapter 9 : Analog signal - Wikipedia

So from (11), the AC signal at the collector is V c =-I c R c This result is negative, which means this circuit operates as an inverting amplifier for small, time varying signals. From (6), c mbe i c = g m v be.