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Digital Signal Processing

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Editorial

Diverse existing and evolving wireless access technologies that complement each other for different application areas and communication conditions are included into wireless systems. Communication systems are evolving towards an era where ubiquitous connection and increased levels of integration will be required for most applications in order to provide seamless and transparent interworking between these various wireless access technologies. For the near future, this revolution will not slowdown in its penetration of society. On the one hand, there is a market pull from a growing global population demanding massive information resources via ubiquitously connected gadgets.

On the other side, there is a market push from a $100 billion business that provides a wide range of communication products and services. Many applications, on the other hand, necessitate changing the signal's sample rate at various stages of the signal processing chain. Multirate Systems are discrete-time systems of this type. There are a variety of reasons why you would want to adjust your sample rate. It is altered in certain applications to reduce computation, while it is adjusted in others to increase accuracy and mitigate quantization problems. The sample rate may be altered at other times to save bandwidth. However, it shall be assumed that the NY Quist condition is always met and that there is no aliasing in all sampling rate changes [1].

Multirate filtering and filter banks

In the early 1970s, the Signal Processing Society proposed multi rate filtering in the context of signal interpolation. While polynomial interpolation of missing data is a traditional numerical analysis problem, novel techniques based on linear digital filtering have been discovered to be appealing. Many authors have looked into the design of decimation and interpolation filters nx. Multi rate filtering has also been utilised implicitly for numerical solution of differential equations in the applied mathematics field, under the name multi grid techniques. The advent of the two-channel quadrature mirror filter bank (QMF) for the compression of speech signals in the early 1970s was one of the milestones in the application of multi rate processing. Using an analysis filter bank, the input signal (e.g., speech) is split into low-pass and high-pass sub bands. Each sub band signal is then quantized after being decimated by a factor of two. As a result, the compressed signal is in the form of quantized sub band signals. To obtain a close approximation of the original signal, these can be recombined using a synthesis filter bank. The approximate nature of the reconstruction is due to compression (sub band quantization) and other flaws introduced by analysis and synthesis filters [2,3].

Components of multirate processing

At various levels of the processing chain, Multirate DSP systems require sampling-rate conversion, which is commonly accomplished using decimation or interpolation. In order to execute decimation and interpolation, we use three basic building elements: linear time invariant low-pass filters, down-samplers, and up-samplers [4,5].

The down-sampler and the up-sampler

By keeping every Mth sample and discarding the others, the sampling rate can be reduced by a factor of M (where M is a positive integer). A down-Sampler is the device that executes this process, and the output of this down-Sampler is a sequence with a sampling rate that is 1/M times that of the input sequence. Down-sampling and up-sampling are time-variant linear processes.

Sampling rate conversion using multirate structures

Decimation: To prevent aliasing when the sampling rate is changed, a decimator incorporates both a down-sampler and a low pass filter. The decimator's output has a lesser bandwidth than the input because it has a lower sampling rate than the input.

Interpolation: "Interpolation" is the inverse of decimation. Up-sampling by an integer factor, which is accomplished by adding zero samples between adjacent input signal samples, is the first stage of the interpolation process. This up sampled signal has a sampling rate that is times the original sampling rate, as well as a spectrum with times the number of repeated images of the original signal spectrum. The up sampled signal must be low-pass filtered in order to retrieve the interpolated signal accurately. This filtering removes all of the spectral pictures that the up sampling introduced.

Rational number sampling rate conversion: So far, the conversions of sampling rates that have been considered have included integer changes in sampling rate (via either decimation or interpolation). Many applications, on the other hand, necessitate changing the sampling rate by a rational value. Even some applications (such as audio signal pitch adjustment) necessitate irrational factor sampling rate conversion.

Implementation of multirate systems

Noble Identities: Two key identities are presented here to aid in the implementation of multi-rate systems with greater flexibility. These are decimation and interpolation identities. These identities can help ease the study and design of complex multi-rate systems. They make it possible to commute the various components of decimators/interpolators as needed. These identities, as will be seen later, lay the way for large computational savings in multi-rate system implementations.

Multistage implementation: The rate of each channel of the input signal is converted from the input sample rate to the output sample rate by this multistage FIR converter. By first lowering the sample rate of the input signal, multistage solutions reduce the amount of computation required by sample-rate conversions.

Design of an interpolated fir filter: The interpolated FIR (IFIR) design technique is a popular method for implementing filters in multistage sampling rate converters. The following is the reasoning for the IFIR design. To get highly fine cut-offs with FIR filters, you usually need many filter taps. Sharp cut-offs can be achieved without a huge number of taps by using an exquisite "sleight of hand [3].

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Conflict of Interest

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