International Journal of
Emerging Research in Management &Technology
ISSN: 2278-9359

Research Article | **February 2013**

ICA Based Improvement In GSM

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ABSTRACT:

ICA is a technique to separate linearly mixed sources. It is a computational method for separating a multivariate signal into additive sub components assuming mutual statistical independence of the source signals. ICA is about performing blind source separation. It is related to other things such as entropy and information maximization, maximum likelihood density estimation and projection pursuit. The inputs to ICA could come from such areas as digital images, document databases, economic indicators and psychometric measurements. In many cases, the measurements are given as a set of parallel signals or time series. Typical examples are mixtures of simultaneous speech signals that have been picked up by several microphones, brain waves recorded by multiple sensors, interfering radio signals arriving at a mobile phone, or parallel time series obtained from some industrial process.

Keywords: ICA algorithm, Gaussian Noise, ICA block

I. Scope of Work

Consider the time series from two independent sources A and B as shown in Figure 1.

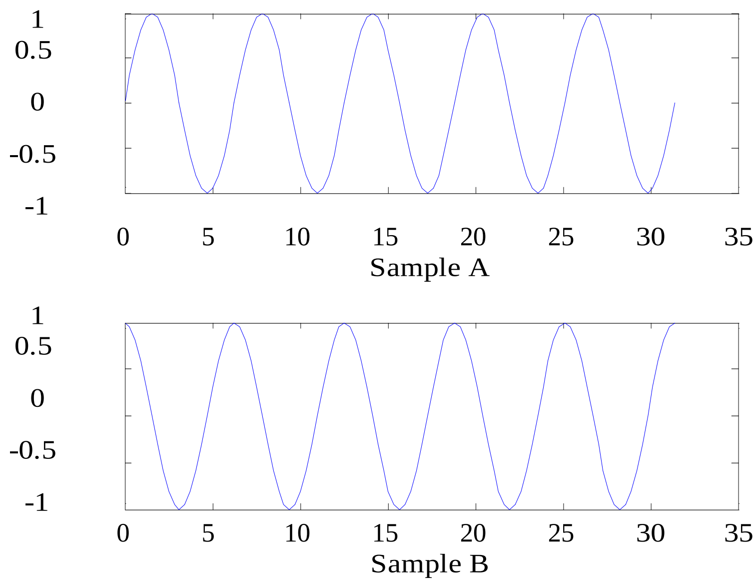
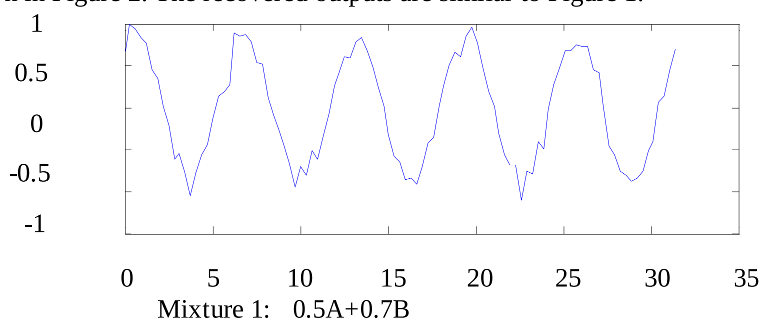


Figure 1 Source Signals

The independent sources are mixed linearly to get two mixtures. The mixtures obtained are 0.5A+0.7B and 1.5A-2B. These are the inputs to the ICA algorithm. At the output, we get the separated sources A and B. The mixtures or the inputs to the ICA algorithm are shown in Figure 2. The recovered outputs are similar to Figure 1.



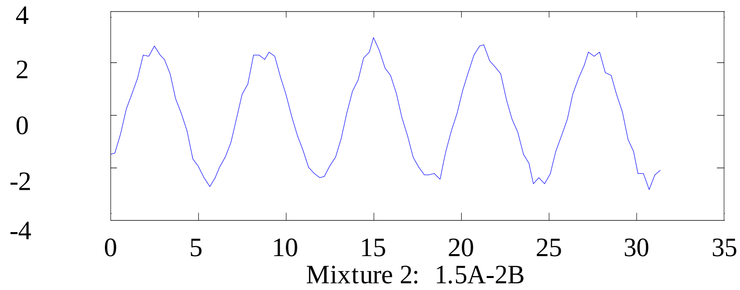


Figure 2 Mixture's (Inputs to ICA algorithm)

II. The Anatomy of our Signal Processing Block:

After understanding the concept of ICA, let's examine how it can be utilized in the SPB to solve our problem. In wireless communications, when the signal is transmitted, it travels through the channel, where noise is added to the signal. Thus, received signal is a mixture of the original signal and noise. The nature of the noise added can be modeled as Additive White Gaussian Noise (AWGN). It is desirable to reduce the effect of this noise. This mixture of signal and noise when input to the SPB reduces the effect of noise.

The first cut at the SPB

If the voice is greater than a certain threshold i.e. if the signal is good quality, then we store the sample of voice and calculate the pitch for the stored voice. Its pitch, which corresponds to frequency with maximum amplitude, is calculated using the harmonic product spectrum (HPS) method. Once the pitch is calculated it is stored for future comparison purposes. If on the other hand, the voice is less than a certain threshold i.e. if the signal quality is not good enough then it is passed through the ICA block, which gives us the separated samples of noise and voice.

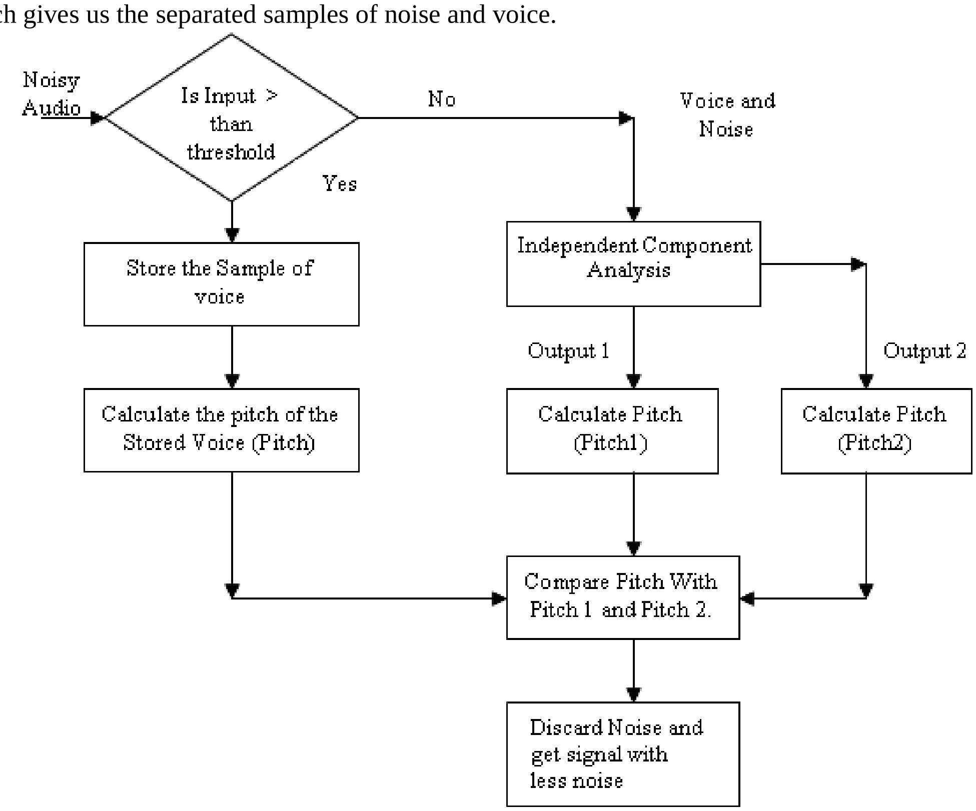


Figure 3 Initial block diagram of the ICA based call drop reduction system

Even though we have a separated version of noise and voice we still need to recognize which is voice and which is noise. In this case we calculate the pitch for the separated samples and compare it with the stored pitch of the good signal. The comparison helps in recognizing which one of the samples is voice and which one is noise.

The ICA block helps in reduction of noise using the gradient ascent rule.

We simulated a part of the block diagram using actual voice as the input. The voice was recorded and mixed with White Gaussian Noise (WGN) to get mixture 1. Mixture 2 was obtained by adding a similar type of noise to mixture 1. These mixtures, where sent through the ICA block. The results of the experiment (shown in Figure 4) proved that the ICA works effectively in the separation of noise and voice.

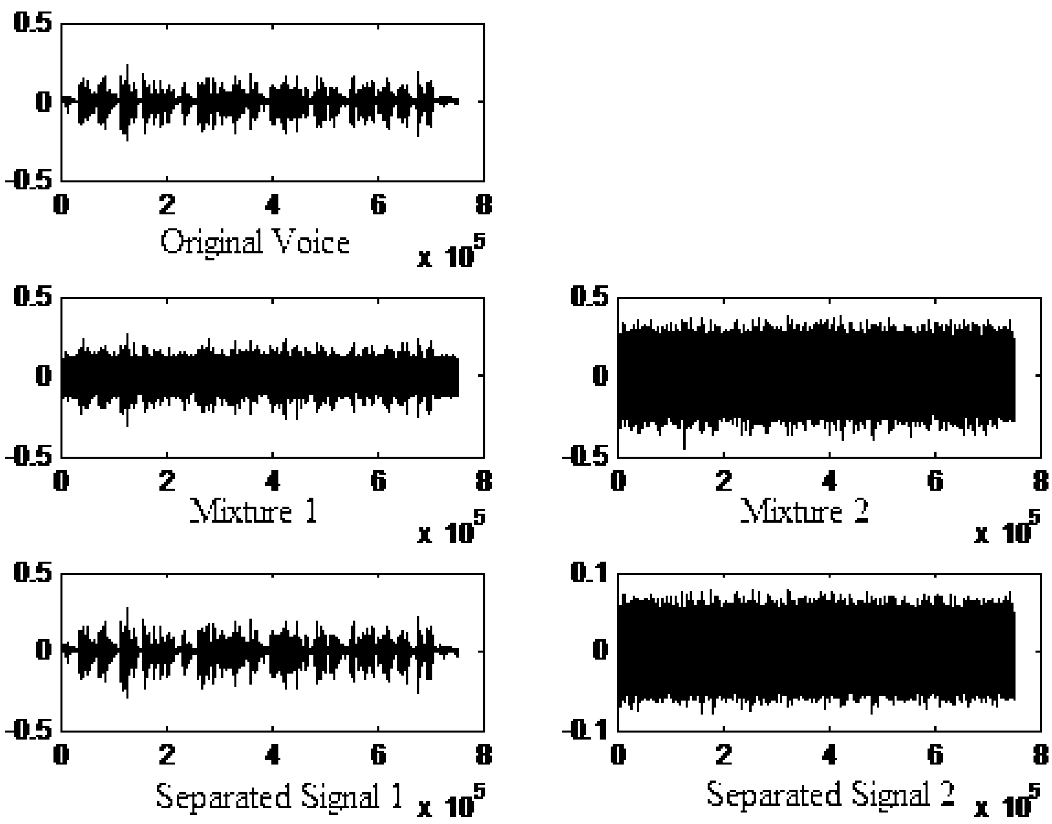


Figure 4 Results for Simulation of ICA

III. Harmonic Product Spectrum

As can be seen in the initial block diagram, HPS is a critical part in the automated identification of signal vs. noise after they are separated by ICA. HPS algorithm is a widely used method to calculate the pitch of speech signals. In HPS, we follow the following steps:

- Divide the input signal into segments by applying a Hanning window.
- For each window, calculate the Fourier Transform.
- Apply down sampling. In other words, compress the spectrum twice in each window by re-sampling. The first time the signal is compressed by two and then by three.
- Multiply the down-sampled version of the signals with the original signal in each window (as shown in Figure 5).
- Find the frequency that corresponds to the peak (maximum value). This frequency represents the dominant frequency (pitch) of that particular window.

The steps described above can be diagrammatically shown as in figure 5

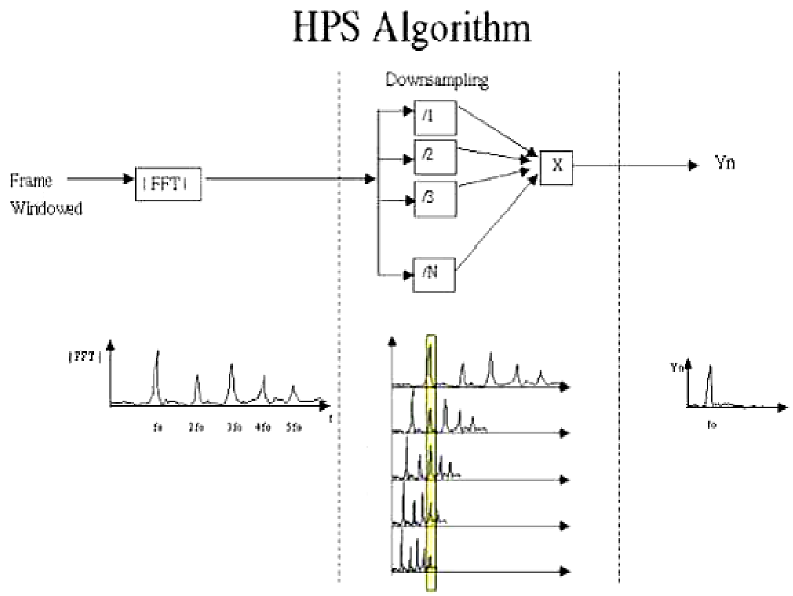


Figure 5 Harmonic Product Spectrum

The initial experiment was with the actual audio baseband signal. But in cell telephony/wireless world, the audio is digitalized, modulated, filtered and then transmitted. The signal at the receiver end is a mixture of noise and data. We wanted to verify the application of our proposed block diagram in reduction of noise in a real wireless system in particular in the physical layer of such a system.

IV. WIRELESS COMMUNICATION SYSTEM:

As discussed previously, in order to validate our proposed block, the physical layer simulation of a wireless communication system is required. In the following, we provide a brief over view of the physical layer of a wireless communication system. A simulated wireless communication system (Figure 6) consists of a transmitter, a receiver and a channel. The transmitter consists of a burst builder (which has the actual "payload"), modulator, up-sampler and filter. The receiver on the other hand, consists of a matched filter, down sampler, demodulator and a BER calculator. The noise added can be modeled as AWGN. The channel should take fading effects into consideration. Each block is explained briefly in the following subsections.

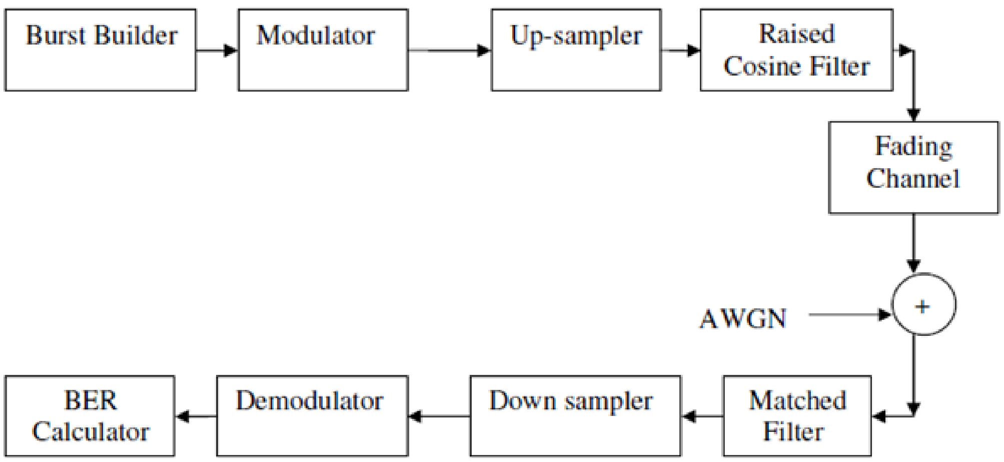


Figure 6 Physical Layer simulation of a wireless communication system

V. Conclusions:

In this paper we introduced ICA Technique which may be used for separating mixture of sources. SPB is also used for noise reduction in wireless communication. In this paper we introduce flow chart which will separate our mixed signal with the help of HPS.

We simulated a transmitter, a channel and a receiver of a WCS. Then, we introduced the SPB at the receiver section and calculated the BER. The BER improvement will:

- Help in solving the problem related to call drop outs because of the high BER.
- improve the overall QoS.
- Potentially help in the extension of the range of a base station.
- provide a coding gain.

The receiver section of the mobile handsets could incorporate the SPB and improve the call reception quality.

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