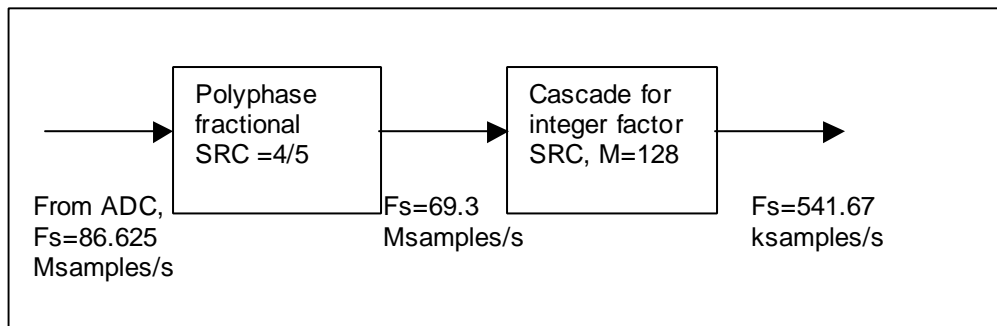


Project 1 Description

In this project, we have a communications system (GSM) which has a digitizer sampling rate of 86.625 Msamples/sec at an intermediate frequency (IF). The sampling rate for the final system should be 541.6 ksamples/sec. Use the method described in the article by Hentschel, T. and Fettweis, G., "Sample Rate Conversion for Software Radio," *IEEE Communications Magazine*, August 2000, 142-151, to break up this system into a fractional and integer sampling rate converter (SRC) as shown in the figure below. The polyphase filter bank will be used for the first part of the sampling rate conversion. A set of cascaded decimators will be used for the second part.



The goal of this project is to explore different methods of changing sampling rates and the effects of using real filters for anti-aliasing filters.

Design Requirements:

- The integer factor SRC cascade should have at least three sections.
- The filters should all have linear phase to avoid phase distortion in the signal.
- The anti-aliasing filter for the fractional SRC should have at least 60 dB of attenuation and the anti-aliasing filters for the cascade should each have at least 40 dB of attenuation.
- The DSP filter requirements should be easier to satisfy (shorter filters) at higher frequencies.
- Calculate the oversampling ratio for each phase at the system. The oversampling ratio is defined as the sampling rate divided by the bandwidth of the signal of interest.
- Verify that your signal is intact after all this processing by generating an appropriate signal and filtering it.