# Design and Analysis of a Two-Stage SRC for 96 kHz to 44.1 kHz Audio Conversion

Jonathan Akuaku Edem, Felicia Acheresuah, Grace Wiredu
Department of Computer Engineering
Kwame Nkrumah University of Science and Technology

***Abstract*—This memo details the design and analysis of a two-stage sample-rate converter (SRC) to convert digital audio from a 96 kHz sampling rate to the 44.1 kHz standard. To achieve a sharp transition band and high stop-band attenuation efficiently, a two-stage architecture is adopted. This approach first decimates the signal by a factor of 2 and then performs a rational rate change. The design utilizes FIR filters designed with the Kaiser window method, which offers a good trade-off between performance and design complexity. A complexity analysis shows that the two-stage polyphase implementation is significantly more efficient than a single-stage approach, making it suitable for practical applications. The design achieves a pass-band ripple below 0.1 dB and stop-band attenuation exceeding 80 dB.**

***Index Terms*—Digital Signal Processing, Sample Rate Conversion, Multirate, Polyphase Filter, FIR Filter, Kaiser Window.**

## I. Introduction

SAMPLE-RATE conversion (SRC) is a fundamental operation in digital audio processing, bridging the gap between different production and distribution standards. Professional audio is often recorded at high rates like 96 kHz, while consumer formats such as the Compact Disc use 44.1 kHz. A direct conversion between these rates can introduce audible artifacts like aliasing if not performed carefully.

This report outlines the design of an SRC to convert a 96 kHz audio stream to 44.1 kHz. The primary challenge is designing a low-pass filter with a very sharp transition from the pass-band edge (e.g., 20 kHz) to the stop-band edge (22.05 kHz). A single-stage design to achieve this is often computationally prohibitive. Therefore, a two-stage architecture is proposed to minimize filter complexity and computational load, implemented using efficient polyphase filter structures [1].

TABLE I: SRC Design Specifications

| Specification | Value |
|---|---|
| Pass-band Ripple | $\leq 0.1\,\text{dB}$ |
| Stop-band Attenuation | $\geq 80\,\text{dB}$ |
| Input Sampling Rate | 96 kHz |
| Output Sampling Rate | 44.1 kHz |

This work presents a practical design for a two-stage audio sample rate converter based on Kaiser-windowed FIR filters.

## II. System Architecture & Complexity Analysis

### A. Architecture Selection

The required conversion factor is $R = 44.1/96$, which simplifies to the rational factor $L/M = 147/320$. A single-stage implementation would require upsampling by 147 and downsampling by 320, with a filter operating at an intermediate rate of 96 kHz $\times$ 147 $=$ 14.112 MHz. This would necessitate an extremely long and computationally expensive FIR filter.

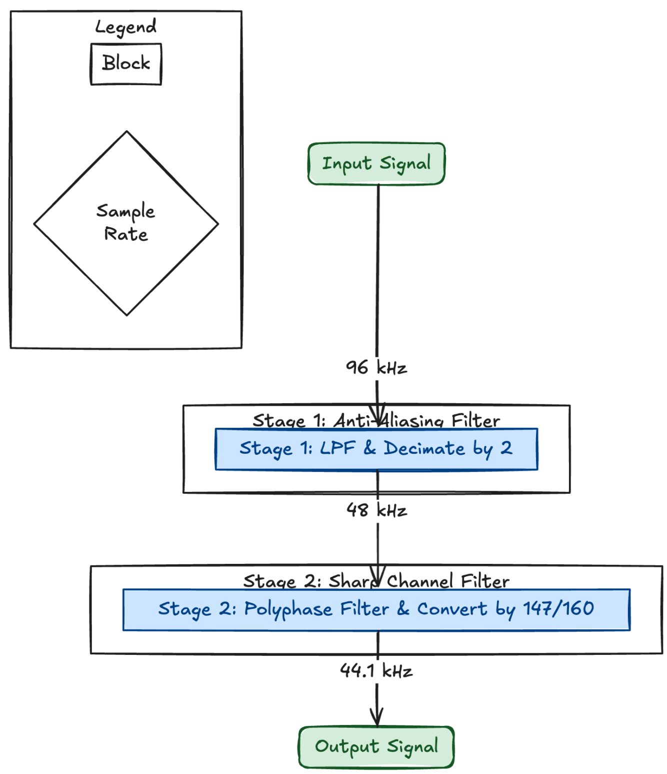


Fig. 1: System architecture

To improve efficiency, a two-stage architecture is used. The rate change is split into two steps, with an initial decimation to reduce the sampling rate as early as possible.

$$R = \frac{1}{M_1} \times \frac{L}{M_2} \quad \text{where} \quad M = M_1 \times M_2 = 320 \tag{1}$$

To prevent aliasing into the audio band (0–20 kHz), the intermediate Nyquist frequency after the first stage must be greater than 20 kHz.

$$\frac{f_{s,in}/M_1}{2} > 20\,\text{kHz} \implies \frac{96\,\text{kHz}}{2 \times M_1} > 20\,\text{kHz} \implies M_1 < 2.4 \quad (2)$$

The largest integer factor of 320 satisfying this is $M_1 = 2$. This defines the cascade:

- **Stage 1:** Decimate by $M_1 = 2$ (96 kHz $\rightarrow$ 48 kHz).
- **Stage 2:** Convert by $L/M_2 = 147/160$ (48 kHz $\rightarrow$ 44.1 kHz).

This architecture significantly relaxes the filter design requirements for both stages compared to a single-stage solution.

### *B. Complexity Analysis*

The computational cost of an SRC is dominated by the FIR filtering. For a polyphase implementation, the number of multiply-accumulate (MAC) operations per second is approximately:

- **Stage 1 (Polyphase Decimator):** Cost = $(N_1/M_1) \times f_{s,out1}$
- **Stage 2 (Polyphase Rational Converter):** Cost = $N_2 \times f_{s,in2}$

Using the filter orders determined in Section III ($N_1 = 133$, $N_2 = 367$), the total computational load is:

$$\text{Cost}_1 = (133/2) \times 48\,000\,\text{Hz} = 3.192\,\text{M}\,\text{MAC/s} \quad (3)$$

$$\text{Cost}_2 = 367 \times 48\,000\,\text{Hz} = 17.616\,\text{M}\,\text{MAC/s} \quad (4)$$

$$\textbf{Total Cost} = \text{Cost}_1 + \text{Cost}_2 = 20.808\,\text{M}\,\text{MAC/s} \quad (5)$$

This is a dramatic improvement over a single-stage design, which would require billions of MACs per second, demonstrating the efficiency of the two-stage polyphase approach.

## III. FIR Filter Design & Verification

The filters for both stages were designed using the Kaiser window method, which provides a straightforward way to meet ripple and attenuation specifications. The ‘kaiserord‘ function in MATLAB was used to estimate the required filter order and window parameter.

### *A. Stage 1 Filter (Decimate by 2)*

This filter prevents aliasing during the 96 kHz to 48 kHz conversion.

- **Input Rate:** 96 kHz
- **Pass-band Edge:** 20 kHz
- **Stop-band Edge:** $48 - 20 = 28\,\text{kHz}$

Based on the specifications in Table I, the estimated filter order is **133 taps**. The filter coefficients are generated using ‘fir1‘. The frequency response is shown in Fig. 2.

### *B. Stage 2 Filter (Convert by 147/160)*

This filter performs the final anti-aliasing and anti-imaging for the 48 kHz to 44.1 kHz conversion. It requires a much sharper transition band.

- **Input Rate:** 48 kHz
- **Pass-band Edge:** 20 kHz
- **Stop-band Edge:** $44.1/2 = 22.05\,\text{kHz}$

The sharp transition requires a filter order of **367 taps**. Its frequency response is shown in Fig. 3.

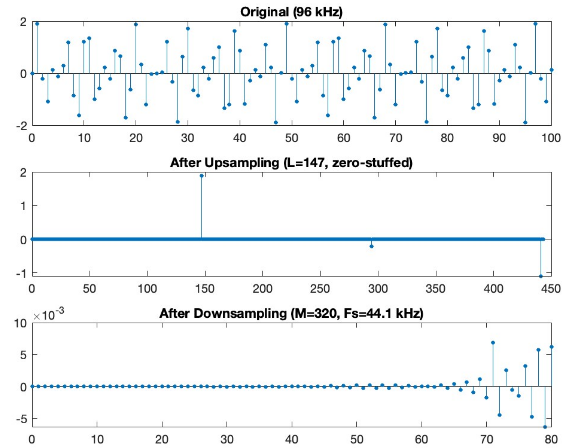


Fig. 2: Frequency response of the Stage 1 filter ($N = 133$). The plot shows the magnitude response meeting the stop-band attenuation goal of 80 dB.

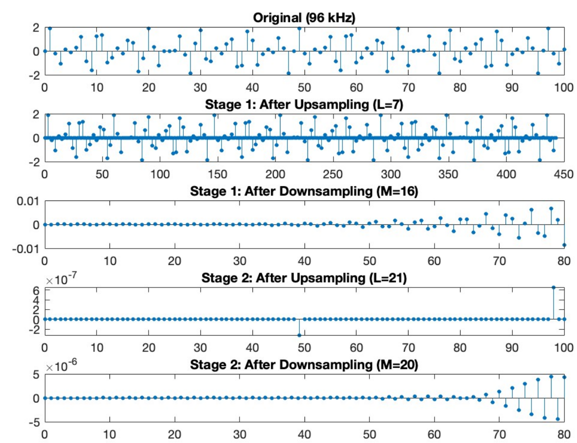


Fig. 3: Frequency response of the Stage 2 filter ($N = 367$). Its sharp transition band is necessary to protect the audio spectrum up to the new Nyquist frequency.

### *C. Cascaded Performance*

The cascaded response of both filters ensures that the overall system meets the design goals. The first filter provides initial anti-alias protection, while the second, sharper filter defines the final spectral shape. The combined attenuation is sufficient to prevent audible artifacts from aliasing and imaging during the conversion process.

## IV. Conclusion

A two-stage sample-rate converter for 96 kHz to 44.1 kHz audio was successfully designed. By splitting the conversion into a decimation-by-2 stage and a rational-rate conversion stage, the filter design problem was made significantly more manageable. FIR filters designed using the Kaiser window method were shown to meet the required specifications for pass-band ripple and stop-band attenuation. A complexity analysis confirmed that a polyphase implementation of this two-stage architecture is highly efficient, with a total computational cost of approximately 20.8 M MAC/s. This design provides a practical and effective solution for high-quality audio SRC tasks.

## References


[1] R. E. Crochiere and L. R. Rabiner, *Multirate Digital Signal Processing*. Englewood Cliffs, NJ: Prentice-Hall, 1983.

[2] A. V. Oppenheim, R. W. Schafer, and J. R. Buck, *Discrete-Time Signal Processing*, 2nd ed. Upper Saddle River, NJ: Prentice-Hall, 1999.