Invariably, real world testing and design assessment must be done to balance system quality versus system complexity and cost.

**Front end filter**

A filter is needed on the front end of the signal path before any DSP occurs to avoid aliasing. If, on a guitar, the 24th fret of the high e string is played (typical highest note), the resulting fundamental frequency is 1.318 kHz. The highest audible frequency by the standard limit of 20 kHz is the 15th harmonic. This harmonic, following typical models of vibrating strings, has a very small amplitude, thus minimal impact on string tonality. This justifies a -3dB amplitude point at 20 kHz, as the attenuation close to that frequency will minimally impact the tonality of an expected guitar input.

1. What filter topology?
   1. Multiple feedback, single amplifier biquad.
      1. PROS
         1. High stability
            1. Justification needed

\*This may be due to the amplifier gain not affecting the Q factor.

* + - * 1. Stability may become critical as one of the goals of this filter is to have a sharp cutoff by the Q factor using as little parts as possible to reduce cost, thus minimizing filter order.
      1. Sensitivity?
      2. Q factor can be chosen, independent of the gain
      3. Less phase delay
    1. CONS
       1. Capacitor values cannot be the same
       2. No guarantee capacitor values will be standard values
          1. Potentially multiple capacitors in parallel/series necessary
  1. Sallen and Key, single amplifier biquad.
     1. PROS
        1. Capacitor and resistor values can be the same
           1. This can reduce bulk cost.
     2. CONS
        1. Low stability?
        2. Impossible to implement for higher order filters due to the required gain for the needed Q factors to produce a correctly shaped response. The gain levels quickly creates signal voltage beyond the rails.
           1. It needs to be investigated if a low order filter implementation is possible (order of 2 for example), providing that the filter still meets spec. This is going to require real world experimentation.

A filter order of 2 is possible

* + - 1. Amplifier gain directly affects the Q factor
         1. \*If resistors are properly matched, gain should remain constant due to sensitivity and TCO.
      2. Gain restrictions greatly limit Q factor to be no more than 1
      3. Negative feedback path requires 1 additional resistor compared to the multiple feedback topology.
  1. Multiple Amplifier Biquads
     1. Out of the question (in general) due to unnecessary extra cost of extra op-amps.

Preliminary specs:

1. Lowpass or Bandpass
   1. Lowpass and coupling capacitor
   2. Bandpass
2. Passband
   1. 0 – 20kHz for lowpass
   2. 20 – 20kHz for bandpass
3. Stopband
   1. TBD

Problems:

1. Coupling capacitor is needed for a lowpass response. Will this affect signal response?
   1. Simulation shows that small values of coupling capacitors (C < 100nF) irreconcilably alters the magnitude response. Capacitor values too large cause large phase shifts that are undesirable, but lack any signal magnitude alteration.
   2. In simulation, a coupling capacitor linking an ideal voltage source to the filter with a value of results in amplitude attenuation in frequencies less than 20Hz, which are not of any use for this audio application, and also resembles the bandpass shape originally desired. Still further investigation is needed on the effect of the phase delay on the filtered signal.
      1. After further thought, a small valued capacitor, relatively speaking, would block dc current
2. DC pass-through?
   1. Bandpass filter is still going to pass some measure of DC, which is undesirable for the MCU. In order to avoid the DC portion, a coupling capacitor could be used. However, considering that low frequencies from 0 – 20Hz will create no aliasing, the justification for using a bandpass diminishes as there is the potential for extra components, when the coupling capacitor for the lowpass filter essentially accomplishes the same job. Thus, a lowpass response will be used
3. Errors in simulation.
   1. Possibly due to bad math.
   2. This has been resolved. Special attention is needed when determining poles due to the error term:

**Group Delay Equalizer**

**Inverter/Amplifier**

Due to this being an audio application, inversion of the signal may be irrelevant.

**General Issues to be Addressed**

1. Gain Bandwidth / Op-Amp attenuation
   1. Gain is low, thus opening up bandwidth
2. Transfer function characteristics?
   1. Butterworth, Chebyshev, etc.
3. Amax ripple
   1. What is sufficient?
      1. -3dB at 20kHz should be sufficient as a prototype.
   2. This will matter more if a Chebyshev response is desired.

**Final Specs**

Front end filter:

1. Filter response type
   1. Butterworth
2. Filter topology
   1. Multiple feedback, single amplifier
3. Passband
   1. -3dB
4. Stopband
   1. 40kHz
   2. -24dB

**\***Credit to Andrew Kwiecinski for giving estimates for passband and stopband specifications

<http://www.maximintegrated.com/en/app-notes/index.mvp/id/1762>