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Designing a Novel Feedback Controlled Acoustic Musical Instrument

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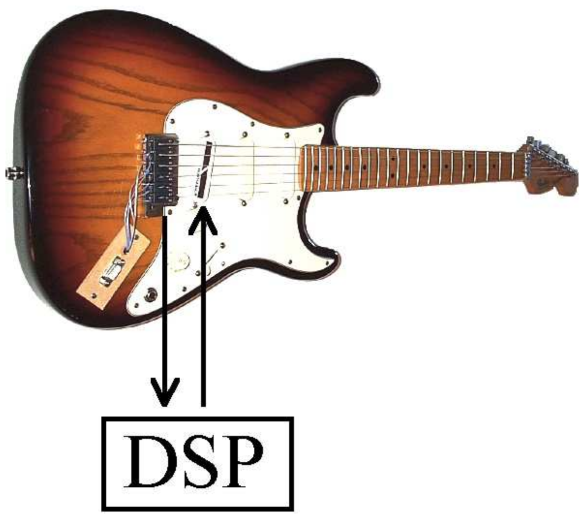

Figure 1: A feedback controlled electric guitar

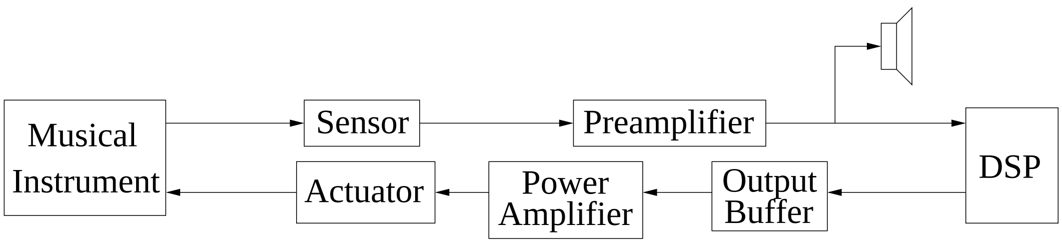

Figure 2: Control loop for a feedback controlled acoustic musical instrument

1 Introduction

This document is meant to be a placeholder until a more permanent publication is produced. At that time, this document will be replaced with a link to the permanent publication.

We hope that this information will be helpful for those developing feedback controlled acoustic musical instruments. For example, in the context of the electric guitar, a DSP can sense the motion of the strings and cause a reaction force to be exerted on the strings (see Figure 1). For more information on the history and theory of feedback controlled acoustic musical instruments, please see the references [2, 3, 1, 11, 4, 5, 7].

Figure 2 shows the technical elements in the feedback control loop. In this document, we assume that a feedback controlled acoustic musical instrument consists of the following main components: acoustic musical instrument, sensor, analog preamplifier, DSP, analog output buffer, power amplifier, and actuator, which is again connected to the same acoustic musical instrument. The feedback changes the acoustical properties of the musical instrument. The output sound can come directly from the acoustic musical instrument if it is loud enough, or it can come from an amplified loudspeaker connected to the preamplifier (see Figure 2).

2 Design Process

We recommend using the following process for developing a novel feedback controlled acoustic musical instrument, as based on collocated feedback control to provide the most freedom in reprogramming the acoustics. The design process involves both aesthetic decisions and practical engineering-oriented decisions. The main engineering challenge is obtaining an appropriate sensor and actuator pair that can be collocated.

The design process consists of the following steps:

1. Choose an acoustic musical instrument.
2. Select an actuator that is powerful enough to strongly actuate the acoustics of the musical instrument. If a sufficiently powerful actuator cannot be reasonably obtained, then go back to #1.
3. Try to find a low-noise sensor that can be collocated with the actuator. If not, go back to #2 or #1.
4. Measure the phase response of the musical instrument and determine up to what frequency the phase response is bounded by +/- 90 degrees. This frequency places a maximum limit on f_c.
5. Build an analog feedback control circuit to test the control system. (For example consider implementing active damping using velocity feedback.) Make a model of the feedback control circuit and **use a signal generator and oscilliscope to verify that the model matches the circuit.**[1]
6. Integrate a digital signal processor (DSP) into the feedback loop. Use the signal generator and oscilliscope to measure the delay introduced by the DSP. Make sure that the delay is short. It must be shorter than 50 microseconds but probably should be much shorter than that.
7. Program the DSP.
8. Make some music!

Note that there should be one, single-pole lowpass filter with cutoff frequency f_c that limits the bandwidth over which the control system operates. This filter needs to roll off the magnitude response starting around f_c so that the magnitude of the control decreases sufficiently before the phase response wraps too far negatively away from 0 degrees [8].

3 Feedback Guitar

3.1 Hardware

We present the example of a feedback controlled electric guitar that was developed according to the design process.

During the design process, we found that only the Sustainiac actuator was powerful enough. When we tried to use a standard electromagnetic pickup as the sensor, we discovered that there was too much electromagnetic interference of the actuation signal into the sensing signal. For this reason, we chose to use a sensor that operates according to a non-electromagnetic principle. We used piezoelectric pickups from Graphtech. Unfortunately the sensor and actuator could not be perfectly collocated; however, we found that by placing the actuator near the end of the string terminated by the piezo sensor, we could achieve approximate collocation.

[1]**The authors cannot emphasize this point enough. Anyone working in this field should spend significant time debugging the phase response using a signal generator and oscilliscope.** Start by replacing the sensor with the signal generator, then display on the scope both the signal at the signal generator and at a test point in the circuit. Using the oscilliscope, you can then visually estimate the phase difference between the two signals as a function of frequency. Start by setting the testing point near the beginning of the circuit. Then debug each additional stage in the circuit by gradually moving the test point from the sensor part of the circuit to the actuator (i.e. check the current through the actuator by measuring the voltage across R_s). For each test point, sweep the frequency across the entire control bandwidth to verify that the circuit is producing the correct phase response.

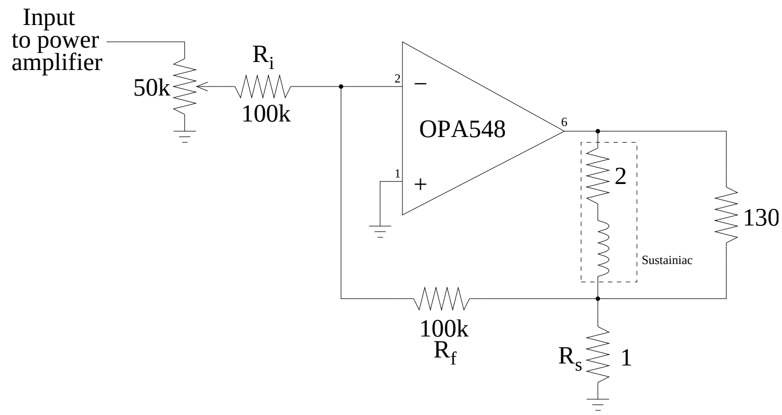

Figure 3: Transconductance power op amp circuit

3.1.1 Power Amplifier

The power amplifier should be of the transconductance type so that the output *current* through the actuator is proportional to the input *voltage* to the power amplifier [9]. This is because the force exerted by the actuator is proportional to the current flowing through it [6]. We employed a simple power amplifier circuit using the OPA548 power operational amplifier. In the circuit in Figure 3, the output current $i_{out} \approx \frac{R_f}{R_i \cdot R_s} V_{in}$. The 130 ohm resistor makes it easier for the amplifier to push a current through R_s at high frequencies [9]. The resulting "lowpass" pole must be configured so that its cutoff frequency is considerably greater than the control system bandwidth f_c.

It is useful to have a power amplifier with an adjustable current limit. This way, during early development, the current limit can be set at a low level (i.e. $R_{limit} = 100k$ for the Sustainiac). This prevents unpleasant things from happening if the control system becomes unstable. Then, once the system is well developed and understood, the current limit can be increased to a level that allows for more control power but which will help prevent the actuator from overheating (i.e. $R_{limit} = 15k$). Different maximum currents may be more appropriate for different actuators, so please see the OPA548 datasheet for more information on setting the current limit. Figure 4 shows the circuit with including the bipolar power supply connections. For the feedback guitar, we employ $V_{1+} = 12V$ and $V_{1-} = -12V$. Significant filtering of the power supply is necessary—please see the OPA548 datasheet for more details.

3.1.2 Remainder Of The Circuit

The rest of the circuit is shown in Figure 5, which incorporates low power operational amplifiers. The operational amplifiers operate using a separate bipolar power supply V^{+2}/V^{-2} from the power amplifier. The first operational amplifier buffers the signal from the piezo sensor. R_{leak} should be chosen as large as possible such that the circuit is still stable. Then the 10μF capacitor and the two 47k resistors add a 1.65V offset to the signal before presenting the signal to the *AnalogIn* input to the mbed (see Figure 5).

The output filter provides a small amount of smoothing of the "stair-step" response of the zero order hold *AnalogOut*, and it also serves as a high-pass filter to remove the DC offset of 1.65V [10]. All of the electronic filters are low order and only affect the system outside of the bandwidth [1Hz 20kHz] to prevent interfering with the feedback control.[2]

While some aliasing is present in the signal sent to the power amplifier, this aliasing is not audible (at least for the guitar string) because the "instrument" filters it out.

[2]Consider for example, if the circuit is setup incorrectly, the system could for example go unstable even at a low frequecy, for example beneath the range of human hearing.

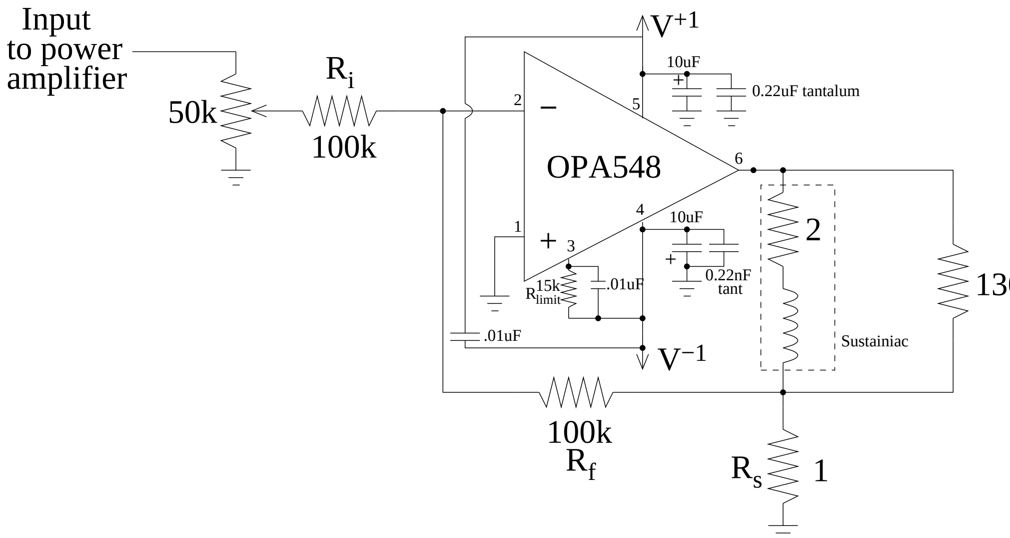

Figure 4: Same circuit as in Figure 3 but showing power supply connections and current limit resistor R_{limit}

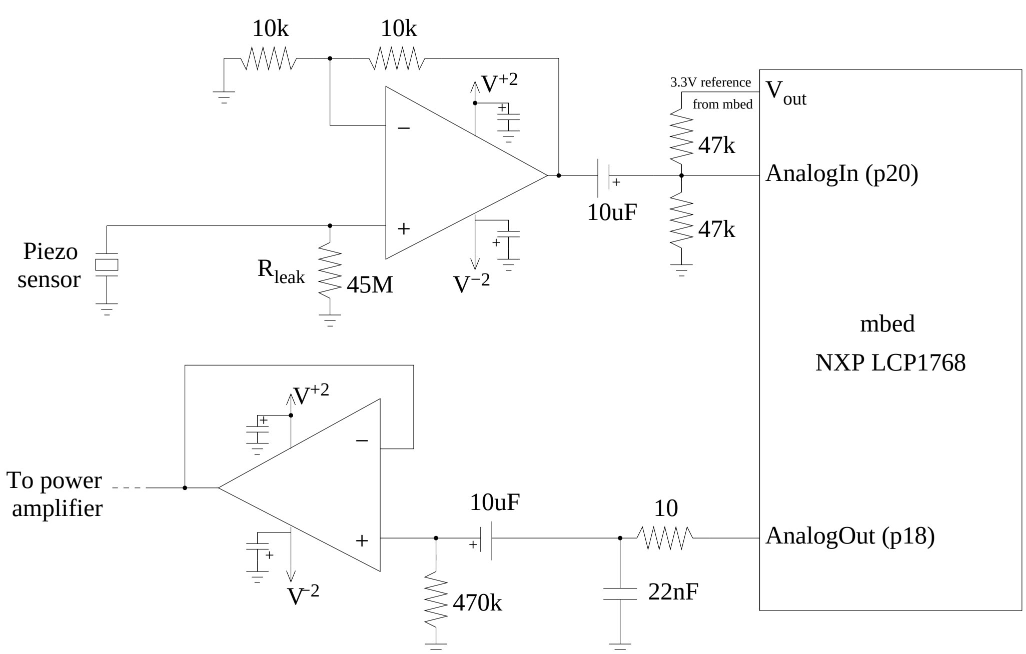

Figure 5: Remainder of the circuit

3.2 Software

3.2.1 Overview

In this section, we demonstrate how to implement the software. During an initial startup configuration phase, the software estimates the average DC offset o of the *AnalogIn* signal. Then, the software runs an infinite loop to essentially do the following:

1. Sample the *AnalogIn* signal once.
2. Subtract out the DC offset o from the sampled *AnalogIn* signal.
3. Then the sensed signal corresponds to the velocity of the guitar string. (For other kinds of sensors, the sensor signal might need to be integrated, differentiated, etc. in order to obtain the velocity.)
4. Compute a reaction force using native floating point operations. (For damping, the force should be proportional to the velocity.)
5. Lowpass-filter the reaction force to limit the control bandwidth to f_c. (For the feedback guitar, we generally choose f_c to be in the range [1kHz 4kHz], primarily due to the sensor and actuator not being perfectly collocated and because of the digital delay.)
6. Add in a mandatory output DC offset of 0.5 to maximize the dynamic range of the output, and write it to the *AnalogOut*.

3.2.2 Bandwidth Limiting Filter

The control bandwidth f_c is adjusted using a digital filter in #5. The filter is developed by applying the bilinear transform to the analog filter

$$H_{LP}(s) = \frac{a}{s+a}, \tag{1}$$

where $a = 2\pi f_c$. From the bilinear transform, we obtain the digital z-domain implementation of the filter

$$H_{LPz}(z) = \frac{a}{2f_s + a} \cdot \frac{1 + z^{-1}}{1 + \frac{a-2f_s}{a+2f_s} z^{-1}}, \tag{2}$$

in which f_s denotes the sampling rate. This filter design is preferrable (for example in comparison with a digital one-pole filter with no zeros) because it places a zero at the Nyquist frequency, which means that the magnitude response is rolled off significantly at higher frequencies.

3.2.3 Example Program

An example program in the appendix shows how to implement active damping using velocity feedback. Active sustain can be implemented with a sign inversion of the feedback.

A Example Program For mbed

```
#include "mbed.h"

// While this file has been assembled by Edgar Berdahl, some pieces have been taken from
// other publicly available mbed code on the Internet (for example for changing the
// ADC rate).

AnalogIn signalIn(p20);
```

```
AnalogOut signalOut(p18);
DigitalOut mypin(p21);
bool ledState=0;

void setADCclockToMaxConversionRate(void) {
    // Set pclk = cclk / 4  (pclk = 96Mhz/4 = 24Mhz)
    LPC_SC->PCLKSEL0 &= ~(0x3 << 24); // Clear bits 25:24

    // software-controlled ADC settings
    LPC_ADC->ADCR = (0 << 0) // SEL: 0 = no channels selected
              | (1 << 8 )    // CLKDIV: ADCCLK = PCLK / (CLKDIV + 1) (12MHz)
              | (0 << 16)    // BURST: 0 = software control
              | (0 << 17)    // CLKS: not applicable
              | (1 << 21)    // PDN: 1 = operational
              | (0 << 24)    // START: 0 = no start
              | (0 << 27);   // EDGE: not applicable}
    return;
}

// Estimate the mean on signalIn so that we can
// subtract it out again digitally for nice
// zero-mean signal processing.
#define NUMBER_MEASUREMENTS   5000
float estimateMean(void){
    float sum = 0.0;
    for (int i=0; i<NUMBER_MEASUREMENTS; i++)
        sum += signalIn.read();
    sum = sum / (float)NUMBER_MEASUREMENTS;

    return sum;
}

float pole = 0.99;
float fs = 30000.0;  // This is just a guess -- the sampling rate depends on the
                     // amount of computation in the loop.

float aS = 2.0*3.14159*1000.0;
float b = aS/(2.0*fs+aS);
float aIntg = (aS-2.0*fs)/(aS+2.0*fs);

int main() {
    float mean, velocity, force=0.0;
    float position=0.0, intgPosition=0.0;

    float lastUnfilteredForceCmd=0.0;
    float lastFilteredForceCmd=0.0;

    //__disable_irq();
    signalOut.write(0.5);

    setADCclockToMaxConversionRate();

    mean = estimateMean();
```

```
    while(1) {
// Using a scope, we can monitor the sampling rate at pin p21
        ledState = !ledState;
        mypin = ledState;

// Estimate velocity and position
        velocity = signalIn.read() - mean;
        position = pole*position + velocity/fs;

// Calculate the output force
// -------------------------
        //force = velocity;   // to implement sustain
        force = -1.0*velocity;  // for damping

        // Implement the one-pole one-zero bandwidth limiting filter.
        lastFilteredForceCmd = b*(force + lastUnfilteredForceCmd) - aIntg*lastFilteredForceCmd;
        lastUnfilteredForceCmd = force;

// Finally write the computed force to the DAC while adding in the required DC offset
        signalOut.write(0.5 + lastFilteredForceCmd);
    }
}
```

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