MCT 4054 Midway Assignment – The Feedbackquencer

Background

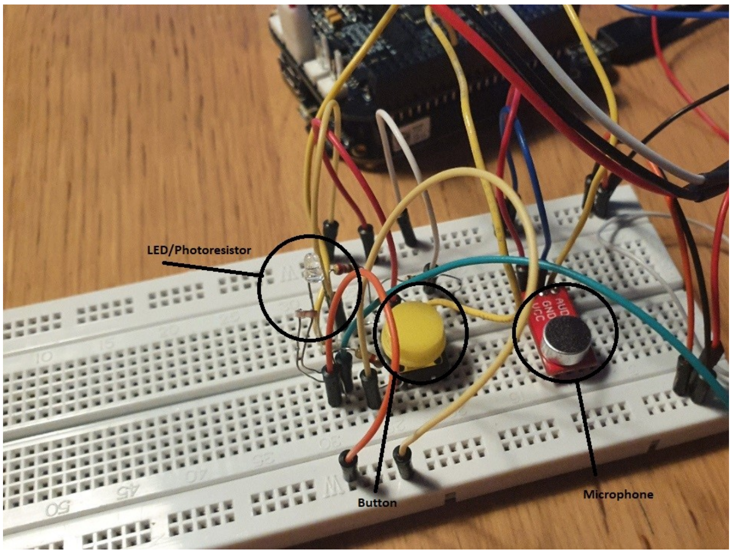
In audio system design, acoustic feedback is a phenomenon that is usually attempted to be avoided at all costs. However, as noted by Cathy van Eck (2017), acoustic feedback is "the sound of microphones and loudspeakers themselves" (4). Making reference to Holmes (2008), she explains that the composer Robert Ashley even went so far as to describe acoustic feedback as "the only sound intrinsic to electronic music" (58). The Feedbackquencer is an attempt to work with this idea.

System Design and Operation

The system comprises two major components. The first is a microphone/loudspeaker feedback component. The acoustic feedback generated by this component is passed to a four note sequencer, created by pitch shifting the feedback signal. The note duration and the semitones of pitch shift applied to each note are set before the Pure Data patch is run.

The second component is an envelope control comprising an LED pointed towards a photoresistor. The brightness of the LED is controlled by a signal (in this case a sawtooth), and the resultant signal captured by the photoresistor is applied to the audio signal. A button on the system enables switching between 5 frequencies for the signal controlling the LED (corresponding to double the note duration, the note duration, one-fifth of the note duration, one-tenth of the note duration, or simply a zero signal which turns the LED off).

Through these two components, the user plays the system by holding the loudspeaker in one hand and a sheet of paper in the other. Through manipulating the position of the loudspeaker in relation to the microphone and moving the sheet of paper between the LED and the photoresistor, the player can control the timbre, amplitude, and envelopes of the notes.



Input

The system takes 2 analogue and 1 digital signals as input; microphone, photoresistor, and button. The microphone and photoresistor inputs are directly linked to the outputs of the system. In terms of pre-processing, the microphone signal is processed with a high and low pass filter in order to remove low frequency noise and harsh high frequency sounds resulting from the acoustic feedback. The photoresistor signal is normalised.

Mapping

In terms of the player control during use of the system, the mappings of the loudspeaker and paper controls follow a one-to-many principle. The position of the loudspeaker controls both amplitude as well as timbre and the position of the paper controls general amplitude as well as the amplitude envelope of each note. The button offers a simple one-to-one mapping, cycling the length of the amplitude envelope applied to the audio signal.

Due to the use of the photoresistor additional mappings are afforded by the environment in which the system is used. For example, the light switch in the room can be used to turn on and off the room's light, switching between 2 general amplitude levels.

Sound Generation

The system does not generate any sound through the use of synthesis. Rather, sound (more precisely feedback) is generated by placing the loudspeaker in proximity to the microphone, and this feedback is manipulated by the system through the use of pitchshifting objects within the Pure Data patch**.**

As output, the audio signal comprises a mix of the raw feedback, the pitchshifted feedback without the amplitude envelope applied, and the pitchshifted feedback with the amplitude envelope applied. The LED control signal is also output to the LED.

References

Holmes, T. (2008). *Electronic and Experimental Music* (3rd ed.). Routledge.

van Eck, C. (2017). *Between air and electricity: Microphones and loudspeakers as musical instruments*. Bloomsbury Academic.